

ActualTest.400-051.270Questions

Number: 400-051
Passing Score: 800
Time Limit: 120 min
File Version: 9.1



CCIE Collaboration



- This is my first share of braindumps questions. Very helpful study center it is. Best Testing VCE it is.
- Fixed the Exhibit size and Drag drops/hot spot questions.
- Still valid , Hurry up guys study and pass this one.
- I have correct many of questions answers. If there is any more then update this vce and re-upload.

Sections

1. Cisco Collaboration Infrastructure
2. Telephony Standards and Protocols
3. Cisco Unified Communications Manager (CUCM)
4. Cisco IOS UC Applications and Features
5. Quality of Service and Security in Cisco Collaboration Solutions
6. Cisco Unity Connection
7. Cisco Unified Contact Center Express
8. Cisco Unified IM and Presence

Exam A**QUESTION 1**

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: Cisco Collaboration Infrastructure

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 2

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

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

QUESTION 3

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Explanation: Four attributes that are needed to determine the time to complete TFTP file transfer process is:

1. File Size
2. Round-trip time
3. Packet loss percentage
4. Response timeout

Reference:

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6882/ps6884/white_paper_c11-583891_ps10451_Products_White_Paper.html

QUESTION 4

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: Telephony Standards and Protocols

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.

QUESTION 5

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Explanation: Upon receiving an OpenReceiveChannel message, the IP 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 6

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: Telephony Standards and Protocols

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, if Hold was 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 7

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: Telephony Standards and Protocols



Explanation

Explanation/Reference:

Explanation: StationInit means 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 8

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

QUESTION 9

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 AUPE 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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

QUESTION 10

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

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

QUESTION 11

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: Telephony Standards and Protocols

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 12

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

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

QUESTION 13

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: Telephony Standards and Protocols

Explanation

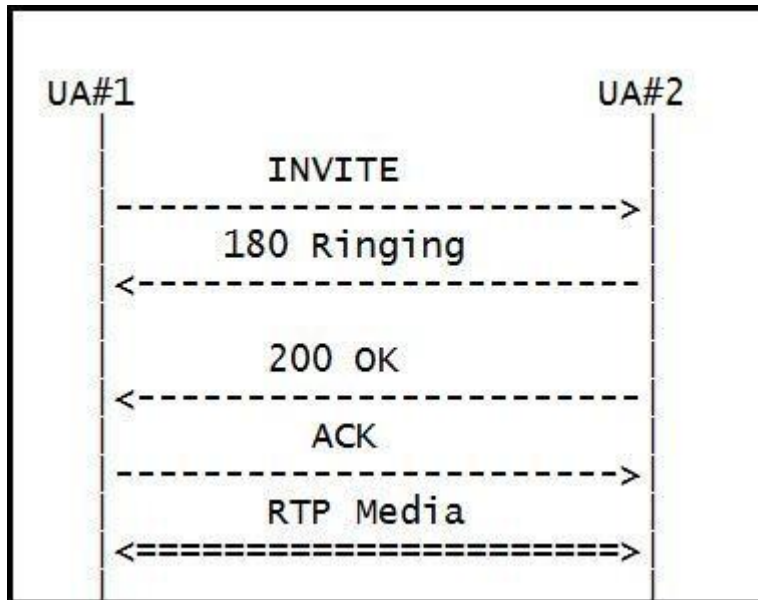


Explanation/Reference:

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

QUESTION 14

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: Telephony Standards and Protocols

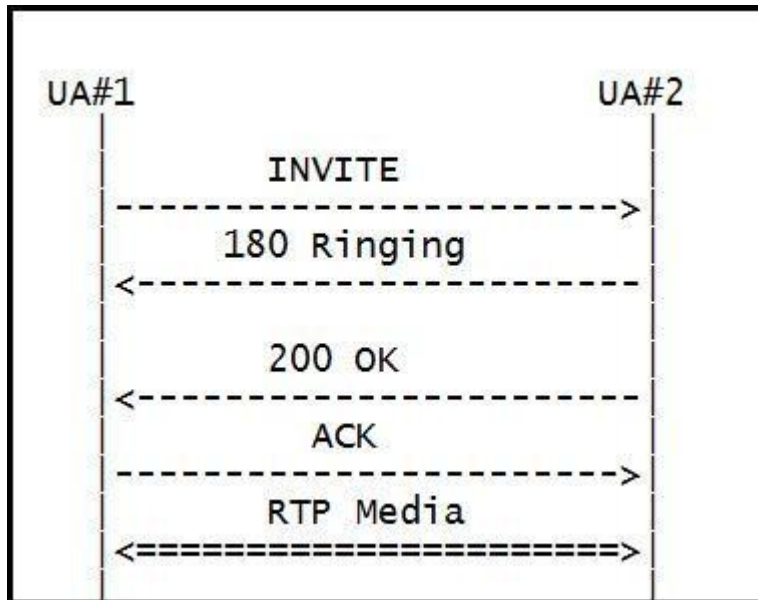
Explanation

Explanation/Reference:

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

QUESTION 15

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: Telephony Standards and Protocols

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 16

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: Telephony Standards and Protocols

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 17

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: Telephony Standards and Protocols

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 18

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: Telephony Standards and Protocols

Explanation

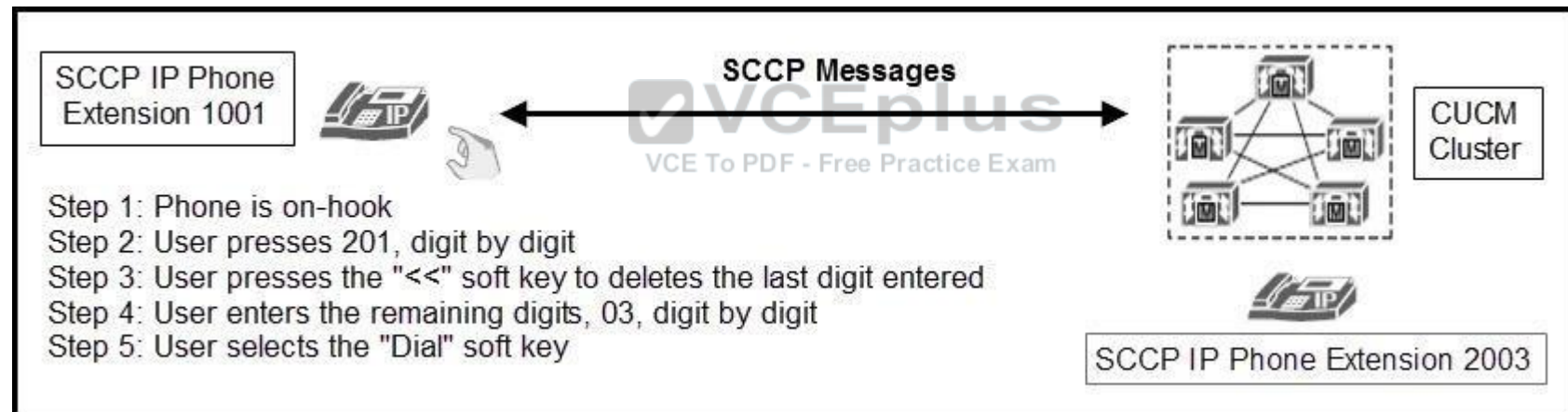
Explanation/Reference:

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 19

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

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

QUESTION 20

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: Telephony Standards and Protocols

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-stampbase as the regular audio channel to simplify the generation of audiowaveforms 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 21

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: Telephony Standards and Protocols

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 22

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, contest, 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: Telephony Standards and Protocols

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 23

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 udpt1 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: Telephony Standards and Protocols

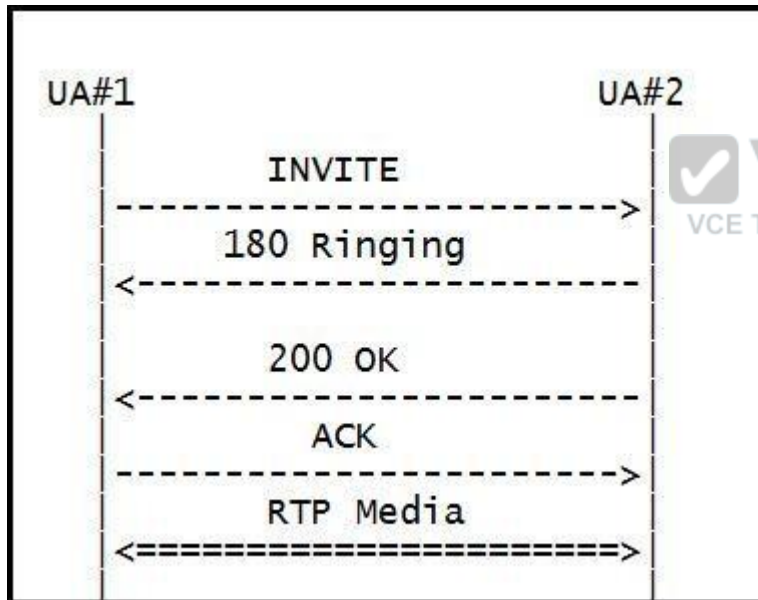
Explanation

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 24

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Explanation: 200 OK Indicates the request was successful.

QUESTION 25

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: Telephony Standards and Protocols

Explanation

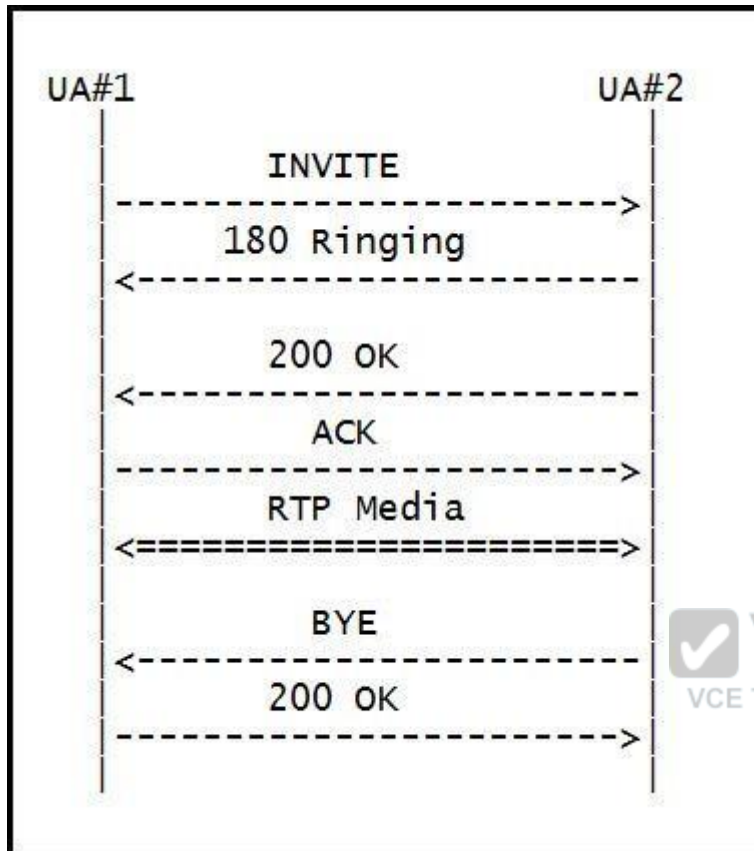


Explanation/Reference:

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 26

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: Telephony Standards and Protocols

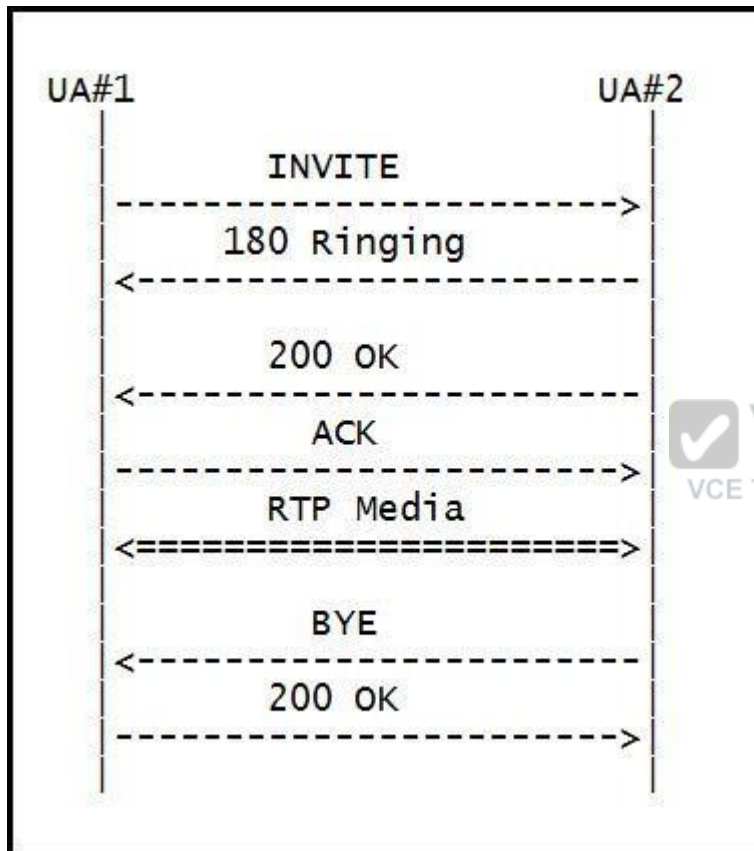
Explanation

Explanation/Reference:

Explanation: Sip transactions are the sip messages which have any sip code associated to it you can see three messages are with code.

