# Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

NEW QUESTION 1
Refer to the exhibit A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type "international' Which configuration accomplishes this goal?
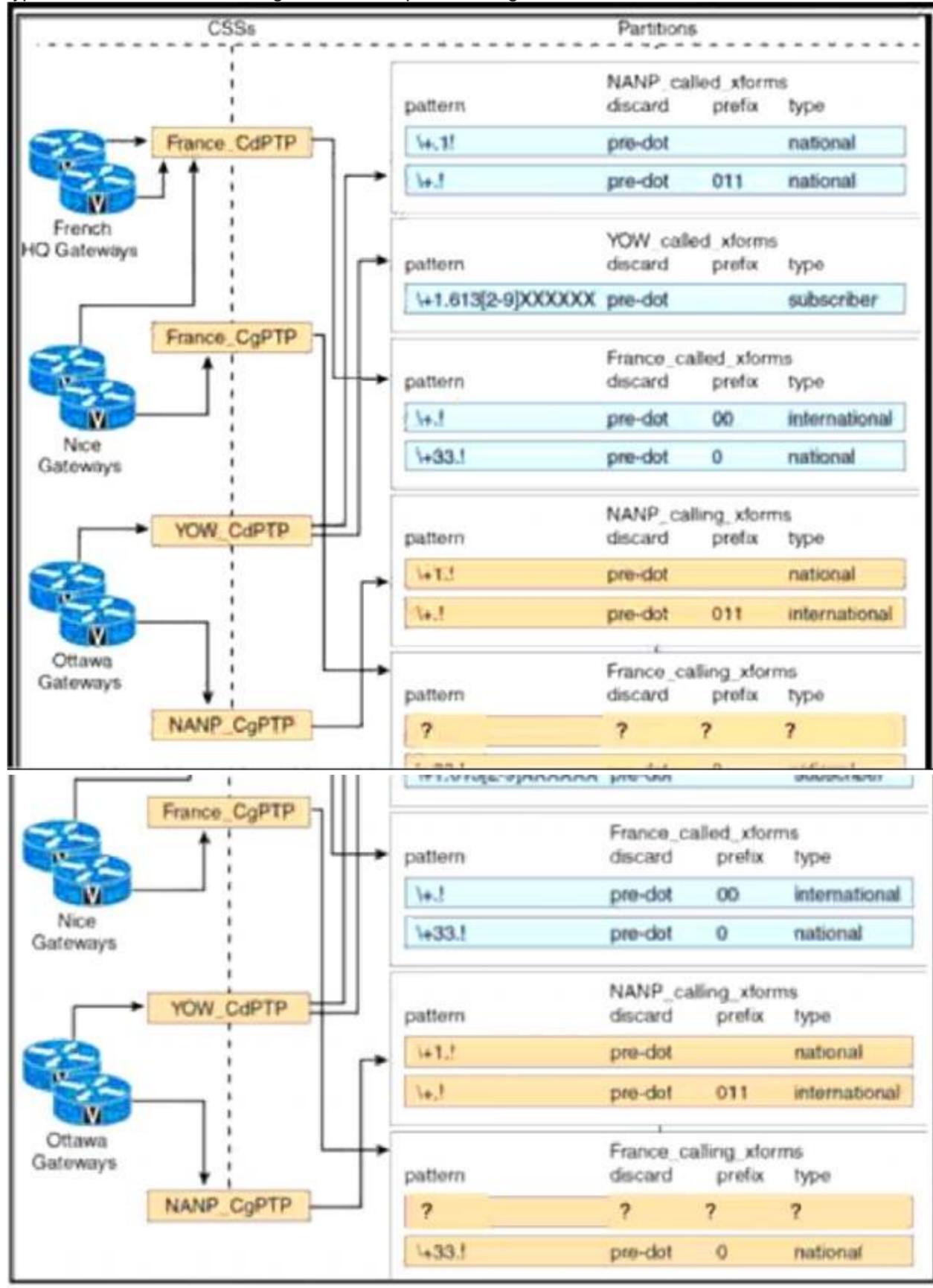
A. \+.001! pre-dot 1 international
B. \+1.1 none pre-dot 001 international
C. \+.! pre-dot 00 international
D. 613XXXXXXX none +011 internationa

Answer: C

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
A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay. Which commands must an engineer run on the dial peer?

