



# Cisco

## Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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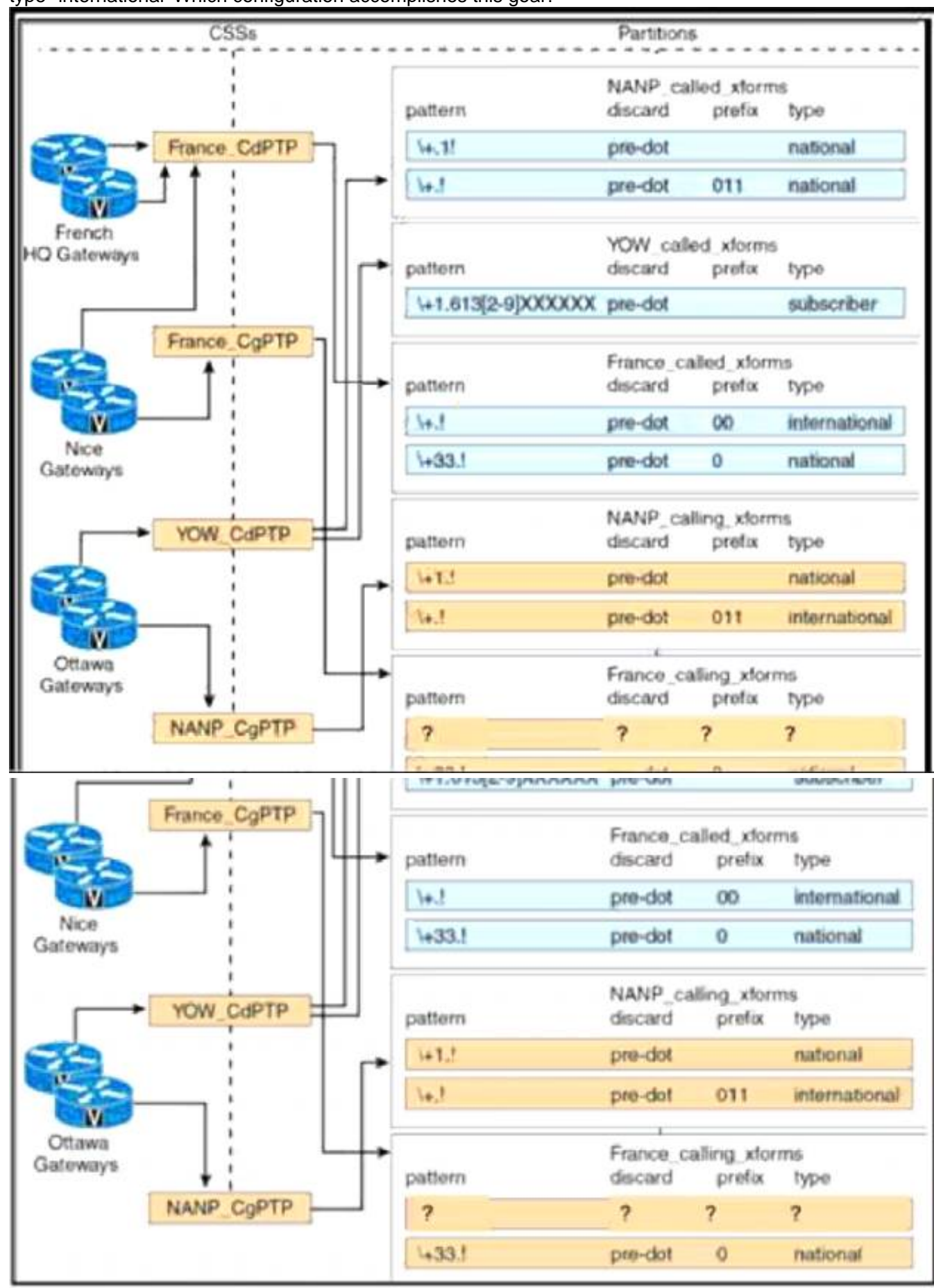
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### NEW QUESTION 1

Refer to the exhibit A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type "international" Which configuration accomplishes this goal?



- A. \+.001! pre-dot 1 international
- B. \+1.1 none pre-dot 001 international
- C. \+! pre-dot 00 international
- D. 613XXXXXXX none +011 international

Answer: C

### NEW QUESTION 2

What happens when a Cisco IP phone loses connectivity to the cluster 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: D

### NEW QUESTION 3

What is an indicator of network congestion in VoIP communications?