QUESTION 27

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Explanation: The two common methods to determine whether a SIP session is active; RTP/RTCP media inactivity timer and session timer have limitations when used with the Cisco UBE. The media inactivity (rtp/rtcp) method will not work if flow around mode is configured as the media is sent directly between endpoints without going through the Cisco UBE and session timer cannot be used if the SIP endpoint does not support session timer. The in-dialog OPTIONS refresh feature introduces a refresh mechanism that addresses these two scenarios, and can be used on SIP-to-SIP and SIP-to-H.323 calls. The refresh with OPTIONS method is meant to only be hop-to-hop, and not end-to-end. Since session timer achieves similar results, the OPTIONS refresh/ping will not take affect when session timer is negotiated. The behavior on the H.323 endpoint is as if it was a TDM-SIP call. The generating in-dialog OPTIONS is enabled at the global level or dialpeer level. The system default setting is disabled. This feature can be use by both a TDM voice gateway and an Cisco UBE.

QUESTION 28

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Explanation: BFCP is a deliverable developed as part of the Internet 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 29

What is the minimum number of TCP sessions needed to complete a H.323 call between two

- A. 323 gateways using slow start?
- B. 0
- C. 1

- D. 2
- E. 3
- F. 4

Correct Answer: C

Section: Telephony Standards and Protocols

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 30

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: Telephony Standards and Protocols

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 31

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: Telephony Standards and Protocols

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 32

Which procedure uses H.225 messages to exchange H.245 Master-Slave 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: Telephony Standards and Protocols

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 33

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Explanation: The Session 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 34

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: Telephony Standards and Protocols

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 35

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: Telephony Standards and Protocols

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.

ARQ--Local zone messages that are sent by H.323 endpoints (usually gateways) to the Cisco gatekeeper. Gatekeepers receive ARQs from an endpoint iF.

o A local zone endpoint initiates a call. OR

o

A local zone endpoint request permission to admit an incoming call.

QUESTION 36

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: Telephony Standards and Protocols

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 37

Which ITU-T recommendation defines the procedures for using more than one video channel in

- A. 320-based systems?
- B. H.324
- C. H.230
- D. H.239
- E. H.264
- F. H.329



Correct Answer: C

Section: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Explanation: H.239 is 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 38

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: Telephony Standards and Protocols

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 39

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: Telephony Standards and Protocols

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 40

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: Telephony Standards and Protocols

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 41

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: Telephony Standards and Protocols

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 42

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: Telephony Standards and Protocols

Explanation

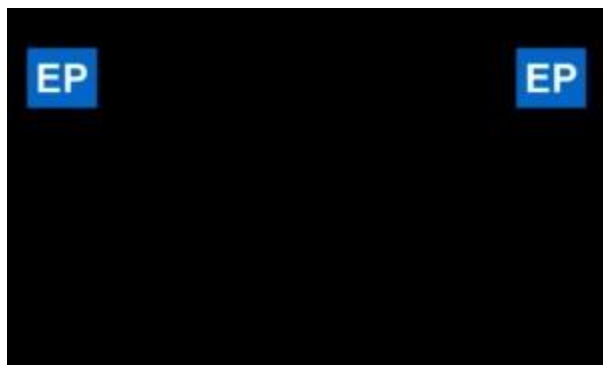
Explanation/Reference:

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 establish bi-directional media 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 43

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: Telephony Standards and Protocols

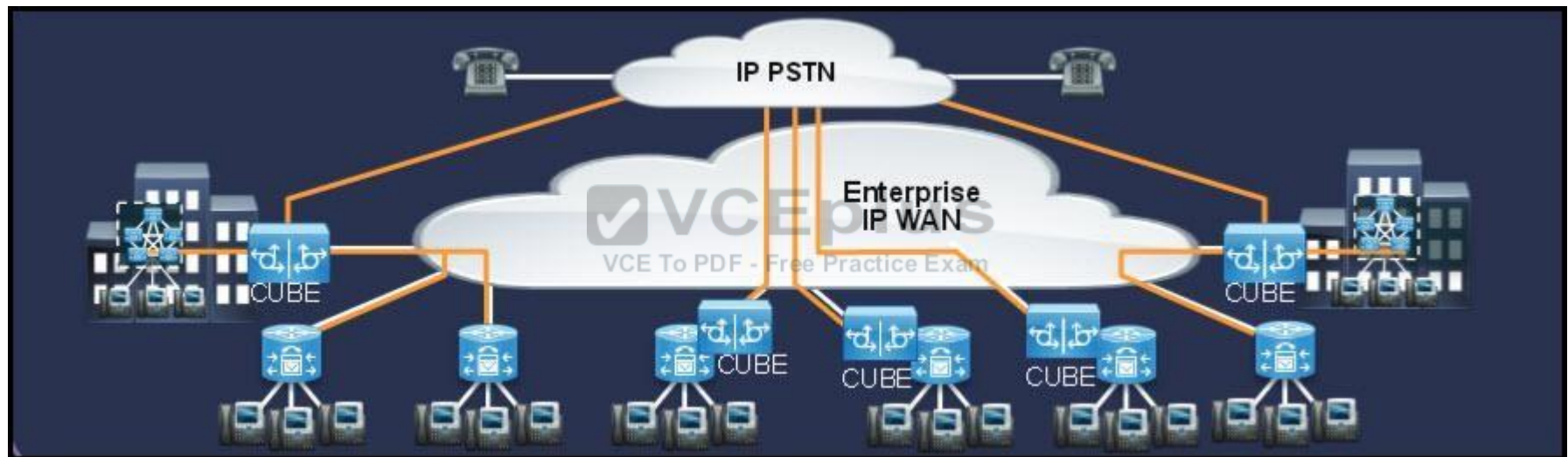
Explanation

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 44

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Reference:

http://www.google.co.in/url?sa=t&rct=j&q=&esrc=s&source=web&cd=1&cad=rja&uact=8&ved=0CBwQFjAA&url=http%3A%2F%2Fwww.cisco.com%2Fcontent%2Fdam%2Fen%2Fus%2Fproducts%2Fcollateral%2FUnified-communications%2FUnified-border-element%2Fcis_45835_cube_assets_wp1e.pdf&ei=p1diVLzKEKOtmAWnhoGgDA&usg=AFQjCNG7F5lsq6teZb_hkhzJP8ENSaG1NA&bvm=bv.79189006,d.c2E

QUESTION 45

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: Telephony Standards and Protocols

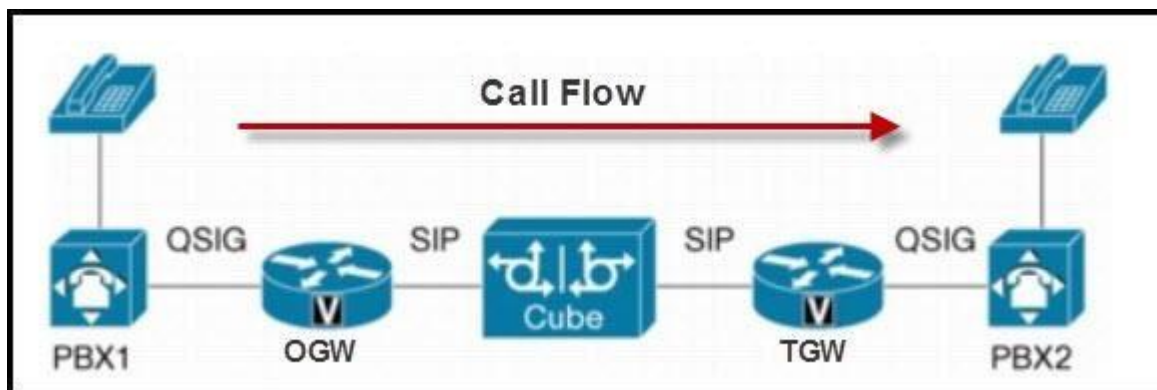
Explanation

Explanation/Reference:

Reference: http://en.wikipedia.org/wiki/List_of_SIP_response_codes

QUESTION 46

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

Correct Answer: B

Section: Telephony Standards and Protocols

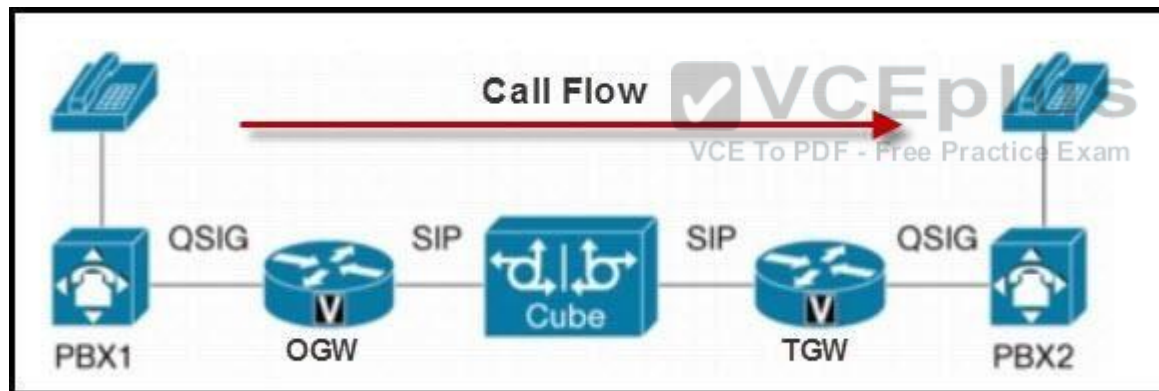
Explanation

Explanation/Reference:

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

QUESTION 47

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: Telephony Standards and Protocols

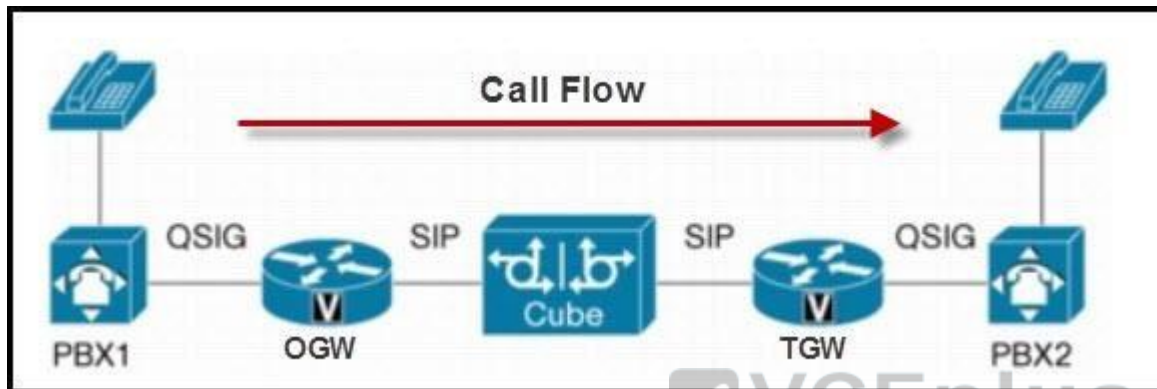
Explanation

Explanation/Reference:

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

QUESTION 48

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Reference: <http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/15-mt/sip-config-15-mt-book/voi-sip-tdm.html>

QUESTION 49

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: Telephony Standards and Protocols

