

## Cisco.350-801.v2023-06-21.q106

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### NEW QUESTION: 1

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 software transcoder on Cisco UCM to support voice calls to and from both remote sites
- C. configure the hardware transcoder on the site A router
- D. configure the hardware transcoder on the hub location router

**Answer: D (LEAVE A REPLY)**

### NEW QUESTION: 2

Which dial plan function restricts calls that are made by a lobby phone to internal extensions only?

- A. calling privileges
- B. manipulation of dialed destination
- C. path selection
- D. endpoint addressing

**Answer: (SHOW ANSWER)**

### NEW QUESTION: 3

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. Unsolicited NOTIFY
- B. TCP port 5060
- C. SIP Register
- D. 200 ok

**Answer: A ([LEAVE A REPLY](#))**

#### **NEW QUESTION: 4**

What is a reason for using a Diffserv value of AF41 for video traffic?

- A. Video traffic can tolerate up to 10% packet loss and latency of 10 seconds
- B. Video traffic can tolerate a packet loss of up to 1% and latency of 150 milliseconds
- C. Video traffic can tolerate up to 5% packet loss and latency of 5 seconds
- D. Video traffic cannot tolerate any packet loss and has a latency of 150 milliseconds

**Answer: ([SHOW ANSWER](#))**

#### **NEW QUESTION: 5**

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 dialled digits?

- A. Dialed Number Analyzer
- B. Dial Plan Analyzer
- C. Cisco Dial Plan Analyzer
- D. Digit Analysis Analyzer

**Answer: ([SHOW ANSWER](#))**

#### **NEW QUESTION: 6**

Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

- A. SRST
- B. SSO
- C. H.323 registration proxy to Cisco Unified Communications Manager
- D. MGCP gateway registration

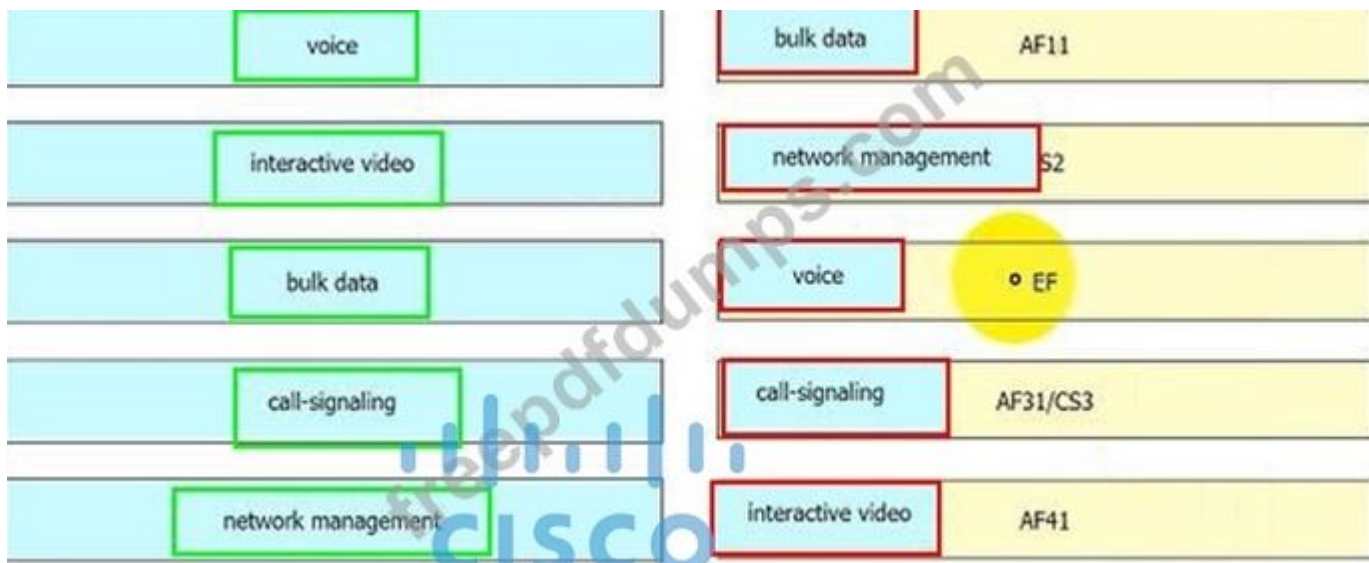
**Answer: ([SHOW ANSWER](#))**

#### **NEW QUESTION: 7**

According to the QoS Baseline Model, drag and drop the applications from the left onto the Per-Hop Behavior values on the right.



**Answer:**



**NEW QUESTION: 8**

Which location must be assigned to the SIP trunk to replicate enhanced location information via a SIP trunk?

- A. shadow
- B. replica
- C. phantom
- D. hub\_none

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 9**

An engineer configures local route group names to simplify a dial plan. Where does the engineer set the route groups according to the local route group names that are configured?

- A. route list
- B. CSS
- C. route pattern
- D. device pool

**Answer: D (LEAVE A REPLY)**

**NEW QUESTION: 10**

A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

- A. Set Call Forward All CSS to restrict international dial patterns.
- B. Configure a Forced Authorization Code on the international route pattern.
- C. Check Route Next Hop By Calling Party Number on the international route pattern.
- D. Block international dial patterns in the SIP trunk CSS.
- E. Set the Call Classification to OnNet for the international route pattern.

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 11**

Refer to the exhibit.

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

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?

- A. Priority = 100, Weight = 90
- B. Priority = 10, Weight = 5
- C. Priority = 10, Weight = 10
- D. Priority = 5, Weight = 70

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 12**

Which DiffServe PHB preserves backward compatibility with any IP precedence scheme?

- A. expedited forwarding
- B. class selector
- C. assured forwarding
- D. default

Answer: B ([LEAVE A REPLY](#))

**NEW QUESTION: 13**

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 Devices service parameter on each node
- B. Maximum Number of Registered and Unregistered Devices service parameter on each node
- C. Maximum Number of Phones service parameter on the Publisher
- D. Minimum Number of Phones service parameter on each node

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

**NEW QUESTION: 14**

How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco UCM?

- A. Call ingress must be normalized before the call being routed.
- B. Normalization is not required.
- C. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.
- D. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.