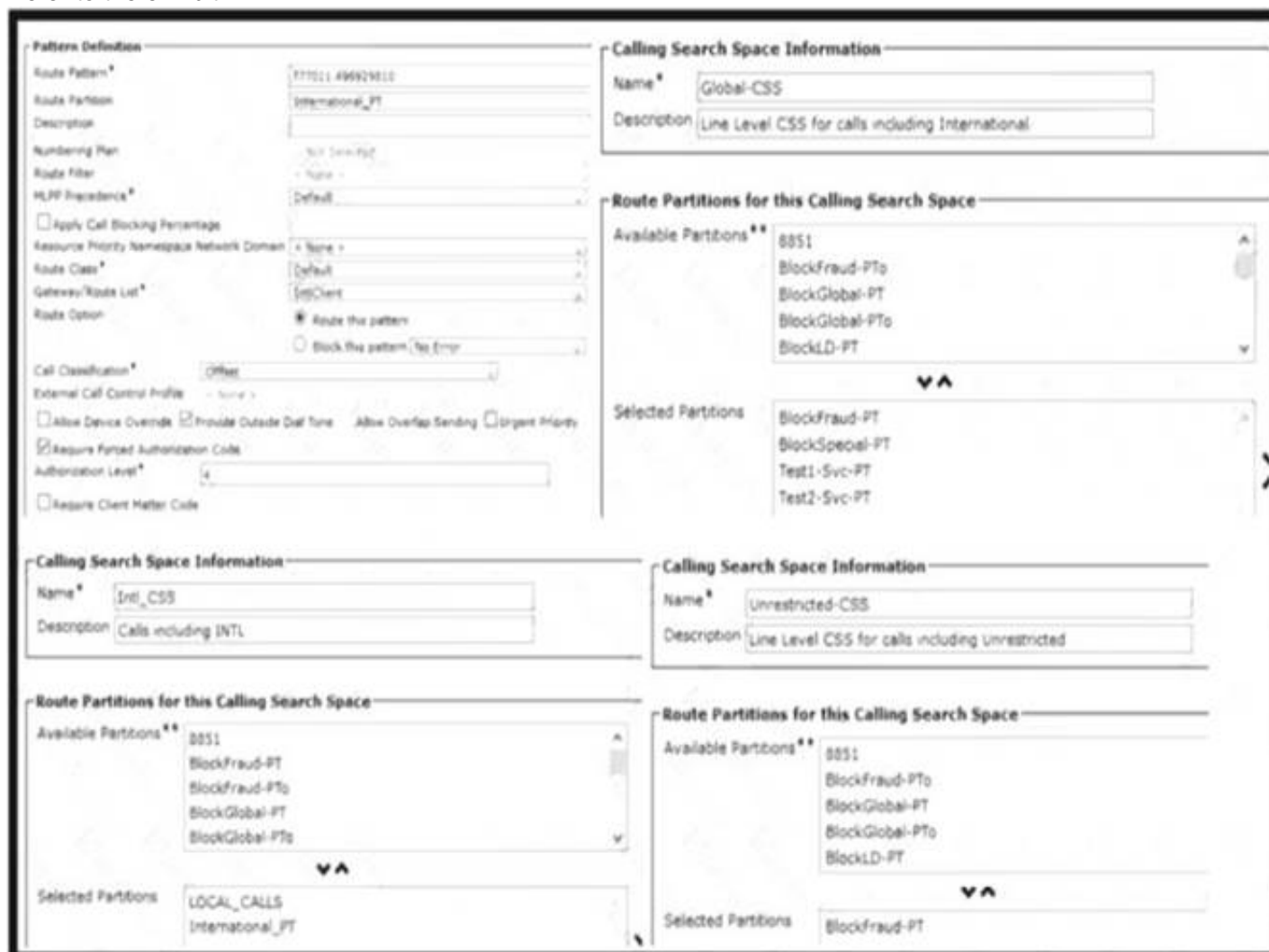
- A. jitter increase due to variable delay

- B. discards in the interface of routers and switches
- C. video loss due to video frame corruption
- D. gaps in the voice due to packet loss

**Answer: A**

#### NEW QUESTION 4

Refer to the exhibit.



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

- ☒ 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 B
- C. Option C
- D. Option D

**Answer: C**

#### NEW QUESTION 5

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

**Answer: A**

#### NEW QUESTION 6

Refer to the exhibit.

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

> set type=SRV
> _collab-edge._tcp.example.com
Server:  dns.example.com
Address:  192.168.100.1

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

You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does not register to cisco Unified Communications Manager while outside of the office. What is a cause of this issue?

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

**Answer:** B

#### NEW QUESTION 7

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:** A

#### Explanation:

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 8

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 stripping or translating the called numbers to internally used directory numbers.
- D. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.

**Answer:** C

#### NEW QUESTION 9

Which two configuration elements are part of the Cisco UCM toll-fraud prevention?(Choose two.)

- A. feature control policy
- B. partition
- C. SIP trunk security profile
- D. SUBSCRIBE Calling Search Space
- E. Calling Search Space

**Answer:** AE

#### Explanation:

The following are the configuration elements that are part of the Cisco UCM toll-fraud prevention:

- Feature control policy - This policy controls the features that are available to users. For example, you can use this policy to prevent users from making international calls.
- Calling Search Space - This space defines the numbers that users can call. For example, you can use this space to prevent users from calling premium-rate numbers.

#### NEW QUESTION 10

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

- A. 323
- B. SIP



- C. SCCP
- D. MGCP
- E. RTP

**Answer:** BC

**Explanation:**

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 10**

An engineer configures a SIP trunk for MWI between a Cisco UCM cluster and Cisco Unity Connection. The Cisco UCM cluster fails to receive the SIP notify messages. Which two SIP trunk settings resolve this issue? (Choose two.)