A. dtmf-relay sip-info
B. dtmf-relay sip-notify
C. dtmf-relay rtp-net
D. no dtmf-relay sip-kpml

Answer: C

NEW QUESTION 4
Which two protocols can be configured for the Cisco Unity Connection and Cisco UCM integration? (Choose two.)

A. 323
B. SIP
C. SCCP
D. MGCP
E. RTP

Answer: BC

Explanation:
The two protocols that can be configured for the Cisco Unity Connection and Cisco UCM integration are SIP and SCCP. SIP, or Session Initiation Protocol, is a signaling protocol used for initiating, maintaining, and terminating real-time sessions, including voice, video, and messaging applications.
SCCP, or Skinny Client Control Protocol, is a Cisco proprietary signaling protocol used for controlling Cisco IP phones.
H.323 is an older signaling protocol that is no longer widely used. MGCP, or Media Gateway Control Protocol, is a protocol used for controlling media gateways. RTP, or Real-time Transport Protocol, is a protocol used for transporting real-time data, such as voice and video

NEW QUESTION 5
Which option must be used when configuring the Local Gateway for a Cisco Webex Calling trunk?

A. local authentication
B. certificate-based
C. mutual TLS
D. Auth-based

Answer: B

Explanation:
A certificate-based trunk is a type of trunk that uses certificates to authenticate the connection between Webex Calling and the Local Gateway1. A Local Gateway is a supported session border controller that terminates the trunk on the premises2. A certificate-based trunk requires a certificate authority (CA) to issue and manage the certificates for both Webex Calling and the Local Gateway1.

NEW QUESTION 6
An engineer configures a SIP trunk for MWI between a Cisco UCM cluster and Cisco Unity Connection. The Cisco UCM cluster fails to receive the SIP notify messages. Which two SIP trunk settings resolve this issue? (Choose two.)

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

Answer: AE

NEW QUESTION 7
Callers from a branch report getting busy tones intermittetly when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco CallManager service traes to examine the situation. The traces show:

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

What should be fixed to resolve the issue?

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

Answer: B

NEW QUESTION 8
An engineer must configure a SIP route pattern using domain routing with destination +13135551212. The domain ciscocm1.jupiter.com resolves to 192.168.1.3.

How must the IPV4 Pattern be configured?

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

Answer: B

NEW QUESTION 9
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 10
Which configuration concept allows for high-availability on IM and Presence services in a UC environment?

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

Answer: D

NEW QUESTION 10
An engineer is configuring Cisco Jabber for Windows and must implement desk phone control mode for some of the users. Which access control group must be configured for those users to enable this functionality?

A. Allow Control of Device from CTI
B. Standard CTI Secure Connection
C. Standard CTI Enabled
D. Standard CTI Allow Reception of SRTP Key Material

Answer: C

NEW QUESTION 11
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 16
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 20
Which two features of Cisco Prime Collaboration Assurance require advanced licensing? (Choose two.)

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

Answer: BC

NEW QUESTION 25

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

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

Answer: AB

NEW QUESTION 27
During the Cisco IP Phone registration process, the TFTP download (ails. What are two reasons (or 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 29
A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?

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

Answer: A

NEW QUESTION 31
Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?

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

Answer: A

NEW QUESTION 33
An employee of company ABC just quit. The IT administrator deleted the employee's user id from the active directory at 10 a m. on March 4th The nightly sync occurs at 10 p.m. daily. The IT administrator wants to troubleshoot and find a way to delete the user id as soon as possible How is this issue resolved?

A. Wart until 10 pm on March 4th when the user is automatically removed from Cisco UCM.
B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
C. Wait until 3 15 a.
D. on March 6th for garbage collection to remove the user from Cisco UCM.
E. Wait until 315am on March 5th for garbage collection to remove the user from Cisco UCM.

