



Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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NEW QUESTION 1

Refer to the exhibit.

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

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

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

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

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

Answer: BC

NEW QUESTION 2

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 3

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 4

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 5

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

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

Answer: C

NEW QUESTION 6

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 7

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 8

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

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

Answer: A

NEW QUESTION 9

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 10

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

- A. The DNS server was not specified, which is needed to resolve the DHCP server IP address.
- B. Option 100 string was not specified, or an incorrect Option 100 string was specified.
- C. The Cisco IP Phone does not know the IP address of the TFTP server.
- D. The Cisco IP Phone does not know the IP address of any of the Cisco UCM Subscriber nodes.
- E. Option 150 string was not specified, or an incorrect Option 150 string was specified.

Answer: CE

NEW QUESTION 10

Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

- A. *
- B. @
- C. !
- D. .

Answer: A

Explanation:

The asterisk (*) wildcard is used to match any sequence of characters, including an empty sequence. Therefore, it can be used to match any domain name in a

SIP Route Pattern.

The other options are not correct because:

- > C. !: The ! symbol is used to negate a character class.
- > D. .: The . symbol is used to match any single character.

NEW QUESTION 13

What should be used to detect common issues on a Cisco IOS XE-based Local Gateway and generate an email?

- A. Real-Time Monitoring Tool
- B. diagnostic signatures
- C. syslog
- D. SNMP

Answer: B

Explanation:

Diagnostic signatures are a feature that proactively detects commonly observed issues in the IOS XE-based Local Gateway and generates email, syslog, or terminal message notification of the event. You can also install the diagnostic signatures to automate diagnostics data collection and transfer collected data to the Cisco TAC case to accelerate resolution time.

NEW QUESTION 16

A customer wants to conduct B2B video calls with a partner using on-premises conferencing solution. Which two devices are needed to facilitate this request?

- A. Expressway-C
- B. Cisco Telepresence Management Suite
- C. Expressway-E
- D. MGCP gateway
- E. Cisco Unified Border Element

Answer: AC

NEW QUESTION 19

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 20

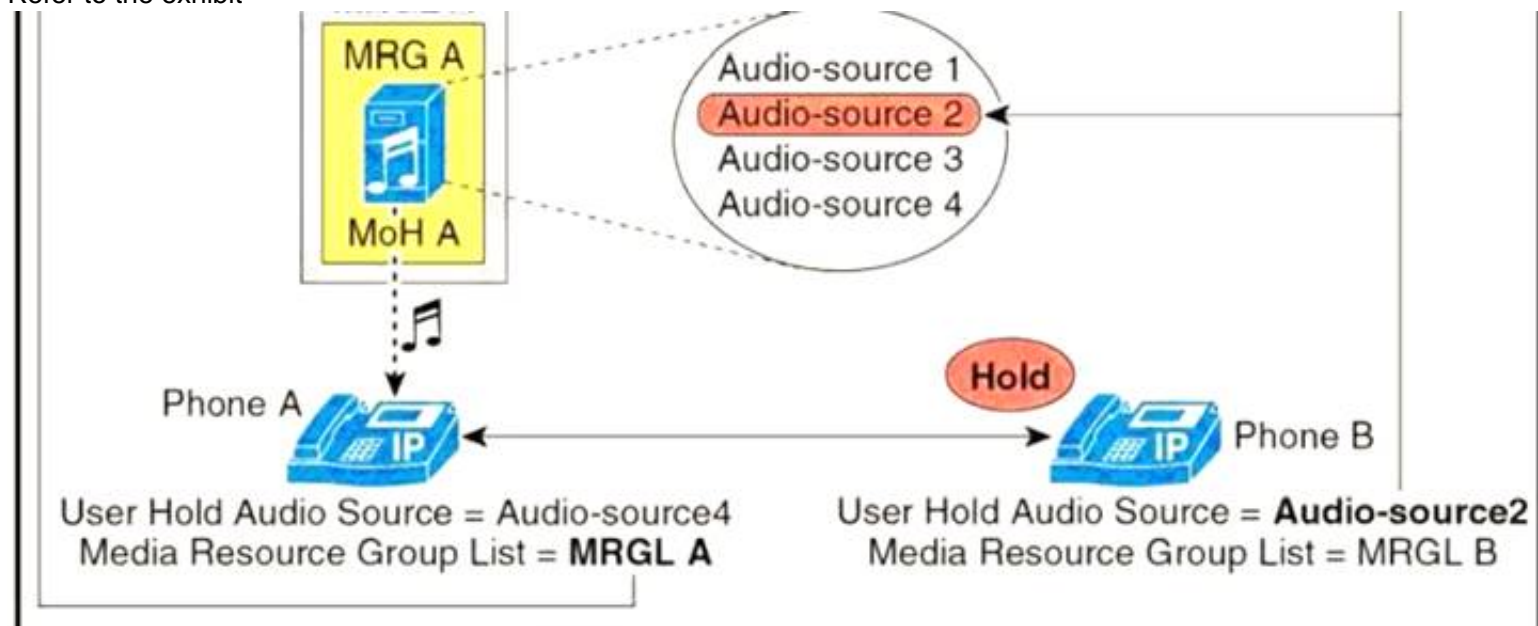
According to QoS guidelines, what is the packet loss for streaming video?