**Answer: D** ([LEAVE A REPLY](#))

**NEW QUESTION: 15**

What happens when a Cisco IP phone loses connectivity to the cluster during an active call?

- A. The call drops immediately.
- B. The call drops after missing two keepalives from Cisco UCM.
- C. The call continues to be active, but features like transfer or hold do not work.
- D. The call continues and all features work.

**Answer: B** ([LEAVE A REPLY](#))

**NEW QUESTION: 16**

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 regions on the Cisco UCM where the appropriate codec to use is set
- C. in DTMF settings in the audio codec preference list under regions in the Cisco UCM
- D. in the dial peer on the Cisco IOS router

**Answer: D** ([LEAVE A REPLY](#))

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**NEW QUESTION: 17**

How does traffic policing respond to violations?

- A. All traffic is treated equally.
- B. Excess traffic is retransmitted.
- C. Excess traffic is dropped.
- D. Excess traffic is queued.

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

#### **NEW QUESTION: 18**

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

- A. Presence Redundancy Groups (configured on Cisco Unified IM and Presence)
- B. IM and Presence subclusters (configured on Cisco UCM)
- C. IM and Presence subclusters (configured on Cisco Unified IM and Presence)
- D. Presence Redundancy Groups (configured on Cisco UCM)

**Answer: ([SHOW ANSWER](#))**

#### **NEW QUESTION: 19**

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 active directory to sync the users who will have a voicemail box.
- B. Configure a SIP integration with Cisco UCM to sync users.
- C. Configure an SCCP integration with Cisco UCM.
- D. Configure an AXL server to access the Cisco UCM users.

**Answer: D ([LEAVE A REPLY](#))**

#### **NEW QUESTION: 20**

Which two access layer switches provide support to provide high-quality voice and take advantage of the full voice feature set. To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support? Choose two

- A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.
- B. Implement IP RTP header compression on the serial interface to reduce the bandwidth required per voice call on point-to-point links.
- C. Deploy RSVP to improve VoIP QoS only where it can have a positive impact on quality and functionality where there is limited bandwidth and frequent network congestion.
- D. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.
- E. Use 808.IQ trunking and 802.Ip for proper treatment of Layer 2 CoS packet marking on ports with phones connected.

**Answer: ([SHOW ANSWER](#))**

**NEW QUESTION: 21**

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?

```
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
```

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
  video codec h264

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 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: ([SHOW ANSWER](#))

### NEW QUESTION: 22

What are the predefined call handlers in Cisco Unity Connection?

- A. opening greeting, welcome, and default system
- B. opening greeting, operator, and goodbye
- C. caller input, greetings, and transfer
- D. greetings, operator, and closed

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 23**

An engineer deploys a Cisco Expressway-E server for a customer who wants to utilize all features on the server. Which feature does the engineer configure on the Expressway-E?

- A. SIP gateway for PSTN providers
- B. VTC bridge
- C. Mobile and Remote Access
- D. H.323 endpoint registrations

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 24**

What is a characteristic of a SIP endpoint configured in Cisco UCM with 'Use Trusted Relay Point' set to 'On'?

- A. If TRP is allocated and MTP is also required for the endpoint, calls fail.
- B. It enables the Use Trusted Relay Point setting from the associated common device configuration.
- C. It enables Cisco UCM to insert an MTP or transcoder designated as a TRP.
- D. It creates a trust relationship with the called party.

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

**NEW QUESTION: 25**

When a remote office location is set up with limited bandwidth resources, which codec would allow the most voice calls with the limited bandwidth?

- A. G.722
- B. G.723
- C. G.711
- D. G.729

Answer: D ([LEAVE A REPLY](#))

**NEW QUESTION: 26**

Where in Cisco UCM are restrictions on audio bandwidth configured?

- A. region
- B. partition
- C. location
- D. serviceability

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

**NEW QUESTION: 27**

Which behavior occurs when Cisco UCM has a CallManager group that consists of two subscribers?

- A. If a subscriber is rebooted, endpoints deregister until the rebooted system is back in service.
- B. Endpoints attempt to register with both subscribers in a load-balanced method.

- C. Endpoints attempt to register with the top subscriber in the list.
- D. Endpoints attempt to register with the bottom subscriber in the list.

**Answer:** ([SHOW ANSWER](#))

**NEW QUESTION: 28**

Which Cisco IM and Presence service handles failover and state changes in the cluster?

- A. XCP router
- B. Cisco XCP Connection Manager
- C. XCP Sync Agent
- D. Cisco Server Recovery Manager

**Answer:** D ([LEAVE A REPLY](#))

**NEW QUESTION: 29**

Which information is needed to restore the backup of a Cisco UCM publisher successfully?

- A. the security password for Cisco UCM
- B. the application credentials for Cisco UCM
- C. the TFTP server details
- D. the FTP server details

**Answer:** A ([LEAVE A REPLY](#))

**NEW QUESTION: 30**

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 can be extended to voice and video devices if the connected PCs are included
- B. QoS trust boundaries can be extended to Jabber running on a PC
- C. QoS trust boundaries exclude Jabber softphone running on a PC
- D. QoS trust boundaries can be extended to voice and video devices exclusively
- E. QoS trust boundaries include all the devices directly attached to the access switch ports

**Answer:** ([SHOW ANSWER](#))

**NEW QUESTION: 31**

The IP phones at a customer site do not pick an IP address from the DHCP. An engineer must temporarily disable LLDP on all ports of the switch to leave only CDP. Which two commands accomplish this task? (Choose two.)

