ActualTest_400-051-GoNawazGo-280Q-15Jun2015

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Wish you all the best.

Go Nawaz Go

Exam A

QUESTION 1
1
Company ABC is planning to migrate from MCS-hosted Cisco Unified Communications Manager applications to Cisco UC on UCS B-Series servers. Which statement about installation media support is true for this migration?

A. The install log can be written to a USB flash drive that is attached to the UCS server.
B. The answer file that is generated by the Answer File Generator (platformConfig.xml) can be read from a USB flash drive to perform an unattended installation on the UCS server.
C. The Cisco Music on Hold USB audio sound card can be mapped to a virtual USB port on a VMware virtual machine on the UCS server.
D. The answer file that is generated by the Answer File Generator (platformConfig.xml) can be read from an FLP image that is mounted in a virtual floppy drive.
E. The Cisco Music on Hold USB audio sound card can be mapped to a virtual serial port on a VMware virtual machine on the UCS server.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Using the AFG will allow you to get this license mac before even touching the server. It is provided after filling in the main form of the AFG but it can also be found by looking at the last few lines of your platformconfig.xml file.
Once you have the xml files, you will need to map those to the floppy drive of the VM (no usb support on the VM OVA). There are many ways to do this. I simply use a freeware virtual floppy app that I drop the platformconfig.xml file on and then copy the*.flp image out to the datastore. I'll end up with a directory on my datastore called AFG that has the host named *.flp images that I will use during install. It also serves as archival of these files in the event the server needs to be reimaged. This is important because the license mac will change if every parameteris not entered exactly as it was prior. If the license mac changes, you will have to go through the process of requesting new license files to be generated.
Reference:http://angryciscoguy.com/jello/cisco-answer-file-generator-to-the-rescue/

QUESTION 2
2
Which statement about the Cisco UC on UCS TRC and the third-party server specs-based virtualization support model is true?

A. Both the UC on UCS TRC and the third-party servers spec-based support models have rulebased approaches.
B. The UC on UCS TRC support model has a rule-based approach and the third-party servers spec-based support model has a configuration-based approach.
C. The UC on UCS TRC support model requires a high level of virtualization experience while the third-party server spec-based support model requires a low to medium level virtualization experience.
D. VMware vCenter is mandatory for the UC on UCS TRC support model but it is optional for the third-party server spec-based support model.
E. VMware vCenter is optional for the UC on UCS TRC support model but it is mandatory for the third-party server spec-based support model.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
VMware vCenter is
Reference:http://docwiki.cisco.com/wiki/Unified_Communications_VMware_Requirements

QUESTION 3
3
Which definition is included in a Cisco UC on UCS TRC?

A. storage arrays such as those from EMC or NetApp, if applicable
B. configuration of virtual-to-physical network interface mapping
C. step-by-step procedures for hardware BIOS, firmware, drivers, and RAID setup
D. server model and local components (CPU, RAM, adapters, local storage) at the part number level
E. configuration settings and patch recommendations for VMware software

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
What does a TRC definition include?
Reference:http://docwiki.cisco.com/wiki/UC_Virtualization_Supported_Hardware#UC_on_UCS_Tested_Reference_Configurations

QUESTION 4
4
Which capability is supported by Cisco Discovery Protocol but not by LLDP-MED?

A. LAN speed and duplex discovery
B. Network policy discovery
C. Location identification discovery
D. Power discovery
E. Trust extension

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Discovery Protocol provides an additional capability not found in LLDP-MED that allows the switch to extend trust to the phone. In this case, the phone is now trusted to mark the packets received on the PC port accordingly. This feature can be used to off-load the switch because now it does not need to police the information being received from the phone.

QUESTION 5
5
Which two mechanisms does Cisco EnergyWise use for neighbor discovery? (Choose two.)

A. multicast
B. LLDP-MED
C. UDP broadcast
D. Cisco Discovery Protocol
E. TCP

Correct Answer: CD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco EnergyWise Neighbor Discovery Process
The Cisco EnergyWise neighbor discovery process is the mechanism by which domain members discover each other and populate their Cisco EnergyWise neighbor tables. Cisco EnergyWise

queries can subsequently be distributed to all domain members using the neighbor relationships to monitor and control the power usage of devices within a domain. Cisco EnergyWise domain members automatically discover their neighbors through one of two mechanisms:
•Cisco EnergyWise UDP broadcast packet
•Cisco EnergyWise CDP packets
UDP broadcast packets are automatically sent out switch ports which support Cisco EnergyWise, regardless of whether the interfaces are configured with the no energywise interface-level command. CDP packets are sent when CDP is configured for the switch ports.
Reference:http://www.cisco.com/en/US/docs/solutions/Enterprise/Borderless_Networks/Energy_Management/energywisedg.html?referring_site=smartnavRD#wp555927

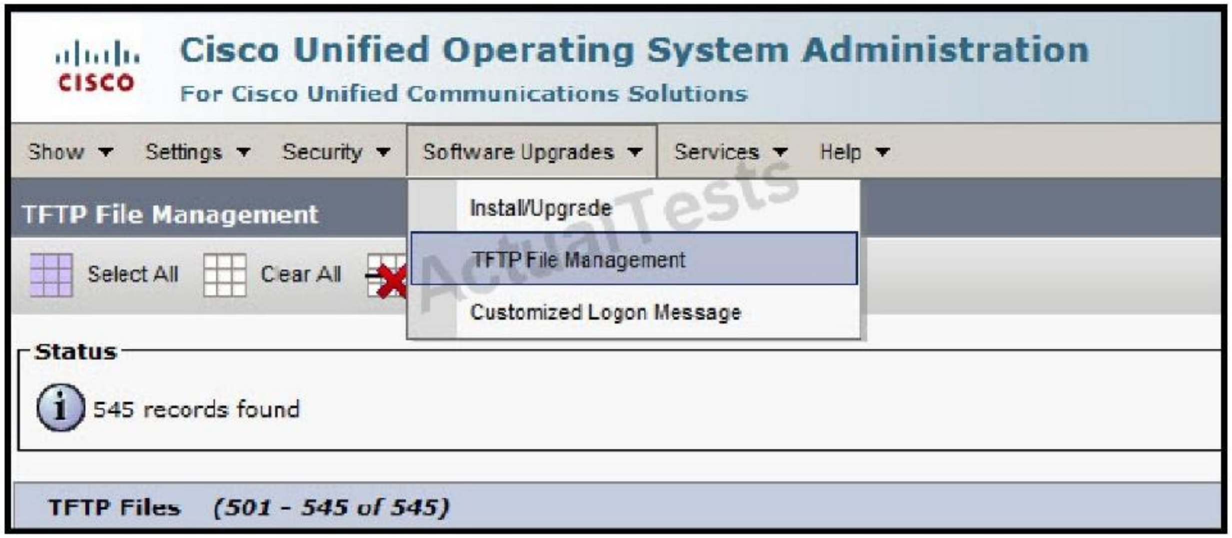
QUESTION 6
6
Which protocol does the Cisco Prime LAN Management Solution application use to communicate with Cisco EnergyWise domain members?

A. UDP broadcast
B. Cisco Discovery Protocol
C. UDP unicast
D. TCP
E. multicast

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Prime LMS 4.1 uses TCP port 43440.

QUESTION 7
7
Refer to the exhibit.



Assuming that the administrator has never performed any manual custom uploads, which two file types can be found when you choose Software Upgrades, followed by TFTP File Management on the Cisco Unified Operating System Administration web page? (Choose two.)

A. IP phone configuration files
B. sample music-on-hold audio files
C. Identity Trust List files
D. IP phone license files
E. Mobile Voice Access audio files
F. softkey template files
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Correct Answer: CE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
We get option for Identity Trust list Files and Mobile Voice Access audio files.

QUESTION 8
8
Which statement describes a disadvantage of using the Cisco TFTP service to serve IP phone load files?

A. The Cisco TFTP services can run on only one Cisco Unified Communications Manager server in a cluster.
B. Because TFTP operates on top of UDP, there is a high risk of corrupted load file delivery at the completion of the TFTP process due to undetected data loss in the network.
C. If a response is not received in the timeout period, the TFTP server will not resend the data packet.
D. Packet loss can significantly increase the TFTP session completion time.
E. Because TFTP operates with an adaptive timeout period, the time to complete the file transfer is unpredictable.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Voice traffic cannot recapture lost packets. Rather than retransmitting a lost network connection, the phone resets and attempts to reconnect its network connection.
Reference:http://www.cisco.com/en/US/docs/voice_ip_comm/cuipph/6921_6941_6961/7_1_2/english/admin/guide/6921trb.html#wp1031181

QUESTION 9
9
Which two statements about using the Load Server option for IP phone firmware distribution are true? (Choose two.)

A. This option must be enabled on at least two servers in a Cisco Unified Communications Manager cluster.
B. This option must be enabled on Cisco Unified Communications Manager service parameters for Cisco TFTP.
C. Phone firmware must be manually copied to any applicable load servers.
D. The load server will not function if its IP address is not in the same subnet as the IP phones.
E. This option is only available for newer IP phone models.
F. This option does not accommodate falling back to Cisco TFTP on error.

Correct Answer: CF
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Choosing the Right Distribution Method
Which of the three different image-distribution methods discussed so far is the best for a customer deployment? Each method has advantages and disadvantages, and they are summarized in Table 1.
Table 1. Summary of Distribution Models
Peer Firmware Sharing
Load Server
Traditional TFTP
Advantages
• Hierarchy is automatic
• One download per phone model on asubnet
• Uses TCP
• Fails back to TFTP
• Speeds up LAN upgrades
• Reduces TFTP CPU load during upgrade
• Has same download time as LAN image distribution
• Distributes TFTP load over multiple TFTPservers
• Proven distribution
• Default behavior
Disadvantages
• Must be enabled on each phone
• Hierarchy is formed for each phone model
• Hierarchy is limited to subnet
• IP must be set on each phone
•Administrator must manually file copy to load server
• No failback to TFTP on error
• More prone to user error
• High-bandwidth requirements
• Multiple requests for same file
• High CPU usage on TFTP server
Reference:http://www.cisco.com/en/US/prod/collateral/voicesw/ps6882/ps6884/white_paper_c11-583891.pdf

QUESTION 10
10
Which two statements about the Peer Firmware Sharing option for IP phone firmware distribution are true? (Choose two.)

A. This option uses a parent-child hierarchy in which a firmware image is downloaded by a parent phone to up to three directly associated child phones.
B. This option must be enabled on Cisco Unified Communications Manager service parameters for

Cisco TFTP.

C. This option mandates that the parent phone and child phones be identical, selected phone models.

D. This option allows firmware transfers between phones in different subnets, as long as the round-trip delay is less than 5 milliseconds.

E. This option uses a parent-child hierarchy that must be manually defined by the Cisco Unified Communications Manager administrator.

F. This option allows falling back to the TFTP server in the Cisco Unified Communications Manager cluster.

Correct Answer: CF
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Peer Firmware Sharingworks by setting up a parent-child hierarchy of the phones in which a firmware image is downloaded by the parent phone to a child phone. The advantage of using Peer Firmware Sharing is that instead of all phones individually retrieving a software image, they pass the image along from one phone to another phone on the same subnet.
Advantage of PFS:
• Hierarchy is automatic
• One download per phone model on a subnet
• Uses TCP
• Fails back to TFTP
• Speeds up LAN upgrades
• Reduces TFTP CPU load during upgrade

QUESTION 11
11
Which two statements about the Cisco UC on UCS specs-based virtualization support model are true? (Choose two.)

A. It has a configuration-based approach.

B. It has a rule-based approach.

C. It has less hardware flexibility compared to the third-party server specs-based support model.

D. It has less hardware flexibility compared to the UC on UCS TRC support model.

E. VMware vCenter is optional with this support model.

Correct Answer: BC
Section: (none)

Explanation

Explanation/Reference:
Reference:http://docwiki.cisco.com/wiki/UC_Virtualization_Supported_Hardware#UC_on_UCS_Te
sted_Reference_Configurations

QUESTION 12
12
Which definition is included in a Cisco UC on UCS TRC?

A. required RAID configuration, when the TRC uses direct-attached storage
B. configuration of virtual-to-physical network interface mapping
C. step-by-step procedures for hardware BIOS, firmware, drivers, and RAID setup
D. configuration settings and patch recommendations for VMware software
E. server model and local components (CPU, RAM, adapters, local storage) by name only; part numbers are not included because they change over time

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Definition of server model and local components (CPU, RAM, adapters, local storage) at the orderable part number level.

QUESTION 13
13
Which capability is support by LLDP-MED but not by Cisco Discovery Protocol?

A. LAN speed discovery
B. network policy discovery
C. location identification discovery
D. power discovery
E. trust extension

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

Explanation:
LLDP-MED supports both LAN speed and duplex discovery. Cisco Discovery Protocol supports duplex discovery only, but this limited support is not seen as a problem because if there is a speed mismatch, LLDP-MED and Cisco Discovery Protocol cannot be exchanged and thus cannot be used to detect the mismatch.

QUESTION 14
14
In a Cisco EnergyWise domain, which two terms describe a Cisco IP phone? (Choose two.)

A. endpoint
B. domain member
C. child domain member
D. EnergyWise agent
E. Cisco power distribution unit

Correct Answer: AC
Section: (none)
Explanation

Explanation/Reference:
Reference:http://www.cisco.com/en/US/docs/switches/lan/energywise/phase2_5/ios/configuration/guide/one_ent.html

QUESTION 15
15
Which statement about Cisco EnergyWise domain member neighbor formation is true?

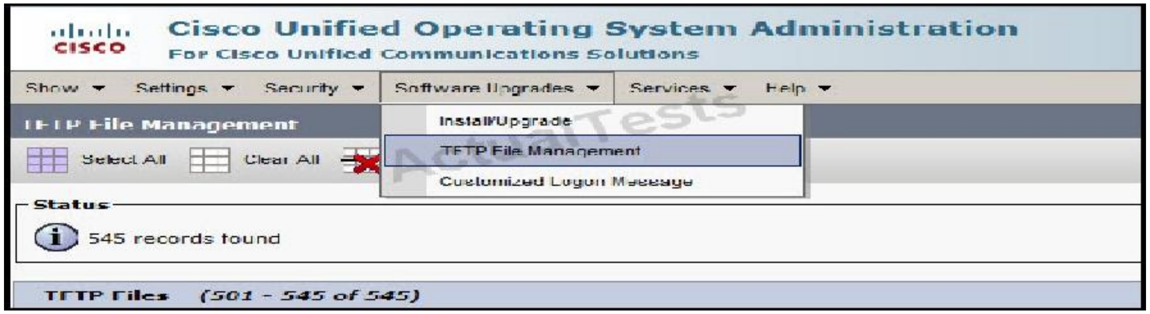
A. Cisco EnergyWise supports static neighbors, but the neighbor relationship is only possible if a noncontiguous domain member and a contiguous domain member have a static neighbor entry pointing to each other.
B. Cisco EnergyWise static neighbors can be formed even if domain members are not physically contiguous.
C. Static neighbors can be manually defined on Cisco EnergyWise domain members, but TCP protocols must be used.
D. Static neighbors can be manually defined on Cisco EnergyWise domain members, but they have a lower priority compared to the autodiscovered members.
E. Static neighbors can be manually defined on Cisco EnergyWise domain members and the TCP or UDP protocol can be used.

Correct Answer: B

Section: (none)
Explanation

Explanation/Reference:
Reference:http://www.cisco.com/en/US/docs/solutions/Enterprise/Borderless_Networks/Energy_Management/energywisedg.html?referring_site=smartnavRD#wp554384

QUESTION 16
16
Refer to the exhibit.



Assuming that the administrator has never performed any manual custom uploads, which two file types can be found when you choose Software Upgrades, followed by TFTP File Management on the Cisco Unified Operating System Administration web page for Cisco Unified Communications Manager? (Choose two.)

A. IP phone configuration files
B. announcement audio files
C. ringer files
D. IP phone license files
E. sample music-on-hold audio files
F. softkey template files

Correct Answer: BC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The two file types that we get are Announcement Audio Files and Ringer Files.

QUESTION 17
17
Which four attributes are needed to determine the time to complete a TFTP file transfer process? (Choose four.)

A. file size
B. file type
C. network interface type
D. round-trip time
E. packet loss percentage
F. response timeout
G. network throughput

Correct Answer: ADEF
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Four attributes that are needed to determine the time to complete TFTP file transfer process is:
Reference:http://www.cisco.com/en/US/prod/collateral/voicesw/ps6882/ps6884/white_paper_c11-583891_ps10451_Products_White_Paper.html

QUESTION 18
18
What is the maximum number of call-processing subscribers in a standard deployment of a Cisco Unified Communications Manager Session Management Edition cluster?

A. 3
B. 4
C. 5
D. 8
E. 16

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:

There is no deployment difference between CUCM & CUCM session management Edition cluster.The only difference is that CUCM SME is designed to support a large number of trunk to trunk connections.Thus,8 subscribers.
Topic 2, Telephony Standards and Protocols

QUESTION 19
19
Which two SCCP call signaling messages are initiated by Cisco Unified Communications Manager to an IP phone? (Choose two.)

A. SoftKeyEvent
B. CloseReceiveChannelAck
C. CallState
D. KeypadButton
E. OpenReceiveChannel
F. Offhook

Correct Answer: CE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Upon receiving anOpenReceiveChannelmessage, the I phone selects the UDP port number it wants to use to receive RTP packets and reports this information to call manager.
With the SCCP protocol architecture, the majority of the H.323 processing power resides in an H.323 proxy — the Cisco CallManager. The end stations (IP phones) run the Skinny client, which consumes less processing overhead. The client communicates with CallManager using connection-oriented (TCP/IP-based) communication to establish a call with another H.323-compliant end station. Once Cisco CallManager has established the call, the two H.323 end stations use connectionless (UDP/IP-based) communication for audio transmissions.

QUESTION 20
20
Which two SCCP call signaling messages are sent by an IP phone to Cisco Unified Communications Manager? (Choose two.)

A. SoftKeyEvent
B. OpenReceiveChannelAck
C. StartMediaTransmission
D. SelectSoftKeys

E. CloseReceiveChannel
F. StopTone

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
This message indicates which soft key was pressed. Upon receipt of this message, CallManager invokes the action associated with the pressed soft key. For example, ifHoldwas the pressed soft key, CallManager places the active call on user hold. In some trace files you might see a soft key number without the corresponding description. The following list defines each soft key number.

QUESTION 21
21
Which device is the initiator of a StationInit message in a Cisco Unified Communications Manager SDI trace?

A. Cisco Unified Communications Manager
B. MGCP gateway
C. Cisco Music on Hold server
D. SCCP IP phone
E. SIP Proxy Server

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Station Initmeans that an inbound Transmission Control Protocol (TCP) message from a Skinny station reached CallManager. A Skinny station is any endpoint that uses the Skinny protocol to communicate with CallManager.

QUESTION 22
22
Refer to the exhibit.

```
Jan 19 17:02:49.998: MGCP Packet received from 10.1.1.1:2427--->
CRCX 83 S0/SU1/DS1-0/2@Router1.CiscoCustomer.com MGCP 0.1
C: D0000000027aef6f000001F500000003
X: 17
L: p:20, a:PCMU, s:off, t:b8, fxr/fx:t38
M: recvonly
R: D/[0-9ABCD*#]
Q: process,loop
```

You received this debug output to troubleshoot a Cisco IOS MGCP gateway problem at a customer site. Which statement about this endpoint on the Cisco MGCP gateway is true?

A. This endpoint is on a T1 Controller 0/1/0.
B. This endpoint is on an E1 Controller 0/1/0.
C. This endpoint is on a T1 Controller 0/1/1.
D. This endpoint is on an E1 Controller 0/1/2.
E. This endpoint is on an T1 Controller 0/1/2.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The s0/Su1/DS1-0 refers to the slot and port information (0/1/0). It is also a DS1 as shown by this output, which means it is a T1 not an E1.

QUESTION 23
23
Refer to the exhibit.

```
Jul 31 18:36:33.201: MGCP Packet sent to 10.1.1.1:2427--->
250 125 OK
P: PS=76, OS=12160, PR=75, OR=12000, PL=0, JI=7, LA=0
```

You received this debug output to troubleshoot a Cisco IOS MGCP gateway media-related problem at a customer site. What is the purpose of this message?

A. The MGCP gateway is responding to an RQNT message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.
B. The MGCP gateway is responding to an AUEP message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.
C. The MGCP gateway is responding to an AUCX message from Cisco Unified Communications Manager to poll the active calls on its endpoints.

D. The MGCP gateway is responding to an MDCX message from Cisco Unified Communications Manager during a call setup.
E. The MGCP gateway is responding to a CRCX message from Cisco Unified Communications Manager during a call setup.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
See MGCP packet debugging examples and their meanings at the Reference link below.
Reference: Sample of Debug MGCP Packets
http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communicationsmanager-callmanager/42104-debug-mgcp.html

QUESTION 24
24
To which SIP response class do the SIP response codes 300 to 399 belong?

A. Provisional
B. Client Failure
C. Server Failure
D. Successful
E. Redirection

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Redirection — further action needs to be taken in order to complete the request. That is what this class implies.

QUESTION 25
25
Which SIP request method enables reliability of SIP 1xx response types?

A. ACK
B. PRACK

C. OPTIONS
D. CANCEL
E. REGISTER

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
In order to achieve reliability for provisional responses, we do nearly the same thing. Reliable provisional responses are retransmitted by the TU with an exponentialbackoff. Those retransmissions cease when a PRACK message is received. The PRACK request plays the same role as ACK, but for provisional responses. There is an important difference, however. PRACK is a normal SIP message, like BYE. As such, its own

QUESTION 26
26
Which SIP response is considered a final response?

A. 183 Session in Progress
B. 199 Early Dialog Terminated
C. 200 OK
D. 180 Ringing
E. 100 Trying

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:

200 OK Indicates the request was successful. Whether other options state the request is still in progress or request is initiated.

QUESTION 27
27
Which two SDP content headers can be found in a SIP INVITE message? (Choose two.)

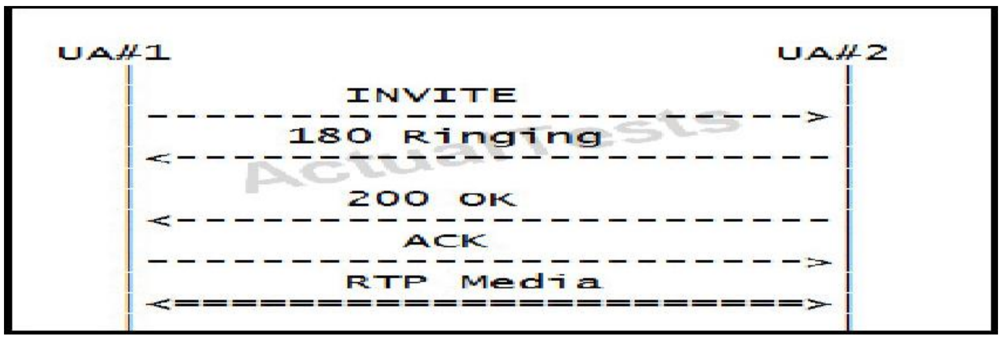
A. Expires

B. Contact
C. Connection Info
D. Media Attributes
E. Allow
F. CSeq

Correct Answer: CD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Connection info is optional field in SDP whether Media attributes decide the codec and media type for that call.

QUESTION 28
28
Refer to the exhibit.



If this SIP call is initiated using early offer, which SIP message will UA#2 use to communicate its media capability to UA#1?

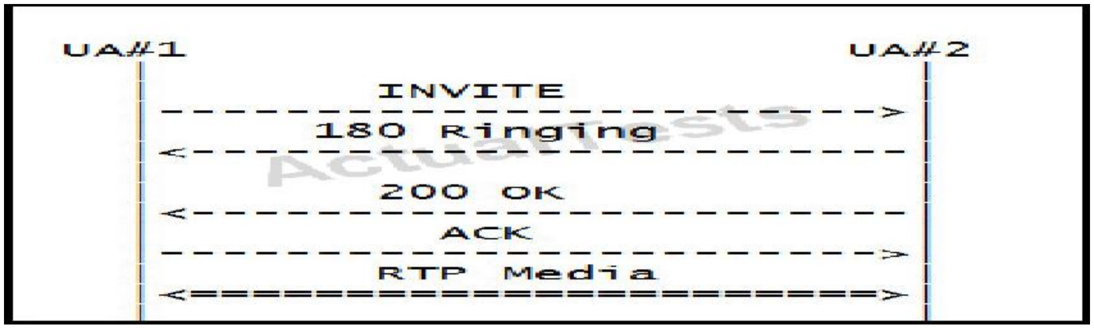
A. INVITE
B. 180 Ringing
C. 200 OK
D. ACK
E. RTP Media

Correct Answer: C
Section: (none)

Explanation

Explanation/Reference:
Explanation:
In Early offer, SIP Send SDP in the invite, the other node will send the SDP in the 200 message.

QUESTION 29
29
Refer to the exhibit.



If this SIP call is initiated using delayed offer, which SIP message will UA#1 use to communicate its media capability to UA#2?

A. INVITE
B. 180 Ringing
C. 200 OK
D. ACK
E. RTP Media

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
In the Delayed Offer process, the calling does not send its offer in the SIP INVITE Message. The callee sends the offer within the SDP fields of its answer (SIP 200 OK). The calling answers within the ACK message.

QUESTION 30
30

Which two responses are examples of client error responses in SIP protocol? (Choose two.)

A. 302 Moved Temporarily
B. 404 Not Found
C. 503 Service Unavailable
D. 502 Bad Gateway
E. 604 Does Not Exist Anywhere
F. 408 Request Timeout

Correct Answer: BF
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Client Error (400 to 499)—Request contains bad syntax or cannot be fulfilled at this server. This class of 400 to 499 contains only error messages.

QUESTION 31
31
Which H.245 information is exchanged within H.225 messages in H.323 Fast Connect?

A. Terminal Capability Set
B. Open Logical Channel
C. Master-Slave Determination
D. Call Setup
E. Call Progress

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
With the standard H.245 negotiation, the two endpoints need three round-trips before they agree on the parameters of the audio/video channels (1. master/slave voting, 2. terminal capability set exchange, and finally, 3. opening the logical channels). In certain situations and especially with high-latency network links, this can last too long and users will notice the delay.

QUESTION 32

32
Which two compression formats for high-definition video have technical content that is identical to H.264? (Choose two.)

A. MPEG-4 Part 10
B. MPEG-4 Part 14
C. MPEG-2 Part 7
D. AVC
E. VC3
F. VP8

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
MPEG-4 Part 10, also known as MPEG-4 AVC (Advanced Video Coding), is actually defined in an identical pair of standards maintained by different organizations, together known as the Joint Video Team (JVT). While MPEG-4 Part 10 is a ISO/IEC standard, it was developed in cooperation with the ITU, an organization heavily involved in broadcast television standards.Since the ITU designation for the standard is H.264, you may see MPEG-4 Part 10 video referred to as either AVC or H.264. Both are valid, and refer to the same standard.

QUESTION 33
33
Refer to the exhibit.

A user is going through a series of dialing steps on an SCCP IP phone (extension 1001) to call another SCCP IP phone (extension 2003). Both phones are registered to the same Cisco Unified Communications Manager cluster. Which user inputs are sent from the calling IP phone to the Cisco Unified Communications Manager, in forms of SCCP messages, after the user pressed the Dial softkey? Note that the commas in answer choices below are logical separators, not part of the actual user input or SCCP messages.

A. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 0, 3.
B. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 1, <<, 0, 3.
C. A single SCCP message is sent to Cisco Unified Communications Manager to report that digits 2003 have been dialed.
D. A single SCCP message is sent to Cisco Unified Communications Manager to report that digits 201<<03 have been dialed.
E. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 1, <<, 2, 0, 0, 3.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation: After the user delete phone stop the digit by digit dialing and send it as a whole setup.

QUESTION 34
34
How are DTMF digits transported in RFC 2833?

A. In the RTP stream with the named telephone events payload format.
B. In the RTP stream with the regular audio payload format.
C. In SIP NOTIFY messages.
D. In SIP INFO messages.
E. In SIP SUBSCRIBE messages.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
DTMF digits and named telephone events are carried as part of the audio stream, and MUST use the same sequence number and time-stamp base as the regular audio channel to simplify the generation of audio waveforms at a gateway. The default clock frequency is 8,000 Hz, but the clock frequency can be redefined when assigning the dynamic payload type.

QUESTION 35
35
Refer to the exhibit.

```
Call-Info: <sip: address>; method="NOTIFY;Event=telephone-event;Duration=msec"
```

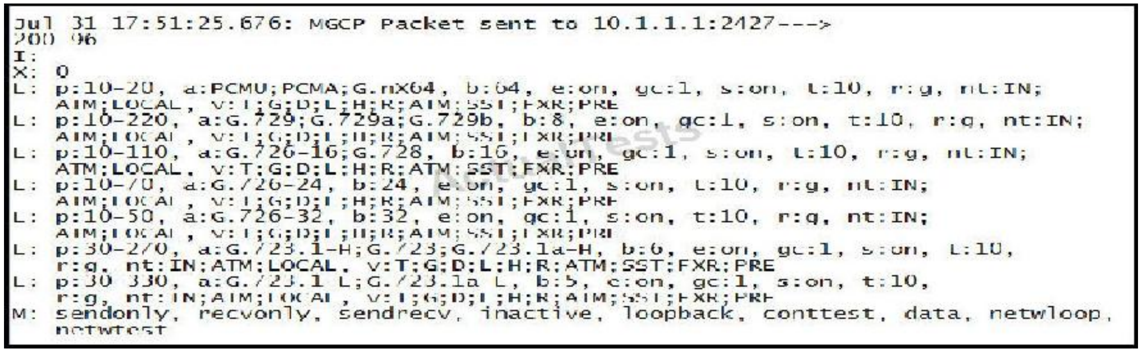
Which DTMF relay method is advertised when the originating SIP gateway sends an INVITE message with a Call-Info header shown?

A. RFC 2833
B. SIP INFO
C. SIP NOTIFY
D. SIP KPML
E. In-band audio

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
You can develop user-specific applications that reside on your network entity and have the ability to subscribe for event services supported by the IMG. If the network entity wants the ability to detect an entered DTMF digit (only telephone event of “###” are currently supported) from the TDM-side of a call to the IP side of a call, the entity can subscribe to the IMG for these events and receive SIP NOTIFY events containing the digit event.

QUESTION 36
36
Refer to the exhibit.



```
Jul 31 17:51:25.676: MGCP Packet sent to 10.1.1.1:2427--->
200 96
I:
X:  0
L:  p:10-20, a:PCMU;PCMA;G.nX64, b:64, e:on, gc:1, s:on, t:10, r:g, nt:IN;
    ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;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;
    ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;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;
    ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L:  p:10-70, a:G.726-24, b:24, e:on, gc:1, s:on, t:10, r:g, nt:IN;
    ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L:  p:10-50, a:G.726-32, b:32, e:on, gc:1, s:on, t:10, r:g, nt:IN;
    ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;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;ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;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;ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
M:  sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop,
    netwtest
```

You received this debug output to troubleshoot a Cisco IOS MGCP gateway problem at a customer site. What is the purpose of this message?