- A. Not more than 8%
- B. Not more than 1%
- C. Not more than 3%
- D. Not more than 5%

Answer: B

NEW QUESTION 22

Refer to the exhibit



There is a call flow between Phone A and Phone B. Phone B (holder) places Phone A (holder) on hold. Which MRGL and Audio Source are played to Phone A?

- A. MRGL A and Audio Source 4
- B. MRGL B and Audio Source 4
- C. MRGL A and Audio Source 2
- D. MRGL B and Audio Source 2

Answer: C

NEW QUESTION 27

How does traffic policing respond to violations?

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

Answer: A

NEW QUESTION 31

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

- A. If a call uses G.729. Cisco UCM subtracts 16k.
- B. If a call uses G.711, Cisco UCM subtracts 64k
- C. If a call uses G.711, Cisco UCM subtracts 80k
- D. If a call uses G.729. Cisco UCM subtracts 24k.
- E. If a call uses G.729. Cisco UCM subtracts 40k

Answer: CD

NEW QUESTION 35

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

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

Answer: AC

NEW QUESTION 40

What are two reasons that AF41 is marked for the audio and video channels of a video call? (Choose two.)

- A. to prioritize video over other high -priority traffic classes
- B. to give video calls a higher priority than AP41 in the QoS policy
- C. to allow high-definition quality calls over low-speed links
- D. to preserve lip synchronization between the audio and video channels
- E. to provide separate classes for audio calls and video calls

Answer: DE

NEW QUESTION 43

Why would we not include an end user's PC device in a QoS trust boundary?

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

Answer: D

NEW QUESTION 45

The chief officer at a company must reduce collaboration infrastructure costs by onboarding all on-premises equipment to the cloud by using CISCO Webex Control Hub. Administrators need the ability to manage upgrades and set up hot desking for on-premises devices. Which action must be taken before on boarding devices by using the Control Hub?

- A. Configure tie Control Hub organization ID on the devices
- B. Acquire a license for each device.
- C. Allow HTTP traffic from each device to Control Hub.
- D. Upgrade all the devices to software version CE9.15 or later

Answer: D

Explanation:

This is a prerequisite for using the Device Connector tool, which allows you to onboard and register several devices simultaneously to the Webex Control Hub1. The Device Connector tool creates a workspace, an activation code, and activates all of your devices in one go1. This way you don't need to be physically present in the same room to activate the devices.

The other options are not required before onboarding devices by using the Control Hub:

- Configuring the Control Hub organization ID on the devices is not necessary, as the Device Connector tool will send the device information to your Webex organization and generate activation codes for them 1.
- Acquiring a license for each device is not necessary, as you can assign licenses to users and devices after they are registered to the Webex Control Hub2.
- Allowing HTTP traffic from each device to Control Hub is not necessary, as HTTPS connectivity is required for the Device Connector tool to communicate with the devices1.

NEW QUESTION 49

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

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

Answer: B

NEW QUESTION 54

An engineer wants to manually deploy a CISCO Webex DX80 Video endpoint to a remote user. Which type of provisioning is configured on the endpoint?

- A. Cisco Unified Border Element
- B. Cisco Unity Connection
- C. Cisco Meeting Server
- D. Edge

Answer: D

Explanation:

The Cisco Webex DX80 Video endpoint can be provisioned in two ways:

- Automatically, using the Cisco Unified Communications Manager (CUCM) or Cisco Video Communication Server (VCS)
- Manually, using the Edge provisioning mode

The Edge provisioning mode is used when the endpoint is not connected to the CUCM or VCS. In this mode, the endpoint is configured with the necessary settings, such as the IP address, SIP/H.323 parameters, and time and date.