- A. Switch(config)# no lldp run
- B. Switch(config)# interface GigabitEthernet1/0/1
- C. Switch# configure terminal
- D. Switch(config)# no lldp transmit
- E. Switch# copy running-config startup-config

**Answer:** A,C ([LEAVE A REPLY](#))

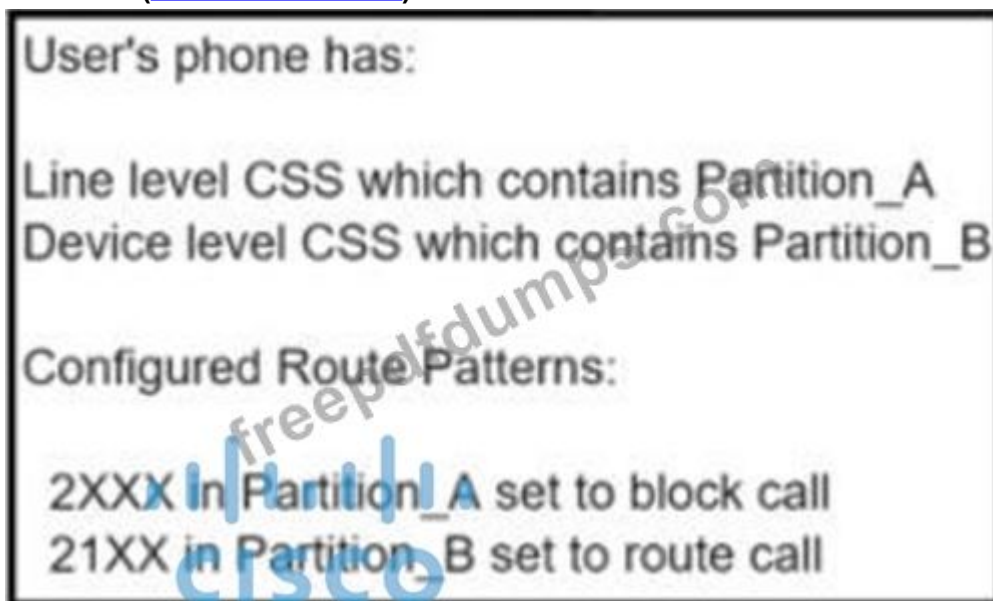
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**NEW QUESTION: 32**

Which two steps are required for bulk configuration transactions on the Cisco UCM database utilizing BAT? (Choose two.)

- A. A data file in Abstract Syntax Notation One format must be uploaded to Cisco UCM
- B. A server template must be created in Cisco UCM
- C. A data file in comma-separated values format must be uploaded to Cisco UCM
- D. A data file in Extensible Markup Language format must be uploaded to Cisco UCM
- E. A device template must be created in Cisco UCM

**Answer: (SHOW ANSWER)**



**NEW QUESTION: 33**

A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?

- A. real-time, call-signaling, critical data, best-effort, and scavenger
- B. call-signaling, real-time, critical data, best-effort, and drop-class
- C. voice, video, signaling, critical data, and best-effort
- D. real-time, signaling, critical data, best-effort and drop-class

**Answer: (SHOW ANSWER)**

**NEW QUESTION: 34**

An engineer implements QoS in the enterprise network. Which command is used to verify the classification and marking on a Cisco IOS switch?

- A. show class-map interface GigabitEthernet 1/0/1
- B. show policy-map
- C. show access-lists
- D. show policy-map interface GigabitEthernet 1/0/1

**Answer: D** ([LEAVE A REPLY](#))

**NEW QUESTION: 35**

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. 87.2 kbps
- B. 55.2 kbps
- C. 31.2 kbps
- D. 38.4 kbps

**Answer: A** ([LEAVE A REPLY](#))

**NEW QUESTION: 36**

What is a description of the DiffServ model used for implementing QoS?

- A. RTP traffic from voice and video calls is marked EF and placed in the same queue.
- B. AF43 has higher drop precedence than AF42 but lower drop precedence than AF41.
- C. Voice and video calls are marked with different DSCP values and placed in different queues.
- D. AF41 has higher drop precedence than AF42. which has higher drop precedence than AF43.

**Answer: D** ([LEAVE A REPLY](#))

**NEW QUESTION: 37**

What are two characteristics of jitter in voice and video over IP communications? (Choose two.)

- A. The packets arrive at varying time intervals.
- B. The packets never arrive due to tail drop.
- C. The packets arrive out of sequence.
- D. The packets arrive with frame errors.
- E. The packets arrive at uniform time intervals.

**Answer: A,C** ([LEAVE A REPLY](#))

**NEW QUESTION: 38**

Refer to the exhibit.

```
voice class codec 20
codec preference 1 g722-64
codec preference 2 ilbc mod 30
!
dial-peer voice 200 voip
destination-pattern ^408555...$
session target ipv4:10.2.3.4
incoming called-number 9T
dtmf-relay h245-alphanumeric rtp-nte
no vad
!
```

Refer to the exhibit. An administrator configured a codec preference list with 0,122 and ILBC codecs. Which change must the administrator make in the dial-peer section of the configuration to use this list?

- A. add session codec 20
- B. add codec preference 20
- C. add voice-codecs 20
- D. add voice-class codec 20

**Answer:** ([SHOW ANSWER](#))

**NEW QUESTION: 39**

End users report bad video quality and voice choppiness on Cisco Collaboration endpoints. The engineer changed the device pool the users were in but did not correct the problem. Which action should be taken to troubleshoot this issue?

- A. Restart the Cisco Location Bandwidth Manager service on the Cisco UCM publisher.
- B. Check for duplex/speed mismatches between the network port settings of the system and network switch.
- C. Use direct IP address calls between two endpoints to troubleshoot call quality issues.
- D. Set the service parameter Use Video Bandwidth Pool for Immersive Video Calls to "false".

**Answer:** D ([LEAVE A REPLY](#))

**NEW QUESTION: 40**

Which DSCP marking is represented as 101110 in an IP header?

- A. CS3
- B. EF
- C. AF41
- D. AF31

**Answer:** ([SHOW ANSWER](#))

**NEW QUESTION: 41**

Which DSCP value and PHB equivalent are the default for audio calls?

- A. 32 and CS4
- B. 32 and AF41
- C. 48 and EF
- D. 34 and AF41

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

#### NEW QUESTION: 42

Refer to the exhibit.

```
Gateway1#show sccp
SCCP Admin State: UP
Gateway Local Interface: Loopback0
  IPv4 Address: 192.168.12.1
  Port Number: 2000

Gateway1#
Gateway1#show ccm-manager
% Call Manager Application is not enabled
Gateway1#

Gateway1#show mgcp
MGCP Admin State DOWN, Oper State DOWN - Cause Code NONE
MGCP call-agent: none Initial protocol service is MGCP 0.1
MGCP validate call-agent source-ipaddr DISABLED
MGCP validate domain name DISABLED
MGCP block-newcalls DISABLED
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
```

Refer to the exhibit. A collaboration engineer adds an analog gateway to a Cisco UCM cluster. The engineer chooses MGCP over SCCP as the gateway protocol. Which two actions ensure that the gateway registers? (Choose two.)