A. The MGCP gateway uses this message to respond to an RQNT message from Cisco Unified Communications Manager.
B. The MGCP gateway uses this message to respond to an AUCX message from Cisco Unified Communications Manager.
C. The MGCP gateway uses this message to respond to an AUEP message from Cisco Unified Communications Manager.
D. The MGCP gateway uses this message to respond to a DLCX message from Cisco Unified Communications Manager.
E. The MGCP gateway uses this message to respond to an NTFY message from Cisco Unified Communications Manager.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
This message requests the status of an endpoint. Information that can be audited with this

includes Requested Events,DigitMap,SignalRequests,RequestIdentifier,QuarantineHandling, Notified Entity, Connection Identifiers, Detect Events, Observed Events, Event States, Bearer Information, Restart Method, Restart Delay, ReasonCode, PackageList, Max MGCP Datagram, and Capabilities. The response will include information about each of the items for which auditing info was requested.

QUESTION 37
37

Refer to the exhibit.

```
Jul 31 18:12:16.640: MGCP Packet sent to 10.1.1.1:2427--->
200 108 OK
I: 2

v=0
c=IN IP4 10.1.1.254
m=audio 18630 RTP/AVP 0 100
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38
```

You received this debug output to troubleshoot a Cisco IOS MGCP gateway call quality issue at a customer site. Which statement about this message is true?

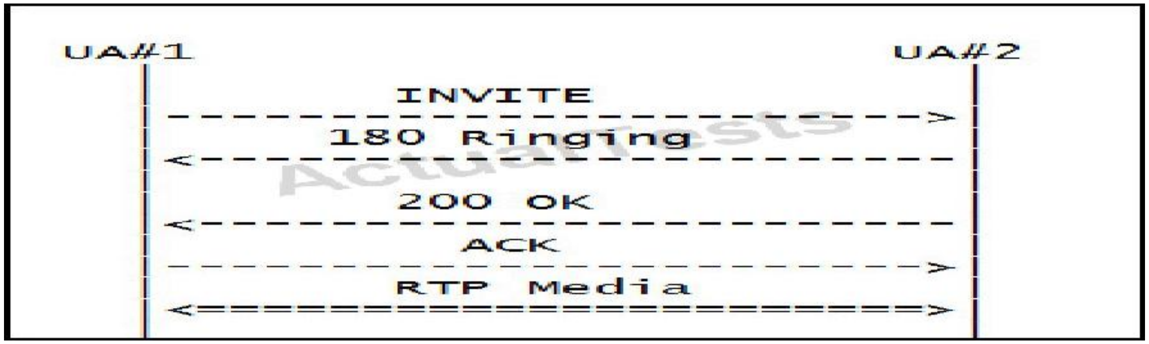
A. The MGCP gateway is responding to an RQNT message from Cisco Unified Communications Manager to poll the call statistics of an active call.
B. The MGCP gateway is responding to an AUEP message from Cisco Unified Communications Manager to poll the call statistics of a terminating call.
C. The MGCP gateway is responding to an MDCX message from Cisco Unified Communications Manager during a call setup.
D. The MGCP gateway is responding to an AUCX message from Cisco Unified Communications Manager about an active call.
E. The MGCP gateway is responding to a DLCX message from Cisco Unified Communications Manager about a terminating call.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
DeleteConnection—used by a call agent to instruct a gateway to delete an existing connection. DeleteConnection can also be used by a gateway to release a connection that can no longer be

sustained.

QUESTION 38
38
Refer to the exhibit.



If this SIP call is initiated using delayed offer, which SIP message will UA#2 use to communicate its media capability to UA#1?

A. INVITE
B. 180 Ringing
C. 200 OK
D. ACK
E. RTP Media

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
200 OK Indicates the request was successful.

QUESTION 39
39
To which SIP response category does 301 Moved Permanently belong?

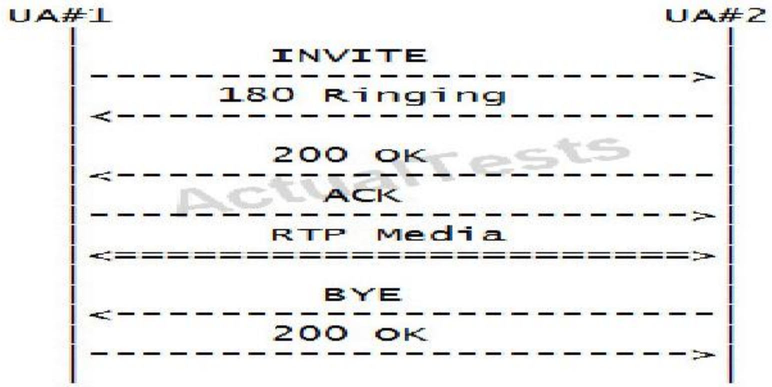
A. Provisional
B. Successful
C. Redirection
D. Client Failure
E. Server Failure

Correct Answer: C

Section: (none)
Explanation

Explanation/Reference:
Explanation:
The 301 response from the Web server should always include an alternative URL to which redirection should occur. If it does, a Web browser will immediately retry the alternative URL. So you never actually see a 301 error in a Web browser, unless perhaps you have a corrupt redirection chain e.g. URL A redirects to URL B which in turn redirects back to URL A. If your client is not a Web browser, it should behave in the same way as a Web browser i.e. immediately retry the alternative URL.

QUESTION 40
40
Refer to the exhibit.



How many SIP signaling transaction(s) took place in this SIP message exchange between two SIP user agents?

A. 1
B. 2
C. 3
D. 4
E. 5
F. 6

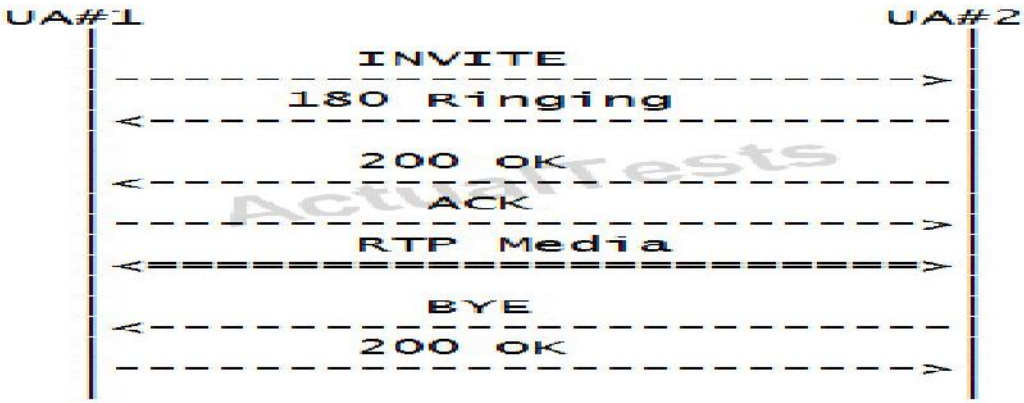
Correct Answer: C
Section: (none)

Explanation

Explanation/Reference:
Explanation:
During the establishment, maintenance and termination of a SIP session, signaling messages are exchanged between the two SIP endpoints. There are two different kinds of signaling “conversations” that those messages take part in: transactions and dialogs.
A transaction is a SIP message exchange between two user-agents that starts with a request and ends with its final response (it can also contain zero or more provisional responses in between).

For example, during the termination of a SIP session, one user releases the call by sending a BYE request and the other party replies back with a 200 OK response. This message exchange is called a transaction.
But what happens in the case of the INVITE request? The establishment of a SIP session starts basically with an INVITE request and is considered as completed upon the receipt of the ACK. In this case, the transaction starts with the INVITE request and ends with the 200 OK, so the ACK is not part of the transaction. The ACK can be considered as a transaction on its own. However, when the final response to an INVITE is not a 2xx response, then the ACK is considered as part of the transaction.A dialog is a complete exchange of SIP messages between two user-agents. That means that transactions are actually parts of a dialog. For example, in the case of a SIP session establishment, a dialog starts with the INVITE-200 OK transaction, continues with the ACK and ends with the BYE-200 OK transaction.
The picture below depicts the dialog and transactions that take place during the establishment of a SIP session:
Note: There can also be subsequent requests that belong to the same dialog, such as a BYE or a re-INVITE message. As out-of-dialog requests are considered messages such as an initial INVITE request for a new session or an OPTIONS message for checking capabilities.
There are different SIP headers/parameters that identify the dialogs and transactions, and they will be analyzed in later posts.
Reference:https://telconotes.wordpress.com/2013/03/13/sip-transactions-vs-dialogs/

QUESTION 41
41
Refer to the exhibit.



How many SIP signaling dialog(s) took place in this SIP message exchange between two SIP user agents?

A. 1
B. 2
C. 3
D. 4
E. 5
F. 6

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
During the establishment, maintenance and termination of a SIP session, signaling messages are exchanged between the two SIP endpoints. There are two different kinds of signaling “conversations” that those messages take part in: transactions and dialogs.
A transaction is a SIP message exchange between two user-agents that starts with a request and ends with its final response (it can also contain zero or more provisional responses in between). For example, during the termination of a SIP session, one user releases the call by sending aBYE request and the other party replies back with a 200 OK response. This message exchange is called a transaction.
But what happens in the case of the INVITE request? The establishment of a SIP session starts basically with an INVITE request and is considered as completed upon the receipt of the ACK. In this case, the transaction starts with the INVITE request and ends with the 200 OK, so the ACK is

not part of the transaction. The ACK can be considered as a transaction on its own. However, when the final response to an INVITE is not a 2xx response, then the ACK is considered as part of the transaction.A dialog is a complete exchange of SIP messages between two user-agents. That means that transactions are actually parts of a dialog. For example, in the case of a SIP session establishment, a dialog starts with the INVITE-200 OK transaction, continues with the ACK and ends with the BYE-200 OK transaction.
The picture below depicts the dialog and transactions that take place during the establishment of a SIP session:
Note: There can also be subsequent requests that belong to the same dialog, such as a BYE or a re-INVITE message. As out-of-dialog requests are considered messages such as an initial INVITE

request for a new session or an OPTIONS message for checking capabilities.
There are different SIP headers/parameters that identify the dialogs and transactions, and they will be analyzed in later posts.
Reference:https://telconotes.wordpress.com/2013/03/13/sip-transactions-vs-dialogs/

QUESTION 42
42
Which two statements describe characteristics of Binary Floor Control Protocol? (Choose two.)

A. Its binary encoding is designed to work in high-bandwidth environments.
B. It is designed for audio or video conference sessions of three or more participants.
C. It enables management of shared content resources independent of video streams.
D. It supports TLS-based authentication.
E. It supports SIP as well as H.323.

Correct Answer: CD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
BFCP is a deliverable developed as part of theInternet Engineering Task Force(IETF) XCON Centralized Conferencing working group. The IETF XCON working group was formed to focus on delivering a standards-based approach to managing IP conferencing while promoting broad interoperability between software and equipment vendors.

QUESTION 43
43
What is the minimum number of TCP sessions needed to complete a H.323 call between two H.323 gateways using slow start?

A. 0
B. 1
C. 2
D. 3
E. 4

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
H.323 has two modes of operation: slow start and fast start. The initiation of a call may proceed in a slow start or fast start in H.323. In a slow start, H.323 signaling consists of Setup, Call

Proceeding, Alerting, and Connect steps. After these steps, the H.245 media negotiation is performed. When a call is initiated in H.323 fast start, the H.245 media negotiation is performed within the initial Setup message. With slow start, multiple TCP connections are needed for anH.323 call, such as one H.225 signaling channel and one H.245 signaling channel if required (minimum of these two).

QUESTION 44
44
Which element was added to H.225 messages to enable Fast Connect in H.323 version 2?

A. fastStart
B. fastConnect
C. H.245 PDU
D. User-User Information
E. Connection Information

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Fast start allows for H323 media connections to be started at the beginning of a call. This is helpful for ringback scenarios, and also reduces the amount of time calls take to establish media. H245 is still negotiated later, but the actual media can be done earlier through H225 messages.

QUESTION 45
45
What is the name of the logical channel proposal that is transmitted from the called entity to the calling entity in H.323 Fast Connect?

A. Forward Logical Channel
B. Backward Logical Channel
C. Reverse Logical Channel
D. Originator Logical Channel
E. Destination Logical Channel

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation: Unlike the OpenLogicalChannel request used by H.323 for video uni-directional logical channels, the request used by H.324 for opening video bi-directional logical channels specifies the temporalSpatialTradeOff Capability in both the forward and reverse directions--in the

forwardLogicalChannelParameters.dataTypeandreverseLogicalChannelParameters.dataType components, respectively. The semantics of temporalSpatialTradeOffCapability used in forward LogicalChannelParameters.dataType is described in the previous section. The semantics for its presence in the reverse direction is described in this section.

QUESTION 46
46
Which procedure uses H.225 messages to exchange H.245 Master-Salve Determination information?

A. H.323 Fast Connect
B. H.245 tunneling
C. H.225 tunneling
D. H.323 early media
E. H.245 terminal capability set

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

Explanation:
The H.245 protocol is a media control protocol that is a part of H.323 protocol suite. The H.245 protocol is used primarily to negotiate master-slave relationship between communicating endpoints. These endpoints exchange terminal capabilities and logical channel manipulations (open, close, modify).The H.245 messages can be encapsulated and carried between H.225 controlled endpoints within H.225 messages. This way of "piggy-backing" an H.245 message to H.225 message is referred to as H245 Tunneling. The H.245 Tunneling method is optional and negotiable between communicating H.323 endpoints. If both endpoints support this option, usually the H.245 Media Controlled messages are exchanged via the Tunneling method.

QUESTION 47
47
Which two VoIP protocols use SDP to describe streaming media sessions? (Choose two.)

A. SCCP
B. H.323
C. SIP
D. MGCP
E. RAS
F. cRTP

Correct Answer: CD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
TheSession Description Protocol(SDP), defined in RFC 2327, describes the content of sessions, including telephony, Internet radio, and multimedia applications. SDP includes information about [8]:
Media streams:A session can include multiple streams of differing content. SDP currently defines audio, video, data, control, and application as stream types, similar to the MIME types used for Internet mail.
Addresses:SDP indicates the destination addresses, which may be a multicast address, for a media stream.
Ports:For each stream, the UDP port numbers for sending and receiving are specified.
Payload types: For each media stream type in use (for example, telephony), the payload type indicates the media formats that can be used during the session.
Start and stop times: These apply to broadcast sessions, for example, a television or radio program. The start, stop, and repeat times of the session are indicated.
Originator: For broadcast sessions, the originator is specified, with contact information. This may be useful if a receiver encounters technical difficulties.

QUESTION 48
48
Which RAS message is used between two gatekeepers to resolve an alias address?

A. GRQ
B. ARQ
C. IRQ
D. LRQ
E. RRQ

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
LRQ—These messages are exchanged between gatekeepers and are used for inter-zone (remote zone) calls. For example, gatekeeper A receives an ARQ from a local zone gateway requesting call admission for a remote zone device. Gatekeeper A then sends an LRQ message to gatekeeper B. Gatekeeper B replies to the LRQ message with either a Location Confirm (LCF) or Location Reject (LRJ) message, which depends on whether it is configured to admit or reject the inter-zone call request and whether the requested resource is registered

QUESTION 49
49
When a Cisco IOS gatekeeper receives an ARQ from a registered endpoint, what is the first step it will take in an attempt to resolve the destination address?

A. Check to see if the destination address is locally registered.
B. Check to see if the destination address matches the technology prefix.
C. Check to see if the destination address matches the local zone prefix.
D. Check to see if the destination address matches the remote zone prefix.
E. Check to see if the destination address matches the default technology prefix.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:

Admission Request (ARQ) and Location Request (LRQ) are the two H.225 Registration, Admission, Status (RAS) messages that trigger a gatekeeper to initiate the call routing decision process.

QUESTION 50
50
When a Cisco IOS gatekeeper receives an LRQ, what is the first step it will take in an attempt to resolve the destination address?

A. Check to see if the LRQ reject-unknown-prefix flag is set.
B. Check to see if the destination address matches the technology prefix.
C. Check to see if the destination address matches the hop-off technology prefix.
D. Check to see if the destination address matches the remote zone prefix.
E. Check to see if the LRQ forward-queries flag is set.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
LRQ—These messages are exchanged between gatekeepers and are used for inter-zone (remote zone) calls. For example, gatekeeper A receives an ARQ from a local zone gateway requesting call admission for a remote zone device. Gatekeeper A then sends an LRQ message to gatekeeper B. Gatekeeper B replies to the LRQ message with either a Location Confirm (LCF) or Location Reject (LRJ) message, which depends on whether it is configured to admit or reject the inter-zone call request and whether the requested resource is registered.

QUESTION 51
51
Which ITU-T recommendation defines the procedures for using more than one video channel in H.320-based systems?

A. H.324
B. H.230
C. H.239
D. H.264
E. H.329

Correct Answer: C
Section: (none)

Explanation

Explanation/Reference:
Explanation:
H.239is the ITU standard for a second video channel; this is supported by all the major vendors, and is the only content channel standard supported by Cisco acquired Codian products or Cisco TelePresence Serial Gateway Series products on H.323 video calls.Cisco acquired Codian products need to be configured to enable H.239. Any H.323 endpoint that also supports the H.239 protocol can source this channel, as can a VNC connection, though some endpoints need to be configured to use H.239 instead of their proprietary solution.

QUESTION 52
52
Which statement about the iSAC on Cisco Unified Border Element is true?

A. It is a narrow-band codec.
B. It has a fixed frame of 30 milliseconds.
C. It has an adaptive frame of up to 60 milliseconds.
D. It is designed to deliver wideband sound quality in high-bit-rate applications only.
E. It is not yet supported on the Cisco Unified Border Element (CUBE)
F. It is not yet supported on Cisco Unified Border Element.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
iSAC-Internet Speech Audio Codec (iSAC) is an adaptive wideband audio codec, specially designed to deliver wideband sound quality with low delay in both low and medium-bit rate applications. Using an adaptive bit rate of between 10 and 32 kb/s, iSAC provides audio quality approaching that of G.722 while using less than half the bandwidth. In deployments with significant packet loss, delay, or jitter, such as over a WAN, iSAC audio quality is superior to that of G.722 due to its robustness. iSAC is supported for SIP and SCCP devices. The Cisco Unified Communications Manager IP Voice Media Streaming App (IPVMSApp), which includes Media Termination Point, Conference Bridge, Music on Hold Server, and Annunciator does not support iSAC. MGCP devices are not supported.

QUESTION 53
53
Which two data frame lengths are supported by iLBC? (Choose two.)

A. 10 milliseconds
B. 20 milliseconds
C. 30 milliseconds
D. 40 milliseconds
E. 50 milliseconds
F. 60 milliseconds

Correct Answer: BC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
iLBC-Internet Low Bit Rate Codec (iLBC) provides audio quality between that of G.711 and G.729 at bit rates of 15.2 and 13.3 kb/s, while allowing for graceful speech quality degradation in a lossy network due to the speech frames being encoded independently. By comparison, G.729 does not handle packet loss, delay, and jitter well, due to the dependence between speech frames. iLBC is supported for SIP, SCCP, H323, and MGCP devices.

QUESTION 54
54
Which two types of devices on Cisco Unified Communications Manager support iSAC? (Choose two.)

A. MGCP
B. SIP
C. SCCP
D. Music on Hold server
E. H.323

Correct Answer: BC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
iSAC-Internet Speech Audio Codec (iSAC) is an adaptive wideband audio codec, specially designed to deliver wideband sound quality with low delay in both low and medium-bit rate applications. Using an adaptive bit rate of between 10 and 32 kb/s, iSAC provides audio quality approaching that of G.722 while using less than half the bandwidth. In deployments with significant packet loss, delay, or jitter, such as over a WAN, iSAC audio quality is superior to that of G.722

due to its robustness. iSAC is supported for SIP and SCCP devices. The Cisco Unified Communications Manager IP Voice Media Streaming App (IPVMSApp), which includes Media Termination Point, Conference Bridge, Music on Hold Server, and Annunciator does not support iSAC. MGCP devices are not supported.

QUESTION 55
55
Which statement about G.722.1 codec support on Cisco Unified Communications Manager is true?

A. It is always preferred by Cisco Unified Communications Manager over G.711.
B. It is a high-complexity wideband codec.
C. It operates at bit rates of 15.2 and 13.3 kb/s.
D. It is supported for SIP and SCCP devices.
E. It is supported for SIP and H.323 devices.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
G.722.1 is a low-complexity wideband codec operating at 24 and 32 kb/s. The audio quality approaches that of G.722 while using at most half the bit rate. As it is optimized for both speech and music, G.722.1 has slightly lower speech quality than the speech-optimized iSAC codec. G.722.1 is supported for SIP and H.323 devices.

QUESTION 56
56
What is the minimum number of H.225 messages required to establish an H.323 call with bidirectional media?

A. 1
B. 2
C. 3
D. 4
E. 5

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
A typical H.245 exchange looks similar to below figure
After this exchange of messages, the two endpoints (EP) in this figure would be transmitting audio in each direction. The number of message exchanges is numerous, each has an important purpose, but nonetheless takes time.
For this reason, H.323 version 2 (published in 1998) introduced a concept called Fast Connect, which enables a device to establishbi-directionalmedia flows as part of the H.225.0 call establishment procedures. With Fast Connect, it is possible to establish a call with bi-directional media flowing with no more than two messages, like in figure 3.
Fast Connect is widely supported in the industry. Even so, most devices still implement the complete H.245 exchange as shown above and perform that message exchange in parallel to other activities, so there is no noticeable delay to the calling or called party.

QUESTION 57
57

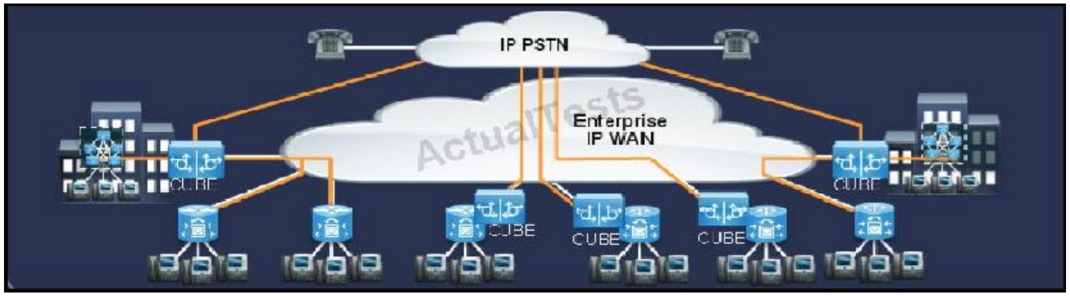
In Key Press Markup Language, which SIP request is used to deliver the actual DTMF digits?

A. SUBSCRIBE
B. INFO
C. NOTIFY
D. INVITE
E. ACK

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
KPML procedures use a SIP SUBSCRIBE message to register for DTMF digits. The digits themselves are delivered in NOTIFY messages containing an XML encoded body.

QUESTION 58
58
Refer to the exhibit.



Which SIP trunk deployment model is shown in this enterprise VoIP topology?

A. mixed TDM and VoIP
B. centralized
C. hybrid
D. traditional TDM
E. distributed

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Hybrid SIP Trunk Model In a hybrid SIP trunk deployment, some of the businesses' sites conform to a distributed SIP trunk deployment model. In this model each site has direct SIP session connectivity to the IP PSTN, and other sites conform to a centralized SIP trunk deployment, accessing the IP PSTN through a central hub, which has SIP session connectivity to the IP PSTN

(Figure 3). The hybrid SIP trunk deployment model may have multiple "central" hubs in different geographic regions, or for specific business functions, such as call centers.
Figure 3 Hybrid SIP Trunk Deployment Mode
Reference: http://www.cisco.com/c/dam/en/us/products/collateral/unified-communications/unifiedborder-element/cis_45835_cube_assets_wp1e.pdf

QUESTION 59
59
Which SIP reason phrase maps to SIP response reason code 181?

A. Ringing
B. Call is Being Forwarded

C. Session in Progress
D. Unknown Number
E. Call Does not Exist

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
1xx—Provisional Responses
100 Trying
Extended search being performed may take a significant time so a forking proxy must send a 100 Trying response.
180 Ringing
Destination user agent received INVITE, and is alerting user of call.
181 Call is Being Forwarded
Servers can optionally send this response to indicate a call is being forwarded.[1]:§21.1.3
182 Queued
Indicates that the destination was temporarily unavailable, so the server has queued the call until the destination is available. A server may send multiple 182 responses to update progress of the queue.
183 Session in Progress
This response may be used to send extra information for a call which is still being set up.
199 Early Dialog Terminated
Can be used by User Agent Server to indicate to upstream SIP entities (including the User Agent Client (UAC)) that an early dialog has been terminated.
Reference: http://en.wikipedia.org/wiki/List_of_SIP_response_codes

QUESTION 60
60
Refer to the exhibit.

Which SIP message header is used to tunnel QSIG messages across the SIP network when the OGW receives a call bound for the TGW?

A. Content-TypE. application/sdp
B. Content-TypE. application/qsig
C. Content-TypE. message/ISUP
D. Content-TypE. message/external-body
E. Content-TypE. application/x-q931

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Tunneling over SIP
The Cisco gateway receives QSIG messages from the PBX side and then identifies the destination of the message (or call). The QSIG messages received from the PBX are encapsulated within SIP messages as Multipurpose Internet Mail Extensions (MIME) bodies and are sent (tunneled) across the IP network to the recipient gateway.
When encapsulating a QSIG message (for switch type primary-qsig) inside a SIP message, Cisco gateways include the QSIG message in a MIME body of the SIP request or response using media

type
Reference:
http://www.cisco.com/c/en/us/solutions/collateral/enterprise-networks/empowered-branchsolution/white_paper_c11_459092.html

Exam B

QUESTION 1
61
Refer to the exhibit.

Which SIP response message should the TGW send if it cannot process the tunneled QSIG messages from the OGW?

A. 405 Method Not Allowed
B. 406 Not Acceptable
C. 412 Conditional Request Failed
D. 415 Unsupported Media Type
E. 485 Ambiguous

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Fallback from QSIG Tunneling
In some situations, QSIG tunneling will fail or need to fall back:
• Remote party does not support multipart MIME body: In this case, the remote side sends a "415 Media Not Supported" response. Upon receiving this response, OGW will fall back to normal mode and send an INVITE request without any tunneled content. This procedure helps ensure that at least the basic call will work normally.
• Remote party does not understand tunneled content: If the remote side does not support the tunneled content, it should drop the tunneled content and continue as a normal call; because all essential parameters are present in the original INVITE, the call can go through without the need for fallback.

Reference:http://www.cisco.com/c/en/us/solutions/collateral/enterprise-networks/empoweredbranch-solution/white_paper_c11_459092.html

QUESTION 2
62
Refer to the exhibit.

During a QSIG tunneling over SIP call establishment, which two types of SIP messages can the OGW use to tunnel a waiting QSIG message? (Choose two.)

A. SIP re-INVITE
B. SIP NOTIFY
C. SIP INFO
D. SIP OPTIONS
E. SIP UPDATE
F. SIP REFER

Correct Answer: AE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The TGW sends and the OGW receives a 200 OK response--the OGW sends an ACK message to the TGW and all successive messages during the session are encapsulated into the body of SIP INFO request messages. There are two exceptions:
When a SIP connection requires an extended handshake process, renegotiation, or an update, the gateway may encapsulate a waiting QSIG message into a SIP re-INVITE or SIP UPDATE message during QSIG call establishment.
When the session is terminated, gateways send a SIP BYE message. If the session is terminated by notice of a QSIG RELEASE COMPLETE message, that message can be encapsulated into the SIP BYE message.
Reference:http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/15-mt/sipconfig-15-mt-book/voi-sip-tdm.html

QUESTION 3

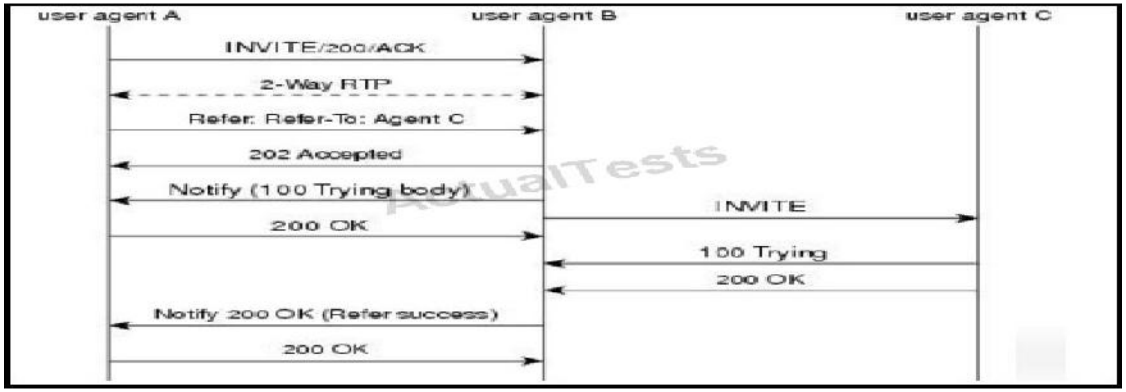
63
In a SIP REFER-based call transfer, which SIP message is being used by the recipient to notify the originator that the final recipient was successfully contacted?

A. 200 OK
B. NOTIFY with a message body of 200 OK
C. 202 Accepted
D. 100 Trying
E. 200 BYE

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The Refer method always begins within the context of an existing call and starts with the originator . The originator sends a Refer request to the recipient (user agent receiving the Refer request) to initiate a triggered Invite request. The triggered Invite request uses the SIP URL contained in the Refer-To header as the destination of the Invite request. The recipient then contacts the resource in the Refer-To header (final-recipient), and returns a SIP 202 (Accepted) response to the originator. The recipient also must notify the originator of the outcome of the Refer transaction--whether the final-recipient was successfully or unsuccessfully contacted. The notification is accomplished using the Notify Method, SIP’s event notification mechanism.A Notify message with a message body of SIP 200 OK indicates a successful transfer, while a body of SIP 503 Service Unavailable indicates an unsuccessful transfer.If the call was successful, a call between the recipient and the final-recipient results.

QUESTION 4
64
Refer to the exhibit.



Which user agent has the recipient role in this SIP REFER call transfer?

A. user agent A
B. user agent B
C. user agent C
D. user agent B and C
E. user agent A and B

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The Refer method has three main roles:
• Originator—User agent that initiates the transfer or Refer request.
• Recipient—User agent that receives the Refer request and is transferred to the final-recipient.
• Final-Recipient—User agent introduced into a call with the recipient.
Reference: http://www.cisco.com/c/en/us/td/docs/ios_xr_sw/iosxr_r3-4/sbc/configuration/guide/sbc_c34/sbc34stx.pdf
Topic 3, Cisco Unified Communications Manager (CUCM)

QUESTION 5
65
Which device is the initiator of a StationD message in a Cisco Unified Communications Manager SDI trace?

A. SCCP IP phone

B. SIP IP phone
C. Cisco Unified Communications Manager
D. MGCP analog gateway
E. digital voice gateway

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
All messages to and from a skinny device are preceded by either the words Stationd or StationInit. StationD messages are sent from call manager to IP phone.Skinny message transmission such as this between the IP phone and CallManger occurs for every action undertaken by the IP phone,includinginitialization,registration,on-hook,off-hook,dialing of the digits,key press on the phone,and so much more.

QUESTION 6
66
What is the maximum length of any numeric geographic area address in ITU recommendation E.164?

A. 15
B. 18
C. 21
D. 22
E. 25

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
E.164 defines a general format for international telephone numbers. Plan-conforming numbers are limited to a maximum of 15 digits. The presentation of numbers is usually prefixed with the character + (plus sign), indicating that the number includes the international country calling code(country code), and must typically be prefixed when dialing with the appropriate international call prefix, which is a trunk code to reach an international circuit from within the country of call origination.

QUESTION 7
67

According to ITU-T E.164 recommendations, which two fields in the National Significant Number code may be further subdivided? (Choose two.)