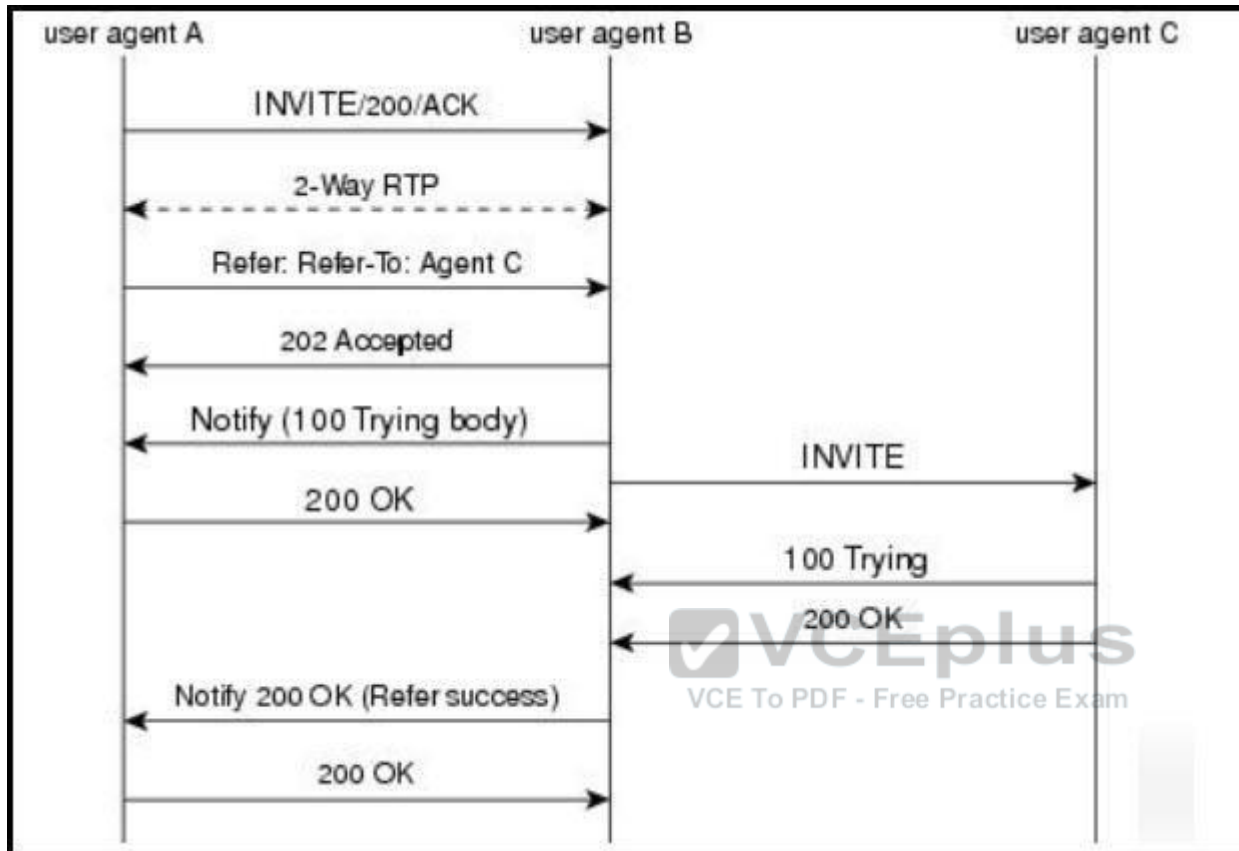
Explanation

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 50

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: Telephony Standards and Protocols

Explanation

Explanation/Reference:

Topic 3, Cisco Unified Communications Manager (CUCM)

QUESTION 51

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: Cisco Unified Communications Manager (CUCM)

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, including initialization, registration, on-hook, off-hook, dialing of the digits, key press on the phone, and so much more.

QUESTION 52

What is the maximum length of any numeric geographic area address in ITU recommendation

- A. 164?
- B. 15
- C. 18
- D. 21
- E. 22
- F. 25

Correct Answer: A

Section: Cisco Unified Communications Manager (CUCM)

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 53

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: Cisco Unified Communications Manager (CUCM)

Explanation

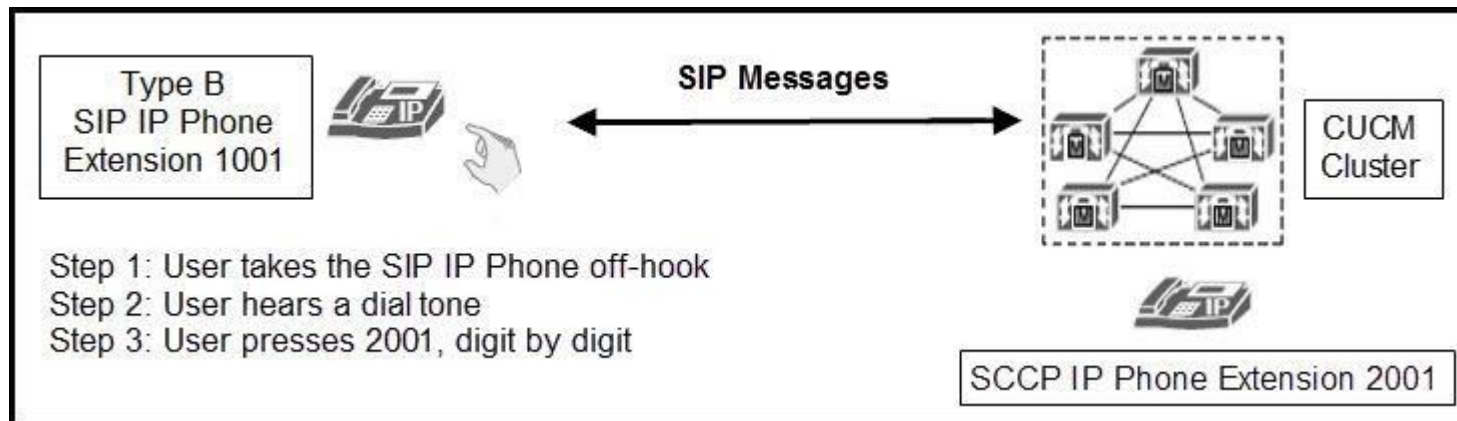
Explanation/Reference:

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 54

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: Cisco Unified Communications Manager (CUCM)

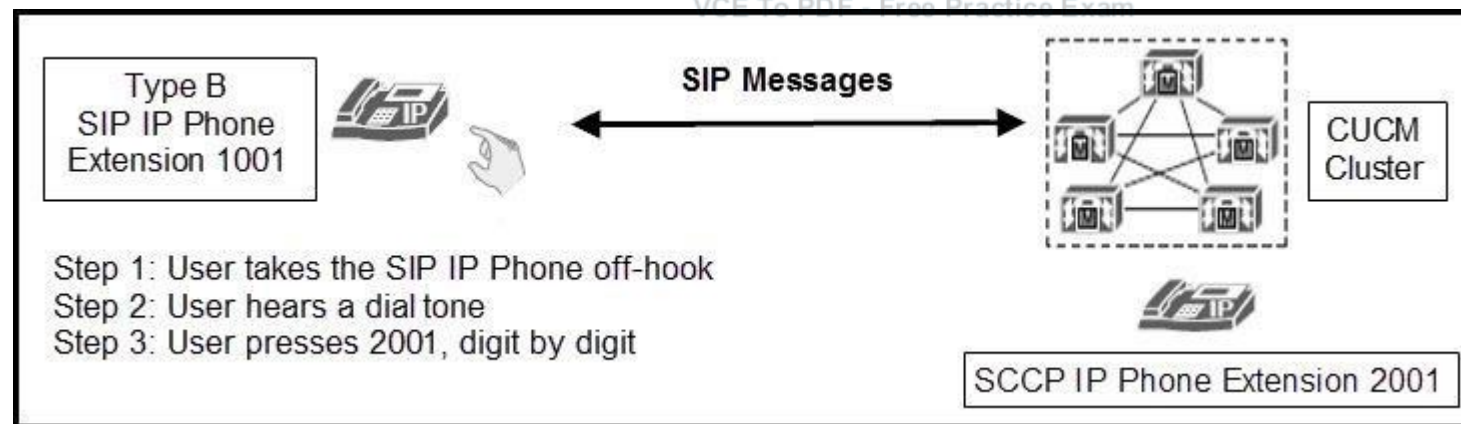
Explanation

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 55

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: Cisco Unified Communications Manager (CUCM)

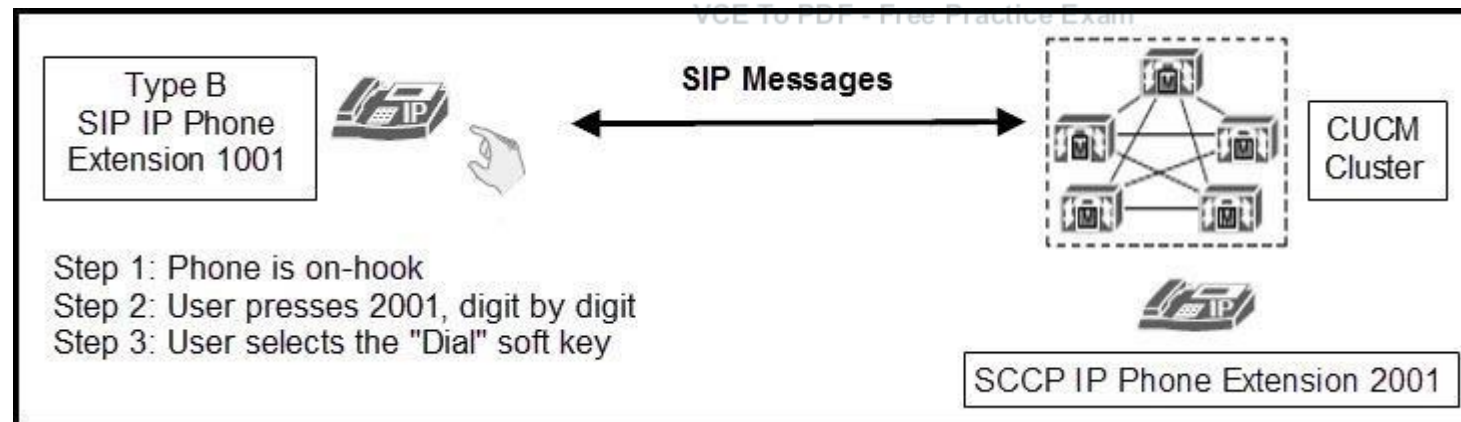
Explanation

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 56

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: Cisco Unified Communications Manager (CUCM)

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 57

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: Cisco Unified Communications Manager (CUCM)

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 58

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: Cisco Unified Communications Manager (CUCM)

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 59

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: Cisco Unified Communications Manager (CUCM)

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 60

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: Cisco Unified Communications Manager (CUCM)

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 61

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: Cisco Unified Communications Manager (CUCM)

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 62

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Explanation: G.728--Low-bit-rate codec that video endpoints support. It supports kilobits per second

QUESTION 63

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: Cisco Unified Communications Manager (CUCM)

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 64

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: Cisco Unified Communications Manager (CUCM)

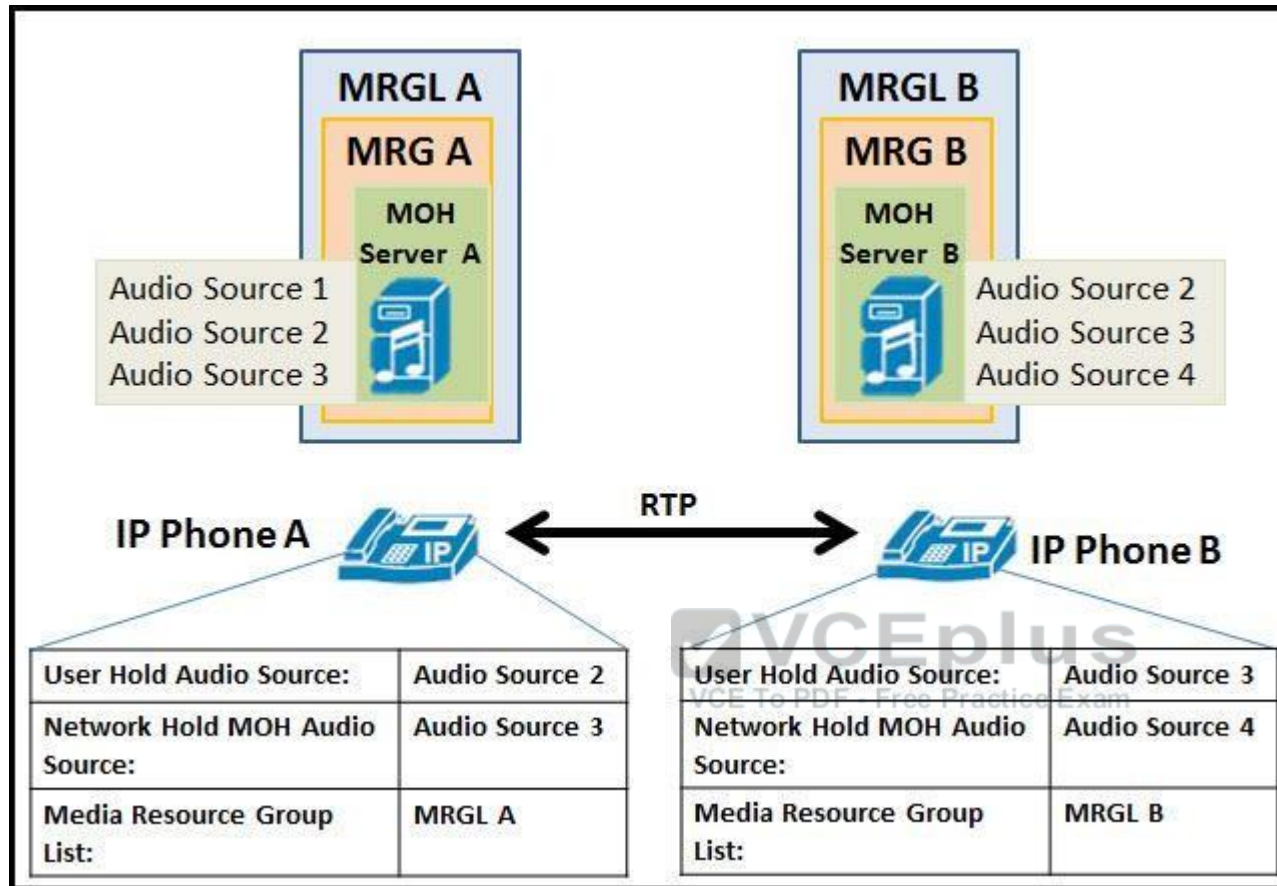
Explanation

Explanation/Reference:

Explanation: The shadow location is 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 65

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: Cisco Unified Communications Manager (CUCM)

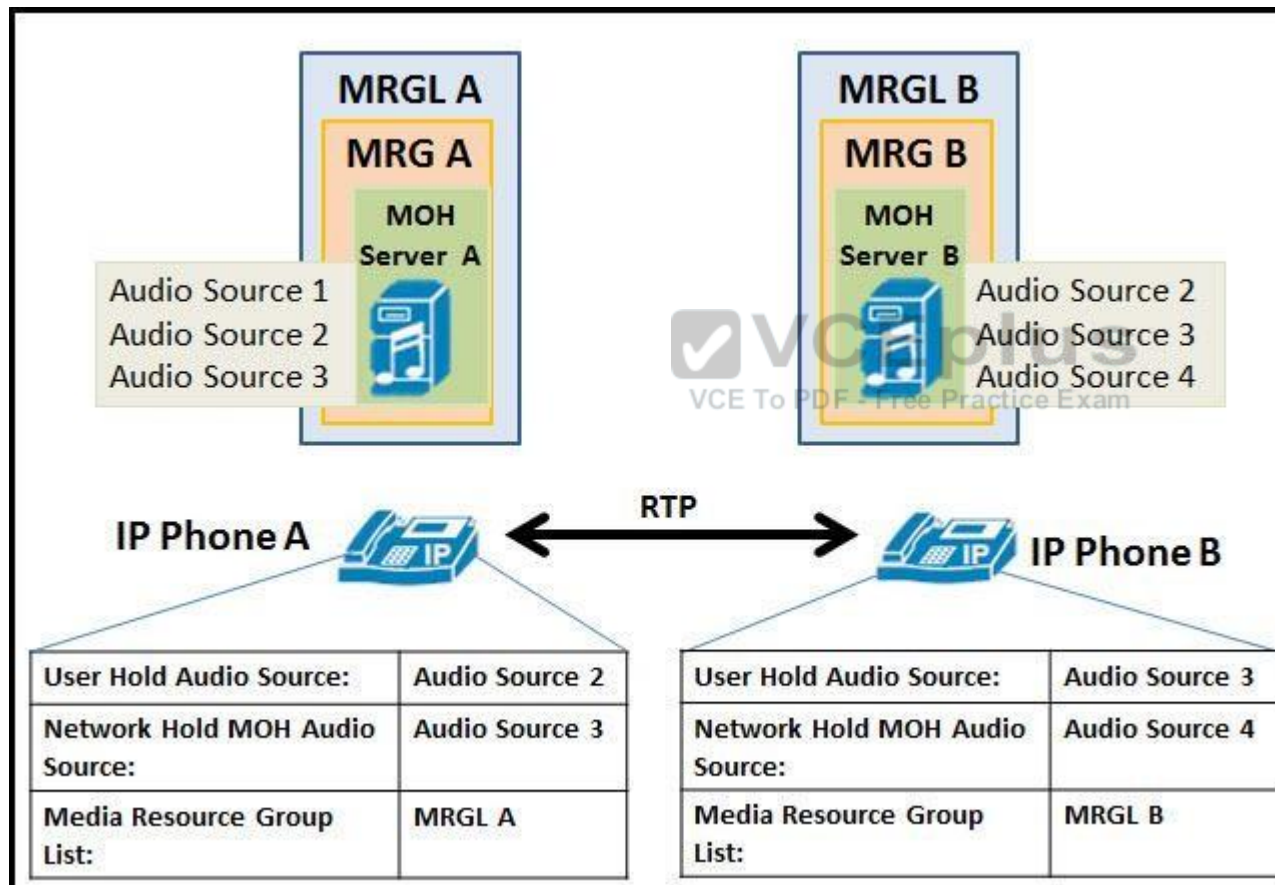
Explanation

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 66

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: Cisco Unified Communications Manager (CUCM)

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 67

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

answer is corrected

QUESTION 68

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: Cisco Unified Communications Manager (CUCM)

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 69

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: Cisco Unified Communications Manager (CUCM)**Explanation****Explanation/Reference:****QUESTION 70**

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: Cisco Unified Communications Manager (CUCM)**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 support presence and EMCC between clusters and maintain CAC.

QUESTION 71

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: Cisco Unified Communications Manager (CUCM)**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 Unified Communications Manager cluster and Unified CME. RSVP Service parameters should be the same on each Unified Communications Manager cluster.

QUESTION 72

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: Cisco Unified Communications Manager (CUCM)

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 73

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: Cisco Unified Communications Manager (CUCM)

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 74

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: Cisco Unified Communications Manager (CUCM)

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 75

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

- A. AUPE
- B. RQNT
- C. NTFY
- D. 200
- E. AUCX

Correct Answer: D

Section: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Explanation:

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

QUESTION 76

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: Cisco Unified Communications Manager (CUCM)

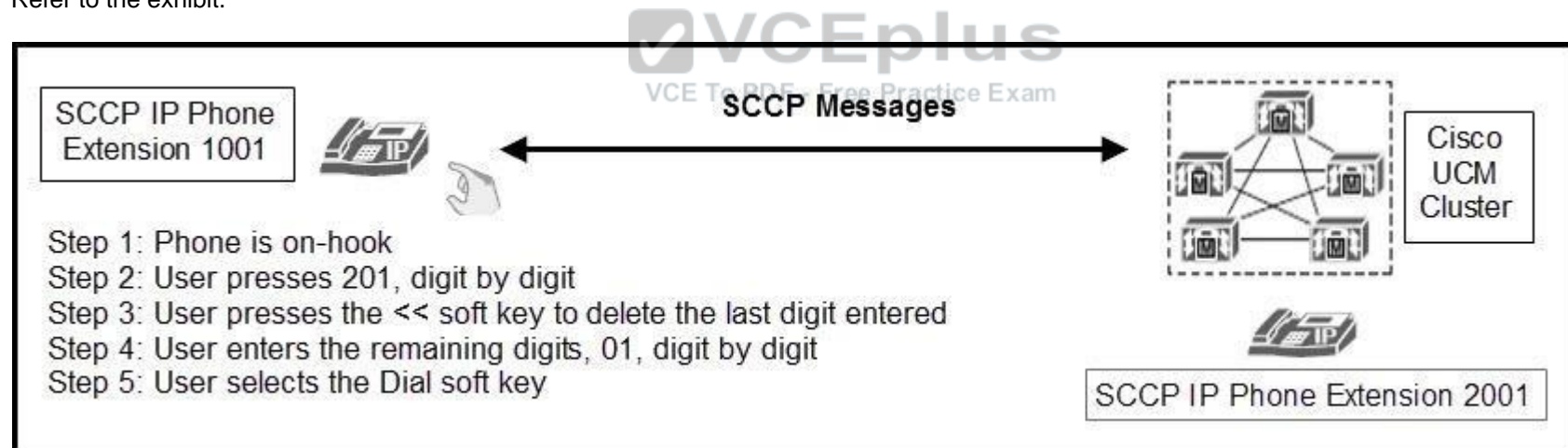
Explanation

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 77

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: Cisco Unified Communications Manager (CUCM)

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-and-dial-rules>

QUESTION 78

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: Cisco Unified Communications Manager (CUCM)

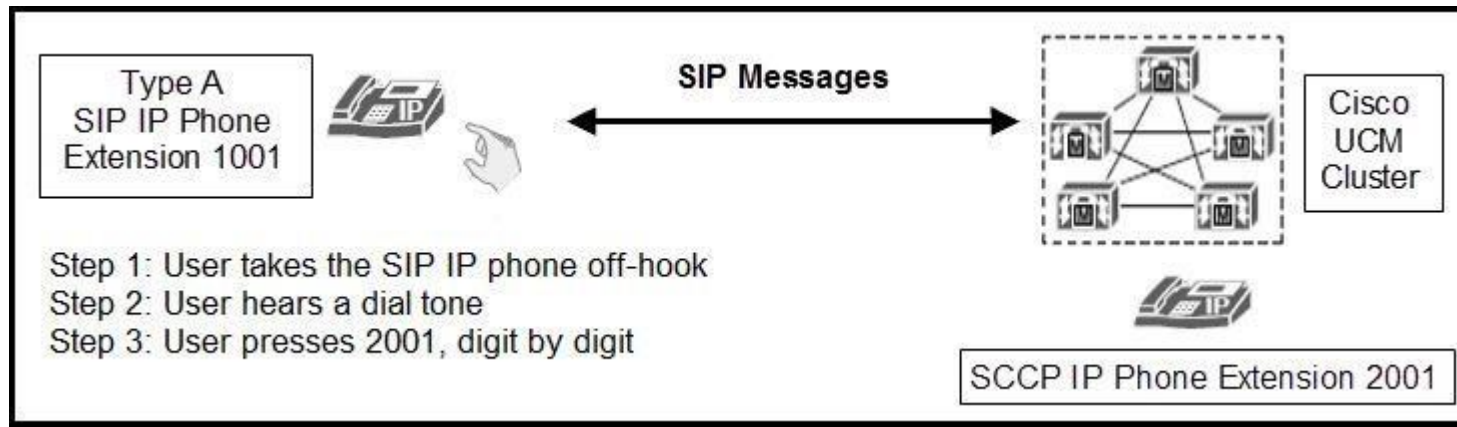
Explanation

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/unified-communications-manager-callmanager/81541-fac-config-ex.html>

QUESTION 79

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: Cisco Unified Communications Manager (CUCM)

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-phones-and-dial-rules>.

QUESTION 80

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

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 81

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/7_1_2/ccmsys/accm-712-cm/a02tftp.html

QUESTION 82

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: Cisco Unified Communications Manager (CUCM)

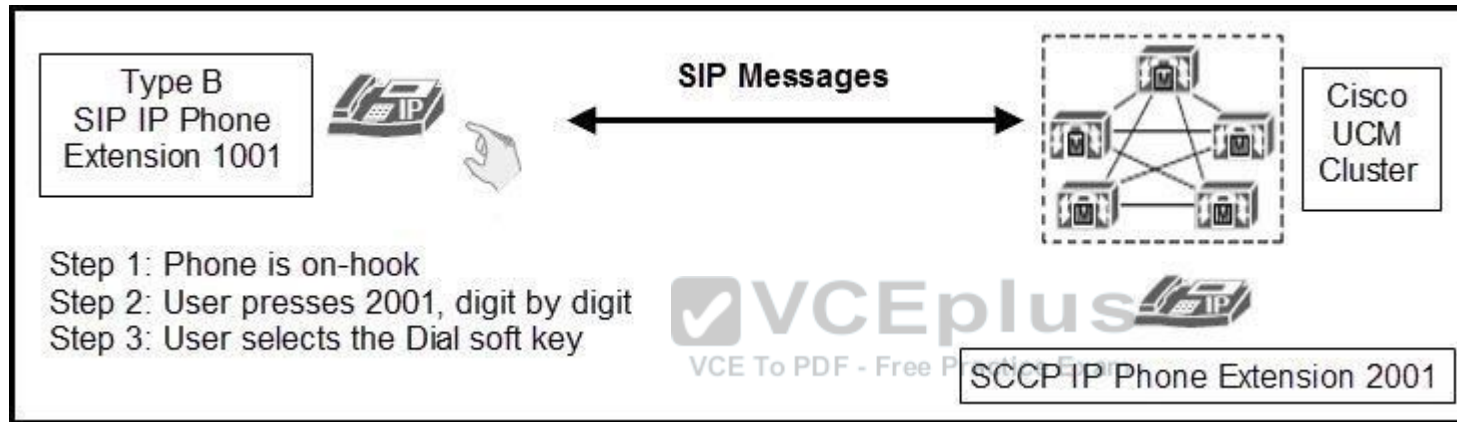
Explanation

Explanation/Reference:

Reference: <https://supportforums.cisco.com/discussion/11874706/forcing-net-calls-pstn-failover> Or
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/dialplan.html

QUESTION 83

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: Cisco Unified Communications Manager (CUCM)**Explanation****Explanation/Reference:**

Reference: <https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phones-and-dial-rules>.

