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Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

Cisco 300-815

Version Demo

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Topic Break Down

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QUESTION NO: 1

Refer to the exhibit. An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco UCM does not complete. The administrator wants to use the Dialed Number

Analyzer on Cisco UCM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability, there is no option for Dialed Number Analyzer under the tool menu.

Which two steps should be taken to resolve this issue? (Choose two.)

A. Restart the subscriber.

B. Activate the Cisco Extended Functions service.

C. Activate the Cisco CallManager service.

D. Activate the Cisco Dialed Number Analyzer service.

E. Activate the Cisco Dialed Number Analyzer Server service.

ANSWER: D E

Explanation:

Cisco Unified Communications Manager Dialed Number Analyzer requires both the **Cisco Dialed Number Analyzer** service and the **Cisco Dialed Number Analyzer Server** service to be activated. These are activated from Cisco Unified Serviceability by selecting the appropriate CUCM server and opening *Tools* > *Service Activation*. After the services are activated and running, the Dialed Number Analyzer entry becomes available from the Unified Serviceability Tools menu.

The Cisco Dialed Number Analyzer service provides the web-based analyzer interface used by an administrator to enter the calling and called-party information, select a device, and run the call-routing analysis. The Cisco Dialed Number Analyzer Server service provides the server-side processing required to evaluate the digit analysis and display the call-routing result. Together, they allow the administrator to determine which translation pattern, route pattern, transformation, or other dial-plan element is matched during call processing. This is particularly useful when a registered endpoint can place a call but the call fails because the dialed digits do not reach the expected routing destination.

For Cisco configuration and service details, see the Cisco Dialed Number Analyzer Guide and the Cisco Unified Serviceability Administration Guide.