- A. Enter "no seep" on the gateway in configuration mode.
- B. Enter "ccm-manager mgcp" on the gateway in configuration mode.
- C. Enter "mgcp" on the gateway in configuration mode.
- D. Enter "ccm-manager config" on the gateway in configuration mode.
- E. Delete and re-add the gateway configuration in Cisco UCM.

**Answer: B,C** ([LEAVE A REPLY](#))

#### NEW QUESTION: 43

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. signaling

- B. video conference
- C. voice
- D. streaming video

**Answer: D** ([LEAVE A REPLY](#))

**NEW QUESTION: 44**

Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

- A. 15 seconds
- B. 10 seconds
- C. 5 seconds
- D. 20 seconds

**Answer: A** ([LEAVE A REPLY](#))

**NEW QUESTION: 45**

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. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer
- B. Advise the user to press the "#" button after dialing the FAC and CMC codes
- C. Change the code if the problem persists
- D. Do not wait for the tones immediately dial the FAC and CMC
- E. Advise the user to hang up and try again

**Answer: A,C** ([LEAVE A REPLY](#))

**NEW QUESTION: 46**

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 bridge is configured to support G.711. The remote user's phone only supports 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. 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: (SHOW ANSWER)**

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**NEW QUESTION: 47**

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

- A. Incorrect partition
- B. Incorrect QoS
- C. Network congestion
- D. Incompatible codec
- E. No region relationship

**Answer: B,C ([LEAVE A REPLY](#))**

**NEW QUESTION: 48**

An administrator is developing an 8-class QoS baseline model. The CS3 standards-based marking recommendation is used for which type of class?

- A. call signaling
- B. best effort
- C. voice
- D. Scavenger

**Answer: D ([LEAVE A REPLY](#))**

**NEW QUESTION: 49**

Refer to the exhibit.

**SIP Trunk Security Profile Information**

Name\* Secure SIP Trunk Profile

Description Non Secure SIP Trunk Profile authenticated by null String

Device Security Mode Encrypted

Incoming Transport Type\* TLS

Outgoing Transport Type TLS

Enable Digest Authentication

Nonce Validity Time (mins)\* 600

Secure Certificate Subject or Subject Alternate Name

Incoming Port\* 5061

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 common name of the Cisco UCM CallManager certificate.
- B. The common name of the remote device certificates.
- C. The fully qualified domain name of the remote device that is configured on the SIP trunk.
- D. The full qualified domain name of all Cisco UCM nodes that run the CallManager service.

**Answer: A (LEAVE A REPLY)**

**NEW QUESTION: 50**

Due to service provider restriction. Cisco UCM cannot send video in the SDR Which two options on Cisco UCM are configured to suppress video in the SDP in outgoing invites? (Choose two.)

- A. Change the Video Capabilities dropdown on the endpoint to Disabled.
- B. Check the Retry Video Call as Audio on the SIP trunk.
- C. Check the Send send-receive SDP in mid-call INVITE check box on the SIP trunk SIP profile.
- D. Add the audio forced command to voice service VoIP on the Cisco Unified Border Element.
- E. Set Video Bandwidth in the Region settings to 0.

**Answer: A,E (LEAVE A REPLY)**

**NEW QUESTION: 51**

An administrator has been asked to implement toll fraud prevention in Cisco UCM Which tool is used to begin this process?

- A. Cisco UCM Access Control Group restrictions
- B. Cisco Unified Mobility
- C. Cisco Unified Real-Time Monitoring Tool

D. Cisco UCM class of service

**Answer: D ([LEAVE A REPLY](#))**

**NEW QUESTION: 52**

Which Cisco unity Connection handler plays a greeting at announces the option to dial a user extension by default?

A. the Goodbye call handler

B. the operator call handler

C. the Interview handler

D. the Directory handler

**Answer: B ([LEAVE A REPLY](#))**

**NEW QUESTION: 53**

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. partitions and calling search spaces

B. calling patterns and route patterns

C. regions and device pools

D. links and pipes

**Answer: A ([LEAVE A REPLY](#))**

**NEW QUESTION: 54**

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: ([SHOW ANSWER](#))**

**NEW QUESTION: 55**

Which QoS marking is used when an administrator configures voice call signaling?

A. EF

B. CS4

C. AF41

D. CS3

**Answer: D ([LEAVE A REPLY](#))**

**NEW QUESTION: 56**

Refer to the exhibit.

The exhibit shows three screenshots of the Cisco Unified Communications Manager configuration interface for Calling Search Spaces (CSS).  
1. **Global-CSS:** Name: Global-CSS, Description: Line Level CSS for calls including International. Route Pattern: 777011496929810, Route Partition: Internationale\_FT. Selected Partitions: BlockFraud-PT, BlockGlobal-PT, Test1-Svc-PT, Test2-Svc-PT.  
2. **Intl\_CSS:** Name: Intl\_CSS, Description: Calls including INTL. Selected Partitions: LOCAL\_CALLS, International\_PT.  
3. **Unrestricted-CSS:** Name: Unrestricted-CSS, Description: Line Level CSS for calls including Unrestricted. Selected Partitions: BlockFraud-PT.

Refer to the exhibit. How must the +E.164 translation pattern be configured to reach international number 496929810?

The exhibit shows four configuration options for the +E.164 translation pattern:

- Pattern= \+.496929810, CSS=Unrestricted-CSS, PreDot, Prefix=777011
- Pattern= \+.777011496929810, CSS=Intl\_CSS
- Pattern= \+.011496929810, CSS=Global-CSS, PreDot, Prefix=777
- Pattern= \+.496929810, CSS=Intl\_CSS, PreDot, Prefix=777011

- A. Option A
- B. Option D
- C. Option C
- D. Option B

**Answer:** (SHOW ANSWER)

**NEW QUESTION: 57**

An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager. Which phone type must be selected?