QUESTION 84

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: Cisco Unified Communications Manager (CUCM)**Explanation****Explanation/Reference:**

Reference:

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

QUESTION 85

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: Cisco Unified Communications Manager (CUCM)**Explanation****Explanation/Reference:**

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 86

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/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 87

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcf/C_UCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide- 91_chapter_0110010.html

QUESTION 88

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

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-and-services-guide_chapter_0111.html

QUESTION 89

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

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-and-services-guide_chapter_0111.html

QUESTION 90

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmfeat/C_UCM_BK_CEF0C471_00_cucm-features-services-guide-90/CUCM_BK_CEF0C471_00_cucm-features-and-services-guide_chapter_0111.html

QUESTION 91

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcfg/C_UCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0100011.html

QUESTION 92

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/6_1_1/ccmfeat/c_mfsgd611/fsdnd.html

QUESTION 93

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/6_1_1/ccmfeat/c_mfsgd611/fsdnd.html

QUESTION 94

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: Cisco Unified Communications Manager (CUCM)

Explanation**Explanation/Reference:**

Explanation:-SDL files log SIP messages from CCM.

QUESTION 95

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: Cisco Unified Communications Manager (CUCM)

Explanation**Explanation/Reference:**

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-features-services-guide-100_chapter_0110011.html#CUCM_RF_D0008C2B_00

QUESTION 96

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: Cisco Unified Communications Manager (CUCM)

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_system-guide_chapter_0101000.html#CUCM_TP_R3F173A9_00

QUESTION 97

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Explanation:-

Link:- http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/7x/uc7_0/models.html

QUESTION 98

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: Cisco Unified Communications Manager (CUCM)

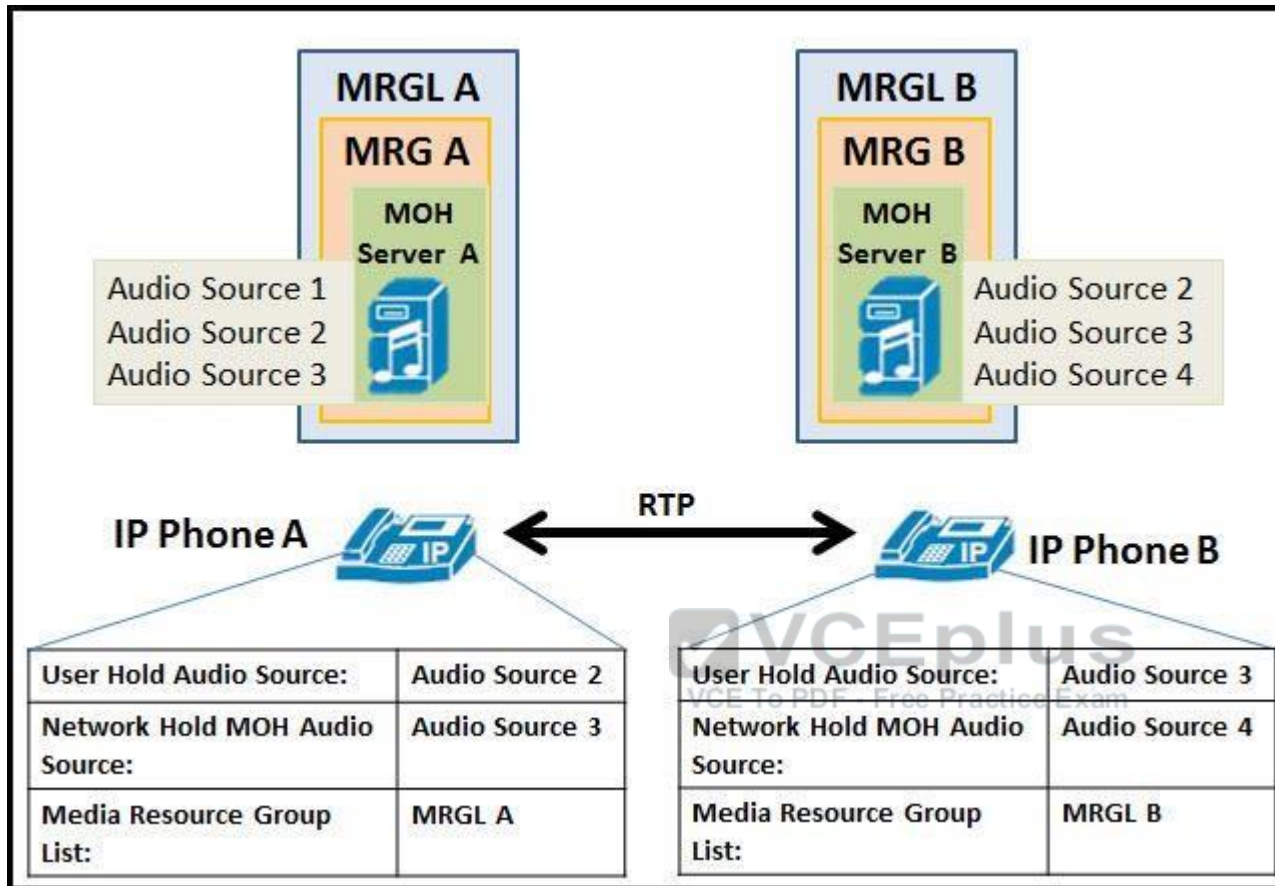
Explanation

Explanation/Reference:

Link:- http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42models.html

QUESTION 99

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.

Correct Answer: C

Section: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

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

QUESTION 100

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/C_UCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_system-guide_chapter_01010.html

QUESTION 101

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Explanation:

Link:-<http://www.techtronicssolution.com/blog/?p=1201>

QUESTION 102

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/C_UCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_system-guide_chapter_0101011.html#CUCM_RF_B5860162_00

And

<https://tools.ietf.org/html/draft-sandbakken-dispatch-bfcp-udp-03>

QUESTION 103

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: Cisco Unified Communications Manager (CUCM)

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 104

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference: http://en.wikipedia.org/wiki/List_of_SIP_response_codes

QUESTION 105

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

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 106

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: Cisco Unified Communications Manager (CUCM)

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 107

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:

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

QUESTION 108

Which two call processing features have a lower priority than the Do Not Disturb settings on a Cisco IP phone? (Choose two.)

- A. park reversion for a locally parked call
- B. hold reversion
- C. intercom
- D. pickup notification
- E. terminating side of a call back
- F. originating side of a call back

Correct Answer: DE

Section: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Explanation: Pickup Notification

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 uses DND Ringer Off, the Callback Available screen will be displayed on the phone after the terminating side goes off hook and on hook.

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 109

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference: <https://supportforums.cisco.com/document/12219631/cucm-sip-trunk-im-presence-server-using-srv>

QUESTION 110

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: Cisco Unified Communications Manager (CUCM)

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 111

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/install/9_0_1/CUCM_BK_I87B437_D_00_installing-cucm-90/CUCM_BK_I87B437D_00_installing-cucm-90_chapter_0100.html#CUP0_TK_G0262E75_00

QUESTION 112

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: Cisco Unified Communications Manager (CUCM)

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 113

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Explanation: Recommendations and Design Considerations for Unified CM Session Management Edition Deployments

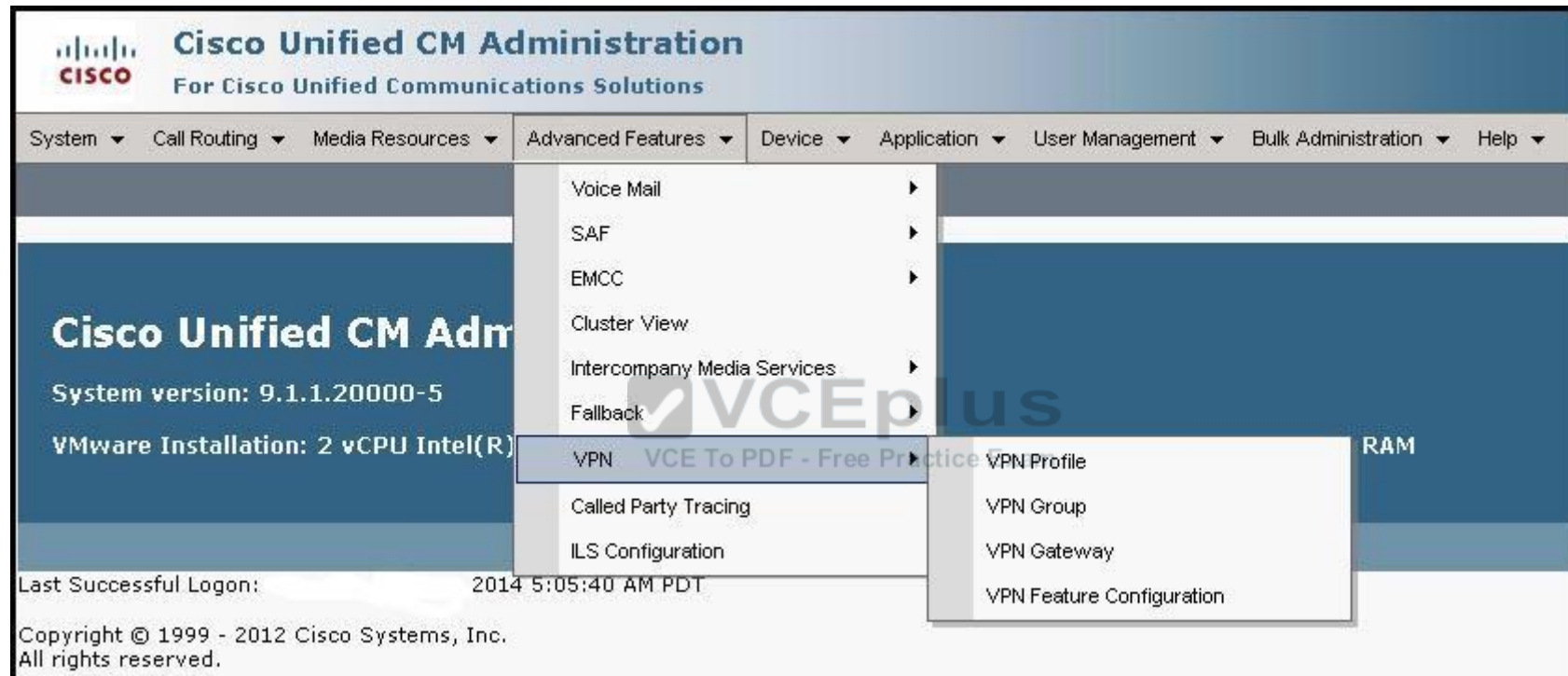
- All leaf clusters that support E-L CAC should be enabled for intercluster E-L CAC with SME.
- SME can be used as a centralized bootstrap hub for the E-L CAC intercluster hub replication network. See LBM Hub Replication Network, for more information.
- All trunks to leaf clusters supporting E-L CAC 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 non-terminating locations; that is, they relay information about locations and are effectively placeholders for user-defined locations on other clusters. Phantom locations are legacy locations 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 114

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: Cisco Unified Communications Manager (CUCM)

Explanation

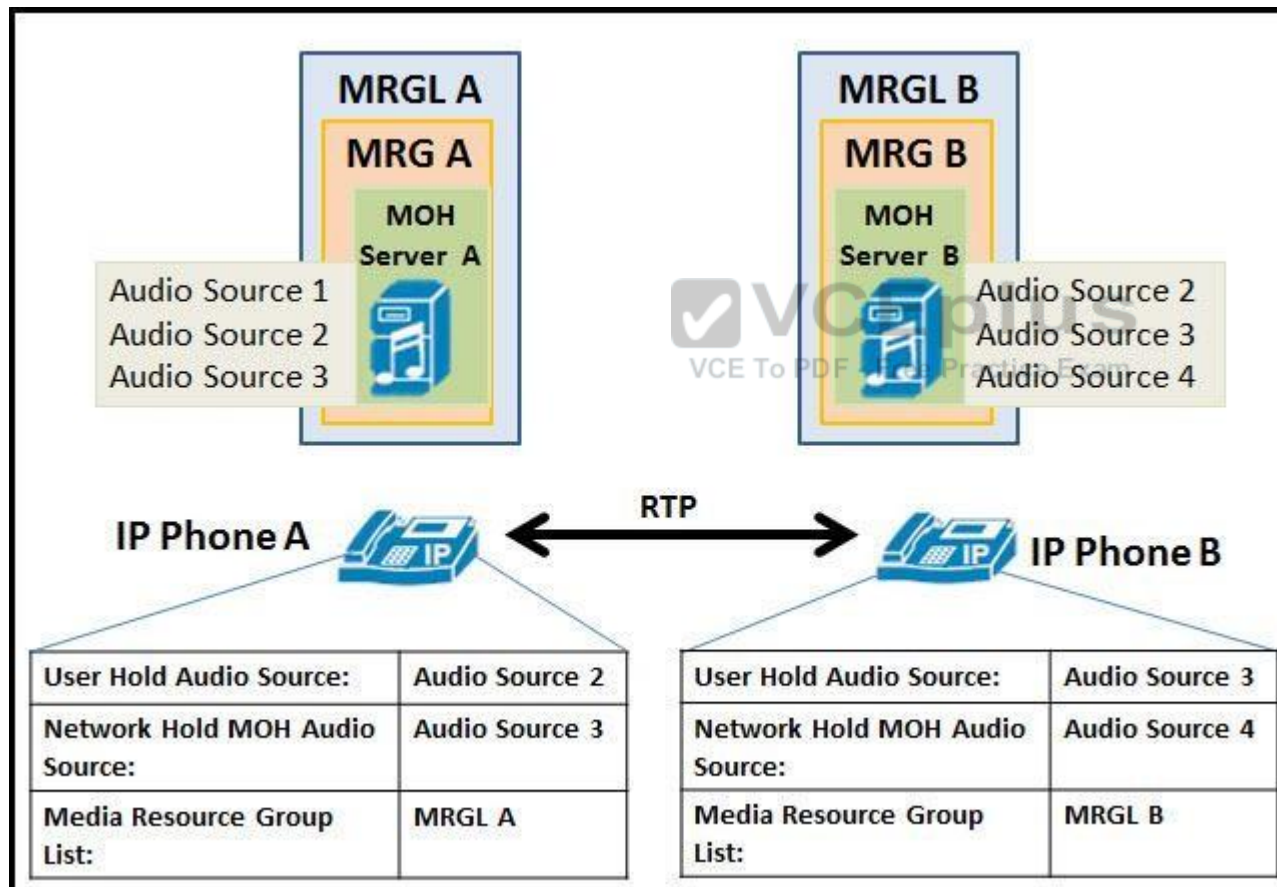
Explanation/Reference:

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 115

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

Answer Incorrect .The correct answer will be option A.