A. Country Code
B. National Destination Code
C. Subscriber Number
D. Regional Significant Number
E. Local User Code
F. National Numbering Plan

Correct Answer: BC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
A telephone number can have a maximum of 15 digits .The first part of the telephone number is the country code (one to three digits) .The second part is the national destination code (NDC). The last part is the subscriber number (SN). The NDC and SN together are collectively called the national (significant) number

QUESTION 8
68
Refer to the exhibit.

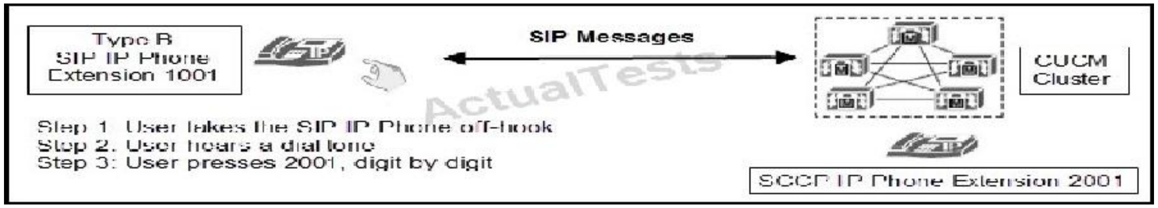
A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming that the calling SIP phone is not associated with any SIP dial rules, which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true?

A. Each digit is sent to Cisco Unified Communications Manager in a SIP NOTIFY message KPML event, at the time that the user enters the digit on the keypad.
B. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending each digit to Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending all digits to Cisco Unified Communications Manager in a SIP INVITE message.
D. The SIP IP phone will wait for the interdigit timer to expire or for the Dial softkey to be selected before sending the first digit in a SIP INVITE and the subsequent digits in SIP INFORMATION messages.
E. The SIP IP phone will send all digits to Cisco Unified Communications Manager in a SIP INVITE message as soon as the fourth digit is pressed.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
KPML procedures use a SIP SUBSCRIBE message to register for DTMF digits. The digits themselves are delivered in NOTIFY messages containing an XML encoded body. And it is Out of Band DTMF

QUESTION 9
69
Refer to the exhibit.



A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming the calling SIP phone is associated with a SIP dial rule with a pattern value of 2001, which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true?

A. As each digit is pressed on the SIP IP phone, it is sent to Cisco Unified Communications

Manager in a SIP NOTIFY message as a KPML event.

B. The SIP IP phone will wait for the interdigit timer to expire, and then send each digit to Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending all digits to Cisco Unified Communications Manager in a SIP INVITE message.
D. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending the first digit in a SIP INVITE and the subsequent digits in SIP INFORMATION messages.
E. The SIP IP phone will wait for the interdigit timer to expire, and then send all digits to Cisco Unified Communications Manager in a SIP INVITE message.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Type B SIP Phones offer functionality based SIP INVITE Message. Every key the end user presses triggers an individual SIP message. The first event is communicated with a SIP INVITE, but subsequent messages use SIP NOTIFY messages. The SIP NOTIFY messages send KPML events corresponding to any buttons or soft keys pressed by the user. Cisco Type B SIP IP Phones with SIP dial rules operate in the same manner as Cisco Type A phones with dial rules.

QUESTION 10
70
Refer to the exhibit.

A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming the calling SIP phone is associated with a SIP Dial Rule with a pattern value of 2001, which statement about the call setup process of this call is true?

A. Each digit will arrive at Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event, and Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.

B. Each digit will arrive at Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event. When the collected digits match the extension of the SCCP IP phone, Cisco Unified Communications Manager will extend the call only if the class of service configuration on both phones permits this action.
C. As soon as the user selects the Dial softkey, the SIP IP phone will forward all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.
D. As soon as the user selects the Dial softkey, the SIP IP phone will forward all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call only if class of service configuration on both phones permits this action.
E. The SIP IP phone will wait for the interdigit timer to expire, and then send all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Type B SIP Phones offer functionality based SIP INVITE Message. Every key the end user presses triggers an individual SIP message. The first event is communicated with a SIP INVITE, but subsequent messages use SIP NOTIFY messages. The SIP NOTIFY messages send KPML events corresponding to any buttons or soft keys pressed by the user. Cisco Type B SIP IP Phones with SIP dial rules operate in the same manner as Cisco Type A phones with dial rules.

QUESTION 11
71
What does a comma accomplish when it is used in a SIP Dial Rule pattern that is associated with a Cisco 9971 IP Phone that is registered to Cisco Unified Communications Manager?

A. It inserts a 500-millisecond pause between digits.
B. It causes the phone to generate a secondary dial tone.
C. It is a delimiter and has no significant dialing impact.
D. It indicates a timeout value of 5000 milliseconds.
E. It is an obsolete parameter and will be ignored.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Comma is accepted in speed dial as delimiter and pause. -Comma used to delineate dial string, FAC, CMC, and post connect digits For post connect digits, commas insert a 2 second delay Commas may be duplicated to create longer delays

QUESTION 12
72

What does a period accomplish when it is used in a SIP Dial Rule pattern that is associated with a Cisco 9971 IP Phone that is registered to Cisco Unified Communications Manager?

A. It matches any single digit from 0 to 9.
B. It matches one or more digits from 0 to 9.
C. It is a delimiter and has no significant dialing impact.
D. It matches any single digit from 0 to 9, or the asterisk (*) or pound (#) symbols.
E. It matches one or more digits from 0 to 9, or the asterisk (*) or pound (#) symbols.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Asterisk (*) matches one or more characters. The * gets processed as a wildcard character. You can override this by preceding the * with a backward slash (\) escape sequence, which results in the sequence *. The phone automatically strips the \, so it does not appear in the outgoing dial string. When * is received as a dial digit, it gets matched by the wildcard characters * and period (.).

QUESTION 13
73
What does a weight represent in the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager?

A. It defines the bandwidth that is available between locations.

B. It defines the bandwidth that is available on a link.
C. It is the amount of bandwidth allocation for different types of traffic.
D. It is used to provide the relative priority of a link between locations.
E. It is used to provide the relative priority of a location.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
A weight provides the relative priority of a link in forming the effective path between any pair of locations. The effective path is the path used by Unified CM for the bandwidth calculations, and it has the least cumulative weight of all possible paths. Weights are used on links to provide a "cost" for the "effective path" and are pertinent only when there is more than one path between any two locations.

QUESTION 14
74

Which statement about the effective path in the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager is true?

A. It is a sequence of links and intermediate locations that connect a pair of locations.
B. It is used to define the bandwidth that is available between locations.
C. Only one effective path is used between two locations.
D. There could be multiple effective paths between a pair of locations.
E. It logically represents the WAN link.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The effective path is the path used by Unified CM for the bandwidth calculations, and it has the least cumulative weight of all possible paths. Weights are used on links to provide a "cost" for the "effective path" and are pertinent only when there is more than one path between any two locations.

QUESTION 15

75
Which configuration component in Cisco Unified Communications Manager Enhanced Location Call Admission Control is designated to participate directly in intercluster replication of location, links, and bandwidth allocation data?

A. an active member of a Location Bandwidth Manager Group
B. a member of a Location Bandwidth Manager Hub Group
C. a standby member of a Location Bandwidth Manager Group
D. all members of a Location Bandwidth Manager Group
E. a shadow member of a Location Bandwidth Manager Hub Group

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
A Location Bandwidth Manager (LBM) service that has been designated to participate directly in intercluster replication of fixed locations, links data, and dynamic bandwidth allocation data. LBMs assigned to an LBM hub group discover each other through their common connections and form a fully-meshed intercluster replication network. Other LBM services in a cluster with an LBM hub participate indirectly in intercluster replication through the LBM hubs in their cluster.

QUESTION 16
76

What is the amount of audio bandwidth, in kilobits per second, that is used in the Cisco Unified Communications Manager location bandwidth calculation for a G.728 call?

A. 8
B. 16
C. 24
D. 29
E. 80

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:

G.728—Low-bit-rate codec that video endpoints support.It supportkilobits per second

QUESTION 17
77
When neither the active or standby Location Bandwidth Manager in the configured LBM group is available, what will the Cisco CallManager service on a subscriber Cisco Unified Communications Manager server do to make location CAC decisions?

A. It will attempt to communicate with the first configured member in the Location Bandwidth Manager hub group.
B. It will use the Call Treatment When No LBM Available service parameter with the default action to allow calls.
C. It will use the Call Treatment When No LBM Available service parameter with the default action to reject calls.
D. It will attempt to communicate with the local LBM service for location CAC decisions.
E. It will allow all calls until communication is reestablished with any configured servers in the LBM group.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
By default the Cisco CallManager service communicates with the local LBM service; however, LBM groups can be used to manage this communication. LBM groups provide an active and standby LBM in order to create redundancy for Unified CM call control.

QUESTION 18
78
Which system location is used for intercluster Enhanced Location CAC on Cisco Unified

Communications Manager?

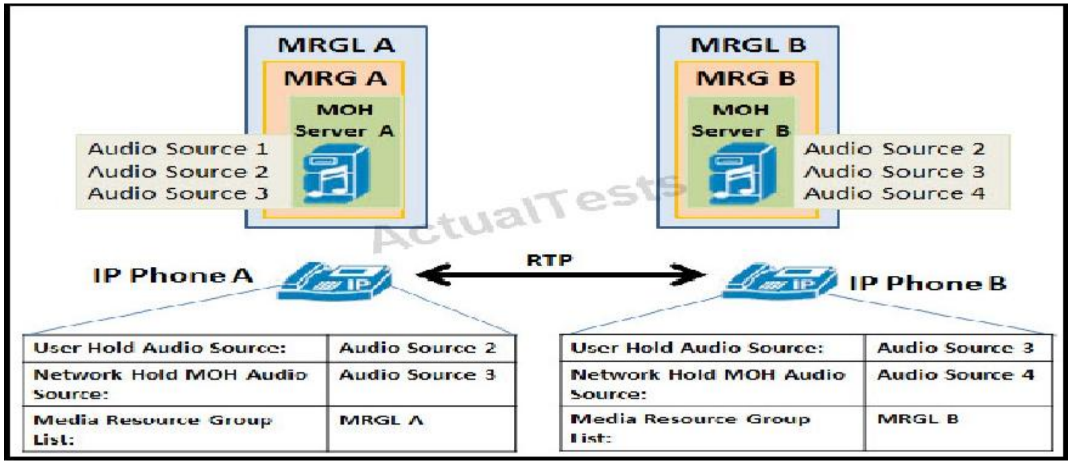
A. Hub_None
B. Default
C. Intercluster
D. Phantom
E. Shadow

Correct Answer: E

Section: (none)
Explanation

Explanation/Reference:
Explanation:
Theshadowlocationis used to enable a SIP trunk to pass Enhanced Location CAC information such as location name and Video-Traffic-Class (discussed below), among other things, required for Enhanced Location CAC to function between clusters. In order to pass this location information across clusters, the SIP intercluster trunk (ICT) must be assigned to the "shadow" location. The shadow location cannot have a link to other locations, and therefore no bandwidth can be reserved between the shadow location and other locations. Any device other than a SIP ICT that is assigned to the shadow location will be treated as if it was associated to Hub_None. That is important to know because if a device other than a SIP ICT ends up in the shadow location, bandwidth deductions will be made from that device as if it were in Hub_None, and that could have varying effects depending on the location and links configuration.

QUESTION 19
79
Refer to the exhibit.



All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assuming the provided configurations exist at the phone line level and multicast MOH is disabled clusterwide, what will happen when the user of IP Phone B presses the Hold softkey?

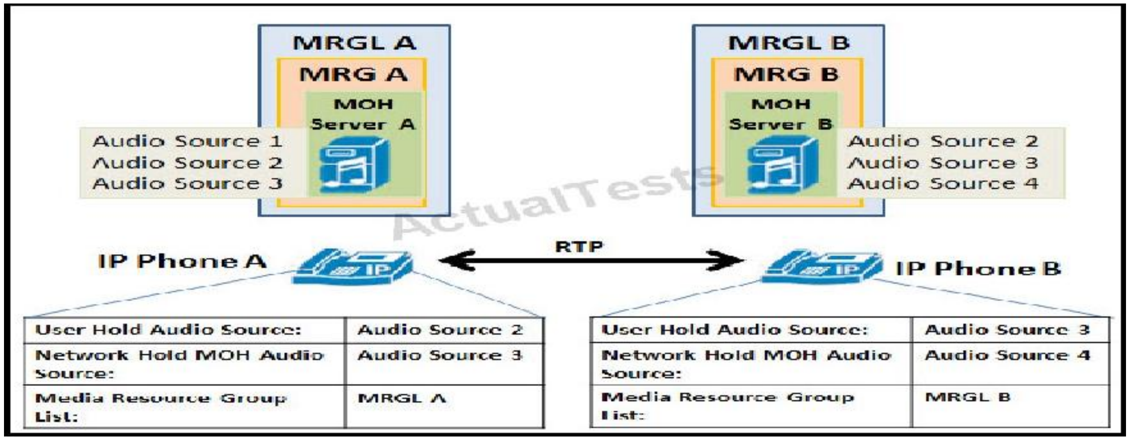
A. IP Phone A receives audio source 2 from MOH Server A.
B. IP Phone A receives audio source 3 from MOH Server A.
C. IP Phone A receives audio source 2 from MOH Server B.

D. IP Phone A receives audio source 3 from MOH Server B.
E. IP Phone A receives audio source 1 from MOH Server A.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Held parties determine the media resource group list that a Cisco Unified Communications Manager uses to allocate a music on hold resource.

QUESTION 20
80
Refer to the exhibit.



All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assuming the provided configurations exist at the phone line level and multicast MOH is disabled clusterwide, what will happen when the user of IP

Phone A presses the Transfer softkey?

A. The IP Phone B user hears audio source 3 from MOH Server A.
B. The IP Phone B user hears audio source 4 from MOH Server B.
C. The IP Phone B user hears audio source 3 from MOH Server B.
D. The IP Phone B user hears audio source 2 from MOH Server B.
E. The IP Phone A user hears no on-hold music.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Held parties determine the media resource group list that a Cisco Unified Communications Manager uses to allocate a music on hold resource.

QUESTION 21
81
Router A and router B are Cisco IOS routers with hardware CFB resources that are registered to the same Cisco Unified Communications Manager server. Which Media Resource Group and Media Resource Group List configuration should be implemented if an administrator wants to make sure that all provisioned DSPs on router A are consumed before router B's DSP is used?

A. Router A's CFB and router B's CFB should each be configured in its own MRG. Both MRGs should then be grouped into the same MRGL, but the MRG that contains router A's CFB should be listed in higher order than the MRG that contains router B's CFB. Finally, associate the MRGL to all conference resource consumers.
B. Router A's CFB and router B's CFB should each be configured in its own MRG. Both MRGs should then be further separated into different MRGLs. Finally, associate the MRGL that contains router A's CFB in higher order than router B's CFB to all conference resource consumers.
C. Router A's CFB and router B's CFB should both be configured in the same MRG with router A's CFB listed higher than that of router B. Then associate the MRG with an MRGL and apply it to all conference resource consumers.
D. Router A's CFB and router B's CFB should both be configured in the same MRG. Make sure router A's CFB is listed in a higher alphabetical order than router B's CFB. Then associate the MRG with an MRGL and apply it to all conference resource consumers.
E. Router A's CFB and router B's CFB should both be configured in the same MRG. Use Cisco Unified Communications Manager service parameters to assign a higher priority to router A's CFB. Then associate the MRG with an MRGL and apply it to all conference resource consumers.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 22

82
Which Device Pool configuration setting will override the device-level settings only when a device is roaming within a device mobility group?

A. Region
B. Location
C. SRST Reference
D. Calling Party Transformation CSS
E. Media Resource Group List

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Device Mobility Related Settings:
The parameters under these settings will override the device-level settings only when the device is roaming within a Device Mobility Group. The parameters included in these settings are:
• Device Mobility Calling Search Space
• AAR Calling Search Space
• AAR Group
• Calling Party Transformation CSS
• Called Party Transformation CSS
The device mobility related settings affect the dial plan because the calling search space dictates the patterns that can be dialed or the devices that can be reached.

QUESTION 23
83
Which two Device Pool configuration settings will override the device-level settings when a device is roaming within or outside a device mobility group? (Choose two.)

A. Adjunct CSS
B. Device Mobility CSS
C. Network Locale
D. Called Party Transformation CSS
E. AAR CSS
F. Device Mobility Group

Correct Answer: CF
Section: (none)

Explanation

Explanation/Reference:
Explanation:
The parameters under these settings will override the device-level settings when the device is roaming within or outside a Device Mobility Group. The parameters included in these settings are:
–

Date/time Group
–
Region
–
Media Resource Group List
–
Location
–
Network Locale
–
SRST Reference
–
Physical Location
–
Device Mobility Group
The roaming sensitive settings primarily help in achieving proper call admission control and voice codec selection because the location and region configurations are used based on the device's roaming device pool.
Reference: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42dvmobl.html

QUESTION 24
84
Which Call Admission Control mechanism is supported for the Cisco Extension Mobility Cross Cluster solution?

A. Location CAC
B. RSVP CAC
C. H.323 gatekeeper
D. intercluster Enhanced Location CAC
E. visiting cluster's LBM hub

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Configuring extension mobility cross cluster (EMCC) is nothing you should take lightly. EMCC requires a lot of configuration parameters including the exporting and importing of each neighbor cluster's X.509v3 digital certificates. EMCC is supported over SIP trunks only. Presence is another feature that's only supported over SIP trunks. If you want to be able to perform scalable Call Admission Control (CAC) in a distributed multi-cluster call processing model, you will need to point an H.225 or Gatekeeper controlled trunk to an H.323 Gatekeeper for CAC… but if you want to

QUESTION 25
85
Which two Cisco Unified Communications Manager SIP profile configuration parameters for a SIP intercluster trunk are mandatory to enable end-to-end RSVP SIP Preconditions between clusters? (Choose two.)

A. Set the RSVP over SIP parameter to Local RSVP.
B. Set the RSVP over SIP parameter to E2E.
C. Set the SIP Rel1XX Options parameter to Disabled.
D. Set the SIP Rel1XX Options parameter to Send PRACK If 1xx Contains SDP.
E. Set the SIP Rel1XX Options parameter to Send PRACK for All 1xx Messages.
F. Check the Fall Back to Local RSVP check box.

Correct Answer: BD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Each Unified Communications Manager cluster and Unified CME should have the same configuration information. For example, Application ID should be the same on each UnifiedCommunications Manager cluster and Unified CME. RSVP Service parameters should be the same on each Unified Communications Manager cluster.

QUESTION 26
86
What is the number of directory URIs with which a Cisco Unified Communications Manager directory number can be associated?

A. 1
B. up to 2

C. up to 3
D. up to 4
E. up to 5

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the username@host format where the host portion is an IPv4 address or a fully qualified domain name. A directory URI is a uniform resource identifier, a string of characters that can be used to identify a directory number. If that directory number is assigned to a phone, Cisco Unified Communications Manager can route calls to that phone using the directory URI. URI dialing is available for SIP and SCCP endpoints that support

directory URIs.

QUESTION 27
87
Which Cisco Unified Communications Manager partition will be associated with a directory URI that is configured for an end user with a primary extension?

A. null
B. none
C. directory URI
D. default
E. any partition that the Cisco Unified Communications Manager administrator desires

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the username@host format where the host portion is an IPv4 address or a fully qualified domain name. A directory URI is auniform resource identifier, a string of characters that can be used to identify a directory number. If that directory number is assigned to a phone, Cisco Unified Communications Manager can route calls to that

phone using the directory URI. URI dialing is available for SIP and SCCP endpoints that support directory URIs.

QUESTION 28
88
Which Call Control Discovery function allows the local Cisco Unified Communications Manager to listen for advertisements from remote call-control entities that use the SAF network?

A. CCD advertising service
B. CCD requesting service
C. SAF forwarder
D. SAF enabled trunks
E. CCD registration service

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
SAF and CCD will allow large distributed multi-cluster deployments to have the directory number

(DN) ranges of each call routing element advertised dynamically over SAF. Cisco routers act as SAF Forwarders (SAFF), while the call routing elements (e.g. CUCM) act as clients that register with the routers to advertise their DN ranges and listen to the advertisements of other routers.

QUESTION 29
89
Which message is used by a Cisco Unified Communications Manager respond to periodic keepalives from a Cisco IOS MGCP gateway?

A. AUEP
B. RQNT
C. NTFY
D. 200
E. AUCX

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
(2xx) Successful completion:The requested transaction was executed normally.

QUESTION 30
90
What is the maximum number of Cisco Unified Communications Manager subscriber pairs in a megacluster deployment?

A. 4
B. 8
C. 12
D. 16
E. 32

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
There can be up to 8 pairs of subscribers, 16 subscribers total and must be in a 1:1 redundancy mode (8 active, 8 standby).

QUESTION 31
91
Refer to the exhibit.

A user is performing a series of dialing steps on a SCCP IP phone (extension 1001) to call another SCCP IP phone (extension 2003). Both phones are registered to the same Cisco Unified Communications Manager cluster.
Which user inputs are sent from the calling IP phone to the Cisco Unified Communications Manager, in the form of SCCP messages, after the user takes the phone off-hook?
(The commas in the options are logical separators, not part of the actual user input or SCCP

messages.)

A. A separate SCCP message is sent to the Cisco Unified Communications Manager for each of these user inputs: 2, 0, 0, 3
B. A separate SCCP message is sent to the Cisco Unified Communications Manager for each of these user inputs: 2, 0, 1, <<, 0, 3
C. The IP phone collects all keypad and soft key events until user inputs stops, then it sends a single SCCP message to report that 2003 has been dialed.
D. The IP phone collects all keypad and soft key events until user inputs stops, then it sends a single SCCP message to report that 201<<03 has been dialed.
E. A separate SCCP message is sent to the Cisco Unified Communications Manager for each of these user inputs: 2, 0, 1, <<, 2, 0, 0, 3

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Because sccp phones send digits DIGIT-by-DIGIT i.e. it sends each digit in real time.
Link:-https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phones-anddial-rules

QUESTION 32
92

You have implemented 5-digit forced authorization codes to all international route patterns on Cisco Unified Communications Manager. Your users report that after entering the FAC codes, they must wait for more than 10 seconds before the call is routed.
Which procedure eliminates the wait time?

A. Check and eliminate any existing route patterns that overlap with the international route pattern.
B. Go to the Cisco Unified Communications Manager Service Parameters and reduce the T-304 number to 5000 milliseconds.
C. Request your long distance telephone service provider to reduce the call setup time to 5 seconds.
D. Configure a # (hash) sign to the end of the forced authorization codes to signal the end of dialing.
E. Educate the users to press # (hash) after entering the forced authorization codes.

Correct Answer: E

Section: (none)
Explanation

Explanation/Reference:
Explanation:
Because it immediately stops taking digits and route the digits to CUCM, otherwise the call occurs after the interdigit timer expire which is 15 seconds by default.
Reference:http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unifiedcommunications-manager-callmanager/81541-fac-config-ex.html

QUESTION 33
93
Refer to the exhibit.

A user is going through a series of dialing steps on a SIP Type A IP phone to call a SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assume that the calling SIP phone is not associated with any SIP dial rules.

Which statement about how digits are forwarded to the Cisco Unified Communications Manager for further call processing is true?

A. As each digit is pressed on the SIP IP phone, it is sent to the Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event.
B. The SIP IP phone waits for the inter-digit timer expiry and then sends each digit to the Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. The SIP IP phone waits for the inter-digit timer expiry or for the Dial soft key to be selected before it sends all digits to the Cisco Unified Communications Manager in a SIP INVITE message.
D. The SIP IP phone waits for the inter-digit timer expiry, or for the Dial soft key to be selected before it sends the first digit in a SIP INVITE and the subsequent digits in SIP NOTIFY messages.
E. The SIP IP phone sends all digits to the Cisco Unified Communications Manager in a SIP INVITE message as soon as the fourth digit is pressed.

Correct Answer: C
Section: (none)

Explanation

Explanation/Reference:
Explanation:
Because Type A SIP phone with no SIP dial rules sends digit in Enbloc style.All digits are sent to CUCM after the user completes the dialing and press the Dial softkey.
Reference: https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phonesand-dial-rules.

QUESTION 34
94
Which call processing feature overrides the Do Not Disturb settings on a Cisco IP phone?

A. park reversion for remotely parked calls by a shared line
B. hold reversion
C. remotely placed pickup request by a shared line
D. pickup notification
E. terminating side of a call back

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Hold Reversion and Intercom
Hold reversion and intercom override DND (both options), and the call gets presented normally.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/7_1_2/ccmfeat/fsgd-712-cm/fsdnd.html#wp1072487

QUESTION 35
95
Which statement describes the Maximum Serving Count service parameter of the Cisco TFTP service on Cisco Unified Communications Manager?

A. It specifies the maximum number of files in the TFTP server disk storage.
B. It specifies the maximum number of TFTP client requests to accept and to serve files at a given time.
C. It specifies the maximum file support by the Cisco TFTP service.
D. It specifies the maximum file counts, in cache as well as in disk, that are supported by the Cisco

TFTP service.

E. It specifies the maximum number of TFTP client requests to accept and to serve files in a 120-minute window.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
This parameter specifies the maximum number of client requests to accept and to serve files at a time. Specify a low value if you are serving files over a low bandwidth connection. You can set it to a higher number if you are serving small files over a large bandwidth connection and when CPU resources are available, such as when no other services run on the TFTP server. Use the default value if the TFTP service is run along with other Cisco CallManager services on the same server. Use the following suggested values for a dedicated TFTP server: 1500 for a single-processor system and 3000 for a dual-processor system. If the dual-processor system is running Windows 2000 Advanced Server, the serving count can be up to 5000.
This is a required field.

QUESTION 36
96
In a Cisco Unified Communications Manager design where +E.164 destinations are populated in directory entries, which call routing practice is critical to prevent unnecessary toll charges caused by internal calls routed through the PSTN?

A. forced on-net routing
B. automated alternate routing
C. forced authorization codes
D. client matter codes
E. tail-end hop-off

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:

It is not uncommon for the dialing habits for on-net/inter-site and off-net destinations to use the same addressing structure. In this case the call control decides whether the addressed endpoint, user, or application is on-net or off-net based on the dialed address, and will treat the call as onnet

or off-net, respectively.
Figure 14-4 shows an example of this forced on-net routing. All four calls in this example are dialed as 91 plus 10 digits. But while the calls to +1 408 555 2345 and +1 212 555 7000 are really routed as off-net calls through the PSTN gateway, the other two calls are routed as on-net calls because the call control identifies the ultimate destinations as on-net destinations. Forced on-net routing clearly shows that the dialing habit used does not necessarily also determine how a call is routed. In this example, some calls are routed as on-net calls even though the used PSTN dialing habit seems to indicate that an off-net destination is called.
Figure 14-4Forced On-Net Routing
Forced on-net routing is especially important if dialing of +E.164 destinations from directories is implemented. In a normalized directory, all destinations are defined as +E.164 numbers, regardless of whether the person that the number is associated with is internal or external. In this case forced on-net routing is a mandatory requirement to avoid charges caused by internal calls routed through the PSTN.
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/dialplan.html

QUESTION 37
97
Refer to the exhibit.

A user is going through a series of dialing steps on a SIP Type B IP phone to call a SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assume that the calling SIP phone is associated with a SIP dial rule with a pattern value of "2001". Which statement about how digits are forwarded to the Cisco Unified Communications Manager for further call processing is true?

A. As each digit is pressed on the SIP IP phone, it is sent to the Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event.
B. The SIP IP phone waits for the inter-digit timer expiry and then sends each digit to the Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. As soon as the user selects the Dial soft key, the SIP IP phone forwards all digits to the Cisco Unified Communications Manager in a SIP INVITE message.
D. As soon as the Dial soft key is selected, the SIP IP phone forwards the first digit in a SIP INVITE and the subsequent digits in SIP INFORMATION messages.
E. The SIP IP phone waits for the inter-digit timer expiry, and then sends all digits to the Cisco

Unified Communications Manager in a SIP INVITE message.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Reference: https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phonesand-dial-rules.

QUESTION 38
98
Which option is a characteristic of the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager?

A. It accounts for network protocol rerouting.
B. It accounts for network downtime and failures.
C. It supports dynamic bandwidth adjustments based on WAN topology changes.
D. It supports asymmetric media flows such that different bit rates in each direction are deducted accordingly.
E. Unidirectional media flows are deducted as if they were bidirectional.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Network Modeling with Locations, Links, and Weights
Enhanced Location CAC is a model-based static CAC mechanism. Enhanced Location CAC involves using the administration interface in Unified CM to configure Locations and Links to model the "Routed WAN Network" in an attempt to represent how the WAN network topology routes media between groups of endpoints for end-to-end audio, video, and immersive calls. Although Unified CM provides configuration and serviceability interfaces in order to model the network, it is still a "static" CAC mechanism that does not take into account network failures andnetwork protocol rerouting. Therefore, the model needs to be updated when the WAN network topology changes. Enhanced Location CAC is also call oriented, and bandwidth deductions are per-call not per-stream, so asymmetric media flows where the bit-rate is higher in one direction than in the other will always deduct for the highest bit rate.In addition, unidirectional media flows will be deducted as if they were bidirectional media flows.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/cac.html

QUESTION 39
99
Which two host portion format conditions are true for directory URI on Cisco Unified Communications Manager? (Choose two.)

A. It is case sensitive.
B. It cannot start with a hyphen.
C. It must have at least one character.
D. It supports IPv4 or IPv6 addresses, or fully qualified domain names.
E. It cannot end with a hyphen.
F. It supports the & character.

Correct Answer: BE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports the following formats in the host portion of a directory URI (the portion after the @ symbol):
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/9_1_1/CUCM_BK_C271A69D_00_cucm-bulk-administration-guide-91/CUCM_BK_C271A69D_00_cucm-bulk-administration-guide-91_chapter_01001110.html

QUESTION 40
100
Which two user portion format conditions are true for directory URI on Cisco Unified Communications Manager 9.1 or later? (Choose two.)

A. It supports the $ character.
B. It support space between characters.
C. It has a maximum length of 50 characters.
D. It has a maximum length of 254 characters.
E. It is always case-sensitive.
F. It cannot be a directory number.

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports the following formats in the user portion of a directory URI (the portion before the @ symbol):
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/9_1_1/CUCM_BK_C271A69D_00_cucm-bulk-administration-guide-91/CUCM_BK_C271A69D_00_cucm-bulk-administration-guide-91_chapter_01001110.html

QUESTION 41
101
Which configuration parameter defines whether or not the user portion of a directory URI is case sensitive on Cisco Unified Communications Manager 9.1 or later?

A. URI Dialing Display Preference in Cisco CallManager Service Parameter
B. URI Lookup Policy in Cisco CallManager Service Parameter
C. URI Dialing Display Preference in Enterprise Parameters
D. URI Lookup Policy in Enterprise Parameters
E. The user portion of a directory URI is always case sensitive and cannot be changed.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation: Cisco Unified Communications Manager supports the following formats in the user portion of a directory URI (the portion before the @ symbol):
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/9_1_1/CUCM_BK_C271A69D_00_cucm-bulk-administration-guide-91/CUCM_BK_C271A69D_00_cucm-bulk-administration-guide-91_chapter_01001110.html

QUESTION 42
102
The number of calls waiting in a Cisco Unified Communications Manager native call queue has reached its maximum limit.
Which statement about what happens to additional incoming calls is true?