- A. Cisco Dual Mode for iPhone
- B. third-party SIP device

- C. Cisco Dual Mode for Android
- D. Cisco Unified Client Services Framework

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

**NEW QUESTION: 58**

Refer to the exhibit.

```
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
```

Which Codec is negotiated?

- A. ILBC
- B. G.728
- C. G.729
- D. G.711ulaw

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 59**

A customer is deploying a SIP IOS gateway for a customer who requires that in-band DTMF relay is first priority and out-of-band DTMF relay is second priority. Which 10\$ entry sets the required priority?

- A. dtmf-relay cisco-rtp
- B. sip-notify dtmf-relay rtp-nte
- C. dtmf-relay rtp-nte sip-notify

D. dtmf-relay sip-kpml cisco-rtp

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 60**

Exhibit.

```
admin:utils ntp status
ntpd (pid 14550) 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
```

Refer 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?

- A. Delete the original NTP server from Cisco UCM
- B. Enable NTP authentication for the new NTP server on Cisco UCM
- C. Stop the NTP service on Cisco UCM
- D. Restart the NTP service on Cisco UCM

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 61**

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. 750 and 15.000 users
- C. 1000 and 10.000 users
- D. 750 and 10.000 users

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

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#### NEW QUESTION: 62

Which two steps should be taken to provision a phone after the Self-Provisioning feature was configured for end users? (Choose two.)

- A. Dial the hunt pilot extension and associate the phone to an end user
- B. Ask the Cisco UCM administrator to associate the phone to an end user.
- C. Plug the phone into the network.
- D. Enter settings menu on the phone and press \* , \*,# (star, star, pound).
- E. Dial the self-provisioning IVR extension and associate the phone to an end user.

Answer: C,E ([LEAVE A REPLY](#))

#### NEW QUESTION: 63

An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)

- A. location
- B. device security profile
- C. CSS
- D. media resources group list
- E. SIP profile

Answer: B,E ([LEAVE A REPLY](#))

#### NEW QUESTION: 64

How are network devices monitored in a collaboration network?

- A. System logs are collected in a Cisco Prime Collaboration Server.
- B. Simple Network Managed Protocol is enabled on each device to poll specific values periodically.
- C. Ping Sweep reports "unmanaged" state devices.
- D. The Cisco Discovery Protocol table is shared among devices.

Answer: ([SHOW ANSWER](#))

#### NEW QUESTION: 65

Refer to the exhibit. An engineer is confining class of control for a user in Cisco UCM. Which change will ensure that the user is unable to call 2143?

- A. Set the user's line CSS to <None>
- B. Change line partition to Partition\_A
- C. Change line CSS to only contain Partition\_B
- D. Set the user's device CSS to <None>

Answer: D ([LEAVE A REPLY](#))

**NEW QUESTION: 66**

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. For extension mobility, logged-out CSS is restricted to internal extensions and emergencies.
- B. Add an additional firewall between the Cisco UCM server and the Expressway Core server.
- C. Forced authorization code is used to recognize a dialing extension and authorize an international call.
- D. Ensure that ad hoc conference calls are dropped if an external user is add.
- E. Call forward settings (ALL/Busy/No Answer) are restricted to internal extensions in the network

**Answer: C,E ([LEAVE A REPLY](#))**

**NEW QUESTION: 67**

Which Cisco Unified communications manager configuration is required for SIP MWI integration?

- A. Enable "Accept unsolicited notification" on the SIP Trunk security profile
- B. Select "Redirecting Diversion Header Delivery- Inbound" on the SIP trunk
- C. Select "Redirecting Diversion Header Delivery - outbound" on the SIP trunk
- D. Enable "Accept presence subscription" on the SIP trunk security profile

**Answer: ([SHOW ANSWER](#))**

**NEW QUESTION: 68**

Refer to the exhibit.

```
isdn switch-type primary-ni
controller t1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-10
```

Refer to the exhibit. An engineer configures ISDN on a voice gateway. The provider confirms that the PRI is configured with 10 channels the engineer ordered and is working from the provider side, but the engineer cannot get a B-channel to carry voice. The rest of the configuration for the serial interface and voice network is functioning correctly. Which actions must be taken to carry voice?

- A. The engineer must activate the controller card on the voice gateway before configuring the device.
- B. The engineer used a T1 interface but must use an E1 interface.
- C. The pri-group timeslots command must be 0-9 for the 10 channels because all values on a router start with 0.
- D. The engineer must manually revert the order of using the channels.

**Answer: ([SHOW ANSWER](#))**

**NEW QUESTION: 69**

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 primary extension.
- B. End users must belong to Standard CCM Admin Users group, the Standard CCM End Users group, and the Standard CCM Self-Provisioning group.
- C. End users must have a secondary extension.
- D. Cisco Extended Functions service must be running
- 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: A,E (LEAVE A REPLY)**

**NEW QUESTION: 70**

Refer to the exhibit.

```
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
```

An engineer must modify the existing QoS policy-map statement to implement LLQ for voice traffic. Which change must the engineer make in the configuration?

- A. bandwidth 170 to LL1 170
- B. bandwidth 170 to priority 170
- C. bandwidth 170 to percent 170
- D. bandwidth 170 to reserve 170

**Answer: B (LEAVE A REPLY)**

**NEW QUESTION: 71**

A remote office has a less-than-optimal WAN connection and experiences packet loss, delay and jitter. Which VoIP codec is used in this situation?

- A. G.711alaw
- B. G722.1
- C. G.729A
- D. iLBC

**Answer: D (LEAVE A REPLY)**

### NEW QUESTION: 72

An engineer must configure switch port 5/1 to send CDP packets to configure an attached Cisco IP phone to trust tagged traffic on its access port. Which command is required to complete the configuration?