Answer: C

NEW QUESTION 35
Refer to the exhibit.

```
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F
From:
<sip:012345678@10.10.10.2>;tag=8D79AF62-DB2
To: <sip:90123456@10.10.4.14>;
tag=811681~ffa80926-5fac-4cc5-b802-
2dbde74ae7w2-
v=0
o=CiscoSystemsCCM-SIP 811681 1 IN IP4
10.10.4.14
s=SIP Call
c=IN IP4 10.5.4.3
t=0 0
m=audio 27839 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Which Codec is negotiated?

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

Answer: C

NEW QUESTION 38
An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

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

Answer: A

NEW QUESTION 39
Which location must be assigned to the SIP trunk to replicate enhanced location information via a SIP trunk?

A. phantom
B. replica
C. hub_none
D. shadow

Answer: D

NEW QUESTION 42
An administrator is configuring a new Cisco UCM with PSTN capabilities. Due to bandwidth constraints, audio compression is used on the codec. DTMF must work as expected because the customer is calling many call centers where the users must select options in the call. Where is DTMF out-of-band in a CCM 12.5 with SIP-based gateway configured?

A. in the DTMF setting under SIP profile on the Cisco Unified Border Element
B. in the dial peer on the Cisco IOS router
C. in regions on the Cisco UCM where the appropriate codec to use is set
D. in DTMF settings in the audio codec preference list under regions in the Cisco UCM

Answer: B

NEW QUESTION 43
A collaboration engineer adds a voice gateway to Cisco UCM. The engineer creates a new gateway device in Cisco UCM. selects VG320 as the device type and selects MGCP as the protocol What must be done next to add the gateway to the Cisco UCM database?

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

Answer: C

NEW QUESTION 48

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

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

Answer: BE

NEW QUESTION 51

A company wants to provide remote user with access to its premises Cisco collaboration features. Which components are required to enable cisco mobile and remote access for the users?

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

Answer: B

NEW QUESTION 56

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 59

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

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

Answer: B

NEW QUESTION 61

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

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

Answer: CD

NEW QUESTION 62

An engineer roust deploy the Cisco Wet*x app to a Windows Virtual Desktop Infrastructure environment that has a roaming database named spark roaming_store stored In a user's AppData\Roaming directory, Which two command line arguments must be used when running the installer? (Choose two.)

A. ALLUSERS=0
B. ENABLEVDI=1
C. ALLUSERS=1
D. ENABLEVDI=2
E. ROAMINGENABLED=1

Answer: BE

Explanation:
The Cisco Webex app can be installed on a Windows Virtual Desktop Infrastructure (VDI) environment by using the following command-line arguments:

> ENABLEVDI=1 - This argument enables VDI mode for the Webex app.
> ROAMINGENABLED=1 - This argument enables roaming for the Webex app.

The ALLUSERS argument is not required when installing the Webex app on a VDI environment. The ENABLEVDI argument must be set to 1, and the ROAMINGENABLED argument must be set to 1.
The following is an example of the command that can be used to install the Webex app on a VDI environment:
Code snippet
msiexec /i WebexApp.msi ENABLEVDI=1 ROAMINGENABLED=1

NEW QUESTION 66
An engineer implements QoS in the enterprise network. Which command is used to verity 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 67
A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

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

Answer: B

NEW QUESTION 70
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 74
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 76
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 78
A Cisco UCM administrator sets up new route patterns to support phones in tour different locations, all with local gateways. The administrator wants to use the same route pattern for all four locations. How must the system be configured to achieve this goal?

A. Use CSS alternate routing rules.
B. Use standard local route groups.
C. Add a CSS to each local gateway.
D. Use transforms in the route groups.

Answer: B

NEW QUESTION 83
What is required for Cisco UCM to accept SIP calls with a URI in the format of 'sip:2001@cucmpub.cisco.com'?

A. Define Cluster Fully Qualified Domain Name under Servers in Cisco UCM.
B. Change the Destination Address to a Fully Qualified Domain Name on the SIP trunk.
C. Define Cluster Fully Qualified Domain Name in Enterprise Parameters.
D. Set the SIPS URI Handling to True in CallManager Service Parameters.

Answer: C

NEW QUESTION 85
Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?

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

Answer: D

NEW QUESTION 90
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 91
Refer to the exhibit.

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

The translation rule is configured on the voice gateway to translate DNIS. What is the outcome if the gateway receives 0255-343-1234 as DNIS?

A. The translation rule is not matched because DNIS does not end with a "$".
B. The translation rule is matched and the translated number is 02553431234.
C. The translation rule is matched and the translated number is 025553431234.
D. The translation rule is not matched because DNIS contains"-".

Answer: B

NEW QUESTION 94
An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

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

Answer: C

NEW QUESTION 98
A customer asked to integrate Unity Connection with Cisco UCM using SIP protocol. Which two features must be enabled on SIP security profiles? (Choose two.)

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

Answer: CE

NEW QUESTION 103
Refer to the exhibit.