A. Calls are handled according to the Forward Hunt Busy settings on the Hunt Pilot configuration page.
B. Calls are handled according to the Forward Hunt No Answer settings on the Hunt Pilot configuration page.
C. Calls are handled according to the Forward Hunt Busy settings on the Line Group members.
D. Calls are handled according to the Hunt Options settings on the Line Group Configuration page.
E. Calls are handled according to the When Queue Is Full settings on the Hunt Pilot Configuration page.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
There are three main scenarios where alternate numbers are used:
When queue is full
Call Queuing allows up to 100 callers to be queued per hunt pilot (the maximum number of callers allowed in queue on a hunt pilot page). Once this limit for new callers been reached on a particular hunt pilot, subsequent calls can be routed to an alternate number.This alternate number can be configured through the Hunt Pilot configuration page (through the “Destination When Queue is Full” settings).
When maximum wait time is met
Each caller can be queued for up to 3600 seconds per hunt pilot (the maximum wait time in queue). Once this limit is reached, that caller is routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the “Maximum wait time in queue” settings).
When no hunt members are logged in or registered
In a scenario where none of the members of the hunt pilot are available or registered at the time of the call, hunt pilot configuration provides an alternate number field (through the “When no hunt members are logged in or registered” settings) where calls can be routed. For Call Queuing, a hunt pilot member is considered available if that member has both deactivated do not disturb (DND) and logged into the hunt group. In all other cases, the line member is considered unavailable or logged off.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmfeat/CUCM_BK_CEF0C471_00_cucm-features-services-guide-90/CUCM_BK_CEF0C471_00_cucm-features-andservices-guide_chapter_0111.html

QUESTION 43
103
A queued call has reached the maximum wait time configured for a Cisco Unified Communications

Manager native call queue.
Which statement about what happens to this queued call is true?

A. Calls are handled according to the Forward Hunt No Answer settings on the Hunt Pilot configuration page.
B. Calls are handled according to the When Maximum Wait Time Is Met settings on the Hunt Pilot Configuration page.
C. Calls are handled according to the When Maximum Wait Time Is Met settings in Cisco Unified Communications Manager Service Parameters.
D. Calls are handled according to the Not Available Hunt Option settings on the Line Group Configuration page.
E. Calls are handled according to the When Queue Is Full settings on the Hunt Pilot Configuration page.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
There are three main scenarios where alternate numbers are used:
When queue is full
Call Queuing allows up to 100 callers to be queued per hunt pilot (the maximum number of callers allowed in queue on a hunt pilot page). Once this limit for new callers been reached on a particular hunt pilot, subsequent calls can be routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the“Destination When Queue is Full” settings).
When maximum wait time is met
Each caller can be queued for up to 3600 seconds per hunt pilot (the maximum wait time in queue). Once this limit is reached, that caller is routed to an alternate number.This alternate number can be configured through the Hunt Pilot configuration page(through the “Maximum wait time in queue” settings).
When no hunt members are logged in or registered
In a scenario where none of the members of the hunt pilot are available or registered at the time of the call, hunt pilot configuration provides an alternate number field (through the “When no hunt members are logged in or registered” settings) where calls can be routed. For Call Queuing, a hunt pilot member is considered available if that member has both deactivated do not disturb (DND) and logged into the hunt group. In all other cases, the line member is considered unavailable or logged off.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmfeat/CUCM_BK_CE

F0C471_00_cucm-features-services-guide-90/CUCM_BK_CEF0C471_00_cucm-features-andservices-guide_chapter_0111.html

QUESTION 44
104
Which statement about what happens to incoming calls to a Cisco Unified Communications Manager native call queue when no hunt members are logged in or registered is true?

A. Calls are handled according to the Forward Hunt No Answer settings on the Hunt Pilot configuration page.
B. Calls are handled according to the Not Available Hunt Option settings on the Line Group Configuration page.
C. Calls are handled according to the Forward Hunt Busy settings on the Hunt Pilot configuration page.
D. Calls are forward to the Forward Busy Calls To destination if configured; otherwise the calls are disconnected.
E. Calls are handled according to the correspondent parameters under the Queuing section on the Hunt Pilot Configuration page.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
There are three main scenarios where alternate numbers are used:
When queue is full
Call Queuing allows up to 100 callers to be queued per hunt pilot (the maximum number of callers allowed in queue on a hunt pilot page). Once this limit for new callers been reached on a particular hunt pilot, subsequent calls can be routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the“Destination When Queue is Full” settings).
When maximum wait time is met
Each caller can be queued for up to 3600 seconds per hunt pilot (the maximum wait time in queue). Once this limit is reached, that caller is routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the “Maximum wait time in queue” settings).
When no hunt members are logged in or registered
In a scenario where none of the members of the hunt pilot are available or registered at the time of the call, hunt pilot configuration provides an alternate number field (through the “When no hunt members are logged in or registered” settings) where calls can be routed. For Call Queuing, a hunt pilot member is considered available if that member has both deactivated do not disturb

(DND) and logged into the hunt group. In all other cases, the line member is considered unavailable or logged off.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmfeat/CUCM_BK_CEF0C471_00_cucm-features-services-guide-90/CUCM_BK_CEF0C471_00_cucm-features-andservices-guide_chapter_0111.html

QUESTION 45
105
Which statement about what happens to a hunt member who does not answer queuing-enabled hunt-list call in Cisco Unified Communications Manager 9.1 is true?

A. The hunt member is logged off automatically and must press HLOG to log back in.
B. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in Cisco Unified Communications Manager Service Parameters.
C. The hunt member is logged off automatically and must manually reset the phone to log back in.
D. The hunt member is logged off if Automatically Logout Hunt Member on No Answer is selected on the Line Group configuration page.
E. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in Hunt Pilot configuration page.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
If a line member does not answer a queue-enabled call, that line member is logged off the hunt group only if the setting "Automatically Logout Hunt Member on No Answer" is selected on the line group page.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcfg/CUCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0100011.html

QUESTION 46
106
Which SIP request is used by a Cisco 9971 IP Phone to signal DND status changes to Cisco Unified Communications Manager?

A. REGISTER
B. NOTIFY

C. INFO
D. PUBLISH
E. UPDATE

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports Do Not Disturb that a SIP device initiates or that a Cisco Unified Communications Manager device initiates. A DND status change gets signaled from a SIP device to Cisco Unified Communications Manager by using the SIP PUBLISH method (RFC3909). A DND status change gets signaled from a Cisco Unified Communications Manager to a SIP device by using a dndupdate Remote-cc REFER request. Cisco Unified Communications Manager can also publish the Do Not Disturb status for a device, along with the busy and idle status for the device.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/6_1_1/ccmfeat/cmfsgd611/fsdnd.html

QUESTION 47
107
Which SIP request is used by Cisco Unified Communications Manager to signal DND status changes to a Cisco 9971 IP Phone?

A. OPTIONS
B. NOTIFY
C. INFO
D. REFER
E. UPDATE

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports Do Not Disturb that a SIP device initiates or that a Cisco Unified Communications Manager device initiates. A DND status change gets signaled from a SIP device to Cisco Unified Communications Manager by using the SIP PUBLISH method (RFC3909). A DND status change gets signaled from a Cisco Unified Communications Manager

to a SIP device by using a dndupdate Remote-cc REFER request. Cisco Unified Communications Manager can also publish the Do Not Disturb status for a device, along with the busy and idle status for the device.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/6_1_1/ccmfeat/cmfsgd611/fsdnd.html

QUESTION 48
108

You are assisting a customer to troubleshoot a SIP early-offer problem with a SIP service provider. You have enabled Cisco CallManager trace and set the debug trace level to Detailed for SIP Call Processing trace on their standalone Cisco Unified Communications Manager 9.1 system. Using the RTMT tool, your customer has remote browsed to the Cisco UCM and asked you which trace file to download.
What is the trace file name syntax in which detailed SIP messages are logged?

A. SDL
B. SDI
C. CCM
D. Call logs
E. Traces

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
SDL files log SIP messages from CCM.

QUESTION 49
109
Which tag in the SIP header is used by Cisco Unified Communications Manager to deliver a blended identity of alpha URI and number?

A. x-cisco-callinfo
B. x-cisco-service-control
C. x-cisco-serviceuri
D. x-cisco-number
E. x-cisco-uri

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports blended addressing of directory URIs and directory numbers. When blended addressing is enabled across the network, Cisco Unified Communications Manager inserts both the directory URI and the directory number of the sending party in outgoing SIP Invites, or responses to SIP Invites. The destination endpoint has the option of using either the directory URI or the directory number for its response—both will reach the same destination.
Cisco Unified Communications Manager uses the x-cisco-number tag in the SIP identity headers to communicate a blended address. When both a directory URI and directory number are available for the sending phone and blended addressing is enabled, Cisco Unified Communications

Manager uses the directory URI in the From fields of the SIP message and adds the x-cisconumber tag with the accompanying directory number to the SIP identity headers. The x-cisconumber tag identifies the directory number that is associated with the directory URI.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-featuresservices-guide-100_chapter_0110011.html#CUCM_RF_D0008C2B_00

QUESTION 50
110
Which SIP header is used by Cisco Unified Communication Manager to support the Redirected Number ID Service?

A. replaces
B. RPID
C. diversion
D. join
E. P-charging-vector

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
CUCM uses sip diversion header in INVITE message to carryout Redirected Number ID service.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_systemguide_chapter_0101000.html#CUCM_TP_R3F173A9_00

QUESTION 51
111
What is the maximum one-way delay, in milliseconds, between any two Cisco Unified Communications Manager servers in a non-Session Management Edition cluster over an IP WAN?

A. 20
B. 40
C. 80
D. 160
E. 250

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The maximum one-way delay between any two Unified CM servers should not exceed 40 msec, or 80 msec round-trip time. Propagation delay between two sites introduces 6 microseconds per kilometer without any other network delays being considered. This equates to a theoretical maximum distance of approximately 3000 km for 20 ms delay or approximately 1860 miles. These distances are provided only as relative guidelines and in reality will be shorter due to other delay incurred within the network
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/7x/uc7_0/models.html

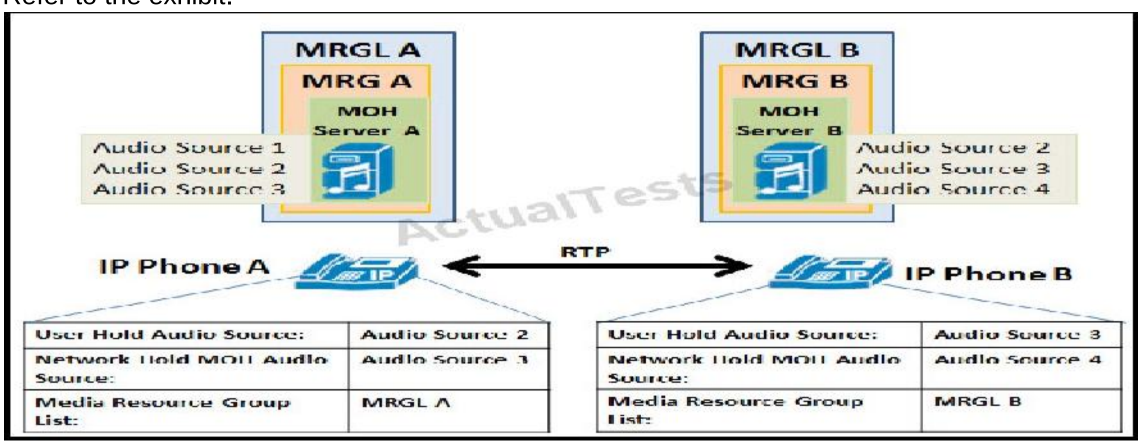
QUESTION 52
112
Which Cisco Unified Communications Manager deployment model for clustering over the IP WAN mandates a primary and a backup subscriber at the same site?

A. multisite with centralized call processing
B. multisite with distributed call processing
C. local failover
D. remote failover
E. remote failover with Cisco Unified Communications Manager Express as SRST

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Clustering Over the IP WAN
You may deploy a single Unified CM cluster across multiple sites that are connected by an IP WAN with QoS features enabled. This section provides a brief overview of clustering over the WAN. For further information, refer to the chapter on Call Processing.
Clustering over the WAN can support two types of deployments:
•Local Failover Deployment Model
Local failover requires that you place the Unified CM subscriber and backup servers at the same site, with no WAN between them. This type of deployment is ideal for two to four sites with Unified CM.
•Remote Failover Deployment Model
Remote failover allows you to deploy primary and backup call processing servers split across the WAN. Using this type of deployment, you may have up to eight sites with Unified CM subscribers being backed up by Unified CM subscribers at another site.
Reference: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42models.html

QUESTION 53
113
Refer to the exhibit.



All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assume that the provided configurations exist at the phone line level and multicast MOH is disabled cluster wide.

Which description of what will happen when the user of IP phone A presses the Hold soft key is true?

A. IP phone B receives audio source 2 from MOH server A.
B. IP phone B receives audio source 3 from MOH server A.
C. IP phone B receives audio source 2 from MOH server B.
D. IP phone B receives audio source 3 from MOH server B.
E. IP phone B receives audio source 1 from MOH server A.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Because audio source 2 is in top of the MRGL List and it will be selected locally first.

QUESTION 54
114
What is the maximum number of option 150 IP addresses that a Cisco IP SCCP phone will accept and use from a DHCP server?

A. 1
B. 2
C. 3
D. 4
E. 5

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Unified IP Phones use the option 150 value as the TFTP server IP address when Alternate TFTP option is set to No. You can assign only IP addresses as Option 150 values. A maximum of two IP addresses get used, and only the first two IP addresses that the DHCP server provides get accepted.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_systemguide_chapter_01010.html

QUESTION 55
115
What is the maximum number of option 66 IP addresses that a Cisco IP SCCP phone will accept and use from a DHCP server?

A. 1
B. 2
C. 3
D. 4
E. 5

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Reference: http://www.techtronicssolution.com/blog/?p=1201

QUESTION 56
116
Which two statements about BFCP with Cisco Unified Communications Manager are true?

(Choose two.)

A. BFCP is supported only on full SIP networks.
B. Cisco Unified Communications Manager allows BFCP only over UDP.
C. BFCP is not supported for third-party endpoints.
D. BFCP is not supported through Cisco Unified Border Element.
E. BFCP is supported between Cisco Unified Communications Manager and a TelePresence MCU.

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
BFCP configuration tips
To enable BFCP in Cisco Unified Communications Manager, check the Allow Presentation Sharing using BFCP check box in the SIP Profile Configuration window. If the check box is

unchecked, all BFCP offers will be rejected. By default, the check box is unchecked.
BFCP is supported only on full SIP networks. For presentation sharing to work, BFCP must be enabled for all SIP endpoints as well as all SIP lines and SIP trunks between the endpoints.
CUCM uses BFCP over user datagram protocol (UDP) in both secure and non-secure BFCP modes.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_systemguide_chapter_0101011.html

QUESTION 57
117
When the Cisco Unified Communications Manager service parameter "Auto Call Pickup Enabled" is selected, which two softkeys on an IP phone connect you to an incoming call? (Choose two.)

A. Pickup
B. Gpickup
C. CallBack
D. Select
E. Join

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Pickup softkey is used to receive a call that is ringing in another phone within the same pickup group and Gpickupsoftkey is used to receive calls that are ringing but that phone is another pickup group.

QUESTION 58
118
On a Cisco Unified Communications Manager SIP trunk with a single remote device and OPTIONS ping feature enabled, which response from the SIP remote peer causes the trunk to be marked as "Out of Service"?

A. 401 Unauthorized
B. 505 Version Not Supported
C. 406 Not Acceptable
D. 408 Request Timeout
E. 500 Server Internal Error

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
408 Request Timeout
Couldn’t find the user in time. The server could not produce a response within a suitable amount of time, for example, if it could not determine the location of the user in time. The client MAY repeat the request without modifications at any later time
Reference:http://en.wikipedia.org/wiki/List_of_SIP_response_codes

QUESTION 59
119
How many music on hold servers are required in a trunk-only megacluster of Cisco Unified Communications Manager Session Management Edition?

A. 0
B. 1
C. 2
D. 3
E. 4

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When considering a megacluster deployment, the primary areas impacting capacity are as follows:
•The megacluster may contain a total of 21 servers consisting of 16 subscribers, 2 TFTP servers, 2 music on hold (MoH) servers (0 required), and 1 publisher

•Server type must be either Cisco MCS 7845-I3/H3 class or Cisco Unified Computing System (UCS) C-Series or B-Series using the 10K Open Virtualization Archive (OVA) template.
•Redundancy model must be 1:1.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_5_1/ccmfeat/fsgd-851-cm/fsmoh.html#wp1027431

QUESTION 60
120

Which two applications can connect directly with a Cisco Unified Communications Manager Session Management Edition cluster? (Choose two.)

A. Cisco Unity
B. Cisco Unified Meeting Place Express
C. Cisco Unified Contact Center Enterprise
D. Cisco Unified Contact Center Express
E. Cisco Unified Communications Manager Attendant Console
F. Cisco Emergency Responder

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The deployment of a Unified CM Session Management Edition enables commonly used applications such as conferencing or videoconferencing to connect directly to the Session Management cluster, thus reducing the overhead of managing multiple trunks to leaf systems. Cisco Unity or other voicemail systems can be deployed at all sites and integrated into the Unified CM cluster.

QUESTION 61
121
Which two applications must be connected to a leaf cluster in a Cisco Unified Communications Manager Session Management Edition deployment? (Choose two.)

A. Cisco Unified Meeting Place
B. Cisco Unified Contact Center Express
C. H.323-based video conferencing systems
D. Cisco Unity
E. Cisco Unified Communications Manager
F. fax servers

Correct Answer: BE
Section: (none)
Explanation

Explanation/Reference:
Explanation:

The deployment of a Unified CM Session Management Edition enables commonly used applications, such as conferencing or videoconferencing to connect directly to the session management cluster, which reduces the overhead of managing multiple trunks to leaf systems.
Unified CM Session Management Edition supports the following applications:

- Unity, Unity Connection
- Meeting Place, Meeting Place Express
- SIP and H.323-based video conferencing systems
- Third Party voice mail systems
- Fax servers
- Cisco Unified Mobility

The following applications must connect to the leaf cluster:

- Unified Contact Centre, CUCM, Unified Contact Centre Express
- Cisco Unified Presence Server
- Attendant Console
- Manager Assistant
- IP IVR
- Cisco Voice Portal

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/session_mgmt/deploy/8_5_1/overview.html

QUESTION 62
122

F. originating side of a call back

A.
B.
C.
D.

Correct Answer:

Section: (none)
Explanation

Explanation/Reference:
Explanation:
For the DND Ringer Off option, only visual notification gets presented to the device.
For the DND Call Reject option, no notification gets presented to the device.
For the terminating side of the call, Do Not Disturb overrides call back:
When the phone that terminates the call has DND Call Reject enabled but the phone becomes available (goes off hook and on hook), a new screen will be presented to the originating device as “<Extension> has become available but is on DND-R”. Callback available notification will be sent only after the terminating side disables DND Call Reject.

QUESTION 63
123
How many destinations can be configured for a SIP trunk on a Cisco Unified Communications Manager 9.1 system when the destination address is an SRV?

A. 1
B. 2
C. 3
D. 8
E. 16

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
SIP trunks can be configured with up to 16 destination IP addresses, 16 fully qualified domain names, or a single DNS SRV entry.
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/uc8x/trunks.html

QUESTION 64
124
Which design restriction applies to Cisco Unified Communications Manager Session Management Edition clustering over the WAN deployment with extended round-trip times in Cisco Unified CM 9.1 and later releases?

A. SIP and H.323 intercluster trunks are supported.
B. Only SIP trunk is supported.

C. SIP trunks and H.323 gateways are supported.
D. A minimum of 1.544 Mb/s bandwidth is required for all traffic between any two nodes in the cluster.
E. Only RSVP agents can be configured and registered to the SME cluster as media resources.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Using only SIP trunks in the SME cluster allows you to deploy a "media transparent" cluster where media resources, when required, are inserted by the end or leaf Unified Communications system and never by SME. Using only SIP trunks also allows you to use extended round trip times (RTTs) between SME nodes when clustering over the WAN.

QUESTION 65
125
Which statement about using the Answer File Generator to load a Cisco Unified Communications virtual machine is true?

A. You must copy the output text to a file named platformConfig.txt.
B. Each host should be copied to its own configuration file.
C. The answer file can be used only when performing the new identity process to load the Cisco Unified Communications virtual machines.
D. The configuration file should be placed inside an ISO file and mounted on the virtual machine.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/install/9_0_1/CUCM_BK_I87B437D_00_installing-cucm-90/CUCM_BK_I87B437D_00_installing-cucm-90_chapter_0100.html#CUP0_TK_G0262E75_00

QUESTION 66
126
Which method does a Cisco Unified 9971 phone use to send keep-alive messages to Cisco Unified Communications Manager?

A. SIP NOTIFY with Event set to keep-alive
B. SIP OPTIONS
C. SIP REGISTER with Expires set to zero
D. SCCP StationRegister
E. SCCP StationServerReq

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Phone registers with primary and establishes keepalive connection with secondary.
Expires = 0 keepalive mechanism allows Cisco SIP Phones to more closely resemble the failover / fallback behavior of SCCP.

QUESTION 67
127
Which three requirements must be met to share Enhanced Location Based Call Admission Control bandwidth usage between clusters? (Choose three.)

A. The Cisco Unified Communications Manager version must be 8.6 or higher.
B. The location name must be the same on both clusters.
C. SIP ICT must use the Shadow location.
D. The Location Bandwidth Manager Service should be started on only two servers in each cluster.
E. A Location Bandwidth Manager Hub Group must be created for each cluster.
F. Links must be created to the Shadow location.

Correct Answer: BCE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Recommendations and Design Considerations for Unified CM Session Management Edition Deployments
•
All leaf clusters that support E-L CAC should be enabled for intercluster E-L CAC with SME.
•
SME can be used as a centralized bootstrap hub for the E-L CAC intercluster hub replication

network. See LBM Hub Replication Network, for more information.

- All trunks to leaf clusters supporting E-L CAC should be SIP trunks placed in the shadow location to enable E-L CAC on the trunk between SME and the leaf clusters supporting E-L CAC.

- For TelePresence video interoperability, see the section on Call Admission Control Design Recommendations for TelePresence Video Interoperability Architectures.

- Connectivity from SME to any trunk or device other than a Unified CM that supports E-L CAC (some examples are third-party PBXs, gateways, Unified CM clusters prior to release 9.0 that do not support E-L CAC, voice messaging ports or trunks to conference bridges, Cisco Video Communications Server, and so forth) should be configured in a location other than a phantom or shadow location. The reason for this is that both phantom and shadow locations are nonterminating locations; that is, they relay information about locations and are effectively

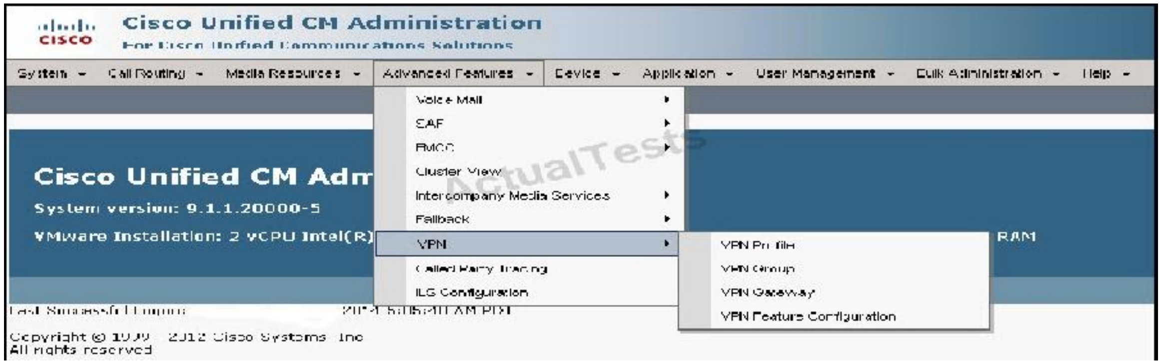
that allow for the transmission of location information in versions of Unified CM prior to 9.0, but they are not supported with Unified CM 9.x Enhanced Locations CAC. Shadow locations are special locations that enable trunks between Unified CM clusters that support E-L CAC to accomplish it end-to-end.

- SME can be used as a locations and link management cluster

QUESTION 68

128

Refer to the exhibit.



On which two Cisco Unified CM Administration pages can a system administrator define MTU for an SSL VPN tunnel connecting between a Cisco IP phone and a Cisco IOS VPN gateway? (Choose two.)

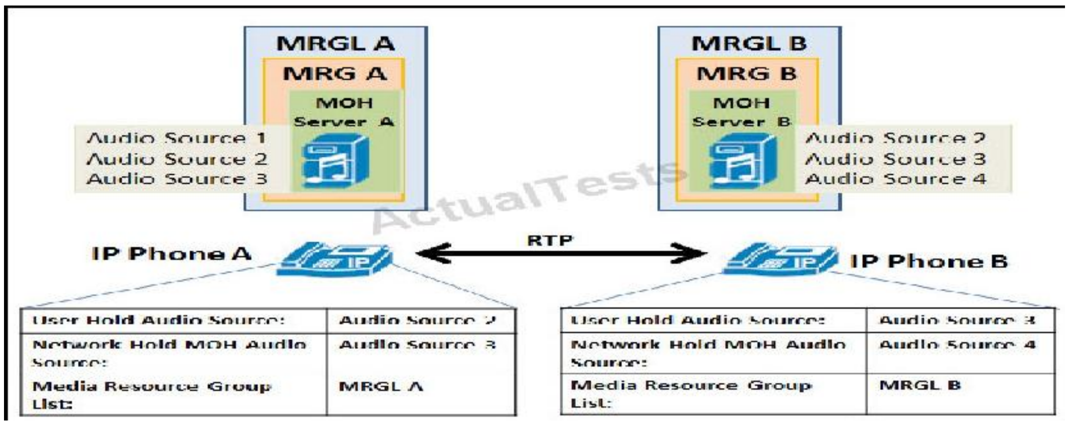
A. VPN Profile

B. VPN Group
C. VPN Gateway
D. VPN Feature Configuration
E. System, followed by Enterprise Parameters
F. System, followed by Enterprise Phone Configuration

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/8_5_1/secugd/sec-851-cm/secvpfet.html

QUESTION 69
129
Refer to the exhibit.



All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assume that the provided configurations exist at the phone line level and multicast MOH is disabled cluster wide.
Which description of what happens when the user of IP phone B presses the Transfer soft key is true?

A. IP phone A user hears audio source 3 from MOH server A.
B. IP phone A user hears audio source 4 from MOH server B.
C. IP phone A user hears audio source 3 from MOH server B.

D. IP phone A user hears tone on-hold beep tones.
E. IP phone A user hears no on-hold music or beep tones.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 70
130
When IP phone A was provisioned in a Cisco Unified Communications Manager, 2001 was configured as the directory number for its first line. Also, bob@cisco.com was defined as the only

directory URI on the Directory Number configuration page for this line. A few days later, an end user was created in the same Cisco Unified Communications Manager and was associated with the same phone with the primary extension set to 2001. Also, bobby@cisco.com was defined as a directory URI for that end user.
Which option about the primary directory URI for IP phone A is true?

A. bob@cisco.com
B. bobby@cisco.com
C. It depends on which radio button was selected next to the Directory URI entries on the Directory Configuration page.
D. Both are primary directory URIs in a manner like a shared line for DNs.
E. Neither are primary directory URIs for IP phone A.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_systemguide_chapter_0101111.html

QUESTION 71
131
Which two statements about Cisco Unified Communications Manager mixed-mode clusters are true? (Choose two.)

A. Cluster security mode configures the security capability for your standalone server or a cluster.
B. The device security mode in the phone configuration file is set to nonsecure.
C. The phone makes nonsecure connections with Cisco Unified Communications Manager even if the device security mode specifies authenticated or encrypted.
D. Security-related settings other than device security mode, such as the SRST Allowed check box, get ignored.
E. Auto-registration does not work when you configure mixed mode.

Correct Answer: AE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cluster security mode configures the security capability for a standalone server or a cluster.
Cluster security mode configures the security capability for a standalone server or a cluster.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/7_0_1/secugd/sec701-cm/secuauth.html#wp1037433

Topic 4, Cisco IOS UC Applications and Features

QUESTION 72
132
Refer to the exhibit.

```
I1 CAS Gateway(config controller)#ds0 group 1 time 1 10 type ?
  e&m-delay-dial          E & M Delay Dial
  e&m-fgd                 E & M Type II FGD
  e&m-immediate-start     E & M Immediate Start
  e&m-lmr                 E & M land mobil radio
  e&m wink start          E & M Wink Start
  ext-sig                 External Signaling
  fgd-eana                FGD-EANA BOC side
  fgd-os                  FGD-OS BOC side
  fxo-ground-start        FXO Ground Start
  fxo-loop-start          FXO Loop Start
  fxs-ground-start        FXS Ground Start
  fxs loop start          FXS Loop Start
  none                    Null Signalling for External Call Control
  <cr>
```

Which ds0-group option should you select to send automated number identification information on outbound calls for this digital T1 voice circuit?

A. e&m-fgd
B. e&m-fgd
C. fgd-eana

D. e&m-delay-dial

E. fgd-os

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
E&M signaling is often the preferred optino for CAS because it avoids glare, it provides answer/disconnect supervision and it can receive Automatic Number Identification (ANI) with FGD and send ANI with FGD-EANA. In other words, you can have 1 channel-group for incoming calls and 1 channel-group for outgoing calls.

QUESTION 73
133
Which codec complexity mode, when deployed on Cisco IOS routers with DSPs using the C5510 chipset, supports the most G.711 calls per DSP?

A. Low

B. Medium

C. High

D. Secure

E. Flex

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The flex parameter allows the complexity to automatically adjust to either medium or high complexity depending on the needs of a call. For example, if a call uses the G.711 codec, the C5510 chipset automatically adjusts to the medium-complexity mode.However, if the call uses G.729, the C5510 chipset uses the high complexity mode

QUESTION 74
134
When DSP oversubscription occurs on a Cisco IOS router using DSP modules that are based on the C5510 chipset, what will happen when an analog phone connected to a FXS port goes offhook?

A. A fast busy tone will be played.
B. A slow busy tone will be played.
C. A network busy tone will be played.
D. A dial tone will be played, but digits will not be processed.
E. No tone will be played.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When DSP oversubscription occurs for both analog ports and digital ports, except PRI and BRI. FXO signaling and application controlled endpoints are not supported. This feature does not apply to insufficient DSP credits due to mid-call codec changes (while a call is already established).

QUESTION 75
135
Refer to the exhibit.

```
!
ephone-dn 1 octo-line
 number 2001
 huntstop channel 6
!
ephone 1
 mac-address 1111.1111.1111
 max-calls-per-button 5
 busy-trigger-per-button 3
 type 7965
 button 1:1
!
ephone 2
 mac-address 2222.2222.2222
 max-calls-per-button 6
 busy-trigger-per-button 4
 type 7965
 button 1:1
!
```

How many simultaneous outbound calls are possible with this Cisco Unified Communications Manager Express configuration on these two phones?

A. 6
B. 7
C. 8
D. 9

E. 11

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Ephone is configured as octo line so maximum call number is 8 and it will be divided between lines.

QUESTION 76
136

Refer to the exhibit.

```
!
ephone-dn 1 octo-line
 number 2001
 huntstop channel 4
!
ephone 1
 mac-address 1111.1111.1111
 max-calls-per-button 5
 busy-trigger-per-button 3
 type 7965
 button 1:1
!
```

Ephone 1 has three active calls. The first two calls were inbound calls, which the user put on hold to place a third call outbound. What will happen on ephone 1 when a fourth call arrives for extension 2001?