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router config-if)# description Cube E41.228-0097
```

- A. platform qos trust extend
- B. platform qos trust extend cos 5
- C. platform qos extend trust
- D. platform qos trust extend cos 3

**Answer: A** ([LEAVE A REPLY](#))

### NEW QUESTION: 73

Which action prevent toll fraud in Cisco Unified Communication Manager?

- A. Configure ad hoc conference restriction
- B. Allow off-net to off-net transfer
- C. Implement toll fraud restriction in the Cisco IOS router
- D. Implement route patterns in Cisco Unified CM

**Answer: A** ([LEAVE A REPLY](#))

### NEW QUESTION: 74

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 > Enterprise Parameters
- B. On Cisco UCM Administrator page system > Device Pool > Cisco UCM group
- C. On Cisco UCM Administration Page Device > Phone > Phone Configuration page
- D. On Cisco UCM Administrator Page server > Cisco UCM

**Answer: B** ([LEAVE A REPLY](#))

### NEW QUESTION: 75

What is a characteristic of video traffic that governs QoS requirements for video?

- A. Voice and video traffic are different, but they have the same QoS requirements.
- B. Voice and video are the same, so they have the same QoS requirements.
- C. Video is typically constant bit rate.
- D. Video is typically variable bit rate.

**Answer: D** ([LEAVE A REPLY](#))

### NEW QUESTION: 76

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 CU 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. Set a username and password under Secure Shell information in Cisco UCM
- D. Disable Web Access under Product Specific Configuration Layout in Cisco UCM
- E. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM

Answer: ([SHOW ANSWER](#))

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#### NEW QUESTION: 77

Why isn't an end user s PC device in a QoS trust boundary included?

- A. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.
- B. The end user could incorrectly tag their traffic to bypass firewalls.
- C. The end user may incorrectly tag their traffic to be prioritized over other network traffic.
- D. There is no reason not to include an end user's PC device in a QoS trust boundary.

Answer: ([SHOW ANSWER](#))

#### NEW QUESTION: 78

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. SBCs can modify headers such as To, From, Contact, and Call-ID. It can introduce new headers into the SIP message
- 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. SIP proxies are SDP-aware and can change the SDP bodies
- D. Stateful proxies are context-aware and can terminate communication sessions by themselves
- 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: 79**

When setting a new primary DNS server in the Cisco UCM CLI what is required for the change to take affect?

- A. restart of DirSync service
- B. restart of the network service
- C. restart of TFTP service
- D. restart of CallManager service

**Answer:** ([SHOW ANSWER](#))

**NEW QUESTION: 80**

A Cisco Unity Connection Administrator must set a voice mailbox so that it is accessed from a secondary device. Which configuration on the voice mailbox makes this change?

- A. Mobile User
- B. Alternate Names
- C. Alternate Extensions
- D. Attempt Forward routing rule

**Answer:** ([SHOW ANSWER](#))

**NEW QUESTION: 81**

An engineer builds the configuration on a Cisco IOS gateway for the dial-peers:

```
dial-peer voice 2 voip
 destination-pattern 5419232188
 session-targeted qos 10.5.5.7
 no protocol 08 no redundancy 0 fallback voice
 no rate voice
```

Which command is required to complete the configuration?

- A. Codec g726r32
- B. Codec g711ulaw
- C. Codec g723ar63
- D. Codec g729cr81

**Answer:** B ([LEAVE A REPLY](#))

**NEW QUESTION: 82**

A customer wants a video conference with five Cisco Telepresence 1X5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

- A. Cisco Meeting Server
- B. Cisco PVDM4-128
- C. software conference bridge on Cisco UCM
- D. Cisco Webex Meetings Server

**Answer:** A ([LEAVE A REPLY](#))

**NEW QUESTION: 83**

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 Expressway-C, and Cisco UCM
- C. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server
- D. Cisco Unified Border Element, Cisco IM and Presence Server, and Cisco Video Communication Server

Answer: B ([LEAVE A REPLY](#))

**NEW QUESTION: 84**

Callers from a branch report getting busy tones intermittently when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco CallManager service traces to examine the situation. The traces show:

```
50805567.000 |07:35:39.676 |Sdl Sig |StationOutputDisplayNotify |restart0
|StatinD(1,100,63,6382) |StionCdpc(1,100,64,4725) |1,100,40,6.709919^*^*
|[R:N-H:0,L:0,V:0,Z:0,D:0] TimeOutValue=10 Status=x807 Unicode Status=Locale=1
50805567.001 |07:35:39.676 |AppInfo |StationD: (0006382) DisplayNotify
timeOutValue=10 notify='x807' content='Not Enough Bandwidth' ver=85720014.
```

What should be fixed to resolve the issue?

- A. geolocation configuration
- B. region configuration
- C. class of service configuration
- D. codec configuration

Answer: B ([LEAVE A REPLY](#))

**NEW QUESTION: 85**

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?

A. 

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#pri-group timeslots 1-7
```

B. 

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-group timeslots 1-7
```

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#interface Serial0/0/0:15
2900(config-controller)#pri-group timeslots 1-7
```

C.