- A. accept out-of-dialog refer
- B. accept out-of-band notification
- C. transmit security status
- D. allow changing header
- E. accept unsolicited notification

**Answer:** AE

**NEW QUESTION 12**

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 traes to examine the situation. The traces show:

```
50805567.000 |07:35:39.676 |Sdl Sig |StationOoutputDisplayNotify |restart0
|StaatinD(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. class of service configuration
- B. region configuration
- C. geolocation configuration
- D. codec configuration

**Answer:** B

**NEW QUESTION 16**

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:** D

**NEW QUESTION 17**

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

**NEW QUESTION 22**

Which command must be defined before an administrator changes the linecode value on an ISDN T1 PRI in slot 0/2 on an IOS-XE gateway?

- A. isdn incoming-voice voice
- B. pri-group timeslots 1-24
- C. card type t1 0 2

D. voice-port 0/2/0:23

**Answer:** C

#### NEW QUESTION 27

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

#### NEW QUESTION 32

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

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

**Answer:** BC

#### NEW QUESTION 33

What is set when using COS to mark an Ethernet frame?

- A. Ipp bits
- B. IP ECN bits
- C. DCSP bits
- D. 802.1 p User Priority bits

**Answer:** D

#### Explanation:

When using COS to mark an Ethernet frame, the 802.1 p User Priority bits are set. These bits are used to indicate the priority of the frame. The higher the priority, the more likely the frame is to be transmitted first.

#### NEW QUESTION 38

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

**Answer:** A

#### NEW QUESTION 39

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. 200 ok
- C. SIP Register
- D. TCP port 5060

**Answer:** A

#### NEW QUESTION 40

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. 750 and 15.000 users
- B. 750 and 10.000 users
- C. 500 and 10.000 users
- D. 1000 and 10.000 users

**Answer:** D

#### NEW QUESTION 42

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

- A. SSO

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

**Answer:** A

#### NEW QUESTION 43

Which Webex Calling construct is used to organize calling features within a physical site?

- A. client settings
- B. locations
- C. service settings
- D. call routing

**Answer:** B

#### Explanation:

A location is a physical site that contains users, devices, and resources. Locations are used to organize calling features within a physical site. For example, you can create a location for each of your offices and then assign users, devices, and resources to that location. This will allow you to manage calling features for each office separately.

#### NEW QUESTION 45

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 802.10 frames, tagged with the voice VLAN ID 221?

- A. Device(config-if)# switchport voice vlan 221
- B. Device(config-if)# switchport vlan voice 221
- C. Device(config-if)# switchport access vlan 221
- D. Device(config-if)# switchport trunk allowed vlan 221

**Answer:** A

#### NEW QUESTION 47

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. UDP161
- B. TCP 161
- C. UDP 162
- D. TCP 80

**Answer:** C

#### NEW QUESTION 48

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

**Answer:** C



#### NEW QUESTION 52

An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

- A. linecode ami
- B. linecode b8zs
- C. linecode hdb3
- D. linecode esf

**Answer:** A

#### NEW QUESTION 54

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

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

**Answer:** D

#### NEW QUESTION 57

Refer to the exhibit.

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

An engineer configures a VoIP dial peer on a Cisco gateway. Which codec is used?

- A. G711alaw
- B. No codec is used (missing codec command)
- C. G.711ulaw
- D. G729r8

**Answer:** D

#### NEW QUESTION 61

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

**Answer:** B

#### NEW QUESTION 62

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

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

**Answer:** AD

#### NEW QUESTION 65

A collaboration engineer adds a voice gateway to Cisco UCM. The engineer creates a new gateway device in Cisco UCM. selects VG320 as the device type and selects MGCP as the protocol What must be done next to add the gateway to the Cisco UCM database?

- A. Select the DTMF relay type for the gateway.
- B. Select a device pool for the new gateway.
- C. Add the FQDN or hostname of the device.
- D. Configure the module in slot 0 of the new gateway.

**Answer:** C

#### NEW QUESTION 66

A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet Link with a bandwidth of 160 kb to the Internet Telephony service provider. Which set of commands allows the engineer to complete the task without compromising voice quality?

A)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec aacld
```

B)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec ilbc mode 20
```

C)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
codec mp4a-latm
```

D)

```
dial-peer voice 1 voip
translation-profile outgoing Strip9
max-conn 5
destination-pattern 91[2-9].[2-9]...$
session protocol sipv2
session target ipv4:142.45.10.1
dtmf-relay rtp-nte sip-notify sip-kpml
```

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

**Answer: B**

#### NEW QUESTION 71

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 in the event of an outag
- B. Active/standby mode provides an always on standbynode in the event of an outage, and it also provides load balancing.
- C. Balanced mode provides user load balancing and user failover only for manually generated failovers.Active/standby mode provides anunconfigured standby node in the event of an outage, but it does not provide load balancing.
- D. Balanced mode provides user load balancing and user failover in the event of an outag
- E. Active/standby mode provides an always on standbynode in the event of an outage, but it does not provide load balancing.
- F. Balanced mode does not provide user load balancing, but it provides user failover in the event of an outag
- G. Active/standby mode provides analways on standby node in the event of an outage, but it does not provide load balancing.

**Answer: C**

#### Explanation:

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.

Here is a more detailed explanation of the two modes:

➤ **Balanced mode:** In balanced mode, the IM and Presence Service nodes are configured to work together to provide high availability. The nodes are configured in a redundancy group, and the system automatically balances the load of users across the nodes in the group. If one of the nodes fails, the system automatically fails over the users to the other nodes in the group.

➤ **Active/standby mode:** In active/standby mode, one of the IM and Presence Service nodes is designated as the active node, and the other nodes are designated as standby nodes. The active node handles all of the user traffic, and the standby nodes are only used if the active node fails. If the active node fails, the system automatically fails over to one of the standby nodes.

#### NEW QUESTION 76

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**

#### Explanation:

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 77

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:** C

#### NEW QUESTION 81

An engineer with ID012345678 must build an international dial plan in Cisco UCM. Which action is taken when building a variable-length route pattern?