A. The fourth call will be delivered to ephone 1 because it only received two inbound calls, one call less than the busy-trigger-per-button setting.
B. The fourth call will be delivered to ephone 1 because the huntstop channel setting is not yet saturated.
C. The fourth call will be delivered to ephone 1 because it can handle up to five calls on each button.
D. The fourth call will be held temporarily by the IOS Software until ephone 1 disconnects one of the active calls.
E. The fourth call will not be delivered and the caller will hear a user busy tone.

Correct Answer: E
Section: (none)

Explanation

Explanation/Reference:
Explanation:
Because on line maximum 4 calls can be placed when user put the call on hold is consume a channel and reach the maximum number of calls on line.

QUESTION 77
137
Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 shared-line
 huntstop channel 6
!
voice register pool 1
 busy-trigger-per-button 4
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
voice register pool 2
 busy trigger per button 5
 id mac 2222.2222.2222
 type 7965
 number 1 dn 1
!
```

How many simultaneous inbound calls can be handled by these two IP phones?

A. 2
B. 4
C. 6
D. 9
E. 10

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The line is configured as shared line so it will support maximum two calls at a time.

QUESTION 78
138
Refer to the exhibit.

```
!
voice register dn 1
 number 7001
 call-forward b2bua busy 2100
 call forward b2bua noan 2200 timeout 20
 shared-line max-calls 5
 huntstop channel 4
!
voice register pool 1
 busy-trigger-per-button 3
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
voice register pool 2
 busy-trigger-per-button 3
 id mac 2222.2222.2222
 type 7965
 number 1 dn 1
!
```

IP phone 1 has the MAC address 11111.1111.1111, while IP phone 2 has the MAC address 2222.2222.2222. The first two incoming calls were answered by IP phone 1, while the third incoming call was answered by IP phone 2. What will happen to the fourth incoming call?

A. Both phones will ring, but only IP phone 2 can answer the call.
B. Both phones will ring and either phone can answer the call.
C. Only IP phone 2 will ring and can answer the call.
D. Neither phone will ring and the call will be forwarded to 2100.
E. Neither phone will ring and the call will be forwarded to 2200.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
In shared line configuration phone share the same line so it is possible for any phone to answer the call.

QUESTION 79
139
Which statement describes the question mark wildcard character in a SIP trigger that is configured on Cisco Unity Express?

A. It matches any single digit in the range 0 through 9.
B. It matches one or more digits in the range 0 through 9.
C. It matches zero or more occurrences of the preceding digit or wildcard value.
D. It matches one or more occurrences of the preceding digit or wildcard value.

E. It matches any single digit in the range 0 through 9, when used within square brackets.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Table 5-2 Trigger Pattern Wildcards and Special Characters
Character
Description
Examples
X
The X wildcard matches any single digit in the range 0 through 9.
The trigger pattern 9XXX matches all numbers in the range 9000 through 9999.
!
The exclamation point (!) wildcard matches one or more digits in the range 0 through 9.
The trigger pattern 91! matches all numbers in the range 910 through
91999999999999999999999999999999.
?
The question mark (?) wildcard matches zero or more occurrences of the preceding digit or wildcard value.
The trigger pattern 91X? matches all numbers in the range 91 through
91999999999999999999999999999999.
+
The plus sign (+) wildcard matches one or more occurrences of the preceding digit or wildcard value.
The trigger pattern 91X+ matches all numbers in the range 910 through
91999999999999999999999999999999.
[ ]
The square bracket ([ ]) characters enclose a range of values.
The trigger pattern 813510[012345] matches all numbers in the range 8135100 through 8135105.
-
The hyphen (-) character, used with the square brackets, denotes a range ofvalues.
The trigger pattern 813510[0-5] matches all numbers in the range 8135100 through 8135105.
^
The circumflex (^) character, used with the square brackets, negates a range of values. Ensure that it is the first character following the opening bracket ([).
Each trigger pattern can have only one ^ character.
The trigger pattern 813510[^0-5] matches all numbers in the range 8135106

QUESTION 80
140

Assume the IP address of Cisco Unity Express is 10.1.1.1. Which URL provides Cisco Unity Express end users with a GUI interface to access and manage their messages and mailbox settings?

A. http://10.1.1.1/Web/Common/Login.do
B. http://10.1.1.1/ciscopca
C. http://10.1.1.1/user
D. http://10.1.1.1/inbox
E. http://10.1.1.1/

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
For user access cisco unity has predefined url and it is http://10.1.1.1/user

Exam C

QUESTION 1
141
Refer to the exhibit.

```
%VOICE_IEC-3-GW: Application Framework Core: Internal Error (Toll fraud call rejected):
IEC=1.1.228.3.31.0 on callID 3 GUID=F146D6B0539C11DF800CA596C4C2D7EF
000183: *Jan 3 11:21:31.251: //3/F146D6B0800C/CCAPI/ccCallSetContext:
   Context=0x49EC9978
000184: *Jan 3 11:21:31.251: //3/F146D6B0800C/CCAPI/cc_process_call_setup_ind:
   >>>>CCAPI handed cid 3 with tag 1000 to app "_ManagedAppProcess_TOLLFRAUD_APP"
000185: **Jan 3 11:21:31.251: //3/F146D6B0800C/CCAPI/ccCallDisconnect:
   Cause Value=21, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disconnect Cause=0)
```

Your customer sent you this debug output, captured on a Cisco IOS router (router A), to troubleshoot a problem where all H.323 calls that originate from another Cisco IOS router (router B) are being dropped almost immediately after arriving at router A. What is the reason for these disconnected calls?

A. Calls were unsuccessful because of internal, memory-related problems on router A.
B. Calls were rejected because the called number was denied on a configured class of restriction list on router A.
C. Calls were rejected because the VoIP dial peer 1002 was not operational.
D. Calls were unsuccessful because the router B IP address was not found in the trusted source IP address list on router A.
E. Calls were rejected by router A because it received an admission reject from its gatekeeper because of toll fraud suspicion.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Trusted source IP address list on router is a list which secure the connectivity of router if it is enabled then we need to give the trusted ebtry for any route to reach.

QUESTION 2
142
Which type of mailbox on Cisco Unity Express can play a user greeting and disconnect the call, but cannot take or send messages?

A. PIN-less mailbox
B. announcement-only mailbox
C. general delivery mailbox
D. call-handling mailbox
E. personal mailbox

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Announcement-only mailbox are set for those user who only want the caller to listen the announcement and leave his message according to the announcement.

QUESTION 3
143
When multiple greetings are enabled on Cisco Unity Express, which greeting will take the highest precedence?

A. standard
B. meeting
C. busy
D. closed
E. internal

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Meeting greeting has highest priority because it is set by user when he don't want to take the call and notice the caller he is online.

QUESTION 4
144
Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 2220
  paramspace english index 1
  param max time call retry 40
  param service-name app-b-acd
  param number-of-hunt-grps 1
  param drop through option 1
  paramspace english language en
  param handoff-string app-b-acd-aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa-pilot 52/2000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop-through-prompt _bacd_welcome.au
  param call-retry-timer 10
  !
  service app b acd
  param queue len 7
  param aa-hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps 1
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

Assume the B-ACD configuration on a Cisco IOS Cisco Unified Communications Manager Express router is operational. What will happen to a call in queue that was not answered by any member of the hunt group after the maximum amount of time allowed in the call queue expires?

A. The call will be forwarded to extension 2120.
B. The call will be forwarded to extension 2220.
C. The call will be forwarded to extension 2003.
D. The call will be disconnected with user busy.
E. The call will be forwarded to 2100.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
As we can see in the configuration 2220 is configured as voice mail forwarding extension so the call will forward to voice mail.

QUESTION 5
145
Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice mail 2220
  paramspace english index l
  param max-time-call-retry 40
  param service-name app-b-acd
  param number-of-hunt-grps 1
  param drop-through-option 1
  paramspace english language en
  param handoff-string app-b-acd-aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa-pilot 5272000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop-through-prompt _bacd_welcome.au
  param call retry timer 10
  !
  service app-b-acd
  param queue-len 2
  param aa-hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps l
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

Assume the B-ACD configuration on a Cisco IOS Cisco Unified Communications Manager Express router is operational. What will happen to a new call that enters the call queue when there are already two calls in queue?

A. The call will be forwarded to extension 2120.
B. The call will be forwarded to extension 2220.
C. The call will be forwarded to extension 2003.
D. The call will be disconnected with user busy.
E. The call will be forwarded to 2100.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
That is because queue over flow is forwarded to 2003 and maximum number of calls in queue is configured as two.

QUESTION 6
146
Which two statements are requirements regarding hunt group options for B-ACD implementation

on Cisco Unified Communications Manager Express routers? (Choose two.)

A. The ephone hunt group is mandatory.
B. Either the ephone hunt group or the voice hunt group is acceptable.
C. Hunt group members must be SCCP IP phones.
D. Hunt group members can include both SCCP or SIP IP phones.
E. Hunt group members must be SIP IP phones.
F. The member hunting mechanism must be set to sequential.

Correct Answer: AC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The ephone hunt group is mandatory, and while ephone hunt groups only support Cisco Unified SCCP IP phones, a voice hunt group supports either a Cisco Unified SCCP IP phone or a Cisco Unified SIP IP phone
Reference:.
http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/command/reference/cme_v1ht.html

QUESTION 7
147
Which call hunt mechanism is only supported by the voice hunt group in a Cisco Unified Communications Manager Express router?

A. sequential
B. peer
C. longest idle
D. parallel
E. overlay

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Parallel Hunt-Group, allows a user to dial a pilot number that rings 2-10 different extensions simultaneously. The first extension to answer gets connected to the caller while all other

extensions will stop ringing. A timeout value can be set whereas if none of the extensions answer before the timer expires, all the extensions will stop ringing and one final destination number will ring indefinitely instead. The final number could be another voice hunt-group pilot number or mailbox
The following features are supported for Voice Hunt-Group:

QUESTION 8
148
Which Cisco Unified Communications Manager Express ephone button configuration separator enables overflow lines when the primary line for an overlay button is occupied by an active call?

A. o
B. c
C. w
D. x
E. :

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
x expansion/overflow, define additional expansion lines that are used when the primary line for an overlay button is occupied by an active call.

QUESTION 9
149

Which two statements describe characteristics of Cisco Unified Border Element high availability, prior to Cisco IOS release 15.2.3T, using a box-to-box redundancy configuration? (Choose two.)

A. It leverages HSRP for router redundancy and GLBP for load sharing between a pair of routers.
B. Cisco Unified Border Element session information is check-pointed across the active and standby router pair.
C. It supports media and signal preservation when a switchover occurs.
D. Only media streams are preserved when a switchover occurs.
E. It can leverage either HSRP or VRRP for router redundancy.
F. The SIP media signal must be bound to the loopback interface.

Correct Answer: BD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Configure box-to-box redundancy when you:
•
Expect the behavior of the CSSs to be active/standby (only the master CSS processes flows)
•
Can configure a dedicated Fast Ethernet (FE) link between the CSSs for the VRRP heartbeat
Do not configure box-to-box redundancy when you:
•
Expect the behavior of the CSSs to be active-active (both CSSs processing flows). Use VIP redundancy instead.
•
Cannot configure a dedicated FE link between the CSSs.
•
Require the connection of an Layer 2 device between the redundant CSS peers.

QUESTION 10
150
Refer to the exhibit.

```
isdn switch-type primary-dms100

controller T1 1/0
framing esf
linecode b8zs
pri group timeslots 1 24 nfas_d primary nfas_int 0 nfas_group 1

controller T1 1/1
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d backup nfas_int 1 nfas_group 1

controller T1 2/0
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d none nfas_int 2 nfas_group 1

controller T1 2/1
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d none nfas_int 3 nfas_group 1
```


In an effort to troubleshoot a caller ID delivery problem, a customer emailed you the voice port configuration on a Cisco IOS router. Which type of voice port is it?

A. FXS
B. E&M

C. BRI
D. FXO
E. DID

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
To configure caller-ID on FXS and FXO voice ports, use the following commands beginning in global configuration mode:
Command
Purpose
Step 1
Router(config)# caller-id enable
Enables caller ID. This command applies to FXS voice ports that send caller-ID information and to FXO ports that receive it. By default caller ID is disabled.
Note
If the station-id or a caller-id alertingcommand is configured on the voice port, these automatically enable caller ID, and thecaller-id enable command is not necessary.
Step 2
Router(config-voiceport)#station-id namename
Configures the station name on FXS voice ports connected to user telephone sets. This sets the caller-ID information for on-net calls originated by the FXS port. You can also configure the station name on an FXO port of a router for which incoming Caller ID from the PSTN subscriber line is expected. In this case, if no caller-ID information is included on the incoming PSTN call, the call recipient receives the information configured on the FXO port instead. If the PSTN subscriber line does provide caller-ID information, this information is used and the configured station name is ignored.
The name argument is a character string of 1 to 15 characters identifying thestation.
Note
This command applies only to caller-ID calls, not Automatic Number Identification (ANI) calls. ANI supplies calling number identification only.
Step 3
Router(config-voiceport)#station-id numbernumber
Configure the station number on FXS voice ports connected to user telephone sets. This sets the caller-ID information for on-net calls originated by the FXS port.

You can also configure the station number on an FXO port of a router for which incoming caller ID from the PSTN subscriber line is expected. In this case, if no caller-ID information is included on the incoming PSTN call, the call recipient receives the information configured on the FXO port instead. If the PSTN subscriber line does provide caller-ID information, this information is used

and the configured station name is ignored.
If the caller-ID station number is not provided by either the incoming PSTN caller ID or by the station number configuration, the calling number included with the on-net routed call is determined by Cisco IOS software by using a reverse dial-peer search. In this case, the number is obtained by searching for a POTS dial-peer that refers to the voice-port and the destination-pattern number from that dial-peer is used.
number is a string of 1 to 15 characters identifying the station telephone or extension number.
Reference:
http://www.cisco.com/c/en/us/td/docs/ios/12_2/voice/configuration/guide/fvvfax_c/vvfclid.html

QUESTION 11
151
Refer to the exhibit.

```
isdn switch-type primary-dms100

controller T1 1/0
framing esf
linecode b8zs
pri group timeslots 1 24 nfas_d primary nfas_int 0 nfas_group 1

controller T1 1/1
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d backup nfas_int 1 nfas_group 1

controller T1 2/0
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d none nfas_int 2 nfas_group 1

controller T1 2/1
framing esf
linecode b8zs
pri group timeslots 1 24 nfas_d none nfas_int 3 nfas_group 1
```


From this NFAS-enabled T1 PRI configuration on a Cisco IOS router, how many bearer channels are available to carry voice traffic?

A. 91
B. 92
C. 93
D. 94
E. 95

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:

Explanation: A T1 circuit typically carries 24 individual timeslots. Each timeslot in turn carries a single telephone call. When a T1 circuit is used to carry Primary Rate ISDN one of the timeslots is used to carry the D channel. A single Primary Rate ISDN circuit is thus sometimes described as 23B + D. There are 23 bearer channels carrying voice or data, and one D channel carrying the Common Channel Signaling. In this case, there are 96 total channels in the group, but only 1 will be needed for use as the D channel, leaving 95 available for bearer channels.

QUESTION 12
152
Refer to the exhibit.

```
isdn switch-type primary-dms100

controller T1 1/0
framing esf
linecode b8zs
pri group timeslots 1 24 nfas_d primary nfas_int 0 nfas_group 1

controller T1 1/1
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d backup nfas_int 1 nfas_group 1

controller T1 2/0
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d none nfas_int 2 nfas_group 1

controller T1 2/1
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d none nfas_int 3 nfas_group 1
```

Assuming this NFAS-enabled T1 PRI configuration on a Cisco IOS router is fully functional, what will the controller T1 1/1 D-channel status be in the output of the show isdn status command?

A. MULTIPLE_FRAME_ESTABLISHED
B. TEI_ASSIGNED
C. AWAITING_ESTABLISHMENT
D. STANDBY
E. INITIALIZED

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
TEI_ASSIGNED, which indicates that the PRI does not exchange Layer 2 frames with the switch. Use theshow controller t1xcommand to first check the controller t1 circuit, and verify whether it is clean (that is, it has no errors) before you troubleshoot ISDN Layer 2 problem with thedebug isdn

q921.

QUESTION 13
153
Which message is used by a Cisco IOS MGCP gateway to send periodic keepalives to its call agent?

A. CRCX
B. AUCX
C. NTFY
D. RQNT
E. 200 OK

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The gateway maintains this connection by sending empty MGCP Notify (NTFY) keepalive messages to Cisco CallManager at 15-second intervals. If the active Cisco CallManager fails to acknowledge receipt of the keepalive message within 30 seconds, the gateway attempts to switch over to the next highest order Cisco CallManager server that is available.
If none of the Cisco CallManager servers respond, the gateway switches into fallback mode and reverts to its default H.323 session application for basic call control support of IP telephony activity in the network.

QUESTION 14
154
Which SIP message element is mapped to QSIG FACILITY messages being tunneled across a SIP trunk between two Cisco IOS gateways?

A. SIP UPDATE
B. SIP OPTIONS
C. SIP SUBSCRIBE
D. SIP INFO
E. SIP NOTIFY

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
This section lists QSIG message elements and their associated SIP message elements when QSIG messages are tunneled over a SIP trunk.

- QSIG FACILITY/NOTIFY/INFO
<—>
SIP INFO
- QSIG SETUP
<—>
SIP INVITE
- QSIG ALERTING
<—>
SIP 180 RINGING
- QSIG PROGRESS
<—>
SIP 183 PROGRESS
- QSIG CONNECT
<—>
SIP 200 OK
- QSIG DISCONNECT
<—>
SIP BYE/CANCEL/4xx—6xxResponse

Reference:
http://www.cisco.com/c/en/us/td/docs/ios/voice/sip/configuration/guide/15_0/sip_15_0_book/tunneling_qsig.html

QUESTION 15
155
In Cisco IOS routers, which chipset is the PVDM-12 DSP hardware based on?

D. C5421
E. C5409

A.
B.
C.

D.

Correct Answer:
Section: (none)
Explanation

Explanation/Reference:
Explanation:
NM-HDV has five SIMM sockets (called Banks) that hold the PVDM-12 cards. Each PVDM-12 card contains three TI 549 DSPs
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/7x/uc7_0/media.html

QUESTION 16
156
Which codec is supported on the Cisco PVDM2 DSP modules but not on the PVDM3 DSP modules?

A. G.728
B. G.729B
C. G.729AB
D. G.723
E. G.726

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
All codecs that are supported on the PVDM2 are supported on the PVDM3, except that the PVDM3 does not support the G.723 (G.723.1 and G.723.1A) codecs. The PVDM2 can be used to provide G.723 codec support or the G.729 codec can be as an alternative on the PVDM3
Reference:http://www.cisco.com/c/en/us/td/docs/routers/access/1900/software/configuration/guide/Software_Configuration/pvdm3_config.html

QUESTION 17
157
Refer to the exhibit.

```
SIP-Gateway#show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED

SIP User Agent for TLS over TCP : ENABLED
SIP User Agent bind status(signaling): ENABLED 10.1.1.1
SIP User Agent bind status(media): ENABLED 10.1.1.1
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 70
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
Reason Header will override Response/Request Codes: DISABLED
Out-of-dialog Refer: DISABLED
Presence support is DISABLED
protocol mode is ipv4

SDP application configuration:
 Version line (v=) required
 Owner line (o=) required
 Timespec line (t=) required
 Media supported: audio video image
 Network types supported: IN
 Address types supported: IP4 IP6
 Transport types supported: RTP/AVP udptl
```

Which option describes how this Cisco IOS SIP gateway, with an analog phone attached to its FXS port, handles an incoming informational SIP 180 response message without SDP?

A. It will enable early media cut-through.
B. It will generate local ring back.
C. It will do nothing because the message is informational.
D. It will terminate the call because this is an unsupported message format.
E. It will take the FXS port offhook.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The Session Initiation Protocol (SIP) feature allows you to specify whether 180 messages with Session Description Protocol (SDP) are handled in the same way as 183 responses with SDP. The 180 Ringing message is a provisional or informational response used to indicate that the INVITE message has been received by the user agent and that alerting is taking place. The 183 Session Progress response indicates that information about the call state is present in the message body media information. Both 180 and 183 messages may contain SDP, which allows an early media session to be established prior to the call being answered.
Prior to this feature, Cisco gateways handled a 180 Ringing response with SDP in the same manner as a 183 Session Progress response; that is, the SDP was assumed to be an indication that the far end would send early media. Cisco gateways handled a 180 response without SDP by

providing local ringback, rather than early media cut-through. This feature provides the capability

to ignore the presence or absence of SDP in 180 messages, and as a result, treat all 180 messages in a uniform manner. The SIP—Enhanced 180 Provisional Response Handling feature allows you to specify which call treatment, early media or local ringback, is provided for 180 responses with SDP.
Reference:
http://www.cisco.com/c/en/us/td/docs/ios/voice/cube/configuration/guide/vb_book/vb_book/vb_1506.html

QUESTION 18
158
In which call state does the Mobility soft key act as a toggle key to enable or disable Single Number Reach for Cisco Unified Communications Manager Express SCCP IP phones?

A. idle
B. seized
C. alerting
D. ringing
E. connected

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Pressing the Mobility soft key during the idle call state enables the SNR feature. This key is a toggle; pressing it a second time disables SNR.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html

QUESTION 19
159
Refer to the exhibit.

```
!
ephone-dn 1 octo-line
 number 2001
 huntstop channel 6
!
ephone 1
 mac-address 1111.1111.1111
 max-calls-per-button 5
 busy-trigger-per-button 3
 type 7965
 button 1:1
!
ephone 2
 mac-address 2222.2222.2222
 max-calls-per-button 6
 busy-trigger-per-button 4
 type 7965
 button 1:1
!
```

How many inbound calls can be handled simultaneously between ephone 1 and ephone 2 before a user busy tone is returned?

A. 6
B. 7
C. 8
D. 9
E. 11

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Because hunt stop channel is set to 6 as it enables call hunting to up to six channels of this ephone-dn and remaining 2 channels are available for outgoing call features.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_e1ht.html

QUESTION 20
160
Refer to the exhibit.

```
!
ephone-dn 1 octo-line
 number 2001
 huntstop channel 6
!
ephone 1
 mac-address 1111.1111.1111
 max-calls-per-button 5
 busy-trigger-per-button 3
 type 7965
 button 1:1
!
ephone 2
 mac-address 2222.2222.2222
 max-calls-per-button 6
 busy-trigger-per-button 4
 type 7965
 button 1:1
!
```

Three calls are active on ephone 1. Assume ephone 2 will remain idle.
How many additional calls can be placed from ephone 1?

A. 0
B. 1
C. 2
D. 3
E. 5

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
As we can see max-calls-per-button set to 5 and 3 calls are active. So, 2 calls remains.

QUESTION 21
161
Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call forward b2bua busy 2100
 call-forward b2bua noan 2200 timeout 20
 shared-line max-calls 4
 huntstop channel 3
!
voice register pool 1
 busy-trigger-per-button 2
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
voice register pool 2
 busy-trigger-per-button 2
 id mac 2222.2222.2222
 type 7965
 number 1 dn 1
!
```

IP phone 1 has MAC address of 1111.1111.1111, and IP phone 2 has MAC address of 2222.2222.2222. The first two incoming calls rang both phones and were answered by IP phone 2. Which option describes what will happen to the third incoming call?

A. Both phones ring, but only IP phone 1 can answer the call.
B. Both phones ring and either phone can answer the call.
C. Only IP phone 1 rings and can answer the call.
D. Neither phone rings and the call is forwarded to 2100.
E. Neither phone rings and the call is forwarded to 2200.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
As we can see busy-trigger-per-button set to 2 in voice register pool 1(IP Phone 1).So ipphone 1's channel is free for receiving incoming calls and right now IPPhone 2 is busy answering call.

QUESTION 22
162
Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 call-forward b2bua noan 2200 timeout 20
 shared-line max-calls 4
 huntstop channel 3
!
voice register pool 1
 busy-trigger-per-button 3
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
voice register pool 2
 busy-trigger-per-button 2
 id mac 2222.2222.2222
 type 7965
 number 1 dn 1
!
```

IP phone 1 has MAC address of 1111.1111.1111, and IP phone 2 has MAC address of 2222.2222.2222. The first two incoming calls were answered by IP phone 1, and the third incoming call was answered by IP phone 2.
Which option describes what will happen to the fourth incoming call?

A. Both phones ring, but only IP phone 2 can answer the call.
B. Both phones ring and either phone can answer the call.
C. Both phones ring, but only IP phone 1 can answer the call.
D. Neither phone rings and the call is forwarded to 2100.
E. Neither phone rings and the call is forwarded to 2200.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
IP Phone 1 & 2 both has busy-trigger-per-button configured to 3 & 2 resectively.So,the 4thincoming call will get forwarded to 2100 as busy-triggers are exceed in IP Phones.
Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_c1ht.html#wp1570384096

QUESTION 23
163
Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 2220
  paramspace english index 1
  param max-time-call-retry 40
  param service-name app-b-acd
  param number of hunt grps 1
  param drop-through-option 1
  paramspace english language en
  param handoff string app b acd aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa pilot 5272000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop-through-prompt _bacd_welcome.au
  param call retry timer 10
  !
  service app-b-acd
  param queue len 2
  param aa-hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps 1
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

Assume the B-ACD configuration on a Cisco Unified Communications Manager Express router is operational.
Which option describes what will happen to an incoming call that entered the call queue but all

members of the hunt group are in Do Not Disturb status?

A. The call is forwarded to extension 2120.
B. The call is forwarded to extension 2220.
C. The call is forwarded to extension 2003.
D. The call is disconnected with user busy.
E. The call is forwarded to extension 2100.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Because all members of hunt group is unavailable or activates DnD and incoming queued call will forward to voicemail using the param voice-mail 2220 command.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/bacd/configuration/guide/cme40tcl/40bacd.html#wpmkr1105714

QUESTION 24
164
Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 2220
  paramspace english index 1
  param max time call retry 40
  param service name app b acd
  param number-of-hunt-grps 1
  param drop-through-option 1
  paramspace english language en
  param handoff string app b acd aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa-pilot 5272000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop through prompt  bacd welcome.au
  param call-retry-timer 20
  !
  service app-b-acd
  param queue len 2
  param aa hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps 1
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

Assume the B-ACD configuration on a Cisco Unified Communications Manager Express router is operational.
How much time does a member of the hunt group have to answer a queue call that is ringing on their extension?

A. 5 seconds
B. 10 seconds
C. 20 seconds
D. 30 seconds
E. 40 seconds

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:

As you can see the timeout 10 sec in ephone-hunt 1 means hunt group membes have to answer the queued call within 10 sec.

QUESTION 25
165
Which two Cisco Unified Communications Manager Express hunt group mechanisms keep track of the number of hops in call delivery decisions? (Choose two.)

A. sequential
B. peer
C. longest idle
D. parallel
E. overlay
F. linear

Correct Answer: BC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Peer configures hunting in a circular manner among the hunt group member DNs and starts with the DN to the right of the last DN to ring.
Longest-idle specify hunting on the DN which is idle for a longest period of time and the call will go to that DN of the hunt Group.
Reference:http://ccievoice.ksiazek.be/?p=690

QUESTION 26
166
Refer to the exhibit.

```
CUBE-2#show voice high-availability summary
======== Voice HA DB INFO ========
Number of calls in HA DB: 28  (MAX:2048)
Number of calls in HA sync pending DB: 12
Number of calls in HA preserved session DB: 9
```

This output was captured on a Cisco IOS gateway shortly after it became the active Cisco Unified Border Element in a box-to-box redundancy failover.

How many calls are native to this Cisco Unified Border Element?

A. 9
B. 12
C. 19
D. 31
E. 40

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Total no of calls =28+12=40.
So, native calls are =40-9=31.
Reference:http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unifiedborder-element/112095-cube-hsrp-config-00.html.

QUESTION 27
167
Which method allows administrators to determine the best match impedance on analog voice ports in Cisco IOS router without having to shut and no shut the ports?

A. THL tone sweep
B. original tone sweep
C. ECAN test
D. inject-tone local sweep
E. remote loop

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
THL tone sweepallows all available impedances for a single test call to a quiet termination point out to the PSTN. You do not need to manually disable ECAN on the voice port under test. The test feature switches impedances automatically for the tester. The test feature calculates the arithmetic mean ERL and reports the mean for each channel profile at each impedance setting. Then, at the end of the test, the feature specifies thebest matchimpedance setting. This test requires minimal

supervision.
Reference:http://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ipvoip/64282-impedance-choice.html

QUESTION 28
168
Which two types of line codes are configurable for an E1 PRI controller on a Cisco IOS router? (Choose two.)

A. CRC4
B. AMI
C. B8ZS
D. HDB3
E. ESF
F. SF

Correct Answer: BD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Perform this task to select and configure an NM-xCE1T1-PRI network module card as E1.
SUMMARY STEPS
1. enable
2. configure terminal
3. card type e1 slot
4. controller e1 slot / port
5. linecode {ami | hdb3}
6. framing {crc4 | no-crc4}
Reference:
http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/interface/configuration/12-4/ir-12-4-book/ir-12-port-chann-nm.html

QUESTION 29
169
In Channel Associated Signaling on a T1 circuit using Extended Super Frame, how many signaling bits does each T1 timeslot have?

A. 1
B. 2

C. 4

D. 12

E. 24

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:

Each T1 channel carries a sequence of frames. These frames consist of 192 bits and an additional bit designated as the framing bit, for a total of 193 bits per frame. Super Frame (SF) groups twelve of these 193 bit frames together and designates the framing bits of the even numbered frames as signaling bits. CAS looks specifically at every sixth frame for the timeslot's or channel's associated signaling information. These bits are commonly referred to as A- and B-bits. Extended super frame (ESF), due to grouping the frames in sets of twenty-four,has four signaling bits per channel or timeslot. These occur in frames 6, 12, 18, and 24 and are called the A-, B-, C-, and D-bits respectively.
Reference: http://www.cisco.com/c/en/us/support/docs/voice/digital-cas/22444-t1-cas-ios.html

QUESTION 30
170
Refer to the exhibit.