```
hostname GATEWAY
ccm-manager config
ccm-manager config server 192.168.1.100
ccm-manager mgcp

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

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

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

Answer: D

NEW QUESTION 108
What is the traffic classification for voice and video conferencing?

A. Voice is classified as CoS 4, and video conferencing is CoS 5.
B. Voice and video conferencing are both classified is CoS 3.
C. Voice is classified as CoS 5, and video conferencing is CoS 4.
D. Video conferencing is classified as CoS 1, and voice is CoS 2.

Answer: B

NEW QUESTION 111
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 114
A collaboration engineer configures Global Dial Plan Replication for multiple Cisco UCM clusters. The local cluster acts as the hub cluster, and the remaining clusters act as spoke clusters Which service must the engineer configure on the local cluster'

A. Intercluster Lookup Service
B. Location Conveyance on intercluster SIP trunks
C. Intra-Cluster Communication Signaling
D. Mobility Cross Cluster

Answer: A

NEW QUESTION 119
Which Cisco Unified communications manager configuration is required for SIP MWI integration?

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

Answer: D

NEW QUESTION 123
Refer to the exhibit.

```
ROUTER-1(config)# policy-map LLQ_POLICY
ROUTER-1(config-pmap)# class VOICE
ROUTER-1(config-pmap-c)# bandwidth 170
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# class VIDEO
ROUTER-1(config-pmap-c)# bandwidth remaining percent 30
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# exit
```

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

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

Answer: C

NEW QUESTION 126
An administrator executes the debug isdn q931 command while debugging a failed call. After a test call is placed, the logs return a disconnect cause code of 1. What is the cause of this problem?

A. The media resource is unavailable.
B. The destination number rejects the call.
C. The destination number is busy.
D. The dialed number is not assigned to an endpoint.

Answer: D

NEW QUESTION 129
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 endpoin
E. calls fail.

Answer: C

NEW QUESTION 131
End users report bad video quality and voice choppiness on Cisco Collaboration endpoints. The engineer changed the device pool the users were in but did not correct the problem. Which action should be taken to troubleshoot this issue?

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

Answer: D

NEW QUESTION 132
Which two recommendations are made to optimize Cisco UCM configuration to reduce the number of toll fraud incidents in an organization? (Choose two.)

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

Answer: BE

NEW QUESTION 134
Which Cisco UCM configuration is required for SIP MWI integrations?

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

Answer: C

NEW QUESTION 136
When a call Is delivered to a gateway, the calling and called party number must be adapted to the PSTN service requirements of the trunk group. If a call is destined locally, the + sign and the explicit country code must be replaced with a national prefix. For the same city or region, the local area code must be replaced by a local prefix as applicable. Assuming that a Cisco UCM has a SIP trunk to a New York gateway (area code 917). which two combinations of solutions localize the calling and called party for a New York phone user? (Choose two.)

A.
Configure two calling party transformation patterns:
    \+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
    \+1.!, strip pre-dot, numbering type: national

B.

Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling nu

```
!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate called 1
!
```

C. Configure the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called nu

```
!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate calling 1
!
```

D. Configure two called party transformation patterns:
\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
\+1.!, strip pre-dot, numbering type: national

E. Configure two calling party transformation patterns:
\+1917.CCCCCC, strip pre-dot, numbering type: subscriber
\+!, strip pre-dot, numbering type: national

Answer: BC

NEW QUESTION 137
An engineer builds the configuration on a Cisco IOS gateway for the dial-peers:
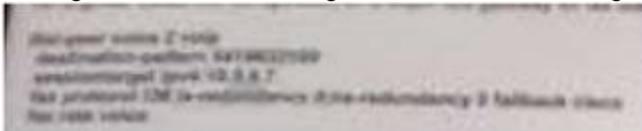
Which command is required to complete the configuration?

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

Answer: D

NEW QUESTION 139
An administrator is developing an 8-class QoS baseline model. The CS3 standards-based marking recommendation is used for which type of class?