QUESTION 116

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: Cisco Unified Communications Manager (CUCM)

Explanation

Explanation/Reference:

answer is valide

QUESTION 117

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: Cisco IOS UC Applications and Features

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

QUESTION 118

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 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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation: E&M signaling is often the preferred option 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 119

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: Cisco IOS UC Applications and Features

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 120

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 off-hook?

- 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: Cisco IOS UC Applications and Features

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 121

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

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

QUESTION 122

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: Cisco IOS UC Applications and Features

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 123

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: Cisco IOS UC Applications and Features

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 124

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-debug 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: Cisco IOS UC Applications and Features

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 125

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-debug 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 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: Cisco IOS UC Applications and Features

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 126

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: Cisco IOS UC Applications and Features

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 127

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: Cisco IOS UC Applications and Features

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:
Calls can be forwarded to Voice Hunt-Group

Calls can be transferred to Voice Hunt-Group

Member of Voice Hunt-Group can be SCCP, ds0-group, pri-group, FXS or SIP phone/trunk

Max member of Voice Hunt-Group will be 32



QUESTION 128

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: Cisco IOS UC Applications and Features

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 129

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: Cisco IOS UC Applications and Features

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 130

Refer to the exhibit.

```
!  
voice-port 1/1/0  
  caller-id enable  
  station-id number 5251234  
  station-id name Cisco  
  ring number 6  
!
```

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

answer is updated

QUESTION 131

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

http://en.wikipedia.org/wiki/Non-Facility_Associated_Signalling

QUESTION 132

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation: TEI_ASSIGNED, which indicates that the PRI does not exchange Layer 2 frames with the switch. Use the show controller t1 xcommand 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 the debug isdn q921.

QUESTION 133

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: Cisco IOS UC Applications and Features

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 134

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: Cisco IOS UC Applications and Features

Explanation**Explanation/Reference:**

Reference:

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

QUESTION 135

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

- A. C542
- B. C549
- C. C5510
- D. C5421
- E. C5409

Correct Answer: B

Section: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/7x/uc7_0/media.html

QUESTION 136

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/td/docs/routers/access/1900/software/configuration/guide/Software_Configuration/pvdm3_config.html

QUESTION 137

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/td/docs/routers/access/1900/software/configuration/guide/Software_Configuration/pvdm3_config.html

QUESTION 138

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmесnr.html

QUESTION 139

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation:-Because huntstopchannel 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/cm_e_cr/cme_e1ht.html

QUESTION 140

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: Cisco IOS UC Applications and Features
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 141

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: Cisco IOS UC Applications and Features

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 142

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation:-IpPhone 1 & 2 both has busy-trigger-per-button configured to 3 & 2 th
respectively. So, the 4 incoming call will get forwarded to 2100 as busy-triggers are exceeded 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 143

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-debug 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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation:-Because all members of hunt group is unavailable or activates DnD and incoming queued call will forward to voicemail using 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 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 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-debug 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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

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

QUESTION 145

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: Cisco IOS UC Applications and Features

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 specifies 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 146

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: Cisco IOS UC Applications and Features

Explanation



Explanation/Reference:

Explanation:-

Total no of calls = 28 + 12 = 40.

So, native calls are = 40 - 9 = 31.

Link: <http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-border-element/112095-cube-hsrp-config-00.html>.

QUESTION 147

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: Cisco IOS UC Applications and Features

Explanation**Explanation/Reference:**

Explanation:-THL tone sweep allows 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 the best match impedance setting. This test requires minimal supervision.

Link:-<http://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/64282-impedance-choice.html>

QUESTION 148

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: Cisco IOS UC Applications and Features

Explanation

**Explanation/Reference:**

Reference:http://docwiki.cisco.com/wiki/Cisco_IOS_Voice_Troubleshooting_and_Monitoring_-_T1_or_E1_Interface_Troubleshooting#Line_Coding_on_Digital_T1.2FE1_Voice_Ports

QUESTION 149

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference: <http://www.cisco.com/c/en/us/support/docs/voice/digital-cas/22444-t1-cas-ios.html>

QUESTION 150

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

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

QUESTION 151

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Link:- http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42media.html

QUESTION 152

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Link:-<http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/vcr1/vcr1-cr-book/vcr-c4.html>

QUESTION 153

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Link:-<http://www.cisco.com/c/en/us/support/docs/voice/digital-cas/22444-t1-cas-ios.html>

QUESTION 154

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: Cisco IOS UC Applications and Features

Explanation



Explanation/Reference:

Link:-

http://www.cisco.com/c/en/us/td/docs/routers/asr1000/configuration/guide/sbcu/2_xe/sbcu_2_xe_book/sbc_h239.html

QUESTION 155

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=2349872847981
To: "SBC" ;tag=2349872938479
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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation:-Because Right now CUCM challenges the 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-security-guide_chapter_011010.html

QUESTION 156

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference:http://en.wikipedia.org/wiki/List_of_SIP_response_codes

QUESTION 157

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: 0x00000000
  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: Cisco IOS UC Applications and Features

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 158

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: Cisco IOS UC Applications and Features

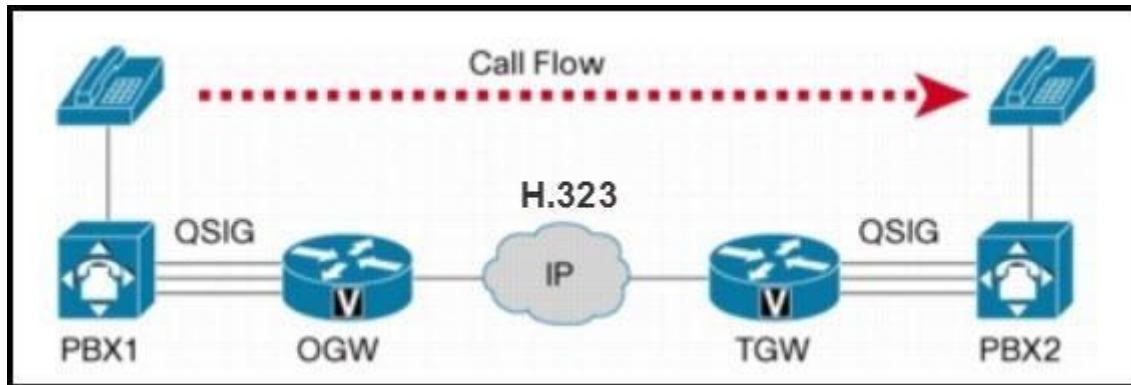
Explanation

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 159

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference: <http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/isdn/configuration/15-mt/vi-15-mt-book/vi-gtd-setup-msg.html>

QUESTION 160

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: Cisco IOS UC Applications and Features**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 E-lead and address information is sent using dual-tone multifrequency (DTMF) digits. Immediate start signaling is vulnerable to glare just like loop-start signaling.

QUESTION 161

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: Cisco IOS UC Applications and Features**Explanation****Explanation/Reference:**

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

QUESTION 162

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation: Three types of video profiles are supported: homogeneous conferences (video switching), heterogeneous conferences (video mixing), and guaranteed audio conferences (best- effort 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 all participants 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- video-conferencing-isr-routers/qa_c67-649850.html

QUESTION 163

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: Cisco IOS UC Applications and Features

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- video-conferencing-isr-routers/data_sheet_c78-649427.pdf

QUESTION 164

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: Cisco IOS UC Applications and Features

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 heterogeneous 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 allows 4-, 8-, or 16-party participants.

Reference: http://www.cisco.com/c/en/us/products/collateral/unified-communications/voice-video-conferencing-isr-routers/data_sheet_c78-649427.html

QUESTION 165

Refer to the exhibit.



```
Router#show dial-peer voice sum
dial-peer hunt 0
```

TAG	TYPE	AD MIN	OPER	PREFIX	DEST-PATTERN	PRE FER	PASS THRU	SESS-TARGET	OUT 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: Cisco IOS UC Applications and Features

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 166

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: Cisco IOS UC Applications and Features

Explanation



Explanation/Reference:

Explanation:

Verifying POTS and VoIP Dial Peer Configurations

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',
in-call-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, lcpif = 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, lcpif = 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 167

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 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: Cisco IOS UC Applications and Features

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 command 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 168

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: Cisco IOS UC Applications and Features

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 169

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Reference: <http://www.cisco.com/c/en/us/support/docs/security/ios-sslvpn/116125-configure-SSLVPN-00.html>

QUESTION 170

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. The 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 ip- helper 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: Cisco IOS UC Applications and Features

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 171

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

answer is corrected

QUESTION 172

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation: Output incomplete to figure out the answer

QUESTION 173

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: Cisco IOS UC Applications and Features
Explanation

Explanation/Reference:

QUESTION 174

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

QUESTION 175

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 udpt1
```

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: Quality of Service and Security in Cisco Collaboration Solutions

Explanation

Explanation/Reference:

Topic 5, Quality of Service and Security in Cisco Collaboration Solutions

QUESTION 176

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: Quality of Service and Security in Cisco Collaboration Solutions

Explanation

Explanation/Reference:

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 177

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: Quality of Service and Security in Cisco Collaboration Solutions

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 178

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: Quality of Service and Security in Cisco Collaboration Solutions

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 179

To which QoS tool category does compressed RTP belong?

- A. classification
- B. marking

- C. link efficiency
- D. queuing
- E. prioritization

Correct Answer: C

Section: Quality of Service and Security in Cisco Collaboration Solutions

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 180

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: Quality of Service and Security in Cisco Collaboration Solutions

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 181

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: Quality of Service and Security in Cisco Collaboration Solutions

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 conversation. WFQ is a flow-based 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 182

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: Quality of Service and Security in Cisco Collaboration Solutions

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 183

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: Quality of Service and Security in Cisco Collaboration Solutions**Explanation****Explanation/Reference:**

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

QUESTION 184

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: Quality of Service and Security in Cisco Collaboration Solutions**Explanation****Explanation/Reference:**

Explanation:

Since the use of the priority keyword was not used in this example 0% is the correct answer.

QUESTION 185

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: Quality of Service and Security in Cisco Collaboration Solutions**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 186

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: Quality of Service and Security in Cisco Collaboration Solutions**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 187

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: Quality of Service and Security in Cisco Collaboration Solutions**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 188

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: Quality of Service and Security in Cisco Collaboration Solutions**Explanation****Explanation/Reference:**

Reference:

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

QUESTION 189

Assume a 30-millisecond 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 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: Quality of Service and Security in Cisco Collaboration Solutions**Explanation****Explanation/Reference:**

Reference:

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

QUESTION 190

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: Quality of Service and Security in Cisco Collaboration Solutions**Explanation****Explanation/Reference:**

answer is accurated

QUESTION 191

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: Quality of Service and Security in Cisco Collaboration Solutions

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 192

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: Quality of Service and Security in Cisco Collaboration Solutions

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 193

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. 323 trunks to account for ephemeral port numbers that are used by H.225 and H.245.
- E. 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.
- F. The adaptive security appliance provides full application inspection for SAF network services.

Correct Answer: C

Section: Quality of Service and Security in Cisco Collaboration Solutions

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 194

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: Quality of Service and Security in Cisco Collaboration Solutions

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 34 or PHB AF41; call signaling as DSCP 24 or PHB CS3) and therefore these devices can be trusted.

QUESTION 195

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: Cisco Unity Connection

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 196

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: Cisco Unity Connection

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 197

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: Cisco Unity Connection

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 198

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: Cisco Unity Connection

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 of CUCM. 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 199

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: Cisco Unity Connection

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 200

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: Cisco Unity Connection

Explanation

Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/unity_exp/rel7_0/administrator/ivr/cueivradmin/httptrig.html

Or

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 201

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: Cisco Unity Connection

Explanation

Explanation/Reference:

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 202

Which two statements about virtual SNR in Cisco Unified Communications Manager Express are true? (Choose two.)

- A. The SNR DN must be configured as SCCP.
- B. Calls cannot be pulled back from the phone associated with the DN.
- C. Ephone hunt groups are supported.
- D. The virtual SNR DN must be assigned to an ephone.
- E. Music on hold is supported for trunk and line side calls.

Correct Answer: AB

Section: Cisco Unity Connection
Explanation

Explanation/Reference:

Explanation: Please go through the below mentioned link for proper explanation.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmearm/cmearm.html#pgfld-1012065

QUESTION 203

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 capabilities 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: Cisco Unity Connection

Explanation

Explanation/Reference:

QUESTION 204

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: Cisco Unity Connection

Explanation



Explanation/Reference:

Explanation:

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 205

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: Cisco Unity Connection

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).

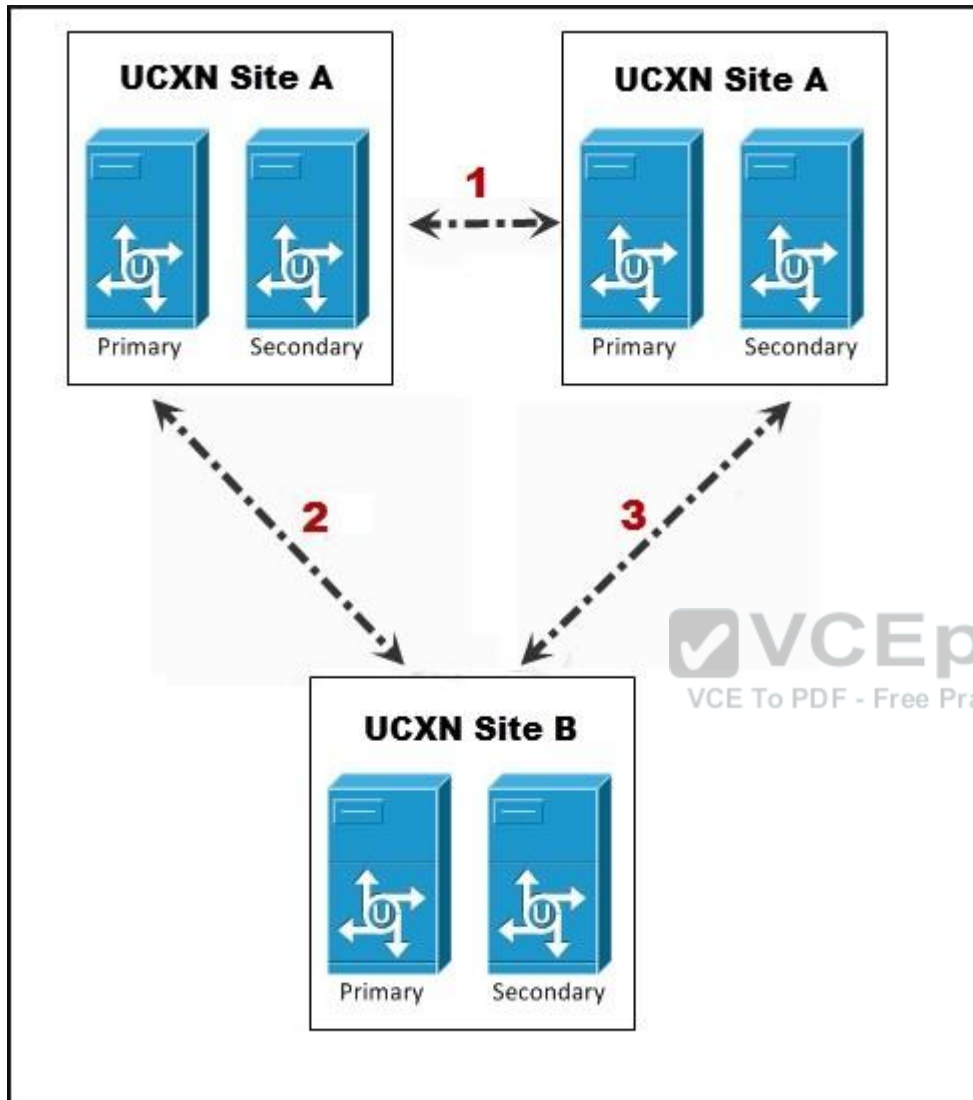
When callers search a directory handler for a particular name, if the scope of the directory handler is set to a search space, Cisco Unity Connection searches each partition in the search space and returns a list of all of the objects that match the name.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/administration/guide/8xcuc_sagx/8xcucsag235.html#pgfId-1058609

QUESTION 206

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: Cisco Unified Contact Center Express

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/8xcucnet_x/8xcucnet010.html

Topic 7, Cisco Unified Contact Center Express

QUESTION 207

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: Cisco Unified Contact Center Express

Explanation

Explanation/Reference:

Explanation: Unified CCX applications might use auxiliary files that interact with callers, such as scripts, pre-recorded prompts, grammars, and custom Java classes. Depending on each implementation, Unified CCX applications use some or all of the following file types. The 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 208

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: Cisco Unified Contact Center Express

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 209

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: Cisco Unified Contact Center Express

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 of Cisco Agent Desktop.

QUESTION 210

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: Cisco Unified Contact Center Express

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 211

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: Cisco Unified Contact Center Express

Explanation**Explanation/Reference:**

Explanation: This data store contains Unified CCX system configuration information.

QUESTION 212

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: Cisco Unified Contact Center Express

Explanation

Explanation/Reference:

Explanation: The Unified CCX Engine enables you to run multiple applications to handle Unified CM Telephony calls or HTTP requests. The Unified CCX Engine uses the Unified CM Telephony subsystem to request and receive services from the Computer Telephony Interface (CTI) manager that controls Unified 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:

Unified CCX Administration web interface: Used to configure system parameters, subsystems, view real-time reports that include total system activity and application statistics, and so on

Unified CCX Serviceability web interface: Used to view alarm and trace definitions for Unified CCX services, start and stop the Unified CCX Engine, monitor Unified CCX Engine activity, and so on

QUESTION 213

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: Cisco Unified Contact Center Express

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-network-design_chapter_0110.html

QUESTION 214

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: Cisco Unified Contact Center Express

Explanation

Explanation/Reference:

Explanation: Desktop monitoring enables Cisco UCCX agent desktop application to obtain a copy of the RTP packet.

QUESTION 215

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: Cisco Unified Contact Center Express

Explanation

Explanation/Reference:

Reference:

http://www.cisco.com/en/US/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_0_2/design/guide/UCCX_BK_C39FDB35_00_cisco-unified-contact-center-express_chapter_010.html

QUESTION 216

Which two statements describe the remote supervisory monitoring feature in ? (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: Cisco Unified Contact Center Express

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 217

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: Cisco Unified Contact Center Express

Explanation

Explanation/Reference:

Explanation: Guidelines for Agent Phone Configuration

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 218

Refer to the exhibit.



Contact Service Queue Configuration

Next
 Delete
 Cancel
 Open Printable Report of this CSQ configuration

Status

Status : Ready

Contact Service Queue Name* Customer Service
 Contact Service Queue Type* Voice
 Contact Queuing Criteria FIFO
 Automatic Work* ☐ Enabled ☒ Disabled
 Wrapup Time* ☐ Enabled Second(s) ☒ Disabled
 Resource Pool Selection Model* Resource Skills
 Service Level* 20
 Service Level Percentage* 80
 Prompt - No Selection -

Contact Service Queue Configuration

Update
 Cancel
 Show Resources
 Open Printable Report of this CSQ

Contact Service Queue Name Customer Service
 Resource Selection Criteria* Most Skilled
 Select Required Skills

Commercial Fuels
 Customer Service
 Dispatch Night
 Dispatch OH

 Add

Skills	Minimum Competence	Delete
Customer Service	3	

Resource Configuration

Update
 Cancel
 Open Printable Report of this Resource Configuration

Resource Name Source 1 OH Dispatch
 Resource ID s1dispatch-OH
 IPCC Extension 7782
 Resource Group -Not Selected-
 Automatic Available* ☒ Enabled ☐ Disabled
 Assigned Skills

Dispatch OH(5)
 Dispatch PA(7)
 Customer Service(3)

 Unassigned Skills

Commercial Fuels
 Dispatch Night
 Energy Billing
 Wholesale Fuels

 Competence Level 3 (1-Beginner, 10-Expert)
 Team Source One Dispatch

Resource Configuration

Update
 Cancel
 Open Printable Report of this Resource Configuration

Resource Name Source 1 PA Dispatch
 Resource ID s1dispatch-PA
 IPCC Extension 7783
 Resource Group -Not Selected-
 Automatic Available* ☒ Enabled ☐ Disabled
 Assigned Skills

Dispatch OH(7)
 Dispatch PA(5)
 Customer Service(4)

 Unassigned Skills

Commercial Fuels
 Dispatch Night
 Energy Billing
 Wholesale Fuels

 Competence Level 4 (1-Beginner, 10-Expert)
 Team Source One Dispatch

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: Cisco Unified Contact Center Express

Explanation

Explanation/Reference:

answer is verified

QUESTION 219

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: Cisco Unified IM and Presence

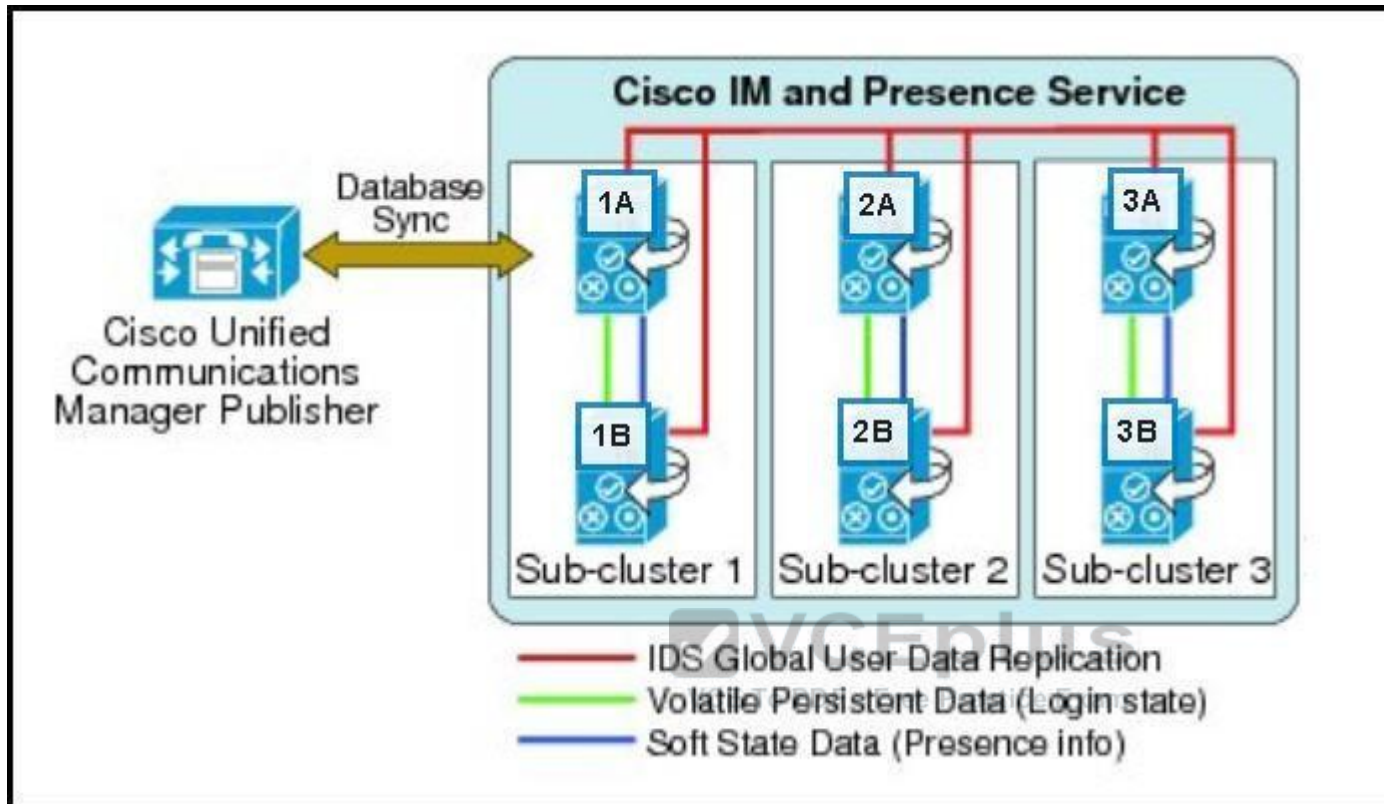
Explanation

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 CUPS to reach the CTI gateway.

QUESTION 220

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: Cisco Unified IM and Presence

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 Presence node. 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 221

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: Cisco Unified IM and Presence

Explanation

**Explanation/Reference:**

Explanation: High availability for XMPP federation differs from the high availability model for other IM 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 222

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: Cisco Unified IM and Presence

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 223

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

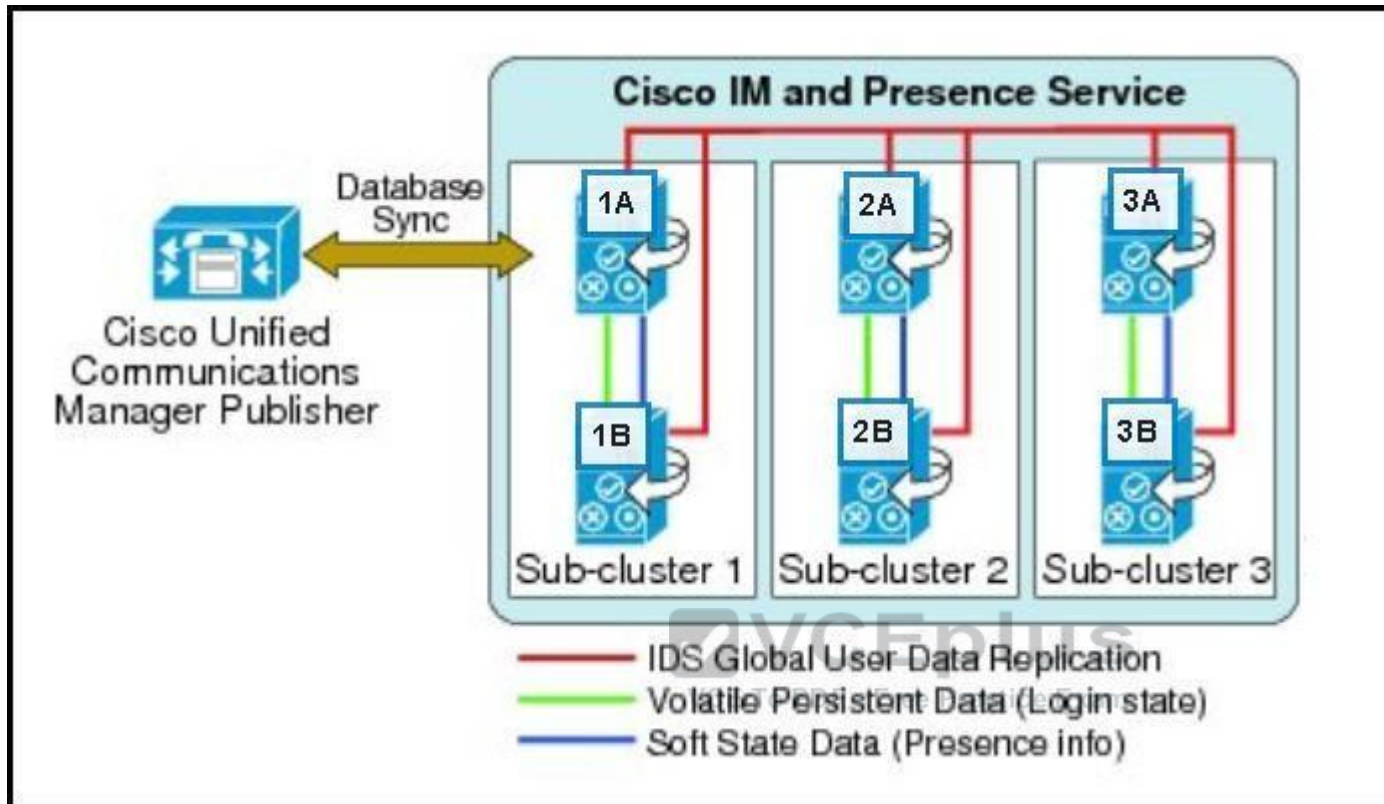
Section: Cisco Unified IM and Presence

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 224

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: Cisco Unified IM and Presence

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.

QUESTION 225

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: Cisco Unified IM and Presence

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 226

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: Cisco Unified IM and Presence

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:

For the Compliance feature, you require at least one external database per cluster. Depending on your deployment requirements, you can also configure a separate external database per node.

For the Permanent Group Chat feature, you require a unique external database per node. Configure and assign a unique external database for each node in your cluster.

If you deploy both the Permanent Group Chat and Compliance features on an IM and Presence node, you can assign the same external database to both features.

QUESTION 227

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: Cisco Unified IM and Presence

Explanation

Explanation/Reference:

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 228

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 Sync Agent
- D. Cisco XCP Connection Manager
- E. Cisco XCP Message Archiver

Correct Answer: E

Section: Cisco Unified IM and Presence

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 229

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: Cisco Unified IM and Presence

Explanation

Explanation/Reference:

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/9_1_1/secugd/CUCM_BK_C0395F44_00_cucm-security-guide-91/CUCM_BK_C0395F44_00_cucm-security-guide-91_chapter_011001.html

QUESTION 230

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: Cisco Unified IM and Presence

Explanation

Explanation/Reference:

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 231

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: Cisco Unified IM and Presence

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 232

Which three issues prevent a customer from seeing the presence status of a new contact in their Jabber contact list? (Choose three.)

- 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: Cisco Unified IM and Presence

Explanation

Explanation/Reference:

Reference: <http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-presence/97443-cups-cupc-ts.html>

QUESTION 233

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: Cisco Unified IM and Presence

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 234

Which option describes how you can show the same contacts in your Jabber for Windows on-premise 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: Cisco Unified IM and Presence

Explanation

Explanation/Reference:

Reference: 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-for-windows-97_chapter_0111.html#CJAB_CN_C2733196_00

QUESTION 235

What is the correct variable type for the "DelayWhileQueued" variable in the "icd.aef" script on Cisco Unified Contact Center Express?

- A. string
- B. user

- C. number
- D. integer
- E. Boolean

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

answer is updated

QUESTION 236

Which statement about embedded Tcl scripts for B-ACD on Cisco Unified Communications Manager Express is correct?

- A. The Tcl scripts that are required for B-ACD services, along with the default audio files, must be available on the router flash memory.
- B. The Tcl scripts and the default audio files for B-ACD services are embedded natively in the Cisco IOS Software, eliminating the requirement to download these files to the router flash memory.
- C. The Tcl scripts that are required for B-ACD services are embedded natively in the Cisco IOS Software; however, the default audio files must still be downloaded to the router flash memory.
- D. The default audio files are embedded natively in the B-ACD Tcl scripts.
- E. The Tcl scripts and the default audio files for B-ACD services must be available on a TFTP server other than the router itself

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 237

Which of these is not a mandatory configuration component to enable B-ACD service on Cisco Unified Communications Manager Express?

- A. an automated attendant Tcl script that handles the welcome prompt and menu choices
- B. a call-queue Tcl script that manages call routing and the queuing behavior number
- C. an ephone hunt group to receive calls from the call-queue service
- D. incoming dial peers for automated attendant pilot numbers
- E. Cisco Unity Express for voice mail to receive undelivered B-ACD calls

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:

QUESTION 238

On Cisco Unified Communications Manager Express with a B-ACD application that is provisioned for four hunt groups—aa-hunt1, aa-hunt2, aa-hunt3, and aa-hunt4—which hunt group will be chosen when a caller dials 0?

- A. aa-hunt0
- B. aa-hunt1
- C. aa-hunt2
- D. aa-hunt3
- E. aa-hunt4

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:



QUESTION 239

What is the maximum number of calls that are allowed in each ephone hunt group call queue that is used by Cisco Unified Communications Manager Express B-ACD?

- A. 10
- B. 15
- C. 20
- D. 25
- E. 30

Correct Answer: E

Section: (none)

Explanation

Explanation/Reference:

QUESTION 240

What is the default number of calls that are allowed in each ephone hunt group call queue that is used by Cisco Unified Communications Manager Express B-ACD?

- A. 5
- B. 10
- C. 15
- D. 20
- E. 30

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 241

What is the maximum number of ephone hunt groups that can be used with a call-queue service by Cisco Unified Communications Manager Express B-ACD?

- A. 3
- B. 5
- C. 10
- D. 15
- E. 20

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

QUESTION 242

Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 3333
  param space english index 1
  param max-time-call-retry 700
  param service-name app-b-acd
  param number-of-hunt-grps 1
  param drop-through-option 1
  param space english language en
  param handoff-string app-b-acd-aa
  param dial-by-extension-option 3
  param max-time-vm-retry 2
  param space english location flash:
  param aa-pilot 5553000
  param second-greeting-time 60
  param welcome-prompt _bacd_welcome.au
  param call-retry-timer 15
!
  service app-b-acd
  param queue-len 15
  param aa-hunt1 3000
  param number-of-hunt-grps 1
  param queue-manager-debug 1
!
dial-peer voice 30 pots
  service app-b-acd-aa
  incoming called-number 5553000
  direct-inward-dial
  port 0/0/0:23
!
```