```
T1-CAS-Gateway(config-controller)#ds0-group 1 time 1-10 type ?
  e&m-delay-dial        E & M Delay Dial
  e&m-fgd               E & M Type II FGD
  e&m-immediate-start   E & M Immediate Start
  e&m-lmr               E & M land mobil radio
  e&m-wink-start        E & M Wink Start
  ext-sig               External Signaling
  fgd-eana              FGD-EANA BOC side
  fgd-os                FGD-OS BOC side
  fxo-ground-start      FXO Ground Start
  fxo-loop-start        FXO Loop Start
  fxs-ground-start      FXS Ground Start
  fxs-loop-start        FXS Loop Start
  none                  Null Signalling for External Call Control
  <cr>
```

Which ds0-group option should you select to support automated number identification information collection on inbound calls for this digital T1 voice circuit?

A. e&m-wink-start

B. e&m-delay-dial

C. e&m-delay-dial
D. e&m-lmr
E. e&m-fgd

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Because it can receive ANI information and sends DNIS info. But can't send ANI.
Reference:
http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/isdn/configuration/15-mt/vi-15-mt-book/viimp-t1cas-voip.html

QUESTION 31
171
In Cisco IOS routers, which chipset is the PVDM2-32 DSP hardware based on?

A. C5441
B. C549
C. C5510
D. C5421
E. Broadcom 1500

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Table 6-2 DSP Resources on Cisco IOS Hardware Platforms with C5510 Chipset
Hardware Module or Chassis
DSP Configuration
Maximum Number of Voice Terminations (Calls) per DSP and per Module
Medium Complexity
(8 calls per DSP)
High Complexity
(6 calls per DSP)
Flex Mode1
(240 MIPS per DSP)

VG-224
Fixed at 4 DSPs
N/A
24 calls per platform
Supported codecs:
•
G.711 (a-law, mu-law)
•
G.729a
N/A
NM-HD-1V2
Fixed at 1 DSP
4 calls per NM
4 calls per NM
240 MIPS per NM
NM-HD-2V
Fixed at 1 DSP

8 calls per NM
6 calls per NM
240 MIPS per NM
NM-HD-2VE
Fixed at 3 DSPs
24 calls per NM
18 calls per NM
720 MIPS per NM
NM-HDV2
NM-HDV2-2T1/E1
NM-HDV2-1T1/E1
1 to 4 of:
PVDM2-83 (½ DSP)PVDM2-16 (1 DSP)PVDM2-32 (2 DSPs)PVDM2-48 (3 DSPs)PVDM2-64 (4 DSPs)
Calls per PVDM:
48162432
Calls per PVDM:
36121824
MIPS per PVDM:
120240480720960
Reference: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42media.html

QUESTION 32
172
Which digital modulation method is used to transmit caller ID information on analog FXS ports on

Cisco IOS routers?

A. DTMF
B. PSK
C. FSK
D. MF
E. pulse dialing

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation: Link:-http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/vcr1/vcr1-cr-book/vcrc4.html

QUESTION 33
173
How many signaling bits are there in each T1 time slot using channel associated signaling with Super Frame?

A. 1
B. 2
C. 4
D. 8
E. 12

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Each T1 CAS has 24 channels that can transmit 8 bits per channel each. This gives us a total of 192 bits. The T1 has one additional bit for framing, bringing the total to 193 bits. Two types of line coding can be used on a T1 CAS. The fi rst type of line coding is called Super Frame (SF). This is an older and less - effi cient type of framing. Super Frame bundles 12 of these 193 - bit frames together for transport. It then uses the even – numbered frames as signaling bits. The T1 CAS signaling then looks at every sixth frame for signaling information. This comes out to be 2 bits that are referred to as the A and B bits, which reside in frames 6 and 12.

QUESTION 34
174
Which two statements about the restrictions for support of H.239 are true? (Choose two.)

A. SIP to H323 video calls using H.239 are not supported.
B. Redundancy for H.323 calls is not supported.
C. H.239 calls are not supported over intercluster trunks with Cisco Unified Communications Manager.
D. H.239 is not supported with third-party endpoints.
E. Cisco Unified Communications Manager supports a maximum of three video channels when using H.239.

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Restriction for Support for H.239
The Support for H.239 feature has the following restrictions:
Reference:http://www.cisco.com/c/en/us/td/docs/routers/asr1000/configuration/guide/sbcu/2_xe/sb

cu_2_xe_book/sbc_h239.html

QUESTION 35
175
Refer to the exhibit.

```
Outgoing SIP UDP message to 10.1.1.1:[5060]:

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.1.1.1:5060;branch=z9hG4bK078cE1C7A
From:"Unknown" ;tag-2319872817981
To: "SBC" ;tag=23498729384/9
Date: Tue, 11 Dec 2012 15:08:29 GMT
Call-ID: 234098d123147652@20.1.1.1
CSeq: 104 OPTIONS
WWW-Authenticate: Digest realm-"StandAloneCluster", nonce-"sdflakjdfjklahsfhkhq", algorithm-MD5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Content-Length: 0
```

The exhibit shows an outgoing SIP 401 response message from Cisco Unified Communications Manager to a SIP VoIP service provider gateway. Which action can the Cisco Unified Communications Manager systems administrator use to change the response to "200 OK"?

A. Make sure the gateway IP address of the SIP VoIP service provider is defined correctly in Cisco

Unified Communications Manager SIP trunk.

B. Enable OPTIONS ping on Cisco Unified Communications Manager SIP trunk.
C. Disable OPTIONS ping on Cisco Unified Communications Manager SIP trunk.
D. Create an SIP response alias to force outgoing 401 messages to "200 OK".
E. Disable digest authentication on Cisco Unified Communications Manager SIP trunk.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Because Right now CUCM challengesthe identity of a SIP user agent and must configure digest credentials for the application user in CUCM or you have to disable it for stop challenging by CUCM.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/9_0_1/secugd/CUCM_BK_CCB00C40_00_cucm-security-guide-90/CUCM_BK_CCB00C40_00_cucm-securityguide_chapter_011010.html

QUESTION 36
176
Which two responses from a SIP device, which is the only remote destination on a Cisco Unified

Communications Manager SIP trunk with OPTIONS ping enabled, cause the trunk to be marked as "Out of Service"? (Choose two.)

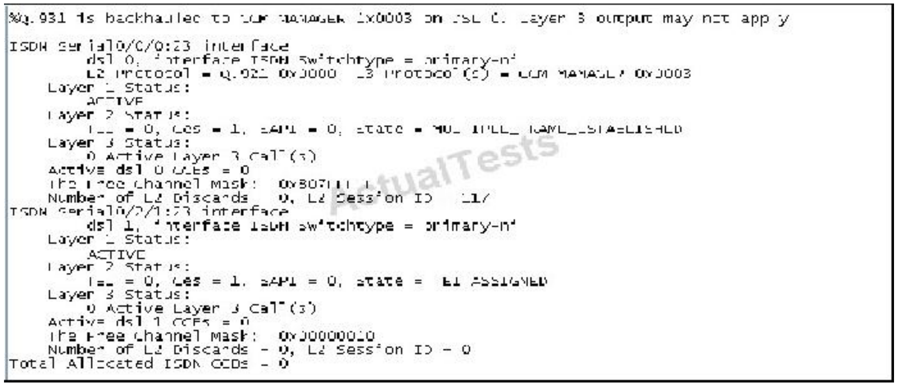
A. 503 Service Unavailable
B. 408 Request Timeout
C. 505 Version Not Supported
D. 504 Server Timeout
E. 484 Address Incomplete
F. 404 Not Found

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:

The remote peer may be marked as Out of Service if it fails to respond to OPTIONS, if it sends 503 or 408 responses, or if the Transport Control Protocol (TCP) connection cannot be established. If at least one IP address is available, the trunk is In Service; if all IP addresses are unavailable, the trunk is Out of Service.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_5_1/ccmcfg/bccm-851-cm/b06siprf.html

QUESTION 37
177
Refer to the exhibit.



```
%Q.931 is backhauled to CCM MANAGER 0x0003 on DSL 0. Layer 3 output may not apply

ISDN Serial0/0/0:23 interface
        dsl 0, interface ISDN Switchtype = primary-ni
        L2 Protocol = Q.921 0x0000  L3 Protocol(s) = CCM MANAGER 0x0003
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Active dsl 0 CCBs = 0
    The Free Channel Mask:  0x807FFFFF
    Number of L2 Discards = 0, L2 Session ID = 117
ISDN Serial0/2/1:23 interface
        dsl 1, interface ISDN Switchtype = primary-ni
    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)
    Active dsl 1 CCBs = 0
    The Free Channel Mask:  0x00000010
    Number of L2 Discards = 0, L2 Session ID = 0
Total Allocated ISDN CCBs = 0
```

Which two statements about the show command output are true? (Choose two.)

A. T1 0/2/1 terminates Q.921 signaling to a Cisco Unified Communications Manager server.
B. T1 0/0/0 terminates Q.921 signaling on the gateway.
C. T1 0/0/0 terminates SIP Signaling to a Cisco Unified Communications Manager server.
D. T1 0/0/0 terminates Q.931 signaling to a Cisco Unified Communications Manager server.
E. T1 0/2/1 terminates Q.931 signaling on the gateway.

Correct Answer: BD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
As you can see the T1 0/0/0:23 interface is active in layer 1,2(multi frame established) & 3,it means Q.931 signaling terminates at gateway and using backhauled technique q931 messages are going to CUCM server.

But in case of T1 0/2/1 port multi frames are not established in layer 2.So, its not configured properly & doesn’t backhauling q931 messages to CUCM

QUESTION 38
178
Refer to the exhibit.

```
voice translation rule 2
   rule 1 /^500..$/ /40877\0/
   rule 2 /^5..[0-9]$/ /+1408777\0/
   rule 3 /^500....$/ /1408777..../
voice translation-profile PSTN
   translate calling 2
```

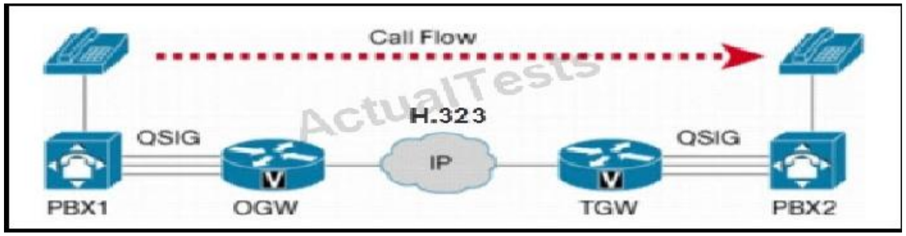
Which number is sent as the caller ID when a user at extension 5001 places a call that matches this translation profile?

A. 14087775001
B. +4087775001
C. 4087750001
D. +14087775001

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When someone dials 5001, it will match rule 2 because it exactly starts with 5(five) using the ^ sign and ends with [0-9] followed by $. In replace pattern you can see +1408777 & \0 means all set of match pattern. Thus, +14087775001.

QUESTION 39
179
Refer to the exhibit.



Which option describes the method used by Cisco IOS gateways to tunnel QSIG signaling messages in H.323 protocol?

A. H.323 Annex M1
B. H.323 Annex M2
C. H.323 Annex A
D. ISDN Generic Transparency Descriptor
E. H.450.1

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation: H.323 is an umbrella recommendation that encompasses various ITU-T recommendations, primarily recommendations H.225.0 and H.245 (basic communication capabilities) and recommendation H.450.1 (generic functional protocol for the support of supplementary services). Tunneling QSIG over H.323 is specified in H.323 Annex-M1. However, Cisco IOS ® Software H.323 QSIG tunneling does not implement Annex-M1 (as the Cisco Unified Communications Manager H.323 implementation does).Instead it uses the ISDN Generic Transparency Descriptor (GTD) to transport QSIG messages in the corresponding H.225 message to another Cisco gateway device on the other side of the network.
Reference: http://www.cisco.com/c/en/us/solutions/collateral/enterprise-networks/empoweredbranch-solution/white_paper_c11_459092.html

QUESTION 40
180

Which two analog telephony signaling methods are most vulnerable to glare conditions? (Choose two.)

A. FXS Loop-start

B. FXO Ground-start
C. E&M Wink-start
D. E&M Delay-dial
E. E&M Immediate-start
F. E&M Feature Group D

Correct Answer: AE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The loop start signaling method is more common and is typically used by residential phone lines. When a voice port is configured with loop start signaling, the device (telephone) closes the circuit loop that signals the CO voice port to provide dial tone; an incoming call is signaled on the CO by supplying a predefined voltage on the line. The loop start signaling method has one main disadvantage in that it has no method of preventing both sides of the connection from attempting to seize the line at the same time; this condition is referred to as glare. Because of this, loop start signaling is typically not used on high demand circuits.
With immediate-start, the calling side of the connection seizes the line by going off hook on the Elead and address information is sent using dual-tone multifrequency (DTMF) digits. Immediate start signaling is vulnerable to glare just like loop-start signaling.

Exam D

QUESTION 1
181
Which two Cisco IOS multipoint video conferencing profiles are supported on the Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3? (Choose two.)

A. homogeneous
B. rendezvous
C. guaranteed-audio
D. scheduled
E. guaranteed-video
F. ad-hoc

Correct Answer: AC
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 2
182
Which Cisco IOS multipoint video conferencing profile is also known as best-effort video on the Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3?

A. homogeneous
B. guaranteed-audio
C. rendezvous
D. heterogeneous
E. flex mode video

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation: Three types of video profiles are supported: homogeneous conferences (video switching), heterogeneous conferences (video mixing), and guaranteed audio conferences (besteffort video).

As the name suggests, when Guaranteed Audio Conferences is configured, the system attempts to display video for all participants; however, it does not guarantee that the video of allparticipants is displayed. For those participants whose video is not displayed, participants are downgraded to audio-only and the profile guarantees preservation of the audio portion of the call. This option gives you added flexibility because the DSPs are not all reserved when the profile is created; the system attempts to reserve them when this profile is activated with an actual conference. For example:

dspfarm profile 1 conference video guaranteed-audio
codec h264 vga
codec h264 4cif

Reference:
http://www.cisco.com/c/en/us/products/collateral/unified-communications/voice-videoconferencing-isr-routers/qa_c67-649850.html

QUESTION 3

183

Which Cisco packet voice and video digital signal processor 3 can be used for video mixing on a

Cisco Integrated Router Generation 2?

A. PVDM3-16

B. PVDM3-32

C. PVDM3-64

D. PVDM3-128

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

Explanation:
All the PVDM3 types (that is, PVDM3-16, PVDM3-32, PVDM3-64, PVDM3-128, PVDM3-192, and PVDM3-256) support switched-only video conferences. Only PVDM3-128 and higher modules support video conferencing with video mixing, transcoding and transrating.
Reference:http://www.cisco.com/c/en/us/products/collateral/unified-communications/voice-videoconferencing-isr-routers/data_sheet_c78-649427.pdf

QUESTION 4

184

Which three options are valid per-session video conference participants supported on the Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3? (Choose three.)

A. 3
B. 4
C. 6
D. 8
E. 9
F. 12
G. 16

Correct Answer: BDG
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The integrated video conferencing services use the same DSP resources on PVDM3s that are used for widely deployed ISR G2 voice capabilities. These modules, in conjunction with Cisco IOS Software, perform audio and video mixing, video transcoding for certain resolutions, and other functions for video endpoints. PVDM3 modules support flexible media resources and conference profile management to maximize capacity with predictable end-user experiences. Both homogenous and heterogonous video conferences are supported. A homogenous conference refers to one in which participants connect to the ISR G2 with devices that support the same video format attributes (for example, the same codec, resolution, frame rate, and bit rate). A heterogeneous conference refers to one in which participants can connect to a conference bridge

with devices that support different video format attributes. Each conference allows4-, 8-, or 16-partyparticipants.
Reference:http://www.cisco.com/c/en/us/products/collateral/unified-communications/voice-videoconferencing-isr-routers/data_sheet_c78-649427.html

QUESTION 5
185
Refer to the exhibit.

```
Router#show dial-peer voice sum
dial-peer hunt 0
               AD                                  PRE PASS                  OUT
TAG     TYPE   MIN  OPER PREFIX   DEST-PATTERN     FER THRU SESS-TARGET      STAT PORT    KEEPALIVE

4300    voip   up   up            4...              0  syst ipv4:10.1.1.4                 active
2300    voip   up   up            [2-3]...          0  syst ipv4:10.1.1.3                 active
1111    voip   up   down          1111              0  syst ipv4:10.1.1.1                 busy-out
20001   pots   up   up            2001$             0                           50/0/1
20002   pots   up   up            2002$             0                           50/0/2
```

Which out-of-dialog SIP OPTIONS ping response put dial-peer tag 1111 into its current operational state?

A. 501 Not Implemented
B. 504 Server Time-out
C. 408 Request Timeout
D. 486 Busy Here
E. 503 Service Unavailable

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
SIP 503 Service Unavailable is commonly seen in a VoIP network when a SIP device (such as a SIP server) is knowingly unable to process a call. Typically when this happens the endpoint that originated the Invite will try the next available host it receives in the SIP Contact header.

QUESTION 6
186
Which statement about what happens to a Cisco IOS SIP VoIP dial-peer that never received any responses to its out-of-dialog OPTIONS ping is true?

A. Its admin state will be up but operational state will be down.
B. Its admin and operational state will be down.
C. Its admin and operational state will remain up.
D. Its admin state will be up but operational state will be "busy-out".
E. Its admin and operational state will be "busy-out".

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
You can check the validity of your dial peer configuration by performing the following tasks:
•
If you have relatively few dial peers configured, you can use the show dial-peer voice command to verify that the configuration is correct. To display a specific dial peer or to display all configured dial peers, use this command. The following is sample output from the show dial-peer voice

command for a specific VoIP dial peer:
router#show dial-peer voice 10
VoiceOverIpPeer10
tag = 10, dest-pat = \Q',
incall-number = \Q+14087',
group = 0, Admin state is up, Operation state is down
Permission is Answer,
type = voip, session-target = \Q',
sess-proto = cisco, req-qos = bestEffort,
acc-qos = bestEffort,
fax-rate = voice, codec = g729r8,
Expect factor = 10,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,
Connect Time = 0, Charged Units = 0
Successful Calls = 0, Failed Calls = 0
Accepted Calls = 0, Refused Calls = 0
Last Disconnect Cause is ""
Last Disconnect Text is ""
Last Setup Time = 0

- To show the dial peer that matches a particular number (destination pattern), use the show dialplan number command. The following example displays the VoIP dial peer associated with the destination pattern 51234:

router#show dialplan number 51234
Macro Exp.: 14085551234
VoiceOverIpPeer1004
tag = 1004, destination-pattern = \Q+1408555....',
answer-address = \Q',
group = 1004, Admin state is up, Operation state is up

type = voip, session-target = \Qipv4:1.13.24.0',
ip precedence: 0 UDP checksum = disabled
session-protocol = cisco, req-qos = best-effort,
acc-qos = best-effort,
fax-rate = voice, codec = g729r8,
Expect factor = 10, Icpif = 30,
VAD = enabled, Poor QOV Trap = disabled
Connect Time = 0, Charged Units = 0
Successful Calls = 0, Failed Calls = 0
Accepted Calls = 0, Refused Calls = 0
Last Disconnect Cause is ""
Last Disconnect Text is ""
Last Setup Time = 0
Matched: +14085551234 Digits: 7

Target: ipv4:172.13.24.0

QUESTION 7
187
Refer to the exhibit.

```
Branch-Router#debug ip dhcp packet

*Aug 15 22:13:23.921: DHCPD: Finding a relay for client 01ec.4476.1e3e.7d on interface Vlan101.
*Aug 15 22:13:23.924: DHCPD: setting giaddr to 10.101.15.1.
*Aug 15 22:13:23.924: DHCPD: BOOTREQUEST from 01ec.4476.1e3e.7d forwarded to 10.100.1.1.
*Aug 15 22:13:23.940: DHCPD: Forwarding BOOTREPLY to client ec44.761e.3e7d.
*Aug 15 22:13:23.940: DHCPD: broadcasting BOOTREPLY to client ec44.761e.3e7d.

Branch-Router#
```

The exhibit shows the Cisco IOS CLI output of debug ipdhcp packet, which was captured on a router that is located at a branch office where a single IP phone is located. There is a standalone Cisco Unified Communications Manager server at the central site, which also provides DHCP services to the IP phone at the branch office. You are troubleshooting a problem where the IP phone could not register to Cisco Unified Communications Manager. You have confirmed that the IP phone received an IP address in the correct subnet and with a correct subnet mask from the DHCP server. Assuming the IP phone is correctly defined on Unified CM, which two statements about the network components are true? (Choose two.)

A. The MAC address of the IP phone is 01ec44761e3e.
B. The IP address of the DHCP server is 10.101.15.1.
C. The MAC address of the VLAN 101 interface is ec44761e3e7d.
D. The IP address of the VLAN 101 interface is 10.101.15.1.
E. There is IP connectivity between the VLAN 101 interface of the branch router and the ip-helper

   address that is configured on this interface.
F. There is IP connectivity between the IP phone and the ip-helper address on the VLAN 101 interface.

Correct Answer: DE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
As we can see from the logs given first line relate that dhcp request is being relayed. So it clarifies there must be ip helper address commend given by the admin on interface vlan 101.Now we can see from the second line that giaddress is set as source address of vlan 101 by the dhcp as 10.101.15.1 to unicast the dhcp request

QUESTION 8
188
Which two are characteristics of jitter buffers? (Choose two.)

A. Jitter buffers are used to change asynchronous packet arrivals into a synchronous stream by turning variable network delays into constant delays at the destination end systems.
B. Jitter buffers are used to change asynchronous packet arrivals into a synchronous stream by turning variable network delays into constant delays at the sending systems.
C. The role of the jitter buffer is to balance the delay and the probability of interrupted playout due to late packets.
D. The role of the jitter buffer is to queue late packets and reorder out-of-order packets.
E. Jitter buffers are used to change asynchronous packet arrivals into a synchronous stream by queuing packets into constant delays at the sending systems.

Correct Answer: AC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Jitter buffers are used to remove the effects of jitter so that asynchronous packet arrivals are changed to a synchronous stream. The jitter buffer trades off between delay and the probability of interrupted playout because of late packets (discard).
Reference:http://www.appneta.com/blog/jitter-voip/

QUESTION 9
189
Which enrollment method does a Cisco IOS VPN router trustpoint use to install a Certificate Authority Proxy Function certificate for LSC validation of a Cisco IP phone client?

A. HTTP proxy server
B. certificate authority server URL
C. terminal
D. self signed
E. registration authority

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Things to Note:
Reference: http://www.cisco.com/c/en/us/support/docs/ios-nx-os-software/authenticationauthorization-accounting-aaa/116313-configure-anyconnect-00.html

QUESTION 10
190
Refer to the exhibit.

```
Branch-Router#debug ip dhcp packet

*Aug 15 22:13:23.924: DHCPD: Finding a relay for client 01ec.4476.1e3e.7d on interface Vlan101.
*Aug 15 22:13:23.924: DHCPD: setting giaddr to 10.101.15.1.
*Aug 15 22:13:23.924: DHCPD: BOOTREQUEST from 01ec.4476.1e3e.7d forwarded to 10.100.1.1.
*Aug 15 22:13:23.940: DHCPD: forwarding BOOTREPLY to client ec44.761e.3e7d.
*Aug 15 22:13:23.940: DHCPD: broadcasting BOOTREPLY to client ec44.761e.3e7d.

Branch-Router#
```

The exhibit shows the Cisco IOS CLI output of debug ipdhcp packet, which was captured on a router that is located at a branch office where a single IP phone is located. There is a standalone Cisco Unified Communications Manager server at the central site, which also provides DHCP services to the IP phone at the branch office. You are troubleshooting a problem where the IP phone received an IP address in the correct subnet and with a correct subnet mask from the DHCP server, but never completed registration with Cisco Unified CM. Assuming the IP phone is correctly defined on Unified CM, which two statements the network components are true? (Choose two.)

A. The MAC address of the IP phone is 01ec44761e3e7d.
B. The IP address of the DHCP server is 10.101.15.1.
C. The MAC address of the VLAN 101 interface is 01ec44761e3e7d.
D. IThe MAC address of the IP phone is ec44761e3e7d.
E. There is no IP connectivity between the VLAN 101 interface of the branch router and the iphelper address that is configured on this interface.
F. Based on the information provided, we cannot conclude if there is IP connectivity between the IP phone and Cisco Unified CM.

Correct Answer: DF
Section: (none)
Explanation

Explanation/Reference:
Explanation:
In the logs the only information that we get is about the mac address of the IP phone because the IP phone is raising the boot request.

QUESTION 11
191
Which statement about a virtual SNR DN-configured Cisco Unified Communications Manager Express-enabled Cisco IOS router is true?

A. Virtual SNR DN supports either SCCP or SIP IP phone DNs.
B. A virtual SNR DN is a DN that is associated with multiple registered IP phones.
C. Calls in progress can be pulled back from the phone that is associated with the virtual SNR DN.
D. The SNR feature can only be invoked if the virtual SNR DN is associated with at least one registered IP phone.
E. A call that arrived before a virtual SNR DN is associated with a registered phone, and still exists after association is made, but cannot be answered from the phone.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Mid-calls are either of the following:
–
Calls that arrive before the DN is associated with a registered phone and is still present after the DN is associated with the phone.
–
Calls that arrive for a registered DN that changes state from registered to virtual and back to registered.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html

QUESTION 12
192
Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 huntstop channel 10
!
voice register pool 1
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
```

How many calls, inbound and outbound combined, are supported on the IP phone?

A. 1
B. 2
C. 8
D. 12
E. 50

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation: Output incomplete to figure out the answer

QUESTION 13
193
Refer to the exhibit.

```
!
ephone-dn 1
 number 1001
!
ephone-dn 2
 number 1002
!
ephone-dn 3
 number 1003
 ephone-hunt login
!
ephone-dn 4
 number 1004
!
ephone-dn 5
 number 1005
 ephone-hunt login
!
ephone-dn 6
 number 1006
!
ephone-hunt 1 peer
 list 1001,1002,1004,*
 hop 6
 final 1100
!
```

Which ephone-dn can join the hunt group whenever a wild card slot becomes available?

A. ephone-dn 1
B. ephone-dn 2
C. ephone-dn 3
D. ephone-dn 4

E. ephone-dn 6

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_e1ht.html

QUESTION 14
194
Refer to the exhibit.

```
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw

!
voice class sip-profiles 102
 request INVITE sip-header Allow-Header modify ", UPDATE" ""
 request REINVITE sip header Allow Header modify ", UPDATE" ""
 response 180 sip-header Allow-Header modify ", UPDATE" ""
 response 200 sip-header Allow-Header modify ", UPDATE" ""
 request INVITE sip-header Diversion remove
 request ANY sip-header Diversion remove
!
dial-peer voice 7 voip
 translation-profile outgoing CALLED_DIGIT_STRIP
 destination pattern 6911
 modem passthrough nse codec g711ulaw
 session protocol sipv2
 session target ipv4:10.0.0.1
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 voice-class sip early-offer forced
 voice-class sip profiles 102
 dtmf-relay rtp-nte
 ip qos dscp af41 signaling
 no vad
```

Which two statements about calls that match dial-peer voice 7 voip are true? (Choose two.)

A. All calls that match dial-peer voice 7 use G.711.
B. All calls that match dial-peer voice 7 have the Diversion header removed from SIP Invites.
C. All calls that match dial-peer voice 7 use NOTIFY-based, out-of-band DTMF relay.
D. All calls that match dial-peer voice 7 are marked with DSCP 32.
E. All calls that match dial-peer voice 7 are marked with DSCP 34.

Correct Answer: BE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Dial peer 7 refers to SIP profile 102, which we can see is configured to have the Diversion header removed from SIP Invites.
Dial peer 7 marks traffic with AF41, which is equivalent to DSCP 34.
Topic 5, Quality of Service and Security in Cisco Collaboration Solutions

QUESTION 15
195
The iLBC codec operates at 38 bytes per sample per 20-millisecond interval. What is its codec bit rate in kilobits per second?

A. 6.3
B. 13.3
C. 15.2
D. 16
E. 24

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation: Explanation;
The internet Low Bit Rate Codec (iLBC) is designed for narrow band speech and results in a payload bit rate of 13.33 kbits per second for 30-millisecond (ms) frames and 15.20 kbits per second for 20 ms frames. When the codec operates at block lengths of 20 ms, it produces 304 bits per block, which is packetized as defined in RFC 3952. Similarly, for block lengths of 30 ms it produces 400 bits per block, which is packetized as defined in RFC 3952.The iLBC has built-in error correction functionality to provide better performance in networks with higher packet loss

QUESTION 16
196
Assume 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and a 40-millisecond voice payload, how much bandwidth should be allocated to the strict priority queue for five VoIP calls that use a G.729 codec over a multilink PPP link?

A. 87 kb/s
B. 134 kb/s
C. 102.6 kb/s
D. 77.6 kb/s

E. 71.3 kb/s

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Voice payloads are encapsulated by RTP, then by UDP, then by IP. A Layer 2 header of the correct format is applied; the type obviously depends on the link technology in use by each router interface: A single voice call generates two one-way RTP/UDP/IP packet streams. UDPprovides multiplexing and checksum capability; RTP provides payload identification, timestamps, and sequence numbering.

QUESTION 17
197
Assume 20 bytes of voice payload, 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and the IP, UDP, and RTP headers are compressed to 2 bytes, how much bandwidth should be allocated to the strict priority queue for six VoIP calls that use a G.729 codec over a multilink PPP link with cRTP enabled?

A. 80.4 kb/s
B. 91.2 kb/s
C. 78.4 kb/s
D. 69.6 kb/s
E. 62.4 kb/s

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Voice payloads are encapsulated by RTP, then by UDP, then by IP. A Layer 2 header of the correct format is applied; the type obviously depends on the link technology in use by each router interface: A single voice call generates two one-way RTP/UDP/IP packet streams. UDP provides multiplexing and checksum capability; RTP provides payload identification, timestamps, and sequence numbering.

QUESTION 18
198
To which QoS tool category does compressed RTP belong?

A. classification
B. marking
C. link efficiency
D. queuing
E. prioritization

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
LLQ is a feature that provides a strict PQ to CBWFQ. LLQ enables a single strict PQ within CBWFQ at the class level. With LLQ, delay-sensitive data (in the PQ) is dequeued and sent first. In a VoIP with LLQ implementation, voice traffic is placed in the strict PQ.

QUESTION 19
199
How are queues serviced in Cisco IOS routers with the CBWFQ algorithm?

A. first-in, first-out
B. weighted round robin based on assigned bandwidth
C. strict priority based on assigned priority
D. last-in, first-out
E. weighted round robin based on assigned priority

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Class Based Weighted Fair queuing is an advanced form of WFQ that supports user defined traffic classes i.e. one can define traffic classes based on match criteria like protocols, access control lists (ACLs), and input interfaces. A flow satisfying the match criteria for a class contributes the traffic for that particular defined class. A queue is allocated for each class, and the traffic belonging to that class is directed to the queue for that class.

QUESTION 20
200

In Cisco IOS routers that use low latency queuing, which algorithm is used to presort traffic going into the default queue?

A. first-in, first-out
B. last-in, first-out
C. weighted round robin
D. fair queuing
E. random processing

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
WFQ is a flow-based queuing algorithm used in Quality of Service (QoS) that does two things simultaneously: It schedules interactive traffic to the front of the queue to reduce response time, and it fairly shares the remaining bandwidth between high bandwidth flows. A stream of packets within a single session of a single application is known as flow or converstion. WFQ is a flowbased method that sends packets over the network and ensures packet transmission efficiency which is critical to the interactive traffic. This method automatically stabilizes network congestion between individual packet transmission flows.

QUESTION 21
201
Which statement describes the Cisco best practice recommendation about priority queue bandwidth allocation in relationship to the total link bandwidth when multiple strict priority LLQs are configured on the same router interface?

A. Each LLQ should be limited to one-third of the link bandwidth capacity.
B. The sum of all LLQs should be limited to two-thirds of the link bandwidth capacity.
C. The sum of all LLQs should be limited to one-half of the link bandwidth capacity.
D. The sum of all LLQs should be limited to one-third of the link bandwidth capacity.
E. Cisco does not recommend more than one strict priority LLQ per interface.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:

Cisco Technical Marketing testing has shown a significant decrease in data application response times when Real-Time traffic exceeds one-third of a link's bandwidth capacity. Cisco IOS Software allows the abstraction (and, thus, configuration) of multiple LLQs. Extensive testing and production-network customer deployments have shown that limiting the sum of all LLQs to 33 percent is a conservative and safe design ratio for merging real-time applications with data applications.

QUESTION 22
202
To which Cisco enterprise medianet application class does Cisco TelePresence belong?

A. VoIP Telephony
B. Real-time Interactive
C. Multimedia Conferencing
D. Broadcast Video
E. Low Latency Data

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Telepresence is used for video conferencing which can be done in Real-time so it is Real-time Interactive.