- A. configure single route pattern for international calls
- B. set up all international route patterns to 0.!
- C. reduce the T302 timer to less than 4 seconds
- D. create a second route pattern followed by the # wildcard

**Answer:** D

#### Explanation:

When building a variable-length route pattern, you need to create a second route pattern followed by the # wildcard. This will allow the user to indicate the end of the number by dialing #. For example, if you want to create a route pattern for international calls, you would create a route pattern like this: 9.011!#

This route pattern will match any number that starts with 9.011, followed by any number of digits, and then ends with #.

The other options are incorrect because:

- Configuring a single route pattern for international calls will not allow the user to indicate the end of the number.
- Reducing the T302 timer to less than 4 seconds will not allow the user to indicate the end of the number.

#### NEW QUESTION 86

Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating? (Choose two.)

- A. software MTP on Cisco IOS Software
- B. software MTP on Cisco UCM
- C. software transcoder on Cisco UCM
- D. hardware transcoder on Cisco IOS Software
- E. hardware MTP on Cisco IOS Software

**Answer:** BE

#### NEW QUESTION 91

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

- 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. Use 802, 1Q trunking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
- C. Implement IP RTP header compression on the serial Interface to reduce the bandwidth required per voice call on point-to-point links.
- D. 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.
- E. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.

**Answer:** AB

#### NEW QUESTION 95

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

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

**Answer:** B

#### NEW QUESTION 99

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 secure PSTN calls by Cisco UCM only while on-premises authentication.
- C. MRA is used to make B2B calls through Expressway registration.
- D. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections

**Answer:** D

#### NEW QUESTION 100

Refer to the exhibit.

```
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
```

Given this "debug isdn q921" output, what is the problem with the PRI?

- A. Nothing, the PRI is sending keepalives.
- B. Layer 2 is down on the controller.
- C. PRI does not have an IP address configured on the interface.
- D. Layer 1 is down on the controller.

**Answer: B**

#### NEW QUESTION 101

Which external DNS SRV record must be present for Mobile and Remote Access?

- A. \_cisco-uds.Jcp.example.com
- B. \_collab-edge.\_tls.example.com
- C. \_collab-edge.\_tcp.example.com
- D. \_cisco-uds.\_tls.example.com

**Answer: B**

#### NEW QUESTION 104

An administrator uses the Cisco Unified Real-Time Monitoring Tool to investigate recent calls on a Cisco UCM cluster. The SIP trace for an on-net. direct-media call shows two 180 Ringing and two 11 BYE messages. Why are there multiples of each message type in the trace?

- A. The source phone sends a 180 Ringing signal to the Cisco UC
- B. which sends a 180 Ringing signal to the destination phon
- C. The same process applies to 11 BYE messages.
- D. The source phone must signal to the destination phone that it is ringing, and the destination phone signals back with a 180 Ringing messag
- E. The same process applies to 11 BYE messages.
- F. The calls have an MTP in the call path due to different codec suppor
- G. The calls are subsequently split into two call legs.
- H. The destination phone signals back to the Cisco UCM that it is ringing, and the Cisco UCM signals back to the source phone.

**Answer: A**

#### NEW QUESTION 106

Which configuration on Cisco UCM is required for SIP MWI to work?

- A. Assign an MWI extension on the mailbox.
- B. The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
- C. The line partition must be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
- D. Set the "Enable message waiting indicator" on the port group.

**Answer: B**

#### Explanation:

The line partition must be inside the inbound CSS assigned to the CUC SIP trunk. This ensures that the SIP MWI messages are sent to the correct destination. The other options are incorrect because:

- Assigning an MWI extension on the mailbox is not required for SIP MWI to work.
- The line partition does not need to be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
- Setting the "Enable message waiting indicator" on the port group is not required for SIP MWI to work.

#### NEW QUESTION 109

A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

- A. SIP003096D2D5CB.cnf.xml
- B. SEP003096D2D5CB.cnf.xml
- C. SEP003096D2D5CB.cnf
- D. SIP003096D2D5CB.cnf

**Answer: B**

#### NEW QUESTION 111

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:** C

#### NEW QUESTION 113

Where in Cisco UCM are codec negotiations configured for endpoints?

- A. under device profiles in device settings
- B. in in-service parameters
- C. under regions using preference lists
- D. in enterprise parameters

**Answer:** C

#### NEW QUESTION 115

An administrator is in the process of moving Cisco Unity Connection mailboxes between mailbox stores. The administrator notices that some mailboxes have active Message Waiting Indicators. What happens to these mailboxes when they are moved?

- A. The move will fail if MWI status is active.
- B. The MWI status is retained after a mailbox is moved from one store to another.
- C. If the source and target mailbox store are not disabled, MWI status is not retained.
- D. Moving the mailboxes from one store to another fails if MWI is turned on.

**Answer:** B

#### NEW QUESTION 118

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

- A. Expressway-C provides encryption (or Mobile and Remote Access but not (or business-to-business communications.
- B. The Expressway-C and Expressway-E pair can enable connectivity from the corporate network to the PSTN via a T1/E1 trunk.
- C. The Expressway-C and Expressway-E pair can interconnect H.323-to-SIP calls for voice.
- D. Expressway-E provides a VPN entry point for Cisco IP phones with a Cisco AnyConnect client using authentication based on certificates.
- E. Expressway-E provides a perimeter network that separates the enterprise network from the Internet.

**Answer:** CE

#### NEW QUESTION 121

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

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

**Answer:** B

#### NEW QUESTION 125

A company wants to provide remote users with access to its on-premises Cisco collaboration features. Which components are required to enable Cisco Mobile and Remote Access for the users?