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

Answer: A

NEW QUESTION 141
Refer to the exhibit.
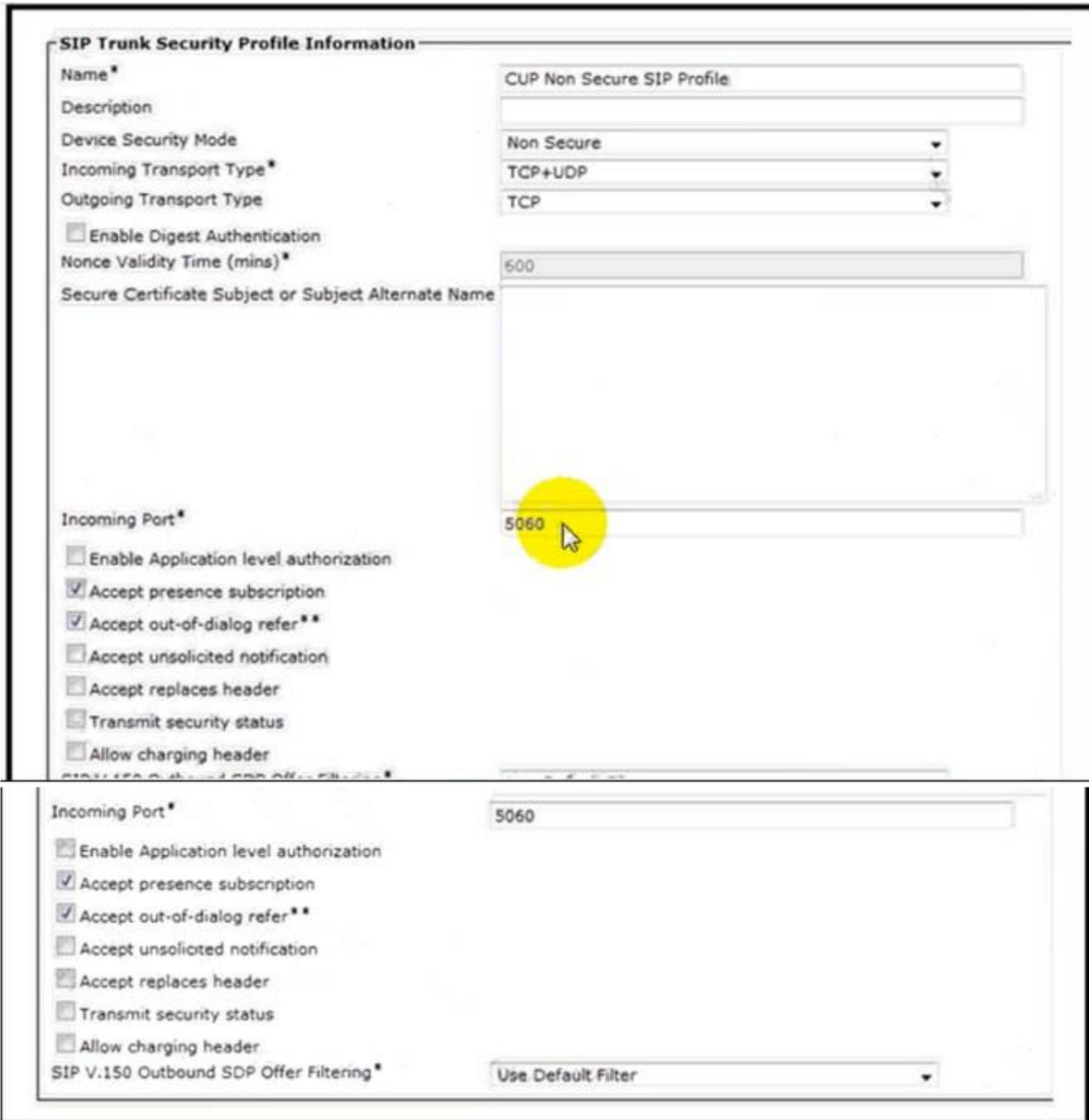
A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

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

Answer: CE

NEW QUESTION 146
Refer to the exhibit.

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

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

Answer: B

NEW QUESTION 151
An engineer troubleshoots outbound can failure on an ISDN-PRI circuit. The engineer ts 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 152
An engineer troubleshoots a Cisco Jabber login problem on a Windows PC. The login fails with the error message "Cannot find your services automatically. Click
advanced settings to set up manually " Which action should the engineer take first?

A. Verify whether the cup-xmpp certificates are valid.
B. Verify the username and password and try again.
C. Perform a manual DNS lookup of SRV record _cisco-uds._tcp.domain.com.
D. Perform a manual DNS lookup of SRV record _collab-edge._tls.domain.com.