QUESTION 23
203
Refer to the exhibit.

```
!
class-map match-any signal
 match ip dscp cs3
class-map match-any rtp
 match ip dscp ef
!
policy-map VoIP
 class rtp
    bandwidth percent 33
    compress header ip rtp
 class signal
    bandwidth percent 5
 class class-default
    fair-queue
!
interface serial 0/1/0
 service-policy output VoIP
!
```

Assume that the serial interface link bandwidth is full T1. What is the maximum amount of bandwidth allowed for priority queuing of RTP packets with a DSCP value of EF?

A. 33% of 1.544 Mb/s
B. 5% of 1.544 Mb/s
C. 38% of 1.544 Mb/s
D. 62% of 1.544 Mb/s
E. 0% of 1.544 Mb/s

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Since the use of the “priority” keyword was not used in this example 0% is the correct answer.

QUESTION 24
204
Which statement describes the key security service that is provided by the TLS Proxy function on

a Cisco ASA appliance?

A. It provides interworking to ensure that external IP phone traffic is encrypted, even if the rest of the system is unencrypted.
B. It only applies to encrypted voice calls where both parties utilize encryption.
C. It manipulates the call signaling to ensure that all media is routed via the adaptive security appliance.
D. It enables internal phones to communicate with external phones without encryption.
E. It protects Cisco Unified Communications Manager from rogue soft clients and attackers on the data VLAN.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
TLS Proxy is typically deployed in front of Cisco Unified Communications Manager and other unified communications application servers that utilize media encryption.TLS Proxy is not

designed to provide remote-access encryption services for remote phones or client endpoints. Other solutions such as Cisco ASA Phone Proxy or IP Security/Secure Sockets Layer (IPsec/SSL) VPN services are more appropriate.TLS Proxy is not designed to provide a secure campus soft phone solution where the requirement is to provide secure data to phone VLAN traversal or for proxying connections to Cisco Unified Communications Manager.

QUESTION 25
205
Which two statements describe security services that are provided by the Phone Proxy function on a Cisco ASA appliance? (Choose two.)

A. It is supported only on phones that use SCCP.
B. It is supported on an adaptive security appliance that runs in transparent mode.
C. It provides interworking to ensure that the external IP phone traffic is encrypted, as long as the Cisco Unified Communications Manager cluster runs in secure mode.
D. It provides a proxy of phone signaling, with optional use of NAT, to hide the Cisco Unified Communications Manager IP address from the public Internet.
E. It proxies phone media so that internal phones are not directly exposed to the Internet.
F. It supports IP phones that send phone proxy traffic through a VPN tunnel.

Correct Answer: DE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
TLS Proxy is typically deployed in front of Cisco Unified Communications Manager and other unified communications application servers that utilize media encryption. TLS Proxy is not designed to provide remote-access encryption services for remote phones or client endpoints. Other solutions such as Cisco ASA Phone Proxy or IP Security/Secure Sockets Layer (IPsec/SSL)

VPN services are more appropriate. TLS Proxy is not designed to provide a secure campus soft phone solution where the requirement is to provide secure data to phone VLAN traversal or for proxying connections to Cisco Unified Communications Manager.

QUESTION 26
206
Which entity signs a Cisco IP phone LSC?

A. Godaddy.com Enrollment Server

B. Manufacturer Certificate Authority
C. Registration Authority
D. Certificate Authority Proxy Function
E. Cisco Certificate Authority

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
By default, LSC certificates are not installed on Cisco IP phones. Cisco IP phones that are required to use LSC certificates must be provisioned to allow TLS transactions before deployment in the field. LSC certificates can be provisioned to the Cisco IP phones through the Certificate Authority Proxy Function (CAPF) process. This process is completed using TLS and USB tokens coupled with the CTL client. Moreover, the Cisco ASA Phone Proxy feature can serve LSC certificates to the Cisco IP phones. Cisco IP phones will only work with the Cisco ASA Phone Proxy and will not establish secure connectivity with the Cisco Unified Communications Manager.

QUESTION 27
207
Assume 18 bytes for the Layer 2 header and a 10-millisecond voice payload, how much bandwidth should be allocated to the strict priority queue for three VoIP calls that use a G.722 codec over an Ethernet network?

A. 331.2 kb/s
B. 347.8 kb/s
C. 261.6 kb/s
D. 274.7 kb/s
E. 238.4 kb/s

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Reference:
http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080094ae2.shtml

QUESTION 28
208

Assume a 30-millisecond voice payload, 6 bytes for the Layer 2 header, 1 byte for the end-offrame flag, and the IP, UDP, and RTP headers are compressed to 2 bytes, how much bandwidth should be allocated to the strict priority queue for eight VoIP calls that use a G.729 codec over a multilink PPP link with cRTP enabled?

A. 121.6 kb/s
B. 92.8 kb/s
C. 88.4 kb/s
D. 83.2 kb/s
E. 78.4 kb/s

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Reference:
http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080094ae2.shtml

QUESTION 29
209
To which Cisco enterprise medianet application class does Cisco Unified Personal Communicator belong?

A. VoIP Telephony
B. Real-time Interactive
C. Multimedia Conferencing
D. Broadcast Video
E. Low Latency Data

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Enterprise Medianet QoS Recommendations

Reference:
http://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND_40/QoSIn

tro_40.html#pgfId-61116

QUESTION 30
210
Refer to the exhibit.

```
!
class-map match-any signal
 match ip dscp cs3
class-map match-any rtp
 match ip dscp ef
!
policy-map VoIP
 class rtp
    bandwidth percent 33
    compress header ip rtp
 class signal
    bandwidth percent 5
 class class-default
    fair-queue
!
interface serial 0/1/0
 service-policy output VoIP
!
```

Assume that the serial interface link bandwidth is full T1. What is the bandwidth that is guaranteed for voice signaling traffic with a DSCP value of CS3?

A. 33 percent of 1.544 Mb/s
B. 5 percent of 1.544 Mb/s
C. 38 percent of 1.544 Mb/s
D. 62 percent of 1.544 Mb/s
E. 0 percent of 1.544 Mb/s

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Under the policy map VOIP the CS3 value falls under the signal class-map, which has been allocated 5 percent of the bandwidth.

QUESTION 31
211
Which two security services are provided by the Phone Proxy function on a Cisco ASA appliance? (Choose two.)

A. It provides interworking to ensure that external IP phone traffic is encrypted, as long as the

Cisco Unified Communications Manager cluster runs in secure mode.

B. It only applies to encrypted voice calls where both parties utilize encryption.
C. It manipulates the call signaling to ensure that all media is routed via the adaptive security appliance.
D. It supports encrypted TFTP operation of IP phone configuration files.
E. It intercepts and authenticates soft clients before they reach Cisco Unified Communications Manager clusters.
F. It requires a remote routing device with an IPsec VPN tunnel.

Correct Answer: CE
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When using TLS Proxy, the Cisco ASA appliance is inserted between the phones and Cisco Unified Communications Manager. The phones will now establish a TLS session with the ASA appliance. The appliance will, in turn, establish a proxy TLS connection with Cisco Unified Communications Manager on the phone’s behalf. This function generates two TLS sessions.

QUESTION 32
212
Which statement about application inspection of SAF network services on an adaptive security appliance is true?

A. The adaptive security appliance can inspect and learn the ephemeral port numbers that are used by H.225 and H.245 on SAF-enabled H.323 trunks.
B. An explicit ACL must be configured on the adaptive security appliance for SAF-enabled SIP trunks.
C. An explicit ACL must be configured on the adaptive security appliance for SAF-enabled H.323 trunks to account for ephemeral port numbers that are used by H.225 and H.245.
D. The adaptive security appliance can inspect and learn the ephemeral port numbers that are used by H.225 on SAF-enabled H.323 trunks, but H.245 ports must be explicitly defined.
E. The adaptive security appliance provides full application inspection for SAF network services.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:

The Adaptive Security Appliances do not have application inspection for the SAF network service. When Unified CM uses a SAF-enabled H.323 trunk to place a call, the ASA cannot inspect the SAF packet to learn the ephemeral port number used in the H.225 signalling. Therefore, in scenarios where call traffic from SAF-enabled H.323 trunks traverses the ASAs, ACLs must be configured on the ASAs to allow this signaling traffic. The ACL configuration must account for all the ports used by the H.225 and H.245 signaling.
Reference: Cisco Collaboration 9.x Solution Reference Network Designs (SRND) page 4-34

QUESTION 33
213
Which option is the default Cisco Wireless Unified Communications endpoints marking for video media traffic or video RTP traffic?

A. DSCP 8
B. DSCP 24
C. DSCP 34
D. DSCP 46

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When configuring network-level quality of service (QoS), Cisco video endpoints (including Cisco Unified IP Phone 8900 and 9900 Series and Cisco TelePresence System EX Series devices) generally mark traffic at Layer 3 according to Cisco general QoS guidelines related to voice and video packet marking (video media as DSCP 34orPHB AF41; call signaling as DSCP 24 or PHB CS3) and therefore these devices can be trusted.

Topic 6, Cisco Unity Connection

QUESTION 34
214
A Cisco Unity Connection administrator receives a name change request from a voice-mail user, whose Cisco Unity Connection user account was imported from Cisco Unified Communications Manager. What should the administrator do to execute this change?

A. Change the user data in the Cisco Unity Connection administration page, then use the Synch User page in Cisco Unity Connection administration to push the change to Cisco Unified Communications Manager.

B. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unity Connection administration to pull the changes from Cisco Unified CM.
C. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unified CM administration to push the change to Cisco Unity Connection.
D. Change the user profile from Imported to Local on Cisco Unity Connection Administration, then edit the data locally on Cisco Unity Connection.
E. Change the user data in Cisco Unity Connection and Cisco Unified Communications Manager separately.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
As we can see user are getting synch from call manager so we first have to change the details of user on call manager so that user will synch the changes from call manager.

QUESTION 35
215
Which message-handling behavior describes how Cisco Unity Connection Single Inbox works for Outlook users who do not have ViewMail installed?

A. Cisco Unity Connection voice messages are treated as emails without a WAV file attachment.
B. Cisco Unity Connection voice messages are treated as voice messages.
C. Cisco Unity Connection voice messages are treated as emails with a WAV file attachment.
D. Cisco Unity Connection adds a Voice Outbox folder to the Outlook mailbox.
E. Replies to Cisco Unity Connection voice messages are sent to Exchange as well as the Cisco Unity Connection mailbox for the recipient.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco unity here acts as an IMAP server for the outlook user who don’t have view mail installed so user send their request as an IMAP client and unity will revert back with email and wav file attached to play.

QUESTION 36
216
When Single Inbox is configured, what will happen to an email message that was moved from any Outlook folder to the Voice Outbox folder?

A. The email message will be delivered to Cisco Unity Connection.
B. The email message will be kept in the Voice Outbox folder.
C. The move will fail because the operation is not supported.
D. The email message will be moved to the Deleted Items folder.
E. The email message will be permanently deleted and will not be retrievable.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Voicemessages queue for delivery in theVoice Outbox folder that is why it shows inDeleted Items folder.

QUESTION 37
217
Which Cisco Unity Connection call handler greeting, when enabled, overrides all other greetings?

A. holiday
B. closed
C. internal
D. busy
E. alternate

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
An Alternate greeting might be enabled to override the Standard Greeting during certain times, because it is a personal greeting used for specific purpose.

QUESTION 38

218
Which three Cisco Unity Connection call handler greetings can be overridden by the internal greeting? (Choose three.)

A. holiday
B. alternate
C. error
D. busy
E. closed
F. standard

Correct Answer: AEF
Section: (none)
Explanation

Explanation/Reference:
Explanation:
This greeting overrides the Standard, Closed, and Holiday greetings but only for internal callers or users defined in Cisco Unity Connection because the mentioned three greetings are defined for externals users.

QUESTION 39
219
Which Cisco Unity Connection call handler message is played when a caller enters a string of digits that is not found in the search scope?

A. error
B. closed
C. internal
D. busy
E. alternate

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
As soon as unity finds the unexpected behavior it prompts the error message to the user.

QUESTION 40
220
What is the default treatment of a message that is left in the opening greeting default call handler in Cisco Unity Connection?

A. It will be sent to the mailbox for the Operator user.
B. It will be sent to the Undeliverable Messages distribution list.
C. It will be sent to the mailbox of the system administrator.
D. It will be sent to the All Voicemail Users distribution list.
E. It will be sent to the General Delivery Mailbox.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Default call handler is selected when we don’t assign any call handler to user and with this default call handler no specific user assigned so it don’t go to any specific mail box and goes to It will be sent to the Undeliverable Messages distribution list

Exam E

QUESTION 1
221
Which statement about system broadcast messages in Cisco Unity Connection is true?

A. The user can skip a system broadcast message to listen to new messages first.
B. The user can forward a system broadcast message only if it has been played in its entirety.
C. System broadcast messages are synchronized between Cisco Unity Connection and Exchange when Single Inbox is configured.
D. System broadcast messages do not trigger MWI.
E. System broadcast messages are played immediately after users sign in and listen to message counts for new and saved messages.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation: Explanation;
System broadcast messages are played immediately after users log on to Cisco Unity Connection by phone even before they hear message counts for new and saved messages. After logging on, users hear how many system broadcast messages they have and Connection begins playing them.

QUESTION 2
222
Which statement describes the supported integration method when Cisco Unity Connection and Cisco Unified Communications Manager are installed on the same server as Cisco Unified Communications Manager Business Edition?

A. Only SCCP integration is supported.
B. Only SIP integration is supported.
C. Both SCCP or SIP integration are supported, but you must choose one or the other.
D. Q-Sig integration is supported through a voice-enabled Cisco ISR router.
E. Circuit-switched integration is supported through PIMG.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When installed on the same server there is no way to create trunk that is why sccp is the only way Cisco Unity Connection and Cisco Unified Communications Manager are installed on the same server.

QUESTION 3
223
Which statement about accessing secure Cisco Unity Connection voice messages in an Exchange mailbox in a Single Inbox deployment is true?

A. Users can listen to a secure voice message if they use the Outlook email client.
B. Users can listen to a secure voice message if they use the Outlook email client with the ViewMail add-in.
C. Users can listen to a secure voice message with email clients other than Outlook if they have installed the ViewMail add-in.
D. Users cannot listen to a secure message in Exchange because it is not supported in Single Inbox.
E. Secure voice messages are stored on the Cisco Unity Connection server and the Exchange server.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Users can listen to a secure voice message if they use the Outlook email client with the ViewMail add-in. Because in this integration outlook integrate with unity as secresmapclient .

QUESTION 4
224
When Cisco Unity Connection users attempt to connect using Web Inbox and receive a Site Is Unavailable error message, which service status should be verified?

A. Tomcat
B. Connection Exchange Notification Web Service
C. Connection Voicemail Web Service
D. Connection Administration
E. Secured Web Server

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Cisco Tomcat service, as the name suggests, is used by the Web Server of CUCM and helps display the administration, operating system, disaster recovery, and other GUI interfaces ofCUCM. The service leverages a built-in CA for Tomcat in that it redirects the incoming HTTP requests to HTTPS using the default self-signed certificate.

QUESTION 5
225
Which statement describes how the digit zero is handled in the predefined restriction tables in Cisco Unity Connection?

A. Zero is listed in the Default Out-Dial Restriction table.
B. Zero is listed in the Default System Transfer Restriction table.
C. Zero is listed in the Default Transfer Restriction table.
D. Zero is listed in the User-Defined and Automatically Added Alternate Extensions Restriction table.
E. Zero is not listed in any default restriction table configuration.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When user dials "0", by default Unity Connection treats it as an operator call and does not block "0" by any restriction table configuration. Only the operator can modify transfer extension associated with operator call.

QUESTION 6
226
In addition to SIP triggers, which two trigger types can invoke applications on Cisco Unity Express? (Choose two.)

A. HTTP
B. IMAP
C. VoiceView

D. JTAPI
E. Cisco Unified CM telephony
F. voice mail

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Triggers are incoming events that invoke application which in turn starts executing the script associated with that application. For example, the incoming event can be an incoming call or an incoming HTTP request.
After you have created and configured your application, you need to create a trigger on the Cisco Unity Express module to point to that application.
Cisco Unity Express supports three types of triggers:
•
SIP triggers—Use this type of trigger to invoke applications in Cisco Unified CME and Cisco SRST mode. This type of trigger is identified by the phonenumber which is dialed to invoke the desired application.
•
JTAPI triggers—Use this type of trigger to invoke applications in Cisco Unified Communications Manager mode. This type of trigger is identified by the phonenumber which is dialed to invoke the desired application.
•
HTTP triggers—Use this type of trigger to invoke applications using an incoming HTTP request. Such a trigger is identified by the URL suffix of the incoming HTTP request. This type of trigger can only be used if an IVR license has been purchased and installed on the system.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/unity_exp/administrator/AA_and_VM/guide/vmadmin_book/syscmp.html#wp1126512

QUESTION 7
227
Which two categories are state-based greetings on Cisco Unity Express? (Choose two.)

A. Meeting
B. Vacation
C. Internal
D. Closed
E. Alternate

F. Extended Absence

Correct Answer: CD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Beginning in version 7.1, you can configure multiple greetings. These greetings fall into the following three categories:
•
Standard greetings
•
Alternate greetings
This category includes the following types of greetings:
–
Alternate
–
Meeting
–
Vacation
–
Extended absence
•
State-based greetings:
This category includes the following types of greetings:
–
Busy
–
Closed
–
Internal
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/unity_exp/administrator/AA_and_VM/guide/vmadmin_book/vmconfig.html

QUESTION 8
228
Which two statements about virtual SNR in Cisco Unified Communications Manager Express are true? (Choose two.)
To configure a virtual SNR DN on Cisco Unified SCCP IP phones, perform the following steps:
Prerequisites
Cisco Unified CME 9.0 or a later version.

Restrictions
Mid-calls are either of the following:
–
Calls that arrive before the DN is associated with a registered phone and is still present after the DN is associated with the phone.
–
Calls that arrive for a registered DN that changes state from registered to virtual and back to registered.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html#pgfId-1012065

A.

B.

C.

D.

Correct Answer:
Section: (none)
Explanation

Explanation/Reference:

QUESTION 9
229
Refer to the exhibit.

```
Jan 29 17:32:01.723: CRYPTO PKI: (A0076) Starting CRL revocation check
Jan 29 17:32:01.723: CRYPTO_PKI: Matching CRL not found
Jan 29 17:32:01.723: CRYPTO_PKI: (A0076) CDP does not exist. Use SCEP to
query CRL.
Jan 29 17:32:01.723: CRYPTO_PKI: pki request queued properly
Jan 29 17:32:01.723: CRYPTO_PKI: Revocation check is complete, 0
Jan 29 17:32:01.723: CRYPTO_PKI: Revocation status = 3
Jan 29 17:32:01.723: CRYPTO_PKI: status = 0: poll CRL
Jan 29 17:32:01.723: CRYPTO_PKI: Remove session revocation service providers
CRYPTO_PKI: Bypassing SCEP capabilies request 0
Jan 29 17:32:01.723: CRYPTO_PKI: status = 0: failed to create GetCRL
Jan 29 17:32:01.723: CRYPTO PKI: enrollment url not configured
Jan 29 17:32:01.723: CRYPTO_PKI: transaction GetCRL completed
Jan 29 17:32:01.723: CRYPTO_PKI: status = 106: Blocking chain verification
callback received status
Jan 29 17:32:01.723: CRYPTO PKI: (A0076) Certificate validation failed
```

The public key infrastructure debugs are generated on a Cisco IOS VPN router for a failed certification validation on an incoming connection from an IP phone client. Which option is a possible solution for this problem?

A. Define a matching Certification Revocation List on the Cisco IOS VPN router.
B. Define a Certification Revocation List in the IP phone certificate.
C. Disable revocation check for the trustpoint.
D. Define an enrollment URL for the trustpoint.
E. Define a matching Certification Revocation List on the Cisco Unified Communications Manager.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
When a certificate is issued, it is valid for a fixed period of time. Sometimes a CA revokes a certificate before this time period expires; for example, due to security concerns or a change of name or association. CAs periodically issue a signed list of revoked certificates. Enabling revocation checking forces the IOS router to check that the CA has not revoked a certificate every time it uses that certificate for authentication.
When you enable revocation checking, during the PKI certificate validation process the router checks certificate revocation status. It can use either CRL checking or Online Certificate Status Protocol or both, with the second method you set in effect only when the first method returns an error, for example, that the server is unavailable.
With CRL checking, the router retrieves, parses, and caches Certificate Revocation Lists, which provide a complete list of revoked certificates. OCSP offers a more scalable method of checking revocation status in that it localizes certificate status on a Validation Authority, which it queries for the status of a specific certificate.

QUESTION 10
230
In Cisco Unity Connection, to which three configuration dialog boxes can a user assign a search space? (Choose three.)

A. Routing Rule
B. Call Handler
C. Interview Handler
D. Contacts
E. Users
F. Port
G. Phone System

Correct Answer: ABE

Section: (none)
Explanation

Explanation/Reference:
Explanation:
In unity connection, user can assign a search space in:
Users
Call Routing Rules
System Distribution Lists
System Call Handlers
Directory Handlers
Interview Handlers
Digital Networking
VPIM Locations
Administrator-Defined Contacts

QUESTION 11
231
Which two search scope options are removed from a directory handler when you check the “voice enabled” check box? (Choose two.)

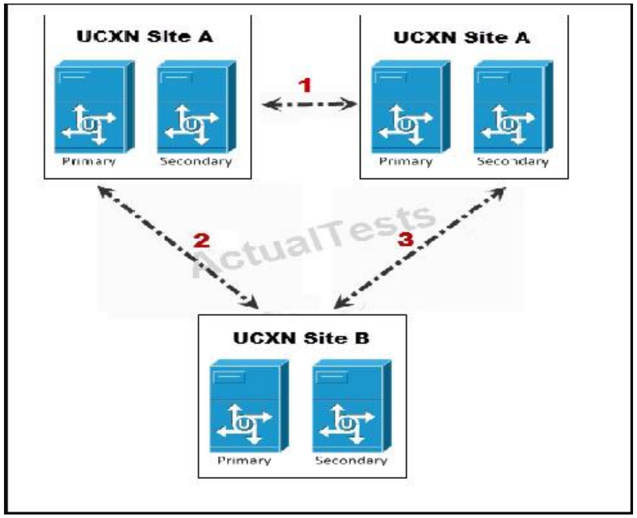
A. Class of Service
B. System Distribution List
C. Search Space
D. Partition
E. Phone System

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
You can configure the scope of a directory handler to define the objects that callers who reach the directory handler can find or hear. For phone directory handlers, you can set the scope to the entire server, to a particular class of service, to a system distribution list, or to a search space (either inherited from the call or specified for the directory handler). For voice-enabled directory handlers, you can set the scope to the entire server or to a search space (either inherited from the call or specified for the directory handler).
When callers search a directory handler for a particular name, if the scope of the directory handler is set to a search space, Cisco Unity Connection searches each partition in the search space and returns a list of all of the objects that match the name.

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/administration/guide/8xcucsagx/8xcucsag235.html#pgfId-1058609

QUESTION 12
232
Refer to the exhibit.



Cisco Unity Connection Site A has two locations. Cisco Unity Connection Site B has one location.
Which protocols connect the locations and servers together for messaging and replication?

A. 1 - SMTP
   2 - HTTP/HTTPS, SMTP
   3 - None
B. 1 - SMTP
   2 - SMTP
   3 - SMTP
C. 1 - HTTP/HTTPS, SMTP
   2 - HTTP/HTTPS, SMTP
   3 - HTTP/HTTPS, SMTP
D. 1 - HTTP/HTTPS, SMTP
   2 - SMTP
   3 – None

Correct Answer: A
Section: (none)

Explanation

Explanation/Reference:
Explanation: You can join two or more Connection servers or clusters (up to a maximum of ten) to form a well-connected network, referred to as a Connection site. The servers that are joined to the site are referred to as locations. (When a Connection cluster is configured, the cluster counts as one location in the site.) Within a site, each location uses SMTP to exchange directory synchronization information and messages directly with every other location. Each location is said to be linked to every other location in the site via an intrasite link.
When you link two Cisco Unity Connection sites with an intersite link, the gateway for each site is responsible for collecting information about all changes to the local site directory, and for polling the remote site gateway periodically to obtain information about updates to the remote site directory. The gateways use the HTTP or HTTPS protocol to exchange directory synchronization updates.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/networking/guide/8xcucnetx/8xcucnet010.html
Topic 7, Cisco Unified Contact Center Express

QUESTION 13
233
Which Cisco Unified Contact Center Express data store contains user scripts, grammars, and documents?

A. configuration data store
B. repository data store
C. agent data store
D. historical data store
E. script data store

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Unified CCX applications might use auxiliary files that interact with callers, such as scripts, prerecorded prompts, grammars, and custom Java classes. Depending on each implementation,
Unified CCX applications use some or all of the following file typesThe Unified CCX Server’s local disk prompt, grammar, and document files are synchronized with the central repository during Unified CCX engine startup and during run-time when the Repository datastore is modified.

QUESTION 14

234
Which Cisco Unified Contact Center Express script media step can invoke a VXML application to retrieve and play prompts on-demand from an off-box location?

A. Play Prompt step
B. Voice Browser step
C. Menu step
D. Recording step
E. Simple Recognition step

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
CRA Voice Browser is fully integrated with the CRA Engine. You can use scripts designed in the CRA Editor to extend VoiceXML applications by providing ICD (Integrated Contact Distribution) call control and resource management.For example, you can use VoiceXML to build a speech dialog as a front end to collect information from the caller. You can then pass this information to the CRA script, and when the agent receives the call, the information collected by VoiceXML will be available.You use the Voice Browser step in the Media palette of the CRA Editor to invoke a VoiceXML application.You can use the bundled voicebrowser.aef script as an example for creating scripts that invoke VoiceXML. (You can create custom scripts to execute other steps in addition to VoiceXML.)

QUESTION 15
235
A company that is using the Cisco Unified Contact Center Express Enhanced version requires that selected types of agent calls are automatically recorded. Which call recording operation can be used to satisfy this requirement?

A. Instruct agents to use the Record button on Cisco IPPA to trigger recording.
B. Instruct supervisors to use the Record button on Cisco Agent Desktop to trigger recording.
C. Instruct supervisors to use the Record button on Cisco Supervisor Desktop to trigger recording.
D. Configure the Cisco Agent Desktop workflow to trigger recording.
E. Recording is not supported on the Cisco Unified CCX Enhanced version. It is supported only on the Premium version.

Correct Answer: D
Section: (none)

Explanation

Explanation/Reference:
Explanation:
On-demand recording of active agent calls, available in Enhanced and Premium versions, improves customer service and encourages appropriate and consistent agent behavior and it is a

feature ofCisco Agent Desktop.

QUESTION 16
236
Which statement describes the call recording operation on Cisco Unified Contact Center Express call agents that use Cisco IPPA?

A. Recording is facilitated via desktop monitoring on supported IP phones.
B. Automatic recording is supported.
C. Only G.711 codec is supported.
D. Only SPAN port monitoring is supported.
E. Call recording is not supported on Cisco Unified CCX call agents that use Cisco IPPA.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
There is no mechanism created as of now to record the call so we first span and record it from packet capture or from third party software.

QUESTION 17
237
Which Cisco Unified Contact Center Express data store contains CSQ information?

A. configuration data store
B. repository data store
C. agent data store
D. historical data store
E. script data store

Correct Answer: A

Section: (none)
Explanation

Explanation/Reference:
Explanation:
The Database component is required for any Unified CCX deployment and manages access to the database. The Unified CCX Database contains four data stores. They are as follows:
The configuration data store contains Unified CCX configuration information such as resources (agents), skills, resource groups, teams, and CSQ information. The repository data store contains user prompts, grammars, and documents. The agent data store contains agent logs, statistics, and pointers to the recording files. The historical data store contains Contact Call Detail Records (CCDRs).

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_02/design/guide/UCCX_BK_C39FDB35_00_cisco-unified-contact-centerexpress/UCCX_BK_C39FDB35_00_cisco-unified-contact-center-express_chapter_010.html

QUESTION 18
238
Which Cisco Unified Contact Center Express core system software component communicates with Cisco Agent Desktop for agent state control and call control?

A. Unified CCX Engine
B. Database
C. Monitoring
D. Recording
E. RmCm

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The Unified CCX Engine enables you to run multiple applications to handleUnified CM Telephony calls or HTTP requests.The Unified CCX Engine uses theUnified CM Telephony subsystem to request and receive services from the Computer Telephony Interface (CTI) manager that controlsUnified CM clusters. The Unified CCX Engine is implemented as a service that supports multiple applications.You can use a web browser to administer the Unified CCX Engine and your Unified CCX applications from any computer on the network. Unified CCX provides you the

following two web interfaces:

QUESTION 19
239
How many RTP streams exist on the network when a Cisco Unified Contact Center Express agent is engaged in a call that is being silently monitored and recorded?

A. 3
B. 4
C. 5
D. 6
E. 8

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
6 RTP streams exist when UCCE agent is engaged in a call when it is being silently monitored.
Reference:http://www.cisco.com/en/US/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_0/design/UCCX_BK_UD5B347F_00_uccx-solution-reference-networkdesign_chapter_0110.html

QUESTION 20
240
Which mechanism enables the Cisco Unified CCX Cisco Agent Desktop application to obtain a copy of the RTP packet stream directly from a supported IP phone?

A. SPAN port monitoring
B. desktop monitoring
C. remote SPAN monitoring
D. reflector port monitoring
E. ESPAN monitoring

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:

Desktop monitoring provides a mechanism for the CAD application to obtain a copy of the RTP packet streams directly from the phone and therefore removes the need for a Monitoring component connected to the SPAN port on the Catalyst switch. A Cisco phone supporting desktop monitoring is required and the agent workstation running CAD must be connected to the data port on the back of the agent phone. The Cisco IP Communicator also supports using desktop monitoring for silent monitoring and recording.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_0/design/UCCX_BK_UD5B347F_00_uccx-solution-reference-networkdesign/UCCX_BK_UD5B347F_00_uccx-solution-reference-network-design_chapter_010.html

QUESTION 21
241
Which statement describes DTMF processing on Cisco Unified Contact Center Express with supported SIP-based agent IP phones that are registered to Cisco Unified Communications Manager?

A. Cisco Unified CCX receives the DTMF digits via SIP NOTIFY messages.
B. Cisco Unified CCX receives the DTMF digits in the RTP payload based on RFC 2833.
C. Cisco Unified CCX receives the DTMF digits via JTAPI messages.
D. Cisco Unified CCX receives the DTMF digits via SIP INFO messages.
E. Cisco Unified CCX receives the DTMF digits as part of the audio encoding in the RTP stream.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Unified CCX CTI ports are notified of caller-entered digits (DTMF input) via JTAPI messages from Unified CM. Unified CCX does not support any mechanism to detect in-band DTMF digits where DTMF digits are sent with voice packets. In deployments with voice gateways or SIP phones that only support in-band DTMF or are configured to use in-band DTMF, an MTP resource must be invoked by Unified CM to convert the in-band DTMF signaling so that Unified CM can notify Unified CCX of the caller-entered digits. Be sure to enable out-of-band DTMF signaling when configuring voice gateways in order to avoid using the previous MTP resources.
Reference:http://www.cisco.com/en/US/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_02/design/guide/UCCX_BK_C39FDB35_00_cisco-unified-contact-centerexpress_chapter_010.html

QUESTION 22
242

Which two statements describe the remote supervisory monitoring feature in Cisco Unified Contact Center Express? (Choose two.)