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

**Answer:** C

#### NEW QUESTION 128

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. Enforce ad-hoc conference restrictions.
- B. Set the appropriate service parameter.
- C. Implement time-of-day routing.
- D. Use the correct route filters.

**Answer:** B

#### Explanation:

To restrict off-net to off-net call transfers, an administrator can set the "Block Offnet to Offnet Transfer" service parameter to "On". This will prevent users from transferring calls from one external number to another external number.

The other options are not correct because:

- A. Enforce ad-hoc conference restrictions: This will prevent users from creating ad-hoc conferences, but it will not prevent them from transferring calls.
- C. Implement time-of-day routing: This will allow calls to be routed to different destinations based on the time of day, but it will not prevent users from transferring calls.
- D. Use the correct route filters: This will allow calls to be filtered based on the destination, but it will not prevent users from transferring calls.



#### NEW QUESTION 130

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: C**

#### NEW QUESTION 133

Refer to the exhibit.

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

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?

- 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 matched and the translated number is 025553431234.
- D. The translation rule is not matched because DNIS contains "-".

**Answer: B**

#### NEW QUESTION 134

Users dial a 9 before a 10-digit phone number to make an off-net call All 11 digits are sent to the Cisco Unified Border Element before going out to the PSTN The PSTN provider accepts only 10 digits. Which configuration is needed on the Cisco Unified Border Element for calls to be successful?

- A. voice translation-rule 1 rule 1 /^9/ //
- B. voice translation-rule 1 rule 1 /^9(.....)/ //
- C. voice translation-rule 1 rule 1 /^9.+/ //
- D. voice translation-rule 1 rule 1 /^9...../ //

**Answer: A**

#### NEW QUESTION 139

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**

#### NEW QUESTION 143

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. allow changing header
- C. accept unsolicited notification
- D. enable application-level authorization
- E. accept replaces header

**Answer: CE**

#### NEW QUESTION 144

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: AD**

#### NEW QUESTION 147

Which Webex Calling dial plan settings restrict a user from placing a particular outbound call type?

- A. Block
- B. Transfer to Number
- C. Reject
- D. Restrict

**Answer:** D

**Explanation:**

The Restrict setting in the Webex Calling dial plan prevents users from placing certain types of outbound calls. For example, you can use the Restrict setting to prevent users from making international calls or calls to premium-rate numbers.

The Block setting in the Webex Calling dial plan prevents users from placing any outbound calls. The Transfer to Number setting in the Webex Calling dial plan transfers all outbound calls to a specified number. The Reject setting in the Webex Calling dial plan rejects all outbound calls.

Here is a table summarizing the different dial plan settings and their effects:

Dial Plan Setting Effect

Block

Prevents users from placing any outbound calls. Transfer to Number

Transfers all outbound calls to a specified number. Reject

Rejects all outbound calls. Restrict

Prevents users from placing certain types of outbound calls.

**NEW QUESTION 148**

An engineer must 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:** B

**Explanation:**

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 152**

How many minutes does it take for automatic fallback to occur in a Presence Redundancy Group if the primary node lost a critical service?

- A. 5 min
- B. 10 min
- C. 30 min
- D. 60 min

**Answer:** C

**NEW QUESTION 155**

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. device pool
- C. CSS
- D. route pattern

**Answer:** B

**NEW QUESTION 159**

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

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

**Answer:** A

**NEW QUESTION 162**

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

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

**Answer:** D

#### NEW QUESTION 167

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

- A. It creates a trust relationship with the called party.
- 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. If TRP is allocated and MTP is also required for the endpoint
- E. calls fail.

**Answer: C**

#### NEW QUESTION 172

A customer reports that the Cisco UCM toll-fraud prevention does not work correctly, and the customer is receiving charges for unapproved international calls as a result. Which two configuration changes resolve the issues? (Choose two.)

- A. Mark patterns as off-net or on-net.
- B. Modify the Block OffNet to OffNet Transfer service parameter.
- C. Disable call forwarding on the phone.
- D. Use Cisco Unified Border Element to debug the calls.
- E. Make the calls route through a firewall.

**Answer: AB**

#### NEW QUESTION 176

Which action prevent toll fraud in Cisco Unified Communication Manager?