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-timeslots 1-7
```

D.

Answer: B ([LEAVE A REPLY](#))

#### NEW QUESTION: 86

Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?

- A. CAPF
- B. Cisco CallManager
- C. IPsec
- D. Cisco Tomcat

Answer: ([SHOW ANSWER](#))

#### NEW QUESTION: 87

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. route pattern
- B. SIP route patterns
- C. CTI route point
- D. route group
- E. SIP trunk

Answer: A,E ([LEAVE A REPLY](#))

#### NEW QUESTION: 88

An engineer must configure a SIP route pattern using domain routing with destination +13135551212. The domain ciscocm1.jupiter.com resolves to 192.168.1.3. How must the IPv4 Pattern be configured?

- A. \+13135551212@192.168.1.3
- B. ciscocm1.jupiter.com
- C. +13135551212@192.168.1.3
- D. 192.168.1.3

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 89**

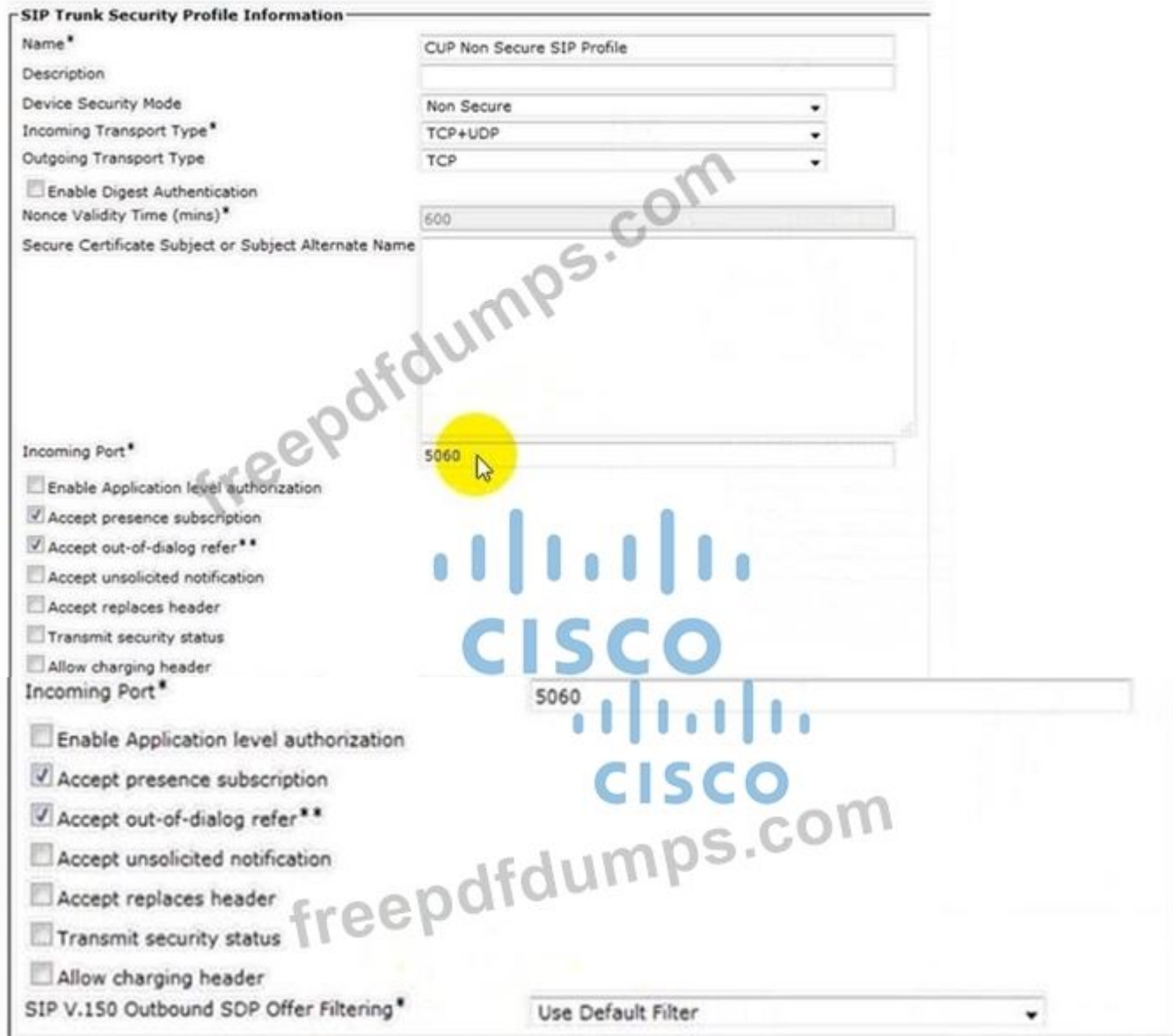
How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

- A. Configure a calling party transformation pattern that keeps only the last 10 digits.
- B. Configure a translation pattern that has a Calling Party Transform Mask of XXXXXXXXXXXX.
- C. On the inbound SIP trunk, change Significant Digits to 10.
- D. Change the service parameter Apply Transformations On Remote Number to True.

Answer: ([SHOW ANSWER](#))

**NEW QUESTION: 90**

Refer to the exhibit.



Refer to the exhibit. A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

- A. Check the box to allow charging header.
- B. Check the box to enable application-level authorization.
- C. Check the box to accept replaces header.
- D. Check the box to transmit security status.
- E. Check the box to accept unsolicited notification.

**Answer: C,E ([LEAVE A REPLY](#))**

#### **NEW QUESTION: 91**

An engineer is integrating Unity Connection with Cisco UCM. Which two actions must be configured so that recording and playback from the IP phones works at all times, including peak traffic hours? (Choose two.)

- A. If it's a Unity Connection Cluster, ensure that replication is fine and not in split-brain mode.
- B. Increase the number of voice ports.
- C. Add dedicated dial-out ports with the allow trap connections setting selected.
- D. Ensure that you have set up SIP Digest Authentication on the SIP trunk security profile.
- E. The phone system to which the phones are registered in Unity Connection has the Default Trap Switch check box enabled.

**Answer: ([SHOW ANSWER](#))**

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#### **NEW QUESTION: 92**

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 SIP Trunk security profile page in CUCM Administration
- B. The software Upgrades page in CUCM OS Administration
- C. The phone configuration page in CUCM Administration
- D. The In-Room control Editor on the webpage of the MX800

**Answer: D ([LEAVE A REPLY](#))**

### NEW QUESTION: 93

Refer to the exhibit.

```
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-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
```

Refer to the exhibit. When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages are examined next to further troubleshoot the issue? (Choose two.)

- A. PRACK
- B. NOTIFY
- C. UPDATE
- D. SUBSCRIBE
- E. REGISTER

Answer: B,D ([LEAVE A REPLY](#))

### NEW QUESTION: 94

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. 143
- B. 150
- C. 166
- D. 66

Answer: B ([LEAVE A REPLY](#))

### NEW QUESTION: 95



- A. Add a transcoder that supports G.711ulaw and G.729.
- B. Nothing; both sides support G.729.
- C. Add a media termination point that supports G.711ulaw and G.729.
- D. Nothing both sides support payload type 101.

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

**NEW QUESTION: 98**

Refer to the exhibit.