The Cisco Unified Border Element (Cisco UBE) is a network element that provides security and call control for IP telephony networks. The Cisco Unity Connection is a unified messaging system that provides voicemail, email, and fax services. The Cisco Meeting Server is a video conferencing system that provides high-quality video and audio conferencing.

NEW QUESTION 57

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 61

Which version is used to provide encryption for SNMP management traffic in collaboration deployments?

- A. SNMPv1
- B. SNMPv3
- C. SNMPv2
- D. SNMPv2c

Answer: B

NEW QUESTION 65

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

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

Answer: D

NEW QUESTION 70

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 73

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 76

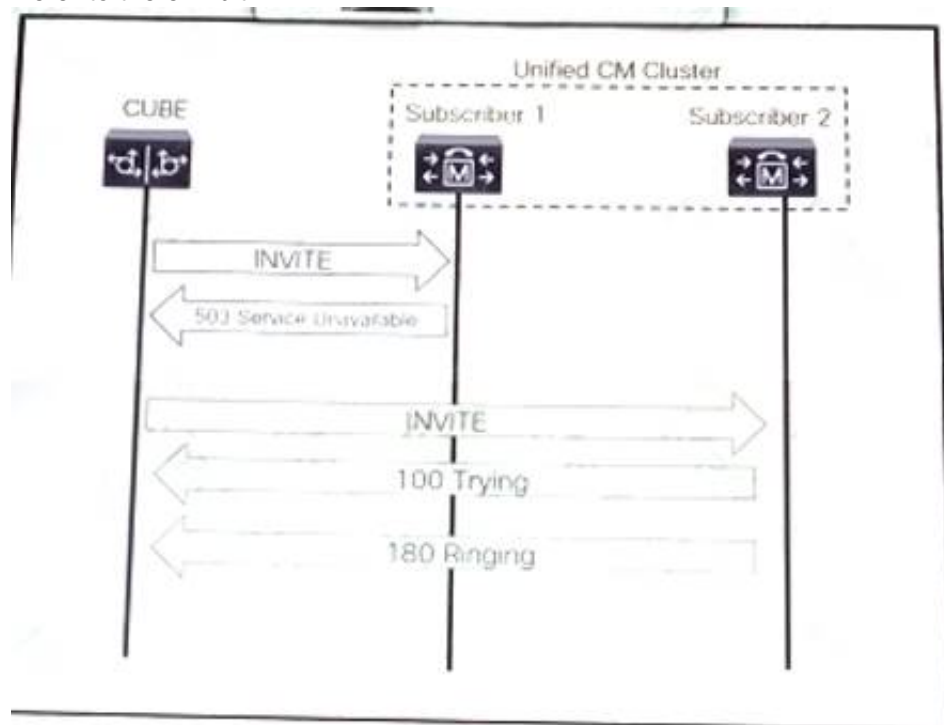
On a cisco catalyst switch which command is required to send CDP packets on a switch port that configures a cisco IP phone to transmit voice traffic in 802.1q frames tagged with the voice VLAN ID 221?

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

Answer: D

NEW QUESTION 79

Refer to the exhibit.



Cisco Unified element is attempting to establish a call with Subscribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subscribers2, and the call proceeds normally.

Which action resolves the issue?

- A. Verify that the correct calling search space is selected for the inbound Calls section
- B. Verify that the run on all active United CM Nodes checkbox is enabled
- C. Verify that the Significant Digits field for inbound Calls is set to All.
- D. Verify that the PSTN Access checkbox is enabled.

Answer: B

NEW QUESTION 80

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 84

Exhibit.

```
admin:utils ntp status
```

```
ntpd (pid 14550) is running...
```

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

Refer the exhibit. A collaboration engineer needs to replace the original, single NTP server that was configured during the initial install of a Cisco UCM server. What is the first step to accomplish this task?

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

Answer: B

NEW QUESTION 85

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 90

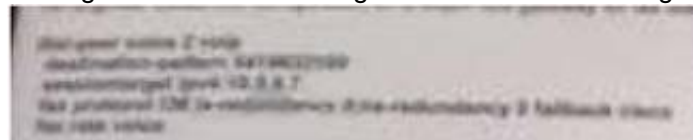
An administrator works with an ISDN PRI that is connected to a third-party PBX. The ISDN link does not come up. and the administrator finds that the third-party PBX uses the OSIG signaling method. Which command enables the Cisco IOS Gateway to use QSIG signaling on the ISDN link?