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

**Answer: A**

#### NEW QUESTION 181

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
```

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

**Answer: BD**

#### NEW QUESTION 184

When a call is delivered to a gateway, the calling and called party number must be adapted to the PSTN service requirements of the trunk group. If a call is

destined locally, the + sign and the explicit country code must be replaced with a national prefix. For the same city or region, the local area code must be replaced by a local prefix as applicable. Assuming that a Cisco UCM has a SIP trunk to a New York gateway (area code 917). which two combinations of solutions localize the calling and called party for a New York phone user? (Choose two.)

- A. Configure two calling party transformation patterns:  
    \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber  
    \+1.!, strip pre-dot, numbering type: national
- B. Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling nu  
    !  
    voice translation-rule 1  
    rule 1 /^1917/ //  
    rule 2 /^[+]1917/ //  
    !  
    voice translation-profile strip+1  
    translate called 1  
    !
- C. Configure the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called nu  
    !  
    voice translation-rule 1  
    rule 1 /^1917/ //  
    rule 2 /^[+]1917/ //  
    !  
    voice translation-profile strip+1  
    translate calling 1  
    !
- D. Configure two called party transformation patterns:  
    \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber  
    \+1.!, strip pre-dot, numbering type: national
- E. Configure two calling party transformation patterns:  
    \+1917.CCCCCC, strip pre-dot, numbering type: subscriber  
    \+!, strip pre-dot, numbering type: national

**Answer:** BC

#### NEW QUESTION 188

An administrator needs to help a 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 UCM?

- A. Set service parameter 'Advanced Ad Hoc Conference' to FALSE.  
B. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves."  
C. Set service parameter "Advanced Ad Hoc Conference" to 2.  
D. Set service parameter "Drop Ad Hoc Conference" to "Do not allow outside parties."

**Answer:** B

#### NEW QUESTION 190

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

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

**Answer:** D

#### NEW QUESTION 191

Refer to the exhibit.



**Outbound Calls**

Called Party Transformation CSS

< None >

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

< None >

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*

Originator

Calling Line ID Presentation\*

Default

Calling Name Presentation\*

Default

Calling and Connected Party Info Format\*

Deliver DN only in connected party

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

< None >

☒ Use Device Pool Redirecting Party Transformation CSS

**Caller Information**

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*

Originator

Calling Line ID Presentation\*

Default

Calling Name Presentation\*

Default

Calling and Connected Party Info Format\*

Deliver DN only in connected party

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

< None >

☒ Use Device Pool Redirecting Party Transformation CSS

**Caller Information**

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

Unanswered calls do not reach the voicemail associated with the phones. Instead, callers receive the default greeting. Which action fixes the configuration?

- A. Reboot Cisco Unity Connection.
- B. Check the box "Redirecting Diversion Header Delivery - Outbound", then reset the trunk.
- C. Check the box "Redirecting Diversion Header Delivery - Outbound".
- D. Review the conversation manager logs on Cisco Unity Connection.

**Answer: B**

#### NEW QUESTION 194

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: C**

#### NEW QUESTION 195

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

**Answer: D**

#### NEW QUESTION 198

Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

- A. \_cisco-uds.\_tls.cisco.com pointing to the IP address of Cisco UCM
- B. \_cuplogin\_tcp.cisco.com pointing to a record of IM and Presence
- C. \_cuplogin.\_tls.cisco.com pointing to the IP address of IM and Presence
- D. \_cisco-uds.tcp.cisco.com pointing to a record of Cisco UCM
- E. \_xmpp.tls.cisco.com pointing to a record of IM and Presence



**Answer:** BD

#### NEW QUESTION 203

An engineer configures Cisco UCM to prevent toll fraud. At which two points does the engineer block the pattern in Cisco UCM to complete this task? (Choose two.)

- A. partition
- B. route partem
- C. translation pattern
- D. CSS
- E. route group

**Answer:** AD

#### NEW QUESTION 207

What is the function of the Cisco Unity Connection Call Handler?

- A. routes calls to a user based on caller input
- B. queues calls
- C. allows customized scripts for IVR capabilities
- D. searches a list of extensions until the call is answered

**Answer:** A

#### Explanation:

A Cisco Unity Connection Call Handler is a software application that answers calls, plays greetings, and routes calls to users based on caller input. Call handlers can be used to create automated attendants, voice menus, and other interactive voice response (IVR) applications.

Call handlers are created and managed using the Cisco Unity Connection Administration interface. When creating a call handler, you can specify a variety of settings, including the greeting that is played, the caller input options that are available, and the destination that calls are routed to.

Call handlers are a powerful tool that can be used to create a variety of IVR applications. By using call handlers, you can improve the efficiency of your organization's communications and provide a better experience for your callers.

Here are some additional tips for using call handlers:

- Use call handlers to create automated attendants that can answer calls and route them to the appropriate person or department.
- Use call handlers to create voice menus that can provide callers with information or options.
- Use call handlers to create interactive voice response (IVR) applications that can collect information from callers and process their requests.

#### NEW QUESTION 212

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

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

**Answer:** C

#### NEW QUESTION 213

Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
voice translation-rule 40
 rule 1 /3...$/ /408555&/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

A.

**Pattern Definition**

Pattern \*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

---

**Called Party Transformations**

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type \*

Called Party Numbering Plan \*

B.

**Pattern Definition**

Pattern \*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

---

**Called Party Transformations**

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type \*

Called Party Numbering Plan \*

C.

**Pattern Definition**

Pattern \*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

---

**Called Party Transformations**

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type \*

Called Party Numbering Plan \*

D.

Pattern Definition	
Pattern *	\+.[1
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> MLPP Preemption Disabled	
Called Party Transformations	
Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type *	Unknown
Called Party Numbering Plan *	Unknown

Answer: C

#### NEW QUESTION 216

An administrator installed a Cisco Unified IP 8831 Conference Phone that is failing to register. Which two actions should be taken to troubleshoot the problem? (Choose two.)

- A. Verify that the switch port of the phone is enabled.
- B. Verify that the RJ-11 cable is plugged into the PC port.
- C. Disable HSRP on the access layer switch.
- D. Check the RJ-65 cable.
- E. Verify that the phone's network can access the option 150 server.

Answer: AE

#### NEW QUESTION 220

Refer to the exhibit.

```
Voice class codec 1
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g729r8

dial-peer voice 13 voip
description incoming dialpeer from ITSP
incoming called-number .
voice-class codec 1

dial-peer voice 19 voip
description outgoing dialpeer to CUCM
destination-pattern T
session protocol sipv2
session-target ipv4.3.3.3
voice-class codec 1

Incoming SDP from ITSP

v=0
o=sip:test@2.2.2.2 1 16 IN IP4 2.2.2.2
s=sip:test@2.2.2.2
c=IN IP4 2.2.2.2
t=0 0
m=audio 5000 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

Which outgoing m-line SDP is sent to Cisco UCM after matching the appropriate dial peers via Cisco Unified Border Element?