Answer: C

NEW QUESTION 156
Refer to the exhibit.
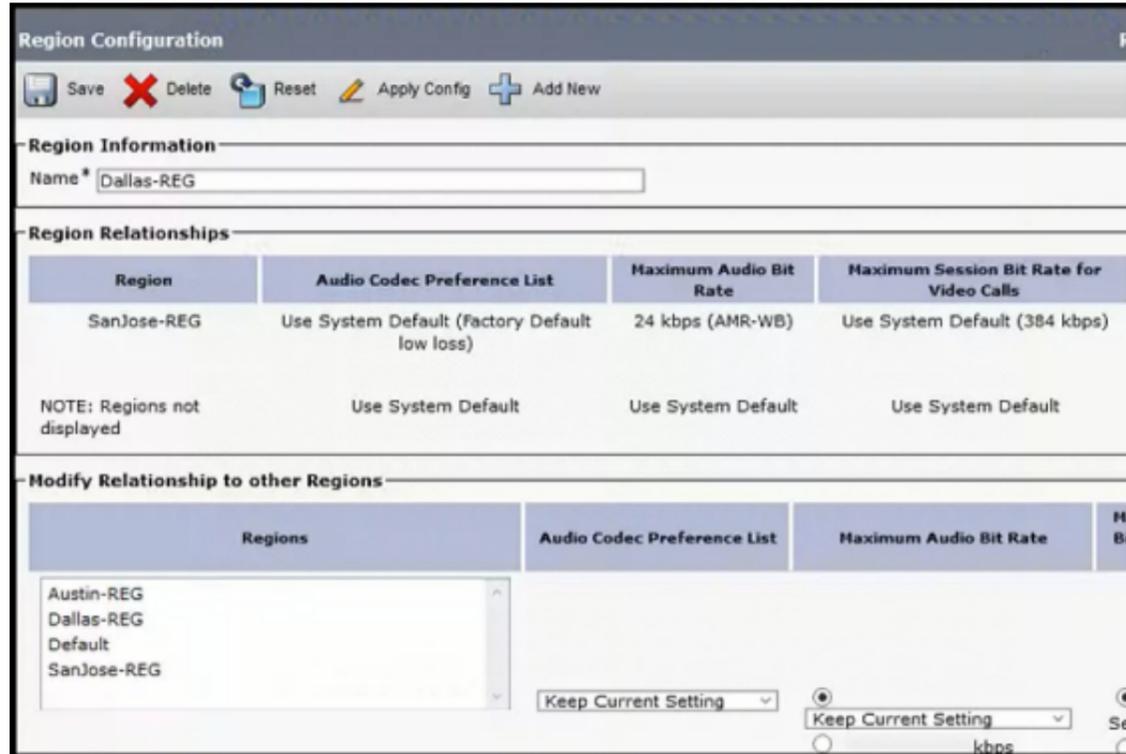
Which codec should an engineer select for a call made between "Dallas-REG' & "Austin-REG'?

A. MP4A-LATM
B. G.711
C. OPUS
D. G.729

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:

≫ 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.

NEW QUESTION 161
An engineer must enable onboarding of on-premises devices by using activation to a Cisco UCM server The engineer activated the CISCO Device Activation Service and set the default registration method to use the codes. Which action completes the configuration?

A. Set Enable Activation Code enterprise parameter to True
B. Manually provision new phones that have an activation code requirement
C. Create a Bulk Administration Tool provisioning template.
D. Generate 16-digit codes by using the Bilk Administration Tool

Answer: A

Explanation:
The engineer must set the Enable Activation Code enterprise parameter to True. This will enable the use of activation codes for onboarding on-premises devices to a Cisco UCM server. The other options are not necessary to complete the configuration.
Here are the steps to complete the configuration:

≫ Log in to the Cisco Unified Communications Manager (CUCM) Administration interface.

≫ Go to System > Enterprise Parameters.

≫ Set the Enable Activation Code enterprise parameter to True.

≫ Click Save.

The activation code onboarding feature is now enabled. You can use it to onboard new phones to the CUCM server.