QUESTION NO: 2

```
!
dial-peer voice 100 voip
 description Outbound to CUCM
 translation-profile outgoing CUCM
 session protocol sipv2
 session target ipv4:192.168.100.200
 voice-class sip transport switch udp tcp
 voice-class sip conn-reuse
 voice-class sip rel1xx disable
 voice-class sip session refresh
 voice-class sip midcall-signaling block
 voice-class sip early-media update block
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

Refer to the exhibit. An engineer is troubleshooting an issue where inbound calls to Cisco UCM with early media fail to establish. While investigating the issue, the engineer finds that Cisco UCM is set to require a PRACK, but the Cisco Unified Border Element is not sending it. Which command is causing this issue?

A. voice-class midcall-signaling block

B. voice-class sip rel1xx disable

C. voice-class sip early-media update block

D. voice-class sip conn-reuse

ANSWER: B

Explanation:

voice-class sip rel1xx disable is the command causing the failure because it disables SIP reliable provisional-response handling on Cisco Unified Border Element. Reliable provisional responses, commonly called 100rel, are the SIP mechanism used for reliable delivery of provisional responses such as 183 Session Progress when early media is negotiated. When a SIP endpoint receives a provisional response that includes the `Require: 100rel` header and an RSeq value, it must acknowledge that response with PRACK before call setup can continue as required by the far-end policy.

With `rel1xx disable` applied, CUBE does not participate in the reliable 1xx/PRACK exchange. Therefore, when Cisco Unified Communications Manager requires PRACK for an early-media call, CUBE does not send the necessary PRACK message. The reliable provisional-response transaction cannot complete, preventing the call from progressing normally to establishment. Removing this command, or otherwise enabling compatible reliable-provisional-response support on the relevant CUBE dial peer, allows CUBE to generate PRACK when CUCM requires it. Cisco documents PRACK as the acknowledgement mechanism for reliable provisional SIP responses in its CUBE SIP Rel1xx configuration guide; SIP's reliable provisional response behavior is also defined in RFC 3262.

QUESTION NO: 3

Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco UCM? (Choose two.)

A. Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.

B. A unicast music on hold audio source must be configured.

C. Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.

D. The maximum number of callers allowed in queue must be 10.

E. The phone button template must have the Queue Status Softkey configured.

ANSWER: A C

Explanation:

Native Call Queuing depends on Cisco Unified Communications Manager services that provide the media treatment and real-time device information required to place, retain, and manage calls in queues. “Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.” is required because the IP Voice Media Streaming application provides the media-streaming capability used for call treatments such as Music On Hold while a caller is queued. The service must be available before Native Call Queuing can supply the expected queued-call experience.

“Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.” is also required. Native Call Queuing uses real-time status information to track queue and device state. Therefore, the Cisco RIS Data Collector service must be operational on each call-processing server that runs the Cisco CallManager service, allowing the queue feature to obtain the real-time information it needs while calls are being processed.

These are service-level prerequisites that should be verified in Cisco Unified Serviceability before configuring queue settings and assigning queue-related call-routing behavior. Cisco documents the prerequisite services as part of its Native Call Queuing configuration guidance. See the Cisco Unified Communications Manager System Configuration Guide and the Cisco Unified Communications Manager System Configuration Guide, Release 14SU1.

QUESTION NO: 4

An engineer must implement call restriction to toll-free numbers using a class of restriction in a branch Cisco UCME. In which two places is the corlist incoming or cor incoming command configured?

(Choose two.)

A. “voice register pool ” configuration mode

B. “ephone-dn ” configuration mode

C. “dial-peer cor custom ” configuration mode

D. “voice register global ” configuration mode

E. “telephony-service ” configuration mode

ANSWER: A B

Explanation:

In Cisco Unified CME, an incoming class-of-restriction assignment is applied at the endpoint or directory-number level so that CME can associate an originating station or line with a COR list before it selects an outbound dial peer. The `cor incoming` command is configured under `ephone-dn` for SCCP endpoints. This associates the COR list with the individual directory number, allowing calling privileges to be controlled on a per-line basis.

For SIP endpoints registered through Cisco Unified CME’s voice-register configuration, the equivalent incoming COR association is configured under `voice register pool`. A voice-register pool represents an individual SIP phone, and the configured COR list determines the calling level applied to calls originating from that registered device. CME then compares the incoming COR privileges with the COR requirements assigned to destination dial peers, preventing calls to destinations such as toll-free numbers when the required privilege is absent.

This placement supports consistent restriction policy for both SCCP and SIP phones while retaining the normal COR model of matching an originating privilege list to outbound dial-peer restrictions. Cisco’s CME documentation and IOS voice command references are available at Cisco Unified CME configuration guides and Cisco IOS voice command reference.

QUESTION NO: 5

Refer to the exhibit.

An engineer is troubleshooting an issue with the caller not hearing a PSTN announcement before the SIP call has completed setup. How must the engineer resolve this issue using the reliable provisional response of the SIP?

A. voice service voip sip send 180 sdp

B. voice service voip sip rehxx require 100rel

C. sip-uadisable-early-media 180

D. voice service voip sip no reMxx

ANSWER: B

Explanation:

The required configuration is `voice service voip sip rehxx require 100rel`, where the command keyword is intended to represent Cisco IOS XE's `rel1xx require 100rel` setting. Reliable provisional responses allow a SIP endpoint or gateway to send provisional signaling, typically a `183 Session Progress` response containing SDP, reliably before the final call-answer response. This is essential when the PSTN supplies an in-band announcement, such as an intercept message, network treatment, or recorded status announcement, during early call progress. The SDP establishes the early-media stream so that the caller can hear that announcement before the call is connected.

Requiring the SIP 100rel extension causes reliable handling of provisional responses through the PRACK mechanism. The receiving SIP side acknowledges the reliable provisional response with PRACK, preventing the provisional response and its early-media negotiation from being lost or handled inconsistently. This provides a dependable early-media path between the PSTN-facing gateway and the SIP peer while final call setup is still pending. SIP reliable provisional-response behavior and PRACK are defined in RFC 3262. Cisco CUBE SIP configuration and command behavior are documented in the Cisco Unified Border Element Configuration Guide.

QUESTION NO: 6

A company is using third-party phones on the internal network at a location behind a Cisco Unified Border Element. These phones seem to work with only two types of codecs, which are high-complexity codecs.

Which configuration ensures that calls are completed with the correct and working media each time?

A. Configure a codec preference list on the Cisco Unified Border Element with the voice class codec command, set the codec preference list codecs, and add it to the dial peer.

B. Configure a codec priority list on the Cisco Unified Border Element with the voice class codec command, set the codec priority list codecs, and add it to the dial-peer voice.

C. Configure a codec preference list on the Cisco Unified Border Element with the VoIP class codec command, set the codec preference list codecs, and add it to the dial peer.

D. Configure a codec preference list on the Cisco Unified Border Element with the voice class codec command, set the codec priority list codecs, and add it to the VoIP peer.

ANSWER: A

Explanation:

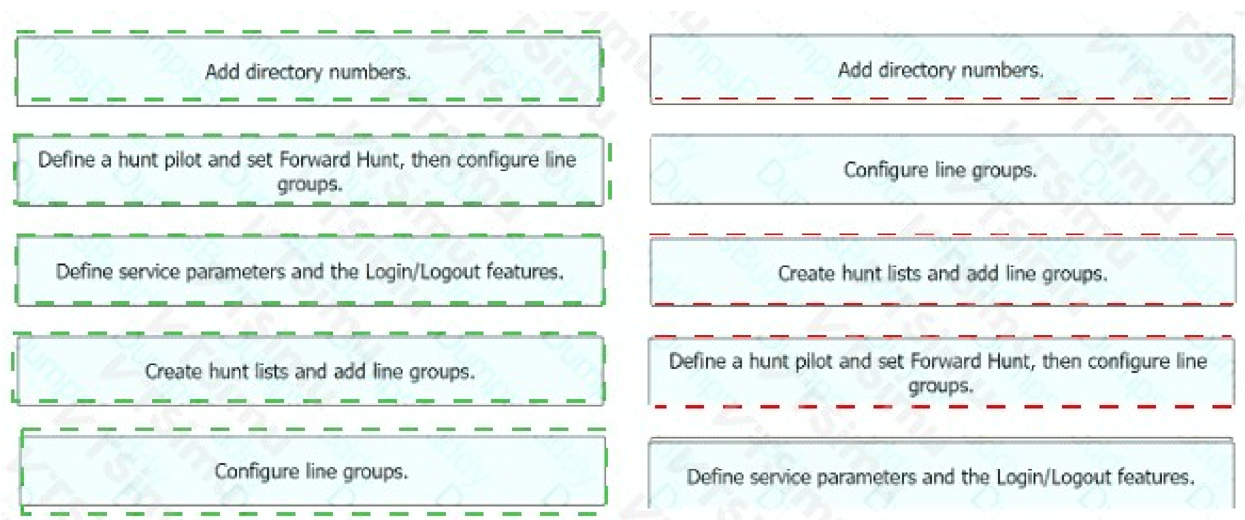
Configure a codec preference list on the Cisco Unified Border Element with the voice class codec command, set the codec preference list codecs, and add it to the dial peer. This is the appropriate approach because CUBE codec negotiation needs an explicit, ordered codec policy when an endpoint has limited codec interoperability. A `voice class codec` defines the acceptable codecs and their preference order, while applying that class to the applicable VoIP dial peer ensures the policy is used for calls matched by that peer. CUBE can then present and negotiate media using the configured codec order, selecting a mutually supported codec that the third-party phones can actually process. This produces predictable media negotiation instead of depending on the codec ordering received in signaling or on a broader default codec configuration. The codec list should contain the two known-working codecs, ordered according to the company's preferred media choice and bandwidth/quality requirements. Cisco documents codec preference configuration through a codec voice class and its application to dial peers in its CUBE configuration guidance. See the [Cisco CUBE codec negotiation documentation]() and the [Cisco IOS Voice Command Reference]().

QUESTION NO: 7 - (DRAG DROP)

A customer wants to configure their team of 10 agents to answer incoming calls to a common central number. The phone for each team member must ring in sequence and ring four times before moving to the next member's phone. Drag and drop the steps from the left into the order on the right to make this happen.

ANSWER:

Answer:



Answer:

Explanation:

The correct configuration sequence starts by adding the directory numbers for the ten team members. Those directory numbers are the actual callable endpoints that will become members of the call-distribution design. Once they exist, configure the line groups. The line group is where the member directory numbers are collected and where the distribution behavior is applied. For this requirement, the line group uses a sequential hunting algorithm, so calls progress through the members in the defined order. Its no-answer handling is also the appropriate place to ensure that a member is tried for four rings before the call proceeds to the next member.

Next, create a hunt list and add the completed line group to it. A hunt list is the ordered container used by Cisco Unified Communications Manager to determine which line group is attempted for an incoming hunt call. With the hunt list available, define the hunt pilot, configure its Forward Hunt behavior, and associate the hunt-list processing needed for calls placed to the common central number. The hunt pilot is the public-facing or internally dialed number callers use; it initiates the hunting process rather than representing an individual agent.

Finally, define the service parameters and Login/Logout features. This completes the operational hunt-group behavior by enabling the appropriate user controls and system treatment for agents who need to sign in or out of hunt-group participation. Cisco documents hunt groups as a relationship of directory numbers in line groups, line groups in hunt lists, and hunt lists reached by hunt pilots. See Cisco's Hunt Group configuration documentation and its Hunt Pilot documentation.

QUESTION NO: 8

A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified SRST gateway to support SIPphones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)

A. three-way conference

B. secure SIP lines

C. T.38 fax relay

D. transcoding

E. SIP trunk

ANSWER: A C

Explanation:

For SIP phones operating in an IPv6 Unified SRST fallback deployment, **three-way conference** and **T.38 fax relay** are supported supplementary features. Three-way conference allows a locally registered SIP endpoint to establish a conference during WAN or Cisco Unified Communications Manager connectivity loss. The SRST gateway supplies the local call-control function needed to maintain this feature while phones are in fallback mode, so users can continue basic collaborative calling without depending on the central call-processing system.

T.38 fax relay is also supported in the IPv6 fallback feature set. T.38 provides a fax-relay method in which fax information is transported as packetized T.38 data rather than relying on an unmodified audio fax signal throughout the IP network. Supporting it at the SRST gateway allows fax-capable endpoints or attached analog fax devices to continue sending and receiving faxes during fallback, subject to the applicable dial-peer, voice-service, and fax-relay configuration.

Cisco's Unified SRST documentation identifies these capabilities in its IPv6 SIP SRST feature-support information. Configuration prerequisites and platform/release-specific details should be verified against the IOS XE release used on the gateway. See Cisco's SCCP and SIP SRST Administration Guide and the Cisco Unified SRST support documentation.

QUESTION NO: 9

An IP Telephony administrator is deploying IP phones The administrator has an existing Cisco UCME router with several SCCP & SIP phones registered. The administrator receives a request for a new SIP phone with MAC address 1111 2222.3333 and directory number 2050 to be added in the Cisco UCME. Which two configurations should be added in CME to support this request? (Choose two)

A. Option A

B. Option B

C. voice register dn 1
number 2050

D. voice register pool 1
id mac 1111.2222.3333
number 1 dn 1

E. Option E

ANSWER: C D

Explanation:

For a Cisco Unified CME SIP endpoint, the directory number and the physical phone registration are configured as separate objects under the `voice register` feature. The `voice register dn 1` configuration creates a SIP directory-number entry, and `number 2050` assigns the requested extension to that entry. The tag number is locally significant and is used to associate the DN with a SIP phone pool; it does not need to match the telephone number.

The `voice register pool 1` configuration creates the phone-specific registration entry. Its `id mac 1111.2222.3333` command identifies the individual SIP phone by its MAC address. The `number 1 dn 1` command then maps the first line appearance on that phone to the previously defined directory-number tag. Together, these configurations provide both information UCME needs: a routable DN of 2050 and a uniquely identified SIP endpoint to which that DN is assigned. SIP phones in CME are provisioned through `voice register dn` and `voice register pool`, rather than SCCP `ephone-dn` and `ephone` objects. Cisco's CME documentation and command references are available from the Cisco Unified Communications Manager Express support page and the Cisco CME command reference.

QUESTION NO: 10

An administrator deployed a third-party H.323 gateway in a voice environment, but users report call failures when using features like call hold or call transfer. What are two reasons that these features fail? (Choose two.)

A. The CSS of the transfer initiating line does not contain the partition of the supplementary feature extension (DirectTransfer or MoH Number).

B. The MTP that is configured for use within the H.323 gateway configuration is configured as a trusted source, but the third-party gateway does not trust the signing root CA certificate of the MTP certificate.

C. The MTP does not support the negotiated codec, and media renegotiating during the call is not supported.

D. The Media Resource Group List of the H.323 gateway contains only transcoders and conference bridges but no MTP.

E. The third-party gateway does not support supplementary features, so Media Termination Point (MTP) must be inserted.

ANSWER: C D

Explanation:

The MTP does not support the negotiated codec, and media renegotiating during the call is not supported is correct because an MTP becomes part of the media path when Cisco Unified Communications Manager must provide supplementary-service interoperability for an H.323 gateway. The selected MTP must be able to process the codec negotiated for the call. If it cannot support that codec and the endpoint or gateway cannot renegotiate to a compatible codec when the feature is invoked, CUCM cannot establish the required media path for the hold or transfer operation.

The Media Resource Group List of the H.323 gateway contains only transcoders and conference bridges but no MTP is also correct. A transcoder and a conference bridge are distinct media resources; neither substitutes for an MTP when MTP insertion is required. Third-party H.323 gateways can require CUCM to allocate an MTP to provide supplementary-service signaling and media handling. CUCM searches the gateway-associated media-resource configuration to allocate an eligible resource. Therefore, an MRGL that has no MTP available can prevent the supplementary feature from completing. Cisco documents the role of MTPs in interworking and feature support in its MTP troubleshooting documentation and describes media-resource assignment in the Cisco Unified Communications Manager Feature Configuration Guide.

QUESTION NO: 11

When locations-based Call Admission Control denies the call, which two masks can AAR apply when routing the call through the PSTN? (Choose two.)

A. AAR destination mask

B. called party transform mask

C. external phone number mask

D. +E.164 alternate number mask

E. enterprise alternate number mask

ANSWER: A C

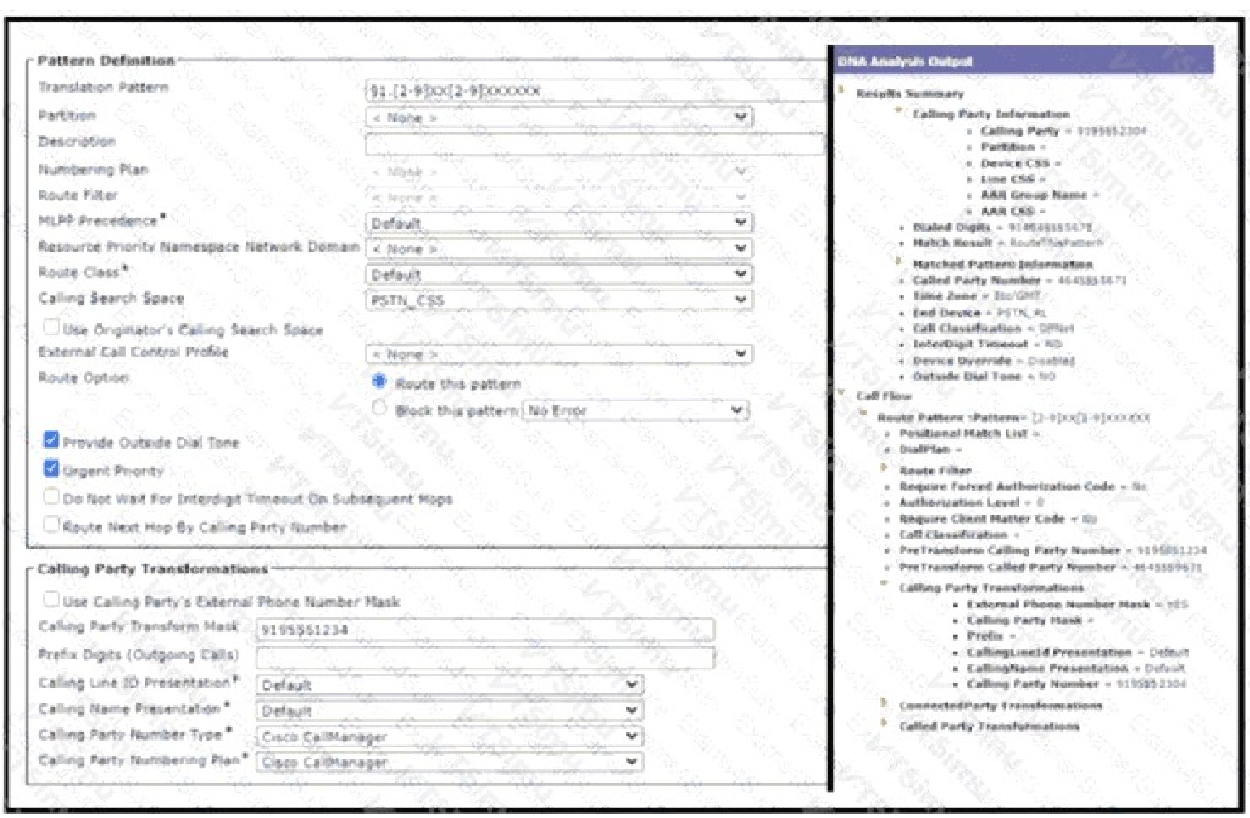
Explanation:

When location-based Call Admission Control determines that insufficient bandwidth is available for an on-net call, Cisco Unified Communications Manager can use Automated Alternate Routing to reroute that call through the PSTN. The **AAR destination mask** supplies the PSTN-routable form of the originally dialed destination. It is configured for the relevant AAR group and is used to transform the called party number into a format that can be sent to the alternate PSTN route, such as a full national or international number.

The **external phone number mask** supplies the externally reachable identity associated with a directory number. During an AAR PSTN reroute, Cisco Unified CM can use this mask to present a valid calling-party number to the PSTN, rather than an internal extension that would not be meaningful or routable outside the enterprise. Together, these masks support a usable PSTN call by providing appropriate externally formatted called-party and calling-party information. Cisco documents AAR configuration and its interaction with the external phone number mask in its dial-plan and call-routing guidance. See Cisco Collaboration SRND—Dial Plan and Cisco Unified CM System Configuration Guide—AAR.

QUESTION NO: 12

Refer to the exhibit.



For long-distance calls, users must prefix their dialed number with “91.” The translation pattern was created to strip the 91 as the PSTN expects a 10- digit number. The PSTN also requires the calling number to be set to 9195551234. However, the service provider has said calls with a different calling number are being received. How is this issue resolved?

A. Change the partition of the translation pattern from none to pstn_pt.

B. Enable Force Authorization Code on the route pattern.

C. Disable Use Calling Party's External Phone Number Mask on the route pattern.

D. Enable Use Calling Party's External Phone Number Mask on the translation pattern.

ANSWER: C

Explanation:

Disable *Use Calling Party's External Phone Number Mask* on the route pattern. A translation pattern can normalize the dialed number by removing the access and long-distance prefix, and it can also apply a calling-party transformation so that the PSTN receives the required calling number, 9195551234. However, when the route pattern is configured to use the calling party’s external phone number mask, Cisco Unified Communications Manager substitutes the external phone number mask associated with the calling device for the calling number that was transformed earlier in call processing. Consequently, the provider can receive a different caller ID than the one configured through the translation pattern’s calling-party transformation. Clearing this route-pattern setting preserves the calling number established by the translation pattern and forwards 9195551234 toward the PSTN. This ensures that the called number is sent in the required 10-digit format while the required calling number is retained. Cisco documents route-pattern digit manipulation and calling-party transformations in its CUCM administration documentation: Cisco Unified Communications Manager Administration Guide. Additional CUCM configuration documentation is available from Cisco’s CUCM installation and configuration guides.

QUESTION NO: 13

Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)

A. TokenID

B. username and secret key

C. TLS certificates

D. passwords

E. FQDN of the servers defined in DNS

ANSWER: C D

Explanation:

Intercluster Lookup Service supports certificate-based authentication and password-based authentication when establishing trust between Cisco Unified Communications Manager clusters. Therefore, **TLS certificates** and **passwords** are the supported authentication types. With certificate authentication, the clusters use certificates exchanged through their trust relationship to authenticate ILS communication securely. This approach relies on the Cisco Unified Communications Manager certificate infrastructure and is commonly used where certificate trust has been established between clusters. With password authentication, administrators configure a shared ILS password on the participating clusters; the password must match so that the clusters can authenticate the ILS connection. The authentication selection is made in the ILS configuration and applies to the relationship used for exchanging directory, route, and other advertised topology information. Cisco documentation describes the ILS authentication type as either Certificate or Password and notes that the configured method must be compatible across the connected clusters. These mechanisms authenticate the ILS peer relationship itself, helping ensure that advertised information is exchanged only with an authorized cluster. See Cisco’s System Configuration Guide and the Cisco Unified Communications Manager System Configuration Guide.

QUESTION NO: 14

An administrator is troubleshooting a one-way audio issue for a call that uses H.323 protocol in slow-start mode. The administrator requests that the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call is provided. The H.225 and H.245 messages for one of the one-way audio calls are gathered and the call flow has not invoked any media resources. Where is the RTP IP and port information for both sides found?

A. H.245 Terminal Capability Set

B. H.245 Open Logical Channel

C. H.225 Connect

D. H.245 Open Logical Channel Ack

ANSWER: B

Explanation:

H.245 Open Logical Channel is correct because H.245 is the H.323 control protocol that establishes logical media channels and communicates the transport addresses used for RTP media. In a slow-start H.323 call, signaling begins with H.225 call setup, while media-channel negotiation occurs after the H.245 control channel is established. The OpenLogicalChannel procedure carries H.225.0 logical-channel parameters, including the media transport address information used to identify the RTP destination IP address and UDP port. The OpenLogicalChannel acknowledgment completes the exchange for that unidirectional channel, so reviewing the OpenLogicalChannel transaction provides the RTP addressing information advertised by each endpoint for the media flow. This is the key information needed to compare the intended RTP source and destination details with packet captures, routing, ACLs, NAT behavior, or firewall state when investigating one-way audio. Because no media resource was invoked, there is no media-resource signaling to inspect; the endpoint-to-endpoint H.245 logical-channel negotiation is the authoritative source for the negotiated RTP transport details. The H.245 specification defines the OpenLogicalChannel and acknowledgment procedure and its H.225.0 media-channel transport parameters. See ITU-T Recommendation H.245 and RFC 882: H.323.

QUESTION NO: 15

An engineer must route all SIP calls in the form of @example.com to the SIP trunk gateway corporate local. Which two SIP route patterns can be used to accomplish this task? (Choose two.)

A. example.com@gateway.corporate.local

B. *@example.com

C. gateway.corporate.local

D. example.com

E. *.*

ANSWER: B E

Explanation:

`*@example.com` is a SIP route pattern that explicitly matches any user portion before the at sign when the SIP URI host/domain is `example.com`. Associating this pattern with the SIP trunk to `gateway.corporate.local` therefore sends every addressed URI of the required form, such as `alice@example.com` or `1001@example.com`, to that trunk. The asterisk supplies the variable user portion, while the specified domain constrains the match to the intended destination domain.

`*.*` also matches these calls because the SIP URI contains a dot in the domain portion, `example.com`. In Cisco Unified Communications Manager SIP route-pattern matching, this pattern is broad rather than domain-specific: it matches SIP URIs containing a dot and can consequently match the required `user@example.com` addresses. When it is configured to use the specified SIP trunk, it accomplishes the requested routing. Cisco recommends considering route-pattern specificity and ordering when broad wildcard patterns are used, so that more-specific destination routing remains preferred where applicable. Cisco documents SIP route patterns as the mechanism for routing SIP requests based on URI patterns and assigning the resulting route to a SIP trunk or route list. See the [Cisco Unified Communications Manager Administration Guide] and Cisco's [Unified Communications Manager documentation portal].

QUESTION NO: 16

What are two configuration features of the Standard Local Route Group deployment? (Choose two.)

A. is associated under the route group

B. is associated only under the route list

C. chooses the route group that is configured under the device pool of the calling-party device

D. chooses the route group that is configured under the device pool of the called-party device

E. is assigned directly to the route pattern

ANSWER: B C

Explanation:

The Standard Local Route Group is a CUCM routing mechanism that enables location-aware PSTN access without requiring separate route patterns for each site. It is associated only under the route list: an administrator adds the special Standard Local Route Group entry as a route group member in a route list. That route list can then be selected by a route pattern, translation pattern, or other dial-plan element. This structure lets a single dialing rule be reused across sites while retaining local egress behavior.

When a call reaches a route list containing the Standard Local Route Group, CUCM chooses the route group that is configured under the device pool of the calling-party device. The calling device's device pool has a Local Route Group setting, which points to the actual route group containing the applicable gateway or trunk. Consequently, users at different locations can dial the same pattern and have calls sent through their respective local gateways. This design simplifies dial-plan administration and supports site-based survivability and toll-bypass strategies. Cisco describes this route-list substitution behavior and device-pool Local Route Group assignment in its CUCM feature-configuration documentation: [Cisco Unified Communications Manager Feature Configuration Guide] and [Cisco Unified Communications Manager documentation].

QUESTION NO: 17

An engineer must configure call queuing under a Hunt Pilot. After the engineer receives the audio file that will be played to callers during queuing, which two steps should be taken to complete the configuration? (Choose two.)

A. Assign the uploaded audio file to the hunting Line Group member's "User Hold MOH Audio Source

B. Assign the uploaded audio file to the hunting Line Group member's "Network Hold MOH Audio Source".

C. Upload the audio file in "TFTP File Managementâ€ via OS Administration GUI

D. Assign the uploaded audio file to "Network Hold MOH Source & Announcements" under Hunt Pilot's Queuing section.

E. Upload the audio file in "MOH Audio File Management" via CM Administration GUI

ANSWER: D E

Explanation:

Call-queuing music is configured as a Cisco Unified Communications Manager Music on Hold resource and then selected for the Hunt Pilot's queueing behavior. The audio file must first be uploaded through *MOH Audio File Management* in Cisco Unified CM Administration. This makes the file available to the Music on Hold subsystem, where it can be associated with an MOH audio source. The uploaded content is therefore managed as an MOH file rather than merely as a general TFTP file.

For calls waiting in a Hunt Pilot queue, Cisco Unified CM places the queued caller on network hold. The Hunt Pilot's Queuing section contains the *Network Hold MOH Source & Announcements* settings used to define what queued callers hear. Selecting the configured MOH source in this area causes the associated uploaded audio to play while the caller waits for an available hunt member. This configuration applies consistently to calls queued by that Hunt Pilot, independent of an individual line-group member's own hold-audio settings.

Cisco's Music on Hold configuration guidance describes managing MOH audio files and sources in Unified CM Administration: Configure Music on Hold on Cisco Unified Communications Manager. Hunt Pilot queueing settings are documented in the Cisco Unified CM Feature Configuration Guide.

QUESTION NO: 18

Cisco SIP IP telephony is implemented on two floors of your company. Afterward, users report intermittent voice issues in calls established between floors. All calls are established, and sometimes they work well, but sometimes there is one-way audio or no audio. It is determined that there is a firewall between the floors, and the administrator reports that it is allowing SIP signaling and UDP ports from

20000 to 22000 bidirectionally. What are two solutions for this issue? (Choose two.)

A. Go to the SIP profile assigned to these IP phones in Cisco UCM and change the range of media ports to 16384-32767

B. Ask the firewall administrator to change the ports to TCP.

C. Ask the firewall administrator to change the range of UDP ports to 16384-32767.

D. Go to the SIP profile assigned to these IP phones in Cisco UCM and change the range of media ports to 20000-22000.

E. Go to System Parameters in Cisco UCM and change the range of media ports to 20000-22000.

ANSWER: C D

Explanation:

SIP call setup and RTP media transport are separate flows. SIP signaling can establish a call successfully while the RTP streams negotiated in SDP are blocked by an intervening firewall, producing intermittent one-way or no-audio symptoms. Cisco SIP endpoints ordinarily use a broad dynamic RTP/RTCP media-port range, commonly 16384 through 32767. Therefore, asking the firewall administrator to change the range of UDP ports to 16384-32767 permits the firewall to pass the normal negotiated Cisco voice-media flows in both directions. This is a common approach when retaining the default endpoint media-port configuration.

Alternatively, configuring the SIP profile assigned to the IP phones so that its media-port range is 20000-22000 constrains the affected SIP endpoints to the range that the firewall already permits. The policy must cover the full negotiated RTP/RTCP range bidirectionally, because each endpoint sends and receives media and port selections can vary from call to call. Matching the endpoint range and firewall policy eliminates the inconsistent reachability that makes only some calls exhibit media failure. Cisco Unified CM SIP profile settings are used to define SIP endpoint behavior, including media-port settings; see Cisco's Unified CM System Configuration Guide. RTP's use as the real-time media transport is defined in RFC 3550.

QUESTION NO: 19

An engineer must configure a Cisco UCM hunt list so that calls to users in a line group are routed to the first idle user and then the next. Which distribution algorithm must be configured to accomplish this task?

A. broadcast

B. top down

C. longest idle time

D. circular

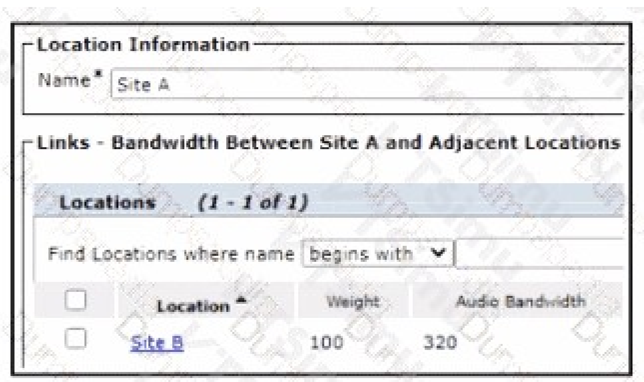
ANSWER: B

Explanation:

top down is the correct Line Group distribution algorithm. With Top Down configured, Cisco Unified CM evaluates the line group members in the order in which they are listed, beginning at the top of the member list for each new call. It selects the first member that is available (idle). If that member is unavailable, CUCM continues down the ordered list until it finds the next available member. This provides a predictable, administrator-defined call-routing order: the first listed idle user is offered the call, followed by the next listed user only when earlier members cannot take the call.

This behavior is appropriate when the intended handling order is based on member position, such as assigning calls to a primary user before overflow users. The line group's distribution algorithm controls the order in which its directory numbers are considered after the hunt pilot and hunt list have directed a call to that line group. Cisco documents Top Down as selecting line group members according to their configured order, starting with the first member. See Cisco's Line Group configuration documentation and the Cisco Unified CM configuration guides.

QUESTION NO: 20



Refer to the exhibit. An engineer deploys CAC to Cisco UCM. UptofiveG.711 calls must be supported on the WAN link between Site A and Site B. Users report that only four concurrent calls are possible between Site A and Site B. Why isn't the filth concurrent call successful?

A. The G.729 audio codec is negotiated between Site A and B instead of the G.711 codec.

B. The QoS configuration on the WAN link causes the fifth call to be dropped.

C. Cisco UCM is using alternative links because of the Weight value.

D. The Audio Bandwidth value is undersized and must be set to 400 kbps.

ANSWER: D

Explanation:

The Audio Bandwidth value is undersized and must be set to 400 kbps. Cisco Unified Communications Manager locations-based call admission control reserves bandwidth according to the codec bandwidth value associated with a call. For G.711 audio, CUCM uses 80 kbps for CAC calculations. Therefore, supporting five simultaneous G.711 calls requires 5 × 80 kbps, or 400 kbps, of configured location audio bandwidth.

With the exhibited audio bandwidth setting below 400 kbps, four calls reserve 320 kbps and can be admitted, but admitting a fifth call would require the cumulative reservation to reach 400 kbps. CUCM denies that additional call because doing so would exceed the available CAC allocation for the WAN path between the locations. Updating the relevant location or location link audio bandwidth allocation to 400 kbps provides the required reservation capacity for five G.711 calls. The configured value is a CAC admission-control budget, so it should also be validated against the actual WAN capacity and the bandwidth reserved for real-time traffic by the network QoS design.

Cisco documents locations-based CAC and its bandwidth reservation behavior in the Cisco Unified Communications Manager System Configuration Guide and provides codec bandwidth values in its Collaboration System SRND.

QUESTION NO: 21

An engineer is configuring a Cisco UCM solution. The requirements state that some users in one location will receive calls from a number during work hours 09AM to 5PM, and another group will get the calls from the same number outside this defined timeslot. Users will also change the outgoing number when reaching out to customers based on the same time-of-day routing rules. Which feature is needed to allow for this type of configuration?

A. partitions

B. route groups

C. route lists

D. CSS

ANSWER: A

Explanation:

partitions is correct because Cisco Unified CM implements time-of-day routing by associating a Time Schedule with a partition. A Time Schedule is built from one or more Time Periods, such as 09:00 through 17:00 on specified working days. While that schedule is active, the partition is available for digit analysis; outside the configured period, it is unavailable. This enables separate routing patterns or translation patterns for the same called number to be placed in partitions that are active at different times. Calls can therefore reach the daytime user group during business hours and the alternate group after hours.

The same design also supports time-dependent outgoing caller ID. Cisco Unified CM can use time-scheduled partitions containing translation patterns or route patterns that apply the appropriate calling-party transformation or calling number presentation for the active period. Calling search spaces determine which partitions a device may search, while the partition time schedule determines when the relevant patterns are available. This allows both inbound destination selection and outbound calling-number treatment to follow the same business-hours policy. Cisco documents the configuration of partitions, including assignment of a time schedule, in the Cisco Unified CM System Configuration Guide. Cisco's Unified Communications Manager documentation portal provides the corresponding release-specific administration guides.

QUESTION NO: 22

If all patterns below are configured in Cisco Unified Communications Manager which would be used when dialing the pattern â€œ123â€?

A. 12!

B. 12X (urgent priority set)

C. 1XX (urgent Priority Set)

D. 12[2-5]

ANSWER: B

Explanation:

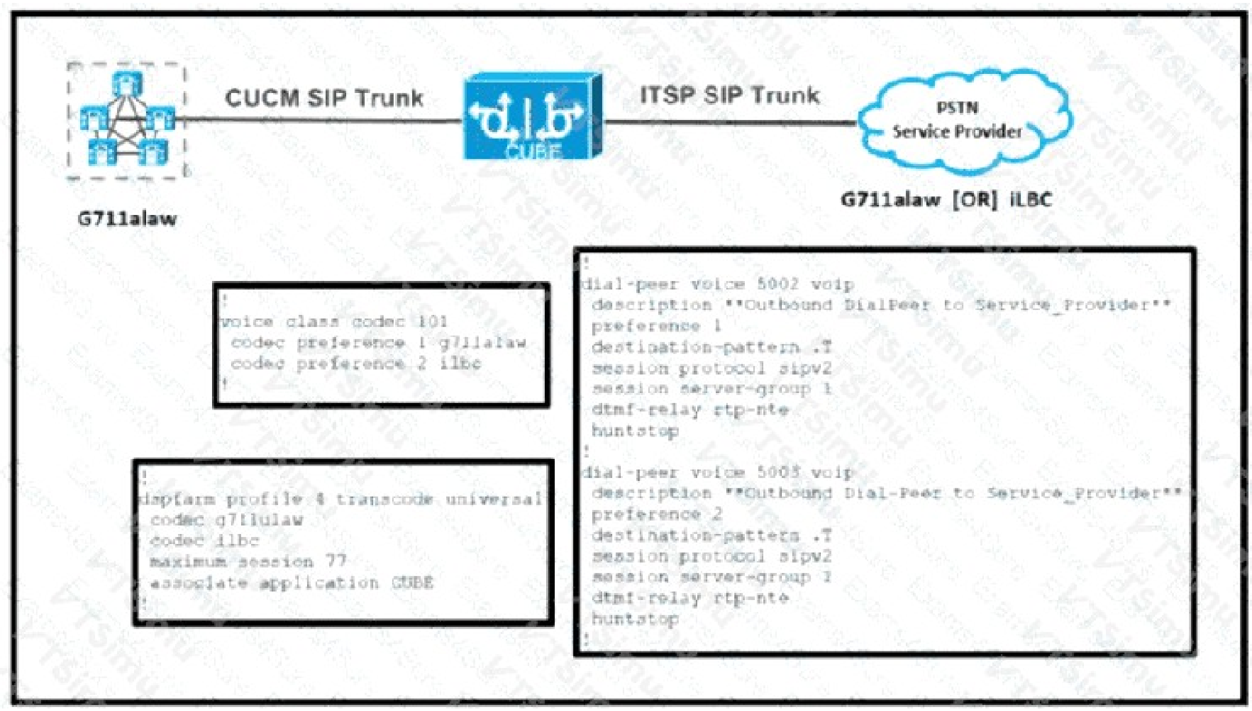
12X (urgent priority set) is selected for the dialed digits 123. The digits fully match this three-digit pattern: the literal digits 1 and 2 match the first two dialed digits, and X matches the final digit, 3. Because Urgent Priority is enabled, Cisco Unified Communications Manager treats completion of this pattern as sufficient to route the call immediately rather than continuing digit collection to determine whether a longer, non-urgent pattern might be dialed.

There is also an overlapping urgent match for these digits, so Cisco Unified CM applies its digit-analysis best-match behavior among the eligible urgent patterns. The pattern 12X is the more specific match because it explicitly fixes the first two digits, whereas the other urgent pattern explicitly fixes only the first digit and uses wildcards for the remaining positions. Consequently, 12X is the selected route pattern for 123. Urgent Priority is commonly used for patterns that must be routed as soon as the required digits are entered, such as short internal numbers or emergency-style dialing requirements.

Cisco documents route-pattern digit analysis, wildcard use, and Urgent Priority in the Cisco Unified CM System Configuration Guide and the Cisco Unified Communications Manager support documentation.

QUESTION NO: 23

Refer to the exhibit.



Outbound calls to the service provider cause intermittent errors due to a codec mismatch. The internal network sends early offer SDP that contains only

G.

G.

C.

```
dial-peer voice 5002 voip
codec g711alaw ilbc
!
dial-peer voice 5003 voip
codec g711alaw ilbc
```

```
dial-peer voice 5002 voip
voice-class codec 101 offer-all
!
dial-peer voice 5003 voip
voice-class codec 101 offer-all
```

```
dial-peer voice 5002 voip
codec g711alaw
!
dial-peer voice 5003 voip
codec ilbc
```

```
dial-peer voice 5002 voip
voice-class codec 101
!
dial-peer voice 5003 voip
voice-class codec 101
```

A. Option A

B. Option B

C. Option C

D. Option D

ANSWER: D

Explanation:

Option D is correct because successful SIP early-offer negotiation requires the outbound INVITE SDP to advertise a codec set that accommodates both codec capabilities reported by the service provider: G.711 A-law and iLBC. Applying an appropriate codec class or codec preference list to outbound dial peers 5002 and 5003 ensures that Cisco Unified Border Element can include both supported codecs in its SDP offer. The service provider can then select G.711 A-law for destinations that require it or iLBC for destinations that support only iLBC, avoiding intermittent codec-negotiation failures caused by offering G.711 A-law alone.

The configuration also enforces the provider's maximum of 20 simultaneous calls. Cisco IOS XE dial-peer call admission controls can limit outbound call connections so that calls above the permitted capacity are not sent to the provider. This protects the SIP trunk from exceeding the contracted call limit while allowing codec negotiation to occur consistently for calls routed through either specified outbound dial peer. Cisco documents codec-class configuration and application to dial peers in its CUBE configuration guidance, and documents dial-peer connection limits in the IOS XE command reference. See Cisco CUBE codec configuration guidance and Cisco IOS XE Voice Command Reference.

QUESTION NO: 24

A locations-based Call Admission Control policy rejects an intersite call because the configured audio bandwidth is exhausted. Automated Alternate Routing is enabled, and the administrator must define the calling search space that Cisco Unified Communications Manager uses for the alternate PSTN call. Where is this calling search space configured?

A. On the calling directory number as the AAR Calling Search Space

B. On the called directory number as the AAR Calling Search Space

C. On the device pool as the Local Route Group

D. On the route pattern as the Calling Party Transform CSS

ANSWER: A

Explanation:

The **AAR Calling Search Space** on the calling directory number is correct. When locations-based CAC rejects an on-net call and AAR is invoked, Cisco Unified Communications Manager uses the AAR Calling Search Space associated with the calling line to perform digit analysis for the alternate destination. This CSS must provide access to the route patterns that can route the transformed destination through the PSTN.

The ordinary calling search space controls the original call-routing attempt. The AAR Calling Search Space is evaluated specifically for the alternate route after a bandwidth denial. This separation permits an organization to control PSTN fallback independently from normal on-net dialing privileges. Cisco documents AAR configuration in the Cisco Unified Communications Manager Feature Configuration Guide.

QUESTION NO: 25

An administrator is configuring an Intercluster Lookup Service between 10 Cisco UCM clusters. Due to security requirements, certificate-based authentication must be used. Due to the deployment size, the administrator wants to avoid manually exchanging certificates between the clusters. Which two steps must be followed to meet these requirements? (Choose two)

A. Set all dusters to standalone on the ILS configuration page.

B. Install multiserver CA-signed Tomcat certificates on all cluster publishers.

C. Enable password-based authentication in conjunction with certificate-based authentication.

D. Ensure that the cluster ID for all clusters is the same.

E. Install multiserver CA-signed CallManager certificates on all cluster publishers.

ANSWER: B C

Explanation:

Installing multiserver CA-signed Tomcat certificates on all cluster publishers establishes the certificate foundation required for Intercluster Lookup Service certificate authentication. A multiserver certificate is appropriate because it can contain the required publisher identities in its Subject Alternative Name entries, while CA signing provides a trust chain that the participating Unified CM clusters can validate. Using a common trusted CA removes the operational burden of individually importing peer certificates among a large number of clusters.

Password-based authentication must also be enabled in conjunction with certificate-based authentication. In an ILS deployment, password authentication supports the initial secure association and certificate exchange process between clusters. Once the ILS peers have established connectivity and trust, certificate-based authentication can be used for the ILS communication without an administrator manually transferring each peer certificate. This combination is particularly useful in a large ILS topology because it preserves authenticated setup while allowing the system to automate the certificate-distribution workflow. Cisco's Unified CM administration documentation covers ILS configuration and its authentication mechanisms in the Cisco Unified CM Administration Guide; Cisco's certificate-management guidance is available in the Cisco Unified CM Security Guide.

QUESTION NO: 26

Region Configuration Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

Region Information

Name* Region A

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Region B	Use System Default (Factory Default low loss)	8 kbps (G.729)	None	None
Region C	Use System Default (Factory Default low loss)	16 kbps (iLBC, G.728)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
Region D	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Media Resource Group Information

Name* MRG2

Description Media Resource Group 2

Devices for this Group

Available Media Resources** MOH_5, MTP_3, MTP_4, MTP_5, XCODER

Selected Media Resources* ANN_2 (ANN), CFB_2 (CFB), MOH_2 (MOH), MTP_2 (MTP)