- A. m=audio 16550 RTP/AVP 8 0 18  
a=rtpmap:0 PCMU/8000/1  
a=rtpmap:8 PCMA/8000/1  
a=rtpmap:18 G729/8000/1
- B.

- m=audio 16550 RTP/AVP 18 0  
a=rtpmap:0 PCMU/8000/1  
a=rtpmap:18 G729/8000/1
- C. m=audio 16550 RTP/AVP 18 0  
a=rtpmap:8 PCMA/8000/1  
a=rtpmap:0 PCMU/8000/1  
a=rtpmap:18 G729/8000/1
- D. m=audio 16550 RTP/AVP 0 8 18  
a=rtpmap:0 PCMU/8000/1  
a=rtpmap:8 PCMA/8000/1  
a=rtpmap:18 G729/8000/1

**Answer:** B

#### NEW QUESTION 222

An administrator is configuring LDAP for Cisco UCM with Active Directory integration. A customer has requested to use "ipphone" instead of "telephoneNumber" as the phone number attribute. Where does the administrator specify this attribute mapping in Cisco UCM?

- A. LDAP Custom Filter
- B. LDAP Directory user fields
- C. LDAP Directory custom user fields
- D. LDAP Authentication

**Answer:** B

#### NEW QUESTION 226

Refer to the exhibit.

```
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
```

The SDP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

- A. Payload type 110 was negotiated rather than type 101.
- B. DTMF was negotiated property in these messages.
- C. DTMF was not negotiated on the call.
- D. G.729 rather than G.711ulaw was negotiated.

**Answer:** C

#### NEW QUESTION 231



A Cisco Telepresence SX80 suddenly has issues displaying main video to a display over HDMI. Which command can you use from the SX80 admin CLI to check the video output status to the monitor?

- A. xStatus HDMI Output
- B. xStatus video Output
- C. xconfiguration video Output
- D. xcommand video status

**Answer:** B

#### NEW QUESTION 232

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

**Answer:** DE

#### NEW QUESTION 237

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

**Answer:** B

#### NEW QUESTION 240

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

**Answer:** AC

#### NEW QUESTION 242

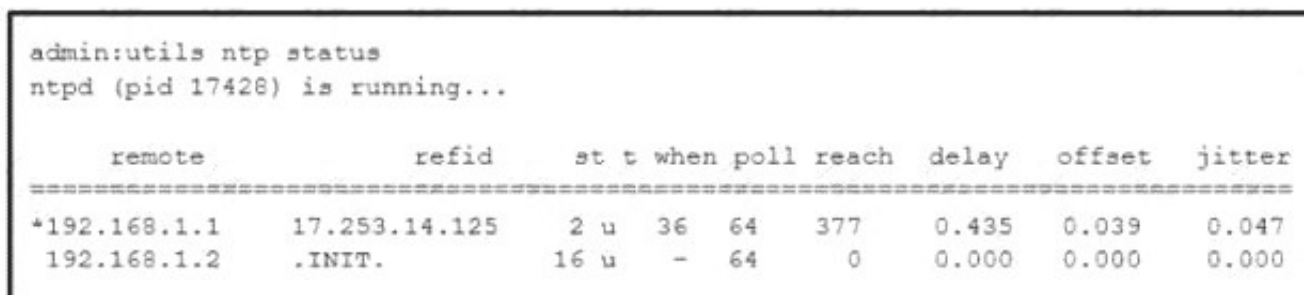
Which behavior occurs when Cisco UCM has a Call Manager group that consists of two subscribers?

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

**Answer:** A

#### NEW QUESTION 244

Refer to the exhibit.



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 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?

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

**Answer:** C

#### NEW QUESTION 247

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



- A. Provide a guaranteed bandwidth of 10 Mbps.
- B. Deploy the VMs to a server running Cisco UCM.
- C. Avoid hardware oversubscription.
- D. Ensure that applications will perform QoS.

**Answer: C**

**Explanation:**

When deploying co-resident VMs by using Cisco UCM, it is important to avoid hardware oversubscription. This means that you should not assign more resources to the VMs than the physical hardware can provide. For example, if you have a server with 16 CPU cores, you should not assign more than 16 CPU cores to the VMs.

If you oversubscribe the hardware, the VMs will not be able to get the resources they need to run properly. This can lead to performance problems and even outages.

To avoid hardware oversubscription, you should carefully plan your VM deployments. You should also monitor the performance of the VMs to make sure that they are not overusing the resources.

Here are some additional tips for deploying co-resident VMs by using Cisco UCM: ➤ Use a virtualization platform that supports Cisco UCM.

- Make sure that the VMs have the correct operating system and software installed.
- Configure the VMs to use the correct network settings.
- Monitor the performance of the VMs to make sure that they are running properly.

**NEW QUESTION 248**

Refer to the exhibit.

```
000142: *Apr 23 19:41:49.050: MGCP Packet received from 192.168.100.100:2427--->
AUEP 4 AALN/S0/SU0/0@VG320.cisco.local MGCP 0.1
F: X, A, I
<---

000143: *Apr 23 19:41:49.050: MGCP Packet sent to 192.168.100.101:2427--->
200 4
I:
X: 2
L: p:10-20, a:PCMU:PCMA:G.nX64, b:64, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-220, a:G.729:G.729a:G.729b, b:8, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-110, a:G.726-16:G.728, b:16, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-70, a:G.726-24, b:24, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-50, a:G.726-32, b:32, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-270, a:G.723.1-H:G.723:G.723.1a-H, b:6, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-330, a:G.723.1-L:G.723.1a-L, b:5, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, contest, data, netwloop, netwtest
<---
```

What is the registration state of the analog port in this debug output?