NEW QUESTION 165
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 168
Refer to the exhibit.
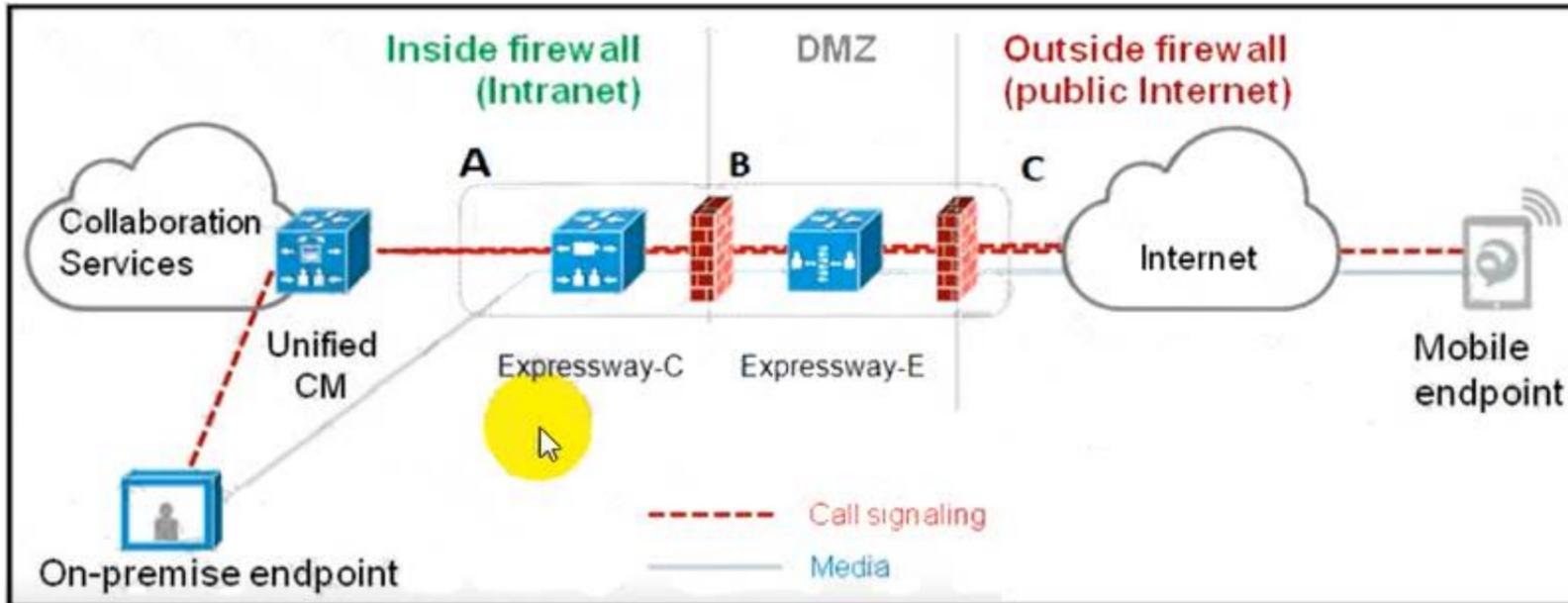
When making a call to a Mobile and Remote Access client, what are the combinations of protocol on each of the different sections A-B-C?

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 173
Refer to the exhibit.

```
!
voice service voip
 ip address trusted list
   ipv4 192.168.100.101
   ipv4 192.168.101.0 255.255.255.128
!
dial-peer voice 1 voip
 destination-pattern +T
 session protocol sipv2
 session target ipv4:192.168.102.102
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

When a call is received on Cisco Unified Border Element. from which IP does it allow a connection?

A. 192.168.100.103
B. 192.168.102.102
C. 192.168.100.102
D. 192.168.101.201

Answer: B

NEW QUESTION 174

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 175
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 176
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.

B.

Pattern Definition

Pattern*	\+.!
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >

☑ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

C. **Pattern Definition**

Pattern*	\+.!
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >

☑ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits	PreDot
Called Party Transformation Mask	
Prefix Digits	9011
Called Party Number Type*	International
Called Party Numbering Plan*	ISDN

D. **Pattern Definition**

Pattern*	\+.!
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >

☑ Urgent Priority

☐ 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 180
Refer to the exhibit.

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

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

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

Answer: C

NEW QUESTION 182
A Cisco Telepresence SX80 suddenly has issues displaying main video to a display over HDMI. Which command can you use from the SX80 admin CLI to check the video output status to the monitor?

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

Answer: B

NEW QUESTION 183
An engineer is asked to implement on-net/off-net call classification in Cisco UCM. Which two components are required to implement this configuration? (Choose two.)

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

Answer: DE

NEW QUESTION 188
Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...

     remote           refid     st t when poll reach  delay   offset   jitter
==============================================================================
*192.168.1.1     17.253.14.125   2 u  36   64   377   0.435   0.039    0.047
 192.168.1.2     .INIT.         16 u   -   64     0   0.000   0.000    0.000
```

A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution On the Cisco UCM OS Administration page, the new NTP server shows as "Not Accessible" Which action resolves this issue?

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

Answer: C

NEW QUESTION 189
Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

A. Endpoint>Expressway-C>Expressway-E>Cisco UCM
B. Endpoint>Expressway-E>Expressway-C> Cisco UCM
C. Endpoint>Expressway-E> Cisco UCM
D. Endpoint>Expressway-C> Cisco UCM

Answer: A

Explanation:
The call flow for a Mobile and Remote Access registered endpoint to central call control is as follows:
≫ The endpoint registers with the Expressway-C.
≫ The Expressway-C forwards the registration request to the Expressway-E.
≫ The Expressway-E forwards the registration request to the Cisco UCM.
≫ The Cisco UCM registers the endpoint.
When the endpoint places a call, the call flow is as follows:
≫ The endpoint sends the call request to the Expressway-C.
≫ The Expressway-C forwards the call request to the Expressway-E.
≫ The Expressway-E forwards the call request to the Cisco UCM.
≫ The Cisco UCM places the call.
The Expressway-C and Expressway-E are used to provide secure access to the Cisco UCM for endpoints that are not located on the corporate network. The Expressway-C is located on the corporate network, and the Expressway-E is located in the DMZ.

NEW QUESTION 194
What are the last two bits of a DS field in DiffServe Byte used for?

A. INC
B. AFxy
C. ECN
D. RMI

Answer: C

NEW QUESTION 197
An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.7llulaw and G.729 for audio and H.264 for video. The preferred voice codec is G.711. Which configuration meets this requirement?

A.
```
voice class codec 10

  codec preference 1 g711ulaw

  codec preference 2 g729r8


dial-peer voice 101 voip

  session protocol sipv2

  destination e164-pattern-map 1

  voice-class codec 10
```

B.

```
voice class codec 10
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 10
```

C.
```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 100
```

D.
```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 10
```

Answer: D

NEW QUESTION 200
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, gc:1, s:on, t:10, 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, gc:1, s:on, t:10, 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, gc:1, s:on, t:10, 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, gc:1, s:on, t:10, 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, gc:1, s:on, t:10, 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, gc:1, s:on, t:10, 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, gc:1, s:on, t:10, r:g, nt:IN, v:T;G;D;L;H;R;ATM;SST;PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, 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 202
Which action prevents toll fraud in Cisco UCM?

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

Answer: D

NEW QUESTION 203
Which call routing pattern is used for phone numbers that are in the E.164 format?

A. /+.! Route Pattern
B. \+.! Route pattern
C. \+.! Translation Pattern
D. \+1.[2-9]XX[2-9]XXXXXXX called Party Transformation Pattern

Answer: B

NEW QUESTION 204
Users want their mobile phones to be able to access their cisco unity connection mailboxes with only having to enter their voicemail pin at the login prompt calling pilot number where should an engineer configure this feature?

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

Answer: C

NEW QUESTION 209
A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?
M=audio 498181 RTP/AVP 0 8 97

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

Answer: A

Explanation:
The SDP offer includes the following line of text: M=audio 498181 RTP/AVP 0 8 97
This line of text indicates that the following codecs are available:

> 0: G.711ulaw

> 8: G.711alaw

> 97: iLBC

The Cisco IP Phone 7841 is registered to a Cisco Unified Communications Manager with default configuration. This means that the phone will negotiate the G.711ulaw codec.
The G.711ulaw codec is a standard codec that is used for voice communication. It is a low-bandwidth codec that provides good quality.
The iLBC codec is a newer codec that is designed for use in low-bandwidth environments. It provides good quality, but it is not as widely supported as the G.711ulaw codec.
The G.722 codec is a high-quality codec that is used for voice communication. It provides excellent quality, but it requires more bandwidth than the G.711ulaw codec.

NEW QUESTION 212
......

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