```

hostname GATEWAY
cme-manager config
cme-manager config server 192.168.1.100
cme-manager mgcp

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

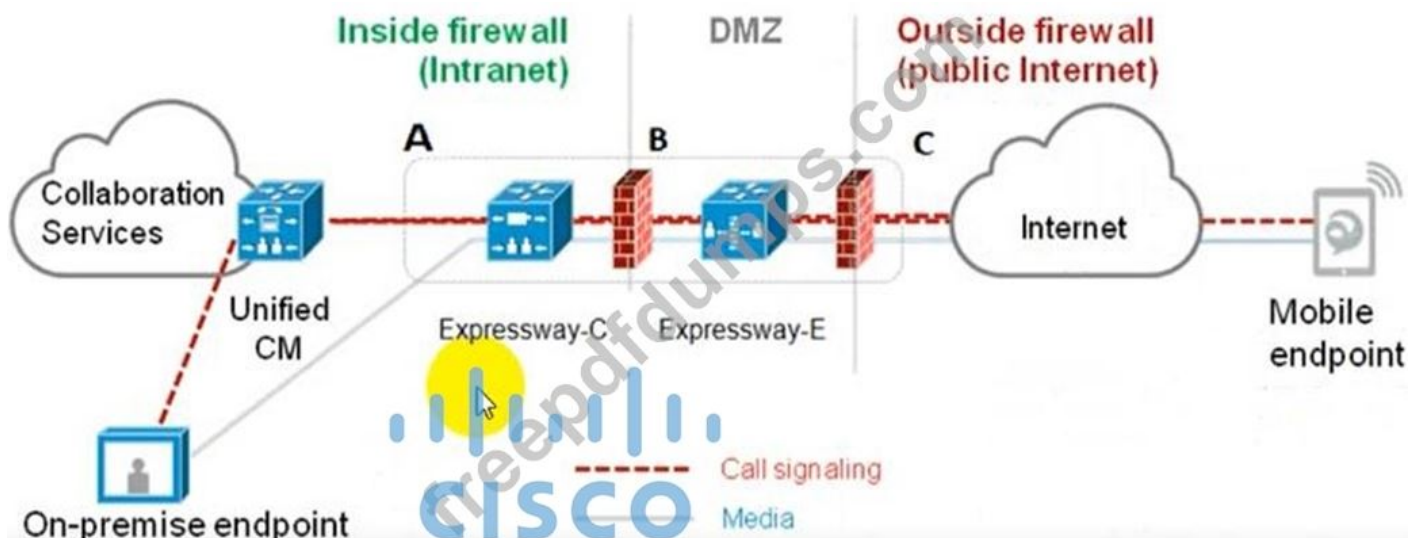
An engineer verifies the configured of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

- A. Device (config)# mgcp
- B. Device(config)# mgcp enable
- C. Device (config) # com-manager active
- D. Device(config)# ccm-manager enable

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

**NEW QUESTION: 99**

Refer to the exhibit.



Refer to the exhibit. When making a call to a Mobile and Remote Access client, what are the combinations of protocol on each of the different sections A-B-C?

- A. SIP TLS (A) + SIP TLS (B) + SIP TLS (C)
- B. SIP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TCP/TLS (C)
- C. IP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TLS (C)
- D. SIP TCP/TLS (A) + SIP TLS (B) + SIP TLS (C)

**Answer: D ([LEAVE A REPLY](#))**

**NEW QUESTION: 100**

What are two features of Cisco Expressway that the customer gets if Expressway-C and Expressway-E are deployed?(Choose two.)

- A. utilization and adoption metrics of all remotely connected devices.
- B. session-based access to comprehensive collaboration for remote workers, without the need for a separate VPN client.
- C. highly secure free-traversal technology to extend organizational reach.
- D. additional visibility of the edge traffic in an organization.
- E. complete endpoint registration and monitoring capabilities for devices that are local and remote.

**Answer: B,C ([LEAVE A REPLY](#))**

**NEW QUESTION: 101**

Which two features of Cisco Prime Collaboration Assurance require advanced licensing? (Choose two.)

- A. call quality monitoring
- B. real time alarm browse
- C. email notifications
- D. customizable events
- E. multicluster support

**Answer: A,E ([LEAVE A REPLY](#))**

**NEW QUESTION: 102**

Which two recommendations are made to optimize Cisco UCM configuration to reduce the number of toll fraud incidents in an organization? (Choose two.)

- A. Classify all route patterns as on-net or off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.
- B. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco UCM service parameters.
- C. Inbound CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.
- D. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.
- E. Classify all route patterns as off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.

**Answer: ([SHOW ANSWER](#))**

**NEW QUESTION: 103**

Which command in the MGCP gateway configuration defines the secondary Cisco UCM server?

- A. ccm-manager redundant-host
- B. mgcpapp
- C. mgcp call-agent
- D. ccm-manager fallback-mgcp

**Answer: A** ([LEAVE A REPLY](#))

**NEW QUESTION: 104**

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 9 011841234567  
configure the Discard Digits as Predot 10-10-Dialing

C)

in the Calling Party Transformation Patterns section, configure the Pattern as 9 011841234567  
configure the Discard Digits as Predot

D)

in the Called Party Transformation Pattern Configuration section, configure the Pattern as 9 011841234567  
configure the Discard Digits as Predot 10-10-Dialing

- A. Option B
- B. Option D
- C. Option A
- D. Option C

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

**NEW QUESTION: 105**

Refer to the exhibit.

```

Sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKFlFED
From: "7000" <sip:7000@192.168.100.200>;tag=43CDE-1A22
To: <sip:2004@192.168.100.100>
Call-ID: 26BCA00-4C4E11EA-80169514-B1C46126@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

```



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?

- A. Cisco Unified Border Element is sending the incorrect media IP address. The IP address of the loopback interface must be advertised in the SDP
- B. Cisco Unity Connection is configured on G.729 only. Cisco Unity Connection must be reconfigured to support iLBC.
- C. 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
- D. Cisco Unified Border Element is not sending any support for DTMF. OTMF configuration must be added to the appropriate dial peer.

**Answer: D (LEAVE A REPLY)**

### NEW QUESTION: 106

A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail. Which action in the Cisco Unity Connection restriction table resolves this issue?

- A. Create a custom restriction table \*\*\*\*\*and block it.
- B. Block PSTN patterns on Default Transfer. Default Outdial. and Default System Transfer.
- C. Create a custom restriction table ?????????? and block it.
- D. Implement password complexity on voicemail boxes to prevent accounts from being compromised.

**Answer: B (LEAVE A REPLY)**

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