A. It is supported on Cisco Unified CCX Enhanced and Premium editions.
B. It does not require a Cisco Supervisor Desktop or any data network connectivity.
C. Agents are aware that they are being silently monitored.
D. Calls can be silently monitored from a PSTN phone.
E. It supports G.711 and G.729 codecs.
F. It works with SPAN port monitoring only.

Correct Answer: BD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Agents use the Cisco Agent Desktop (commonly referred to as CAD) to login to the Unified CCX server and control their ACD state, control incoming and outgoing calls, chat with supervisors and other agents on their team, view their own real-time statistics, and view their own recent call activity.
Supervisors use the CSD to view real-time queue and agent statistics, view recent call activity for agents, change agent states, chat with agents, and send marquee messages to all agents on the selected team. With the Enhanced or Premium packages, the supervisor can also barge-in or intercept ACD calls, silently monitor agents, and record agent calls.

QUESTION 23
243
Which two guidelines are recommended when configuring agent phones for Cisco Unified CCX agents? (Choose two.)

A. In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to 2.
B. In the Multiple Call/Call Waiting Settings section, set the Busy Trigger value to 2.
C. The Unified CCX extension for the agent must be listed within the top four extensions on the device profile.
D. In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to at least 3.
E. Always enable SRTP when configuring an agent phone.

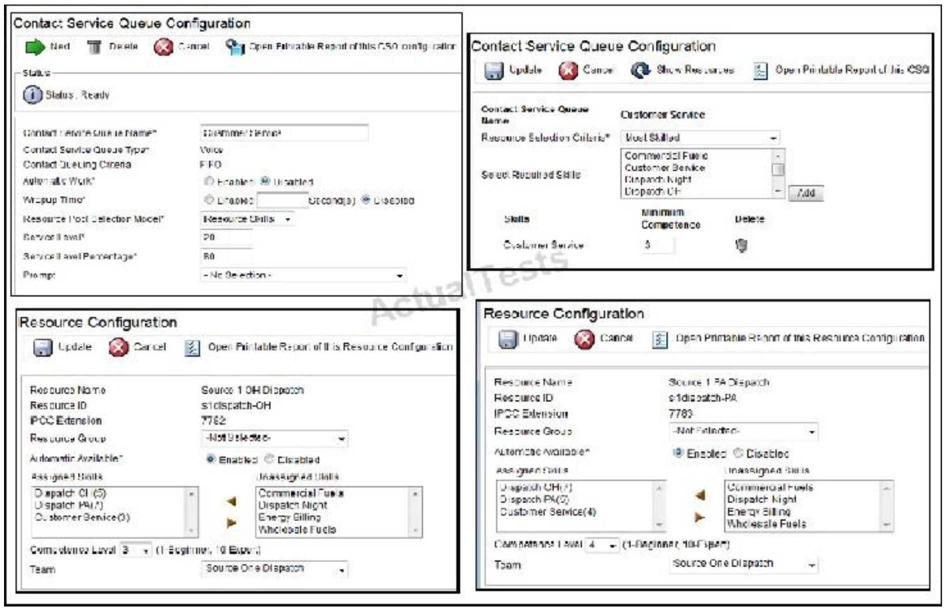
Correct Answer: AC
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Follow these guidelines when configuring agent phones for Unified CCX agents:
Choose Device > Phone in Unified Communications Manager Administration. The Find and List Phones window is displayed.
Enter search criteria to locate a specific phone and click Find. A list of phones that match the search criteria is displayed. Click the device name of the phone to which you want to add a directory number. The Phone Configuration window is displayed.
In the Unified Communications Manager Administration Phone Configuration web page, select the required Association Information (on the left) to get to the Directory Number Configuration web page. On this page, make the following changes:
o In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to 2 (default is 4) for Cisco Unified IP Phones 7900 Series and 3 for Cisco Unified IP Phones 8961, 9951, and 9971.
o In the Multiple Call/Call Waiting Settings section, set the Busy Trigger value to 1 (default is 2).
o In the Call Forward and Call Pickup Settings section, verify that you do not forward any Unified Communications Manager device to the Unified CCX extension of an agent.
o In the Call Forward and Call Pickup Settings section, verify that you do not configure the Unified CCX extension of an agent to forward to a Unified CCX route point.
Always disable (turn off) Secure Real-Time Transport Protocol (SRTP) when configuring a Cisco Unified Communications product. You can disable SRTP for a specified device or for the entire Unified Communications Manager:
o For a specified device—Choose Device > Phone. In the Find and List Phone page, select the required phone device. In the Phone Configuration page for the selected phone, scroll down to the Protocol Specific Information section.To turn off SRTP on the phone device, select any one of the Non Secure SCCP Profile auth by choices from the drop-down list in SCCP Phone Security Profile or SCCP Device Security Profile field.

o For the entire Unified Communications Manager cluster—Choose System > Enterprise Parameters. In the Enterprise Parameters Configuration page, scroll down to the Securities Parameters section, to verify that the corresponding value for the Cluster Security Mode field is 0. This parameter indicates the security mode of the cluster. A value of 0 indicates that phones will register in nonsecure mode (no security).
The Unified CCX extension for the agent must be listed within the top 4 extensions on the device profile. Listing the extension from position 5 on will cause Unified CCX to fail to monitor the device, so the agent will not be able to log in.
Do not forward any Unified Communications Manager device to the Unified CCX extension of an agent.
Do not configure the Unified CCX extension of an agent to forward to a Unified CCX route point.
Do not use characters other than the numerals 0 to 9 in the Unified CCX extension of an agent.

Do not configure two lines on an agent phone with the same extension when both lines exist in different partitions.
Do not assign a Unified CCX extension to multiple devices.
Do not configure the same Unified CCX extension in more than one device or device profile. (Configuring a Unified CCX extension in one device or device profile is supported.)
To use Cisco Unified IP Phones 9900 Series, 8900 Series, and 6900 Series as agent devices, the RmCm application user in Unified Communications Manager needs to have "Allow device with connected transfer/conference" option assigned to itself

QUESTION 24
244
Refer to the exhibit.



Assume that all shown agents are available to take a call. Which agent will receive the call when a select resource step is triggered in the script for the Customer Service CSQ?

A. s1dispatch-PA
B. s1dispatch-OH
C. the agent that has been idle the longest
D. the agent with the shortest handled time

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The Contact Service Queue (CSQ) controls incoming calls by determining where an incoming call should be placed in the queue and to which agent the call is sent.
After you assign an agent to a resource group and assign skills, you need to configure the CSQs. You assign agents to a CSQ by associating a resource group or by associating all skills of a particular CSQ. Agents in the selected resource group or who have all the selected skills are assigned to the CSQ.
Skills within the CSQ can be ordered. This means, when resources are selected, a comparison is done based on the competency level (highest for "most skilled" and lowest for "least skilled") of the first skill in the list. If there is a "tie" the next skill within the order is used, and so on.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/crs/express_10_0/configuration/guide/UCCX_BK_W1AF9DDD_00_uccx-admin-guide-10-0/UCCX_BK_W1AF9DDD_00_uccx-admin-guide-10-0_chapter_0111.html#UCCX_TP_C6155D52_00

Topic 8, Cisco Unified IM and Presence

QUESTION 25
245
Which protocol is used by presence-enabled users in Cisco IM and Presence to control phones that are registered to Cisco Unified Communications Manager?

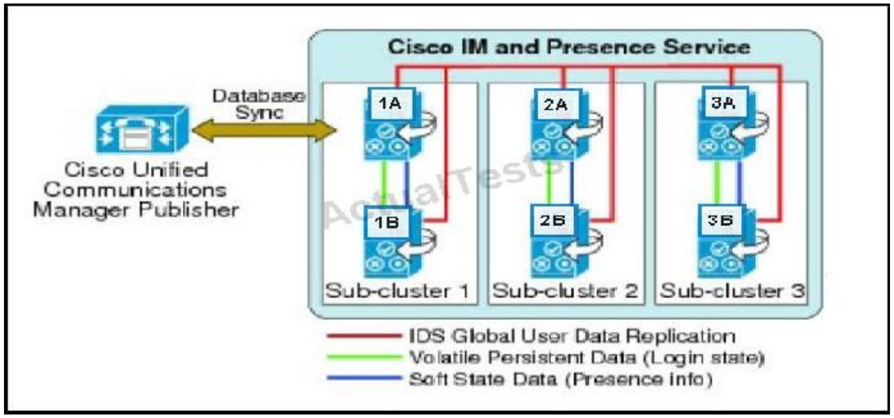
A. AXL/SOAP
B. CTI/QBE
C. SIP/SIMPLE
D. LDAP
E. XMPP

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The CTI gateway provides desk phone control when users are configured for phone association

mode. Proper installation calls upon information to specify CTI gateway server names,addresses, ports, and protocols on CUPS.Configured correctly, the CTI gateway enables users logging in to CUPC to reach the CTI gateway.

QUESTION 26
246
Refer to the exhibit.



In this high-availability Cisco IM and Presence deployment with three subclusters, the first user is assigned to server 1A; the second user is assigned to server 2A; and the third user is assigned to

server 3A. Assume that Cisco IM and Presence is set to active-active mode. To which server will the fourth user be automatically assigned?

A. 1A
B. 3B
C. 1B
D. 2A
E. 3A

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
You can achieve a balanced mode High Availability deployment by evenly balancing users across all nodes in the subcluster, but only using up to 35% of the CPU of each IM and

Presencenode.The balanced mode High Availability deployment option in a redundant mode supports up to fifteen thousand users per cluster. For example, if you have six IM and Presence nodes in your deployment, and fifteen thousand users, you assign 2.5 thousand users to each IM and Presence node.When you use the balanced mode High Availability deployment option in a redundant mode, as compared to a non-redundant mode, only half the number of users are assigned to each node. However, if one node fails, the other node will handle the full load of the additional 50% of users in the subcluster, even at peak traffic. In order to support this failover protection, you must turn on High Availability in each of the subclusters in your deployment.

QUESTION 27
247
Which statement about high availability for XMPP federation in Cisco IM and Presence is true?

A. A maximum of two Cisco IM and Presence nodes can be enabled for XMPP federation.
B. Cisco IM and Presence load balances outbound requests across all nodes that are enabled for XMPP federation.
C. Cisco IM and Presence load balances outbound requests across both nodes that are enabled for XMPP federation in a subcluster.
D. The XMPP federation-enabled nodes should have different priorities and weights on the published DNS SRV for proper inbound request node selection.
E. A single DNS SRV record that resolves to an XMPP federation-enabled node must be published on a public DNS server for inbound request routing.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:
High availability for XMPP federation differs from the high availability model for otherIM and Presence Service features because it is not tied to the two node sub-cluster model.To provide high availability for XMPP federation, you must enable two or more IM and Presence Service nodes in
your cluster for XMPP federation; having multiple nodes enabled for XMPP federation not only adds scale but it also provides redundancy in the event that any node fails

QUESTION 28
248
Which protocol does the Cisco Jabber client use, in conjunction with Cisco IM and Presence, to deliver enterprise-class instant messaging services?

A. SIP
B. CTI/QBE

C. XMPP

D. IRC

E. ICQ

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Many federated IM networks communicate using an open standard, such as Jabber, that leverages the Extensible Messaging and Presence Protocol (XMPP). Networks using XMPP provide open communications with other XMPP-based networks.

QUESTION 29
249
Which protocol that is used between Cisco IM and Presence and Cisco Unified Communications Manager is responsible for the exchange of phone state presence information?

A. AXL/SOAP

B. CTI/QBE

C. SIP/SIMPLE

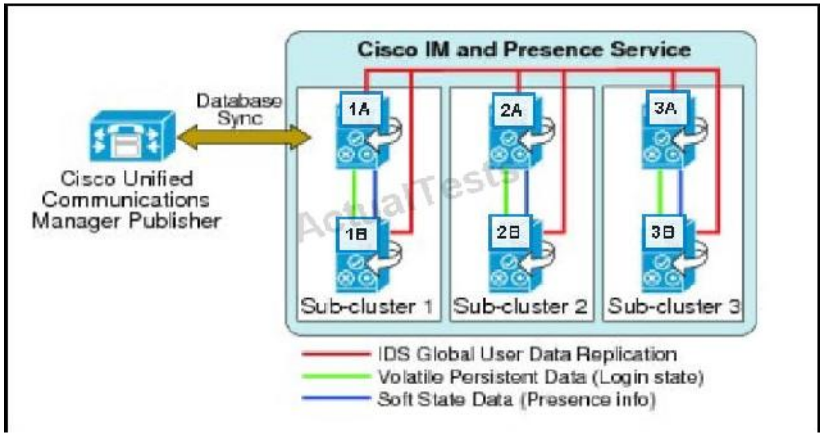
D. LDAP

E. XMPP

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:
To provide interoperability between communications systems, SIP is the protocol leveraged. Enterprise Presence solutions need to provide for a uniform definition of the main communication services such as IM, voice, video, e-mail, web calendaring, and so on, while SIP delivers the necessary features.

QUESTION 30
250
Refer to the exhibit.



In this high-availability Cisco IM and Presence deployment with three subclusters, the first user is assigned to server 1A; the second user is assigned to server 2A; and the third user is assigned to server 3A. Assume that the Cisco IM and Presence is set to Active/Standby mode, to which server should the fourth user be assigned?

A. 1A
B. 3B
C. 1B
D. 2A
E. 3A

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
This deployment model provides the same level of redundancy and high availability as outlined in the "Balanced Redundant High-Availability Deployment" section in this chapter.
The only difference is that users are not deployed in a balanced fashion, but rather all reside on the primary server in the subcluster, and the backup server is there as a standby option if a node failure occurs.

Exam F

QUESTION 1
251
Which two enterprise presence domains can federate with Cisco IM and Presence by using SIP? (Choose two.)

A. AOL
B. Microsoft OCS
C. IBM Sametime
D. Cisco WebEx Connect
E. Google Talk
F. Cisco Unified Presence 8.X Releases

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Microsoft Lync and OCS support presence services with sip as well as AOL so to sip is easy to troubleshoot and feasible for signaling that's why cisco federate these with sip.

QUESTION 2
252
Which statement describes the external database requirement for the Cisco IM and Presence permanent group chat feature?

A. All nodes in a Cisco IM and Presence cluster can share a physical external database.
B. All nodes in a Cisco IM and Presence cluster can share a logical external database.
C. Each node in a Cisco IM and Presence cluster must have its own physical external database.
D. Each node in a Cisco IM and Presence cluster must have its own logical external database.
E. An external database is not mandatory.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:

When you configure an external database entry on IM and Presence, you assign the external database to a node, or nodes, in your cluster as follows:

QUESTION 3
253
Which external database software is required for the Cisco IM and Presence compliance feature?

A. MySQL
B. EnterpriseDB
C. MSSQL
D. SQLite
E. PostgreSQL

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The following Cisco Unified Presence features require an external database:
•
Permanent Group Chat feature - Cisco Unified Presence supports two types of group chat, temporary (ad-hoc) chat and permanent chat. You do not require an external database for temporary chat to work. However, if you require permanent chat rooms on Cisco Unified Presence, you must configure an external database.
•
Instant Messaging Compliance - If you deploy the native Message Archiver (MA) component on Cisco Unified Presence for compliance logging, you require an external database.
Requirements for Configuring an External Database
•
Hardware requirements:
A remote server on which you install the PostgreSQL database(s).
•
Software requirements:
–
Cisco Unified Presence, release 8.x.
–
PostgreSQL database, versions 8.3.x through 9.1.1
–
You can install the PostgreSQL database on either a Linux or a Windows operating system. See the PostgreSQL documentation for details on the supported operating systems and platform requirements.

Reference:http://www.cisco.com/en/US/docs/voice_ip_comm/cups/8_0/english/install_upgrade/database/guide/Preparing_database_setup.html#wp1053954

QUESTION 4
254
Which Cisco IM and Presence service is responsible for logging all IM traffic that passes through the IM and Presence server to an external database for IM compliance?

A. Cisco Presence Engine
B. Cisco Serviceability Reporter
C. Cisco XCP Connection Manager
D. Cisco XCP Message Archiver

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The Cisco Unified Presence XCP Message Archiver service supports the IM Compliance feature. The IM Compliance feature logs all messages sent to and from the Cisco Unified Presence server, including point-to-point messages, and messages from adhoc (temporary) and permanent chat rooms for the Chat feature. Messages are logged to an external Cisco-supported database.

QUESTION 5
255
Which two settings should be configured on the SIP Trunk Security Profile for the IM & Presence Service SIP Trunk? (Choose two.)

A. Check to enable Accept Presence Subscription.
B. Verify that the setting for Incoming Transport Type is TCP+UDP.
C. Configure Device Security Mode to Encrypted.
D. Check to enable Enable Application Level Authorization.
E. Configure the Outgoing Transport Type to TLS.

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:

Explanation:
Configure SIP Trunk Security Profile for IM and Presence Service
Procedure
Step 1
Choose Cisco Unified CM Administration > System > Security > SIP Trunk Security Profile.
Step 2
Click Find.
Step 3
Click Non Secure SIP Trunk Profile.
Step 4
Click Copy and enter CUP Trunk in the Name field.
Step 5
Verify that the setting for Device Security Mode is Non Secure.
Step 6
Verify that the setting for Incoming Transport Type is TCP+UDP.
Step 7
Verify that the setting for Outgoing Transport Type is TCP.
Step 8

Check to enable these items:
Step 9
Click Save.
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/9_0/
CUP0_BK_CFF5B189_00_config-admin-guide-imp-90/CUP0_BK_CFF5B189_00_config-adminguide-
imp-90_chapter_0101.html

QUESTION 6
256
Which three services must be stopped to change the IM & Presence service default domain setting of DOMAIN.NOT.SET? (Choose three.)

A. Cisco XCP Router
B. Cisco Intercluster Sync Agent
C. Cisco XCP Authentication Service
D. Cisco SIP Proxy
E. Cisco Presence Engine
F. Cisco AXL Web service

Correct Answer: ADE
Section: (none)

Explanation

Explanation/Reference:
Explanation:
Change the Domain Value
Follow this procedure if you want to change the domain value (from one valid domain value to another valid IP proxy domain value).
This procedure is applicable if you have a DNS or non-DNS deployment.
Procedure
Step 1
Stop the Cisco SIP Proxy, Presence Engine and XCP Router services on IM and Presence on all nodes in your cluster.
Step 2
On the publisher node, perform the following steps to configure the new domain value:
a.
Select IM and Presence Administration > System > Cluster Topology.
b.
In the right pane, select Settings.
c.
Configure the Domain Name value with the new domain.
a.

Configure the Federation Routing IM and Presence FQDN with the new domain.
c.
You will be prompted to confirm these configuration changes. Select OK for both prompts, and then select Save.
Step 3
On all nodes in the cluster, use this CLI command to set the new domain:
set network domain <new_domain>
This CLI command invokes a reboot of the servers.
Step 4
On all nodes in the cluster, manually start the Cisco Presence Engine and Cisco XCP Router services after the reboot is complete (if required).
Step 5
Manually regenerate all certificates on each node in the cluster.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/ip_address_hostname/9_0_1/IM_P_IPChange/sip_domain.html

QUESTION 7
257
Two Jabber clients are unable to pass instant messages between each other. What is the appropriate next step?

A. Review XCP router logs.
B. Open port 5060 on the firewalls between the PCs and the IM&P servers.
C. Review SIP proxy logs.
D. Review Help > Show Connection Status in each Jabber client, and pull logs as necessary.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:
The XCP Router is the core communication functionality on the Cisco Unified Presence server. It provides XMPP-based routing functionality on Cisco Unified Presence; it routes XMPP data to the other active XCP services on Cisco Unified Presence, and it accesses SDNS to allow the system to route XMPP data to Cisco Unified Presence users. The XCP router manages XMPP sessions for users, and routes XMPP messages to and from these sessions.

QUESTION 8
258
Which three issues prevent a customer from seeing the presence status of a new contact in their

A. incoming calling search space on SIP trunk to IM&P
B. IM&P incoming ACL blocking inbound status
C. subscribe calling search space on SIP trunk to IM&P
D. PC cannot resolve the FQDN of IM&P
E. Owner user ID is not set on device.
F. Primary DN is not set in end user configuration for that user.
G. Subscriber calling search space is not defined on user's phone.

Correct Answer: BCD
Section: (none)
Explanation

Explanation/Reference:
Reference:http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unifiedpresence/97443-cups-cupc-ts.html

QUESTION 9
259
Which three statements about configuring partitioned intradomain federation to Lync are true?

(Choose three.)

A. Intradomain federation to Lync is only possible using SIP.
B. IM&P and Lync should federate to any required remote domains.
C. You must update the URIs of any users migrated from Lync to IM&P to match the Cisco Unified Presence Server SIP URI format.
D. A static route must be added to point the local presence domain to the Lync server.
E. Microsoft RCC must be enabled.
F. The Enable use of Email Address when Federating option can be turned on if SIP URIs are different between IM&P and Lync.

Correct Answer: ACD
Section: (none)
Explanation

Explanation/Reference:
Explanation:
Please refer to the link for more information:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cups/8_6/english/integration_notes/Federation/Intradomain_Federation/Partitioned_Intradomain_Federation/InterdomainFederation.html

QUESTION 10
260
Which option describes how you can show the same contacts in your Jabber for Windows onpremise client as you do on the corporate directory of your IP phone?

A. Switch your Jabber client to use UDS instead of EDI.
B. Switch your Jabber client to use EDI instead of UDS.
C. Update your IM&P server to sync off of the same LDAP directory as your Cisco Unified Communications Manager.
D. Add Jabber to your inbound/outbound firewall rules on your PC.
E. Jabber can only pull directly from LDAP and cannot directly search the Cisco Unified Communications Manager user database.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

Explanation:
LDAP contact resolution —The client cannot use LDAP for contact resolution when outside of the corporate firewall. Instead, the client must use UDS for contact resolution.
When users are inside the corporate firewall, the client can use either UDS or LDAP for contact resolution. If you deploy LDAP within the corporate firewall, Cisco recommends that you synchronize your LDAP directory server with Cisco Unified Communications Manager to allow the client to connect with UDS when users are outside the corporate firewall.
Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/JABW_BK_C4C679C9_00_cisco-jabber-for-windows-97/JABW_BK_C4C679C9_00_cisco-jabber-forwindows-97_chapter_0111.html#CJAB_CN_C2733196_00
Topic 9, Mix Questions

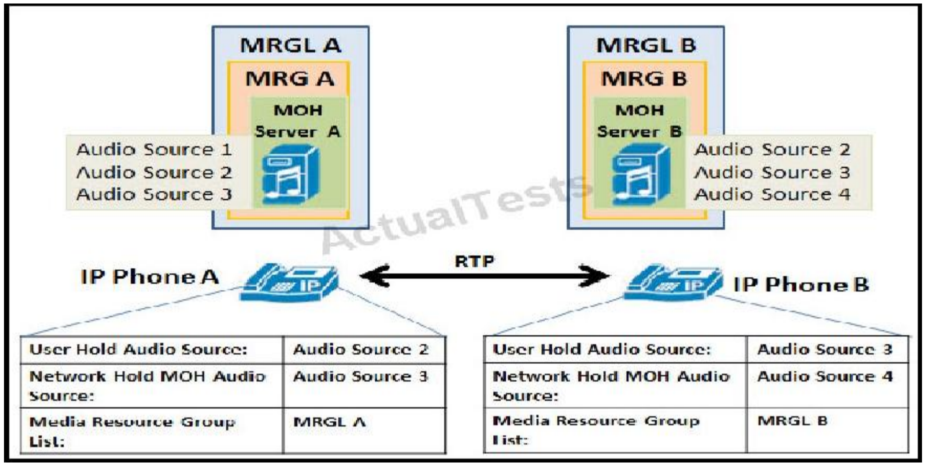
QUESTION 11
261
Company ABC is planning to migrate from Cisco MCS-hosted Cisco Unified Communications Manager applications to Cisco UC on UCS B-Series servers. Which statement about external media supportability is true for this migration?

A. The Cisco Music on Hold USB audio sound card on the MCS servers will continue to work on the USB ports on the UCS server.
B. The Cisco Music on Hold USB audio sound card on the MCS servers will continue to work through the USB ports on the Cisco UCS server KVM dongle cable adaptor connected to the front of the UCS server.
C. The Cisco Music on Hold USB audio sound card on the MCS servers will not work on the UCS server.
D. The Cisco Music on Hold USB audio sound card can be mapped to a virtual USB port on a VMware virtual machine on the UCS server.
E. The Cisco Music on Hold USB audio sound card can be mapped to a virtual serial port on a VMware virtual machine on the UCS server.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 12
262
Refer to the exhibit.



All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assume that the provided configurations exist at the phone line level and multicast MOH is disabled cluster wide. Which description of what will happen when the user of IP phone B presses the Transfer soft key is true?

A. IP phone A user hears audio source 3 from MOH server A.
B. IP phone A user hears audio source 4 from MOH server B.
C. IP phone A user hears audio source 3 from MOH server B.
D. IP phone A user hears tone on-hold beep tones.
E. IP phone A user hears no on-hold music or beep tones.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 13
263

Which codec complexity mode, when deployed on Cisco IOS routers with DSPs using the C5510 chipset, supports the most G.711 calls per DSP?

A. Low
B. Medium
C. High
D. Secure
E. Flex

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 14
264
When DSP oversubscription occurs on a Cisco IOS router using DSP modules that are based on the C5510 chipset, what will happen when an analog phone connected to a FXS port goes offhook?

A. A fast busy tone will be played.
B. A slow busy tone will be played.
C. A network busy tone will be played.
D. A dial tone will be played, but digits will not be processed.
E. No tone will be played.

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 15
265

```
isdn switch-type primary-dms100

controller T1 1/0
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d primary nfas_int 0 nfas_group 1

controller T1 1/1
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d backup nfas_int 1 nfas_group 1

controller T1 2/0
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d none nfas_int 2 nfas_group 1

controller T1 2/1
framing esf
linecode b8zs
pri-group timeslots 1-24 nfas_d none nfas_int 3 nfas_group 1
```

Refer to the exhibit. From this NFAS-enabled T1 PRI configuration on a Cisco IOS router, how many bearer channels are available to carry voice traffic?

A. 91
B. 92
C. 93
D. 94
E. 95

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 16
266
To which QoS tool category does compressed RTP belong?

A. classification
B. marking
C. link efficiency
D. queuing
E. prioritization

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 17
267
In a Cisco Unified Communications Manager system, which three locations does the TFTP server search when a device requests a configuration file from a TFTP server? (Choose three.)

A. internal caches
B. local disk
C. alternate file server
D. NFS server
E. FTP server
F. load server

Correct Answer: ABC
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 18
268
Which three message types for RTCP are valid? (Choose three.)

A. sender report
B. end of participation
C. source description
D. sender codec
E. receiver packets
F. average MOS

Correct Answer: ABC
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 19
269
Which three components are required when configuring the Cisco Unified Communications Manager for time-of-day routing? (Choose three.)

A. Partition
B. Time Period
C. Time Schedule
D. Time Zone
E. Date Time Group

Correct Answer: ABC
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 20
270
Which two steps must be taken when configuring EMCC? (Choose two.)

A. An SFTP server that all clusters share must be set up.
B. Certificates from all remote clusters must be imported into each cluster.
C. Cross-cluster Enhanced Location CAC must be configured.
D. End users must be configured to only exist in their home cluster.
E. A device pool for EMCC phones must be configured.
F. Define MLPP domains.

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 21
271
Which two settings must be the same between the backup source and restore target with DRS in Cisco Unified Communications Manager? (Choose two.)

A. Server Hostname
B. Server IP Address
C. Cluster Security Password
D. NTP Servers
E. Domain Name
F. Certificate Information

Correct Answer: AB
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 22
272
Which two rules apply to MMOH in SRST? (Choose two.)

A. A maximum of three MOH groups are allowed.
B. Cisco Unified SRST voice gateway allows you to associate phones with different MOH groups on the basis of their IP address to receive different MOH media streams.
C. A maximum of five media streams are allowed.
D. Cisco Unified SRST voice gateway allows you to associate phones with different MOH groups on the basis of their MAC address to receive different MOH media streams.
E. Cisco Unified SRST voice gateway allows you to associate phones with different MOH groups on the basis of their extension numbers to receive different MOH media streams.

Correct Answer: CE
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 23

273
Which two QoS guidelines are recommended for provisioning interactive video traffic? (Choose two.)

A. Latency should be no more than 4–5 seconds.
B. Overprovision interactive video queues by 20% to accommodate bursts.
C. Loss should be no more than 5%.
D. Interactive video should be marked to DSCP CS4.
E. Jitter should be no more than 30 ms.

Correct Answer: BE
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 24
274
Which two clock rates does Performance Monitor use to calculate RTP jitter values? (Choose two.)

A. PCMU (G.711 mu-law) , 8000 Hz
B. PCMU (G.711 mu-law) , 32000 Hz
C. PCMA (G.711 A-law) , 16000 Hz
D. H.263 , 90,000 Hz
E. H.263 , 64,000 Hz

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 25
275

Which TFTP server address selection option has the highest precedence on Cisco SCCP IP phones using firmware release 8.0(4) or later?

A. a manually configured alternate TFTP option on the phone
B. the first Option 150 IP address received from the DHCP server
C. the first Option 66 dotted decimal IP address received from the DHCP server
D. the first IPv6 TFTP Server address received from the DHCP server
E. the value of next-server IP address in the boot-up process

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 26
276
Which Cisco IOS multipoint video conferencing profile attempts to reserve DSPs only when it is activated with an actual conference?

A. homogeneous
B. guaranteed-audio
C. rendezvous
D. heterogeneous
E. flex mode video

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 27
277
Which Cisco IOS multipoint video conferencing profile reserves DSPs when it is created in the configuration?

A. flex mode video
B. guaranteed-audio
C. rendezvous

D. heterogeneous
E. guaranteed-video

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:

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QUESTION 28
278
Which statement describes how much of the DSP resources are reserved for video conference when voice-service dsp-reservation 40 is configured on a Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3?

A. 60% of the total available DSP resources
B. 40% of the total available DSP resources
C. 50% of the total available DSP resources
D. Video conferencing resources are reserved dynamically by Cisco IOS and cannot be changed.
E. This command is used for voice resource reservation only.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 29
279
Which statement describes a video conference viewing mode on a Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3 that is configured to work with Cisco Unified Communications Manager?

A. Video of one participant is displayed to all other video capable participants in a round-robin manner.
B. Video of the loudest speaker is displayed across all video capable participants.
C. Video of one participant, except for those with mute enabled, is displayed to all other video

capable participants in a round-robin manner.

D. The dedicated conference lecturer can one participant at a time, while all others can only see the lecturer.

E. Video of one participant is displayed to all other video capable participants in a random manner using an algorithm hard-coded in Cisco IOS.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 30
280

```
Router#show dial-peer voice sum
dial-peer hunt 0
                 AD                                          PRE PASS                  OUT
TAG    TYPE   MIN  OPER PREFIX    DEST-PATTERN       FER THRU SESS-TARGET       STAT PORT     KEEPALIVE

4300   voip   up   up             4...                 0  syst ipv4:10.1.1.4                  active
2300   voip   up   up             [2-3]...             0  syst ipv4:10.1.1.3                  active
1111   voip   up   down           1111                 0  syst ipv4:10.1.1.1                  busy-out
20001  pots   up   up             2001$                0                              50/0/1
20002  pots   up   up             2002$                0                              50/0/2
```

Refer to the exhibit. Which out-of-dialog SIP OPTIONS ping response put dial-peer tag 1111 into its current operational state?

A. 401 Unauthorized

B. 505 Version Not Supported

C. 406 Not Acceptable

D. 482 Loop Detected

E. 500 Server Internal Error

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation: