Free Questions for 350-801 by certsdeals

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Question 1

Question Type: MultipleChoice

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

Options:

A- voice and video transcoding

B- voice and video conferencing

C- interworking of SIP and H.323

D- intercluster extension mobility

E- endpoint registration

Answer:

A, C

Explanation:

The Cisco Expressway Series provides the following functions:

Voice and video transcoding

Interworking of SIP and H.323

Firewall traversal

Session border controller (SBC) functionality

Endpoint registration

Call admission control (CAC)

Quality of service (QoS)

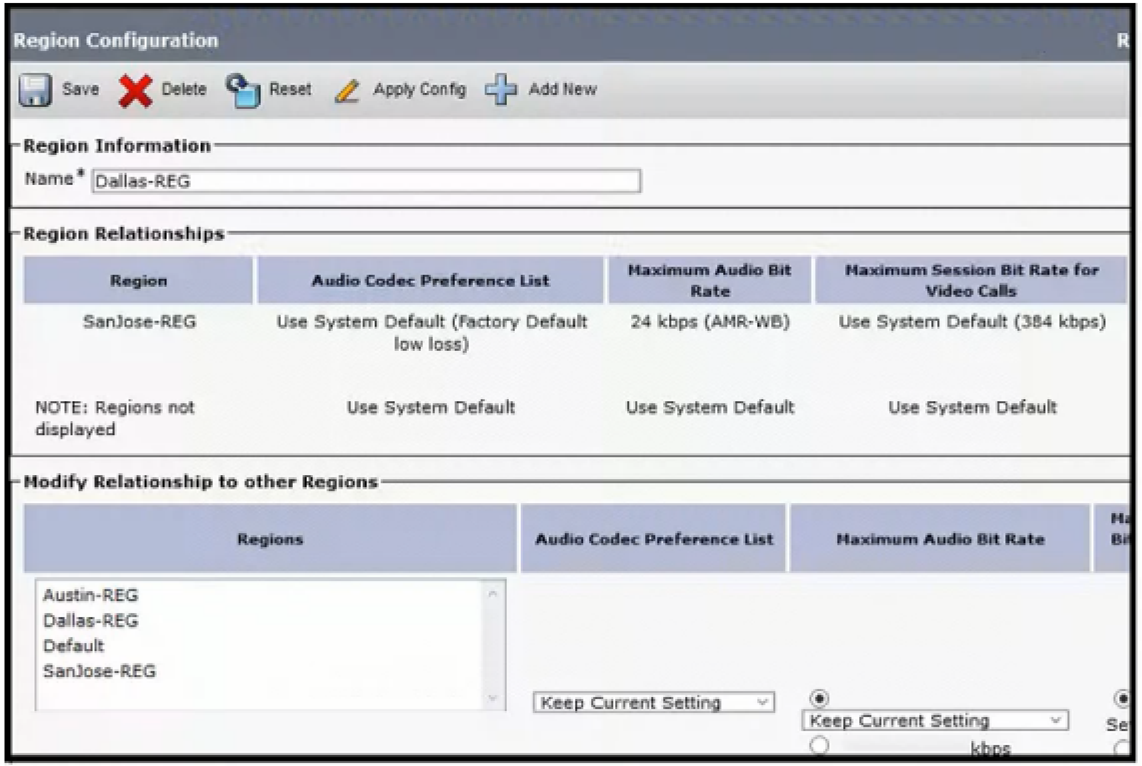
Security

The Cisco Expressway Series does not provide voice and video conferencing or intercluster extension mobility.

Question 2

Question Type: MultipleChoice

Refer to the exhibit.



Which codec should an engineer select for a call made between "Dallas-REG' & "Austin-REG'?

Options:

A- MP4A-LATM

B- G.711

C- OPUS

D- G.729

G- 729

G- 711
OPUS
MP4A-LATM
The codec preference list for the 'Austin-REG' region is 'Factory Default low loss'. This list includes the following codecs in order of preference:

G- 729

G- 711
OPUS
MP4A-LATM
Since both regions have the same codec preference list, the codec that will be used for a call made between 'Dallas-REG' and 'Austin-REG' is G.729.

G- 729 is a narrowband speech codec that was developed by the ITU-T in 1988. It is a low-bitrate codec that provides good quality speech at a bitrate of 8 kbps. G.729 is widely used in VoIP applications and is the default codec for many VoIP systems.

G- 711 is a wideband speech codec that was developed by the ITU-T in 1972. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 64 kbps. G.711 is not as widely used as G.729 due to its high bitrate requirements.
OPUS is a lossy audio codec that was developed by the IETF in 2012. It is a low-bitrate codec that provides good quality speech at a bitrate of 6 kbps. OPUS is widely used in VoIP applications and is the default codec for many VoIP systems.
MP4A-LATM is a lossy audio codec that was developed by the IETF in 1999. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 24 kbps. MP4A-LATM is not as widely used as G.729 or OPUS due to its high bitrate requirements.

Answer:

D

Explanation:

The codec preference list for the 'Dallas-REG' region is 'Factory Default low loss'. This list includes the following codecs in order of preference:

Question 3

Question Type: **MultipleChoice**

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

Options:

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.

Question 4

Question Type: MultipleChoice

Where is urgent priority enabled to bypass the T302 timer?

Options:

A- route partition

B- transformation pattern

C- directory number

D- CTI port

Answer:

C

Explanation:

Urgent priority is enabled on the directory number configuration page. This allows the call to be routed at once to the fully qualified DN without any necessity to wait for inter-digit-timeout. If the Urgent Priority checkbox is disabled and you have overlap patterns configured, then CUCM waits for the user to dial further digits.

The other options are incorrect because:

Route partitions are used to group route patterns and route lists.

Transformation patterns are used to convert dialed digits into a different format.

CTI ports are used to connect Cisco Unified Communications Manager to third-party applications.

https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/200477-Urgent-Priority-Configuration-on-Directo.html

Question 5

Question Type: **MultipleChoice**

If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

Options:

A- SNMP

B- ICMP

C- SMTP

D- DNS

Answer:

D

Explanation:

According to the Cisco Community website1, the phone usesDNSto resolve the hostname of the CUCM server (cucm1.cisco.com) to its IP address. DNS is a network service that translates domain names into IP addresses.

Question 6

Question Type: **MultipleChoice**

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

Options:

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.

Question 7

Question Type: **MultipleChoice**

What is the major difference between the two possible Cisco IM and Presence high-availability modes?

Options:

A- Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standbynode in the event of an outage, and it also provides load balancing.

B- 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.

C- Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standbynode in the event of an outage, but it does not provide load balancing.

D- Balanced mode does not provide user load balancing, but it provides user failover in the event of an outage. 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.

Question 8

Question Type: MultipleChoice

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

Options:

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.

Setting up all international route patterns to 0.! 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.