- A. The analog port failed to register to Cisco UCM with an error code 200.
- B. The MGCP Gateway is not communicating with the Cisco UCM.
- C. The analog port is currently shut down.
- D. The analog port is registered to Cisco UCM.

**Answer: D**

**NEW QUESTION 253**

What happens to voice packets from a Cisco 8845 IP phone in the QoS trust boundary?

- A. The voice packets are not trusted, and the access layer switch reclassifies the packets.
- B. The voice packets are classified by the phone, and the classification is accepted
- C. The voice and access layer switch negotiate the classification of packets
- D. Cisco UCM determines how the voice packers are classified.

**Answer: B**

**NEW QUESTION 258**

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

**Answer: CD**

**NEW QUESTION 259**

Which DHCP option must be set up tor new phones to obtain the TFTP server IP address?

- A. option 66
- B. option 15
- C. option 6
- D. option 120

**Answer: A**

#### NEW QUESTION 262

Refer to the exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD31SE7
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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

This INVITE is sent to an endpoint that only supports G.729. What must be done for this call to succeed?

- 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

#### NEW QUESTION 263

An engineer is deploying the Webex app on a Microsoft Windows computer that has multiple user accounts. Which CLI argument allows the engineer to install the application as a "per machine" installation?

- A. ACCEPT\_EULA
- B. INSTALL\_ROOT
- C. ALLUSERS
- D. FORCELOCKDOWN

**Answer:** C

#### Explanation:

The ALLUSERS property is a command-line argument that can be used to install the Webex app as a "per machine" installation. This means that the app will be installed for all users on the computer, regardless of which user account is currently logged in.

The ACCEPT\_EULA property is a command-line argument that can be used to accept the end-user license agreement (EULA) for the Webex app. The INSTALL\_ROOT property is a command-line argument that can be used to specify the installation directory for the Webex app. The FORCELOCKDOWN property is a command-line argument that can be used to prevent users from uninstalling the Webex app.

#### NEW QUESTION 265

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:** C

#### NEW QUESTION 270

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

**Answer:** C

#### NEW QUESTION 272

In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)

- A. VPN-based enterprise access for a subset of Cisco Unified IP Phone models
- B. secure reverse proxy firewall traversal connectivity
- C. the ability to register third-party SIP or H.323 devices on Cisco UCM without requiring VPN
- D. the ability of Cisco IP Phones to access the enterprise through VPN connection
- E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

**Answer:** BE

#### NEW QUESTION 276

Users want their mobile phones to be able to access their Cisco Unity connection mailboxes with only having to enter their voicemail pin at the login prompt calling pilot number. Where should an engineer configure this feature?

- A. transfer rules
- B. message settings
- C. alternate extensions
- D. greetings

**Answer:** C

#### NEW QUESTION 280

Which two types of trunks can be used when configuring a hybrid Local Gateway for Cisco Webex Calling? (Choose Two.)

- A. TLS-based
- B. certificate-based
- C. registration-based
- D. authentication-based
- E. OAuth-based

**Answer:** AC

#### Explanation:

These are the two types of trunks that can be used when configuring a hybrid local gateway for Cisco Webex Calling. 1. A TLS-based trunk uses Transport Layer Security (TLS) to secure the SIP signaling between the hybrid local gateway and Webex Calling. 2. A registration-based trunk uses SIP registration to authenticate the hybrid local gateway with Webex Calling and receive calls from the cloud.

#### NEW QUESTION 281

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)#interface Serial0/0/0:15
2900(config-controller)#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-timeslots 1-7
```
- C. 

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

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

**Answer:** C

#### NEW QUESTION 286

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:** B

#### NEW QUESTION 290

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. Reboot the VCS server and attempt reregistration.
- B. Change the SIP trunk configuration.
- C. Correct the restriction policy settings.
- D. Upload a new policy in VCS.

**Answer:** C

#### Explanation:

The error message "AOR is not permitted by Allow/Deny list" indicates that the endpoint is not allowed to register with the VCS server because it is not on the Allow List or it is on the Deny List. To resolve this problem, you must correct the restriction policy settings.

#### NEW QUESTION 295

A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?

M=audio 498181 RTP/AVP 0 8 97

- A. G.711ulaw
- B. iLBC
- C. G.711alaw
- D. G.722

**Answer:** A

#### Explanation:

The SDP offer includes the following line of text: M=audio 498181 RTP/AVP 0 8 97

This line of text indicates that the following codecs are available:

- 0: G.711ulaw
- 8: G.711alaw
- 97: iLBC

The Cisco IP Phone 7841 is registered to a Cisco Unified Communications Manager with default configuration. This means that the phone will negotiate the G.711ulaw codec.

The G.711ulaw codec is a standard codec that is used for voice communication. It is a low-bandwidth codec that provides good quality.

The iLBC codec is a newer codec that is designed for use in low-bandwidth environments. It provides good quality, but it is not as widely supported as the G.711ulaw codec.

The G.722 codec is a high-quality codec that is used for voice communication. It provides excellent quality, but it requires more bandwidth than the G.711ulaw codec.

#### NEW QUESTION 299

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## Relate Links

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