- A. isdn incoming-voice voice
- B. isdn switch-type basic-ni
- C. isdn switch-type basic-qsig
- D. isdn switch-type primary-qsig

Answer: D

NEW QUESTION 95

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



```
dial-peer voice 1 voip
description-pattern 94194327000
destination-pattern 94194327000
session-target 10.10.10.1
no protocol 100 is-redundancy 100-redundancy 1 fallback 100
no rate-limit
```

Which command is required to complete the configuration?

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

Answer: D

NEW QUESTION 100

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 101

An engineer troubleshoots outbound call failure on an ISDN-PRI circuit. The engineer is suspecting the 'Incomplete Destination'. Which debugs or commands are run in the voice gateway to troubleshoot the issue?

- A. debug isdn q921term mon
- B. debug voip ecapi inout show controller ti
- C. debug isdn q931 show isdn status
- D. debug isdn q921 debug voip ecapi inout

Answer: C

Explanation:

The engineer should run the following debugs or commands in the voice gateway to troubleshoot the issue: ➤ debug isdn q931 - This debug will show the ISDN Q.931 messages that are being exchanged between the voice gateway and the ISDN switch. This can be used to identify the cause of the "Incomplete Destination" error.

➤ show isdn status - This command will show the status of the ISDN PRI circuit. This can be used to verify that the circuit is up and running.

The other options are not correct. The debug isdn q921 command will show the ISDN Q.921 messages that are being exchanged between the voice gateway and the ISDN switch. This is not necessary for troubleshooting the issue. The term mon command will show the terminal monitor output. This is not necessary for troubleshooting the issue. The debug voip ecapi inout command will show the VoIP ECAP messages that are being exchanged between the voice gateway and the VoIP server. This is not necessary for troubleshooting the issue. The show controller ti command will show the status of the T1 controller. This is not necessary for troubleshooting the issue.

NEW QUESTION 103

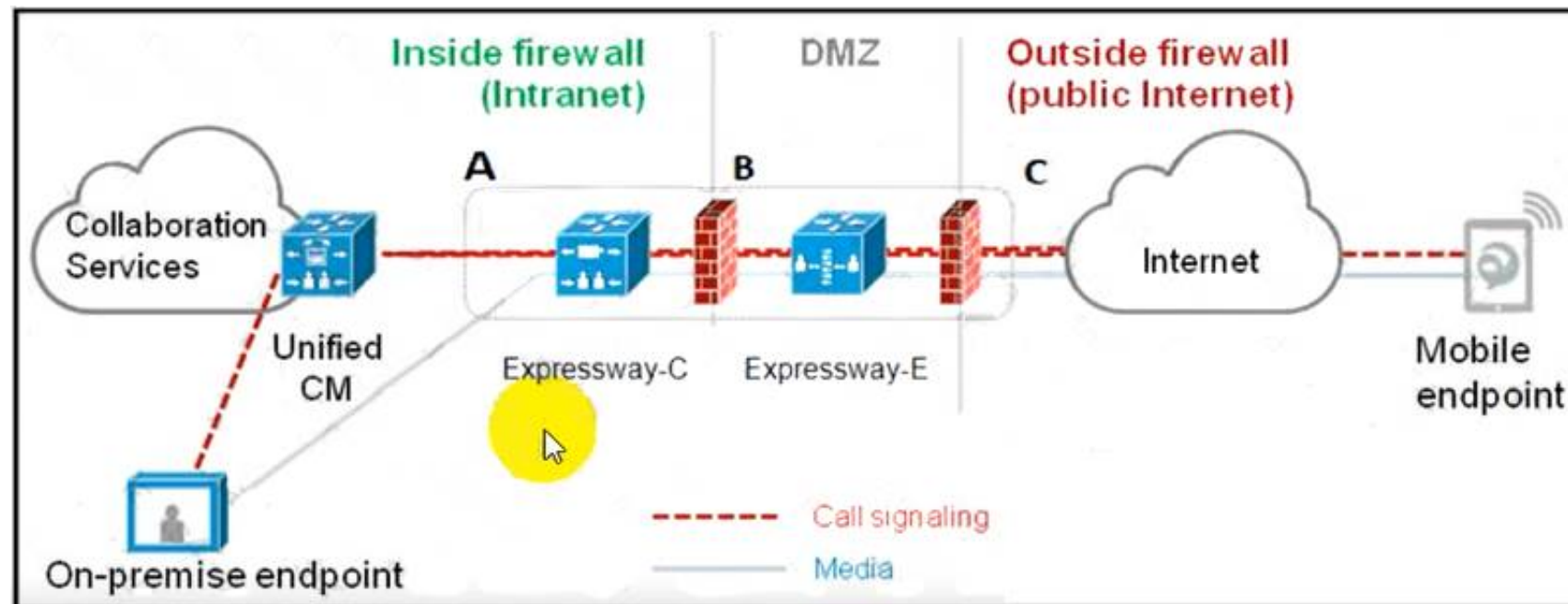
Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

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

Answer: C

NEW QUESTION 104

Refer to the exhibit.



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

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

Answer: D

NEW QUESTION 106

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 111

An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

- A. Configure path-thru content sdp on the voice service.
- B. Configure a hardcoded codec on the dial peers.
- C. Configure a transcoder for video protocols.
- D. Configure codec transparent on the dial peers.

Answer: D

NEW QUESTION 114

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

- A. Change line partition to Partition_A
- B. Change line CSS to only contain Partition_B
- C. Set the user's line CSS to <None>
- D. Set the user's device CSS to <None>

Answer: D

NEW QUESTION 118

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

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

Answer: C

NEW QUESTION 121

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 122

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*	\+.
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 127

What is a possible cause of the PRI issue?

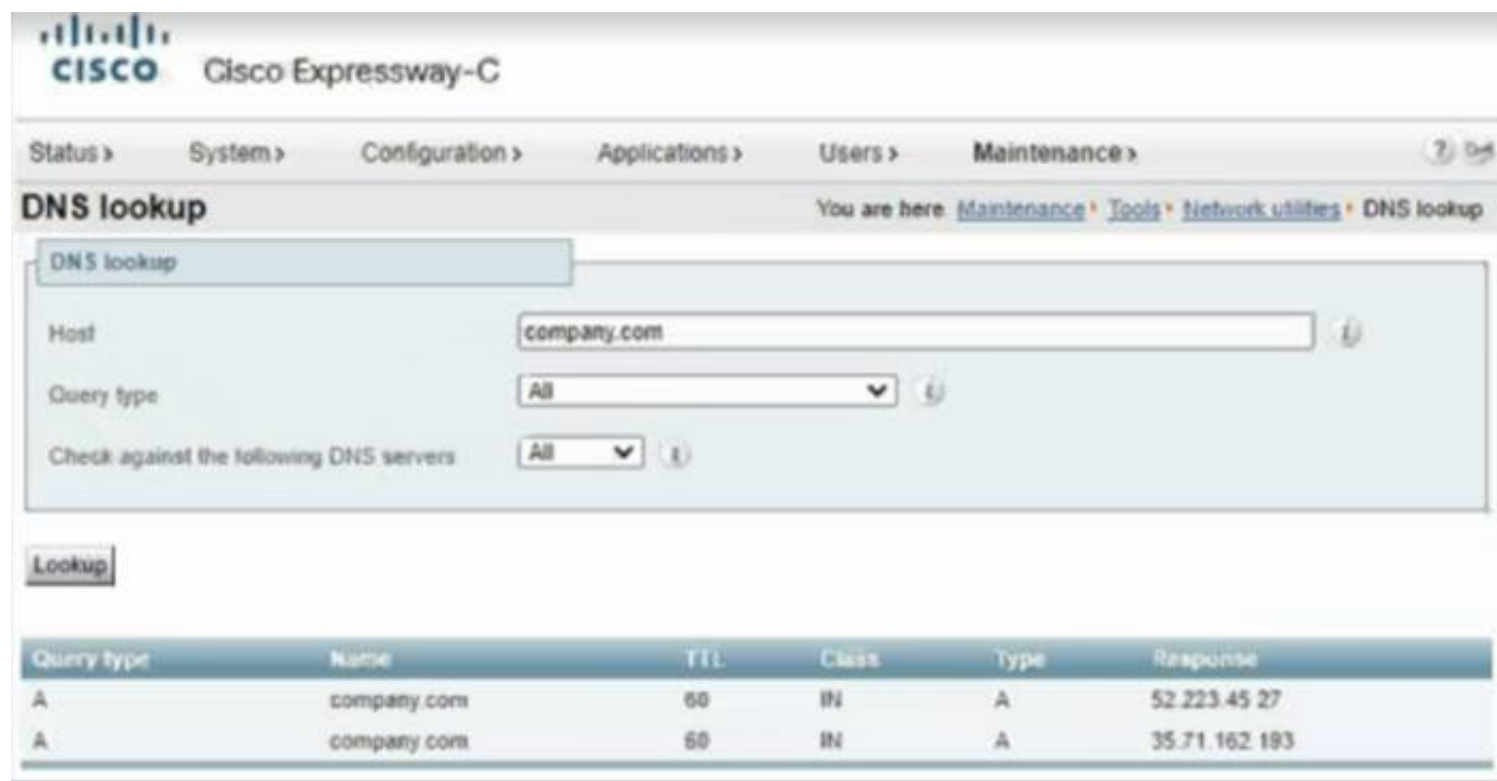
```
ISDN Serial1:23 interface
    dsl 1, interface ISDN Switchtype =
primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask: 0x807FFFFF
    Total Allocated ISDN CCBs = 5
```

- A. The cable is unplugged.
- B. The controller shut down.
- C. The clock source is incorrect.
- D. The framing is configured incorrectly.

Answer: D

NEW QUESTION 128

Refer to exhibit.



The screenshot shows the Cisco Expressway-C web interface. The top navigation bar includes links for Status, System, Configuration, Applications, Users, and Maintenance. The 'Maintenance' link is active, and the breadcrumb trail shows 'You are here: Maintenance > Tools > Network utilities > DNS lookup'. The 'DNS lookup' section has a form with the following fields:

- Host:** A text input field containing 'company.com'.
- Query type:** A dropdown menu set to 'All'.
- Check against the following DNS servers:** A dropdown menu set to 'All'.

Below the form is a 'Lookup' button. The results are displayed in a table with the following columns: Query type, Name, TTL, Class, Type, and Response.

Query type	Name	TTL	Class	Type	Response
A	company.com	60	IN	A	52.223.45.27
A	company.com	60	IN	A	35.71.162.193

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

- A. The company.com domain must be resolved only in Expressway-E
- B. There is a missing SRV record for the company.com domain.
- C. The TTL value for the company.com is too short.
- D. There must be only one response for the company.com domain

Answer: B

NEW QUESTION 130

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 131

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

- A. device pool selection
- B. Call Admission Control
- C. business continuity
- D. ringtone selection

Answer: C

Explanation:

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

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

NEW QUESTION 132

An engineer is deploying Webex app on Microsoft Windows computers. The engineer wants to ensure that the end users do not receive pop-up dialogues when they start the application. Which two actions ensure the end users are not prompted to accept the end-user license (Choose two)

- A. Set the DELETEUSERDATA=r installation argument
- B. Set the "HKEY_LOCAL_MACHINE\Software\Wow6432Node\CiscoCollabHost\Eula_disable
- C. Set the "HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting registry Eula_disable
- D. Set the DEFAULT^THEMES=Dark" installation argument
- E. Set the "/quiet installation argument

Answer: BC

Explanation:

The correct answers are B and C.

To ensure that end users are not prompted to accept the end-user license agreement (EULA) when they start the Webex app, the engineer must set the following two registry keys:

- > HKEY_LOCAL_MACHINE\Software\Wow6432Node\CiscoCollabHost\Eula_disable
- > HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting\Eula_disable

Setting these registry keys will disable the EULA prompt for all users who start the Webex app.
The other options are not valid actions to ensure that end users are not prompted to accept the EULA.

NEW QUESTION 136

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 139

Refer to the exhibit.

SIP Trunk Security Profile Information	
Name*	Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null string
Device Security Mode	Encrypted
Incoming Transport Type*	TLS
Outgoing Transport Type	TLS
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5061

An administrator configures a secure SIP trunk on Cisco UCM.
Which value is needed in the secure certificate subject or subject alternate name field to accomplish this task?

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

Answer: B

NEW QUESTION 143

Refer to the exhibit.

```
voice translation-rule 1
rule 1 /^[2-9].....$/ /\0/ type any subscriber
rule 2 /^1[2-9]..[2-9].....$/ /\0/ type any subscriber
```

What is the result of applying these two rules to a voice translation profile for use with an ISDN T1 PRI on a Cisco Voice Gateway?

- A. The leading Plus is stripped from the numeric phone number.
- B. The ISDN Plan is modified to the administrator's defined value.
- C. Any zero is stripped from the numeric phone number.
- D. The ISDN Type is modified to the administrator's defined value.

Answer: D

NEW QUESTION 147

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=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no;privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto: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 150

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 152

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 155

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

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

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

Answer: B

NEW QUESTION 159
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