```
#CallManager SDL Logs
|AppInfo  |DET-MediaManager- (22401) : :preCheckCapabilities, caps mismatch!
Xcoder Reqd. kbps(8),
     filtered A[capCount=0 {Cap,ptime}=], B[capCount=4 {Cap,ptime} = {11,220}
{12,220} {15,220} {9,270} ]
     allowMTP=0 numXcoderRequired=1 xcodingSide=1
|SdlSig        |MrmAllocateMtpResourceErr    |waitResourcesAllocated
     |MediaManager (6,100,144,22401)   MediaResourceManager (6, 100, 142, 1)
```

Refer to the exhibit. Regions have been configured for all major branches based on the available circuit bandwidth. Some calls from Region A endpoints to Region B endpoints are failing to connect. How is the issue resolved?

A. Update the calling search space for affected endpoints to none.

B. Update all regions to 8 kbps maximum audio bitrate.

C. Increase the number of available media termination points.

D. Add a media resource to transcode between available capabilities.

ANSWER: D

Explanation:

Add a media resource to transcode between available capabilities. A Cisco Unified Communications Manager region relationship imposes a maximum audio bit rate for calls between the two regions. CUCM uses that limit when it selects a codec, but a call can succeed only when both endpoints have a mutually usable codec or CUCM can insert an appropriate media resource. In this scenario, the bandwidth-based region configuration can require a lower-bandwidth codec for the interregion leg while one of the endpoint capabilities is limited to a different codec. The resulting codec mismatch prevents establishment of the media connection.

A transcoder provides the required conversion between the codec used by the Region A endpoint and the codec permitted or supported for the Region B side. The transcoder must be configured as a CUCM media resource and made reachable through the applicable Media Resource Group List or device pool so that CUCM can allocate it during call setup. This preserves the intended region bandwidth policy while allowing endpoints with dissimilar codec capabilities to communicate. Cisco documents regions as the mechanism for defining interregion audio codec bandwidth and documents transcoders as media resources used to translate between incompatible codec streams. See the Cisco Collaboration System SRND and the Cisco Unified Communications Manager System Configuration Guide.

QUESTION NO: 27

Refer to the exhibit.

An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco Unified Communications Manager does not complete. The administrator wants to use the Dialed Number Analyzer on Cisco Unified CM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability there is no option for Dialed Number Analyzer under the tool menu. Which two steps must be performed to resolve this issue? (Choose two.)

A. Restart the subscriber

B. Activate the Cisco Extended Functions service.

C. Activate the Cisco CallManager service.

D. Activate the Cisco Dialed Number Analyzer service.

E. Activate the Cisco Dialed Number Analyzer Server service.

ANSWER: D E

Explanation:

The Cisco Dialed Number Analyzer service and the Cisco Dialed Number Analyzer Server service must both be activated to make Dialed Number Analyzer available and operational in Cisco Unified Communications Manager. The Cisco Dialed Number Analyzer service supplies the application-facing functionality that exposes the analyzer through the Cisco Unified Serviceability Tools menu. Activating it is what enables administrators to access the Dialed Number Analyzer interface rather than seeing the tool absent from the menu.

The Cisco Dialed Number Analyzer Server service provides the server-side processing used by the analyzer to evaluate dialed digits against the configured call-routing logic. With both services active, an administrator can submit a dialed number and calling context, then inspect the route-plan resolution process, including matching translation patterns, partitions, calling search spaces, and subsequent routing decisions. This is particularly useful when a call fails before completion because it validates the effective configuration used for that specific call-routing attempt.

Cisco documents Dialed Number Analyzer as requiring activation of these related services before the tool can be used. Service activation is performed in Cisco Unified Serviceability under Tools > Service Activation, on the appropriate Unified CM node. See Cisco's Dialed Number Analyzer Administration Guide and Serviceability Administration Guide.

QUESTION NO: 28

Single Number Reach calls to a cell phone that not answered are leaving voicemails on the cell phone rather than the corporate mailbox. Which two options will resolve this issue? (Choose two.)

A. Check the Enable Extend and Connect checkbox.

B. Check the Enable Unified Mobility features checkbox.

C. Decrease the T302 timer.

D. Decrease the T301 timer.

E. Decrease the Answer Too Late timer.

ANSWER: B E

Explanation:

Checking the Enable Unified Mobility features checkbox is required because Single Number Reach is a Unified Mobility service. The user must be enabled for Unified Mobility so that Cisco Unified Communications Manager can apply the user's remote destination configuration and control the simultaneous or sequential ringing behavior for the cell phone.

Decreasing the Answer Too Late timer prevents the mobile carrier's voicemail system from answering the call before Cisco Unified Communications Manager can recover it. This timer defines how long Unified CM waits for the remote destination—in this case, the cell phone—to answer. Set it to a value shorter than the time at which the cellular provider normally forwards an unanswered call to its voicemail. When the remote destination does not answer within that interval, Unified CM stops treating the mobile destination as answerable and can route the unanswered call according to the corporate call-forward and voicemail configuration, allowing the corporate mailbox to receive the message.

Cisco documents the Answer Too Late timer as a key Remote Destination/Profile setting for Single Number Reach behavior. See Cisco's Single Number Reach feature documentation and the Cisco Unified CM Feature Configuration Guide.

QUESTION NO: 29

Refer to the exhibit.

ILShas been configured between two hubs using this configuration. The hubs appear to register successfully, butILSis not functioning as expected. Which configuration step is missing?

A. A password has never been set for ILS.

B. Use TLS Certificates must be selected.

C. Trust certificates for ILS have not been installed on the clusters

D. The Cluster IDs have not been set to unique values

ANSWER: D

Explanation:

The missing step is to assign unique Cluster ID values to the two Cisco Unified Communications Manager clusters. Intercluster Lookup Service uses the Cluster ID as a core identity value when it builds the ILS network and exchanges advertised information, such as remote cluster service and directory data. Each cluster participating in an ILS network must therefore have an ID that is unique within that network. If two hubs use the same Cluster ID, they are not reliably distinguishable as separate clusters. As a result, apparent hub registration alone does not ensure that ILS advertisements, discovery, and lookups operate correctly.

Configure the Cluster ID under Cisco Unified CM enterprise parameters before establishing or troubleshooting the ILS topology. The value should be planned as a stable, unique identifier for each cluster, rather than being duplicated across clusters with similar names or roles. After assigning distinct IDs, verify the ILS topology and advertisements from the ILS configuration and monitoring pages to confirm that each hub is represented as a separate cluster. Cisco documents ILS configuration and cluster-level system settings in the Cisco Unified Communications Manager System Configuration Guide and its Unified Communications Manager documentation resources.

QUESTION NO: 30

What are two configuration features of the Client Matter Code setting in the route pattern configuration? (Choose two.)

A. The Client Matter Code feature does not support overlap sending since the Cisco UCM cannot determine when to prompt the user for the code.

B. Selecting the Allow Overlap Sending setting disables the Require Client Matter Code setting.

C. Selecting the Allow Overlap Sending setting allows a user to select the Require Client Matter Code setting.

D. The Client Matter Code feature supports overlap sending since the Cisco UCM can determine when to prompt the user for the code.

E. The Client Matter Code feature provides the option to configure Authorization Level such as in the Forced Authorization Code.

ANSWER: A B

Explanation:

Client Matter Code is a route-pattern feature used to require a caller to enter a code that identifies the client, project, or matter associated with an outgoing call. Cisco Unified Communications Manager must know that the dialing sequence is complete before it can present the Client Matter Code prompt. Consequently, the Client Matter Code feature does not support overlap sending, because overlap sending delivers dialed digits progressively and does not provide the required point at which CUCM can reliably prompt for the code.

The route-pattern settings enforce this dependency in the configuration interface. When *Allow Overlap Sending* is selected, CUCM disables the *Require Client Matter Code* setting. This prevents an unsupported combination from being configured and ensures that matter-code collection occurs only after CUCM has fully matched and processed the dialed pattern. These behaviors are documented in Cisco's CUCM features documentation for Client Matter Codes and route-pattern call-routing configuration. See the Cisco Unified Communications Manager Features and Services Guide and Cisco Unified Communications Manager documentation.

QUESTION NO: 31

Which configuration must an administrator perform to display Translation Pattern operations in Cisco Unified Communications Manager SDL traces?

A. Enable the Detailed Call Analysis option under Enterprise Parameters for Unified CM.

B. Set up the Digit Analysis Complexity in Service Parameters for Cisco Unified CM to TranslationAndAlternatePatternAnalysis.

C. Check the Translation Patterns Analysis check box in Micro Traces on the Cisco Unified CM Serviceability page.

D. By default, the Translation Patterns operations are printed in SDL traces, so no additional configuration is necessary.

ANSWER: B

Explanation:

Set up the Digit Analysis Complexity in Service Parameters for Cisco Unified CM to TranslationAndAlternatePatternAnalysis. This Cisco CallManager service parameter controls the level of digit-analysis detail written to Cisco Unified Communications Manager SDL traces. Selecting `TranslationAndAlternatePatternAnalysis` specifically enables trace output for translation-pattern processing and alternate-pattern analysis, allowing an administrator to follow how the called or calling number is evaluated and transformed during call routing. This is especially useful when diagnosing unexpected number manipulation, translation-pattern matches, or routing behavior that depends on digit analysis. After changing the service parameter for the applicable Cisco CallManager service, collect the SDL traces through Real-Time Monitoring Tool or Cisco Unified Serviceability and review the digit-analysis messages associated with the call. Cisco's trace-collection guidance provides context for enabling and retrieving CUCM call traces: Cisco Community: Taking SIP Call Trace on Cisco Unified CM Using RTMT. The Cisco Unified Communications Manager documentation and configuration-guide resources are available at Cisco Unified CM Configuration Guides.

QUESTION NO: 32

An engineer is implementing QoS for Cisco Unified Communications Manager signaling and RTP voice media across a WAN. Which two DSCP markings are recommended for these traffic types? (Choose two.)

A. EF for RTP voice media

B. CS3 for call signaling

C. AF41 for RTP voice media

D. CS1 for call signaling

E. BE for RTP voice media

ANSWER: A B

Explanation:

EF is the recommended DSCP marking for RTP voice media. Expedited Forwarding is intended for low-loss, low-latency, low-jitter traffic and is commonly placed in the strict-priority queue on WAN links after the queue is properly provisioned and policed.

CS3 is the commonly recommended DSCP marking for call-signaling traffic, including Cisco Unified CM signaling. Signaling is important to call setup and teardown, but it does not require the same strict-priority treatment as RTP media. Separating signaling from RTP permits the network to protect call control while reserving priority-queue capacity for real-time media. Cisco QoS marking concepts are described in the Cisco Quality of Service solution documentation.

QUESTION NO: 33

An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which debug must the Administrator turn on?

A. debug H.323 messages

B. debug H.224 asn1

C. debug H.246 asn 1

D. debug H.225 media

E. debug H.323 asn 1

ANSWER: E

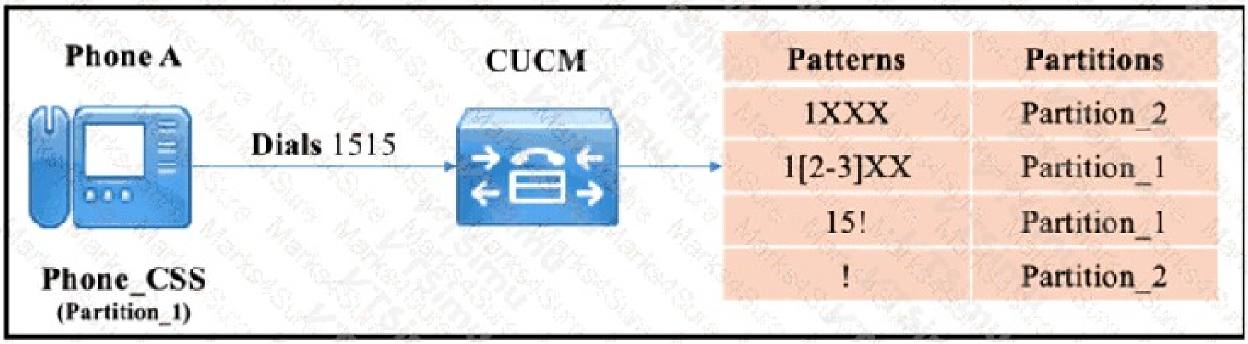
Explanation:

`debug H.323 asn 1` is the appropriate debug because H.323 signaling messages are encoded using ASN.1. Enabling this debug displays decoded protocol information relevant to both call establishment and media negotiation. In an H.323 call, H.225 signaling handles call setup and clearing, while H.245 exchanges capabilities and opens or closes logical channels used for media. Viewing the ASN.1 content therefore gives an administrator detailed visibility into the signaling parameters exchanged by the gateway, including call-control messages, negotiated codecs, and media-channel details that can identify where a call is failing.

This level of output is particularly useful when basic call-state information is insufficient and the engineer needs to inspect the actual H.323 protocol exchanges. Because protocol debugs can generate substantial output on an active gateway, they should be enabled during a controlled troubleshooting window and captured to a terminal or logging destination. Cisco's H.323 gateway configuration documentation describes H.225 and H.245 as the call-control and media-control components of H.323: Cisco IOS H.323 Configuration Guide. Cisco also provides H.323 troubleshooting guidance at Troubleshooting H.323 Gateways.

QUESTION NO: 34

Refer to the exhibit.



The Cisco UCM is configured with four route patterns. When phone A dials 1515, which pattern does Cisco UCM choose as the correct match?

A. 1XXX

B. 15

C. !

D. 1[2-3]XX

ANSWER: B

Explanation:

15 is selected because it is configured as an urgent-priority route pattern in the exhibit. Urgent priority changes normal digit-collection behavior: as soon as Cisco Unified Communications Manager receives digits that completely match an urgent-priority pattern, it immediately performs digit analysis and routes the call. Therefore, after the user enters the first two digits, **15**, CUCM has a complete urgent match and commits the call to that route pattern; it does not wait for additional digits that could produce a longer match. The subsequently dialed digits are not used to select a different route pattern because routing has already begun. This behavior is useful for patterns that must be routed without waiting for the interdigit timeout, such as emergency or otherwise time-critical destinations. Cisco documents urgent priority as a route-pattern setting that causes CUCM to route a call immediately when the entered digits match the configured pattern. See Cisco's Unified CM digit analysis and route-pattern documentation and the Cisco Unified Communications Manager documentation portal.

QUESTION NO: 35

When route patterns are defined as precisely as possible on a Cisco UCM, the control and reliability of the calling plan is increased. A route pattern must be added to support calls to this number range: 9135200 to 9135205. The calls are on-net and no translation patterns are configured. Which configuration meets these requirements?

A. 913520[012345]

B. 913520X

C. 913520[1â€“5]

D. 913520!

ANSWER: A

Explanation:

913520[012345] is the correct route pattern because it precisely matches every number in the required contiguous range: 9135200, 9135201, 9135202, 9135203, 9135204, and 9135205. Cisco Unified Communications Manager digit-pattern syntax treats the digits outside the brackets as fixed, required digits. The bracketed expression represents a single digit selected from the explicitly listed values. Consequently, the fixed prefix `913520` is followed by exactly one digit from `0` through `5`, with no additional digits permitted by this pattern.

This is appropriately specific for a controlled on-net dial plan because it admits only the six stated directory numbers. The route pattern can then be associated with the appropriate route list or gateway configuration for the on-net destination without relying on a translation pattern to normalize or change the called number. Cisco documents route patterns as digit patterns used to match called numbers and supports bracketed digit lists for matching one digit from a defined set. See the [Cisco Unified Communications Manager Administration Guide] and Cisco's [Collaboration SRND] for dial-plan and route-pattern design guidance.