What is the pilot number for the ephone hunt group that is configured on this Cisco Unified Communications Manager Express with B-ACD?

- A. 1
- B. 30
- C. 3000
- D. 3333

E. 555300

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:

answer is modified

QUESTION 243

Which attribute is not associable with a device profile on Cisco Unified Communications Manager?

- A. User Hold MOH Audio Source
- B. phone button template
- C. softkey template
- D. directory URL
- E. expansion module information

Correct Answer: D

Section: (none)

Explanation



Explanation/Reference:

QUESTION 244

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Reference:

http://www.cisco.com/en/US/prod/collateral/voicesw/ps6882/ps6884/white_paper_c11-583891.pdf

QUESTION 245

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: Cisco Collaboration Infrastructure

Explanation**Explanation/Reference:**

Explanation: Peer Firmware Sharing works 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 246

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: Cisco Collaboration Infrastructure

Explanation**Explanation/Reference:**

Reference:

http://docwiki.cisco.com/wiki/UC_Virtualization_Supported_Hardware#UC_on_UCS_Testing_Reference_Configurations

QUESTION 247

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: Cisco Collaboration Infrastructure

Explanation**Explanation/Reference:**

Explanation:



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

Required RAID configuration (e.g. RAID5, RAID10, etc.) - including battery backup cache or SuperCap - when the TRC uses DAS storage

Guidance on hardware installation and basic setup.

Configuration of Virtual-to-physical network interface

mapping is design-dependent and not included in TRC

definition.

Configuration of adapters (such as Cisco VIC, 3rd-party

CNA / NIC / HBA) is design-dependent and not included in

TRC definition.

Design, installation and configuration of external hardware is not included in TRC definition, such as:

Network routing and switching (e.g. routers, gateways,

MCUs, ethernet/FC/FCoE switches, Cisco Catalyst/Nexus/MDS, etc.)

QoS configuration of route/switch network devices

Cisco UCS B-Series chassis and switching components

(e.g. Cisco UCS 6100/6200, Cisco UCS 2100/2200, Cisco UCS 5100)

Storage arrays (such as those from EMC, NetApp or other

vendors)

Configuration settings, patch recommendations or step by step procedures for VMware software are not included in TRC definition.

Infrastructure solutions such as Vblock from Virtual Computing Environment may also be leveraged for configuration details not included in the TRC definition.

QUESTION 248

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: Cisco Collaboration Infrastructure

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 249

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Reference:

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

QUESTION 250

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Reference:

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

QUESTION 251

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: Cisco Unity Connection

Explanation

Explanation/Reference:

Explanation: Cisco unity here acts as a imap server for yhe 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 252

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: Cisco Unity Connection

Explanation

Explanation/Reference:

Explanation:Voice messages queue for delivery in the Voice Outbox folder that is why it shows in Deleted Items folder.

QUESTION 253

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: Cisco Unity Connection

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 254

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: Cisco Unity Connection

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 external users.

QUESTION 255

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: Cisco Unity Connection

Explanation

Explanation/Reference:

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

QUESTION 256

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: Cisco Unity Connection

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

QUESTION 257

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: Cisco Collaboration Infrastructure

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 258

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Reference:

http://www.cisco.com/en/US/docs/solutions/Enterprise/Borderless_Networks/Energy_Management/energywisedge.html?referring_site=smartnavRD#wp555927

QUESTION 259

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Explanation: Cisco Prime LMS 4.1 uses TCP port 43440.

QUESTION 260

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

Correct Answer: CE

Section: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

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

QUESTION 261

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: Cisco IOS UC Applications and Features

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 262

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

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

QUESTION 263

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 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 1111.1111.1111, 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: Cisco IOS UC Applications and Features

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 264

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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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

Explanation:

Table 5-2 Trigger Pattern Wildcards and Special Characters

Chara

cter Description Examples

X The X wildcard The trigger pattern 9XXX matches all matches any single numbers in the range 9000 through 9999. digit in the range 0 through 9.

! The exclamation mark (!) wildcard matches one or more digits in the range 0 through 9.

? The question mark The trigger pattern 91X? matches all numbers (?) wildcard in the range 91 through
matches zero or 919999999999999999999999999999.
more occurrences of
the preceding digit
or wildcard value.

+ The plus sign (+) The trigger pattern 91X+ matches all numbers wildcard matches in the range 910 through one or more 91999999999999999999999999999999.

occurrences of the preceding digit or wildcard value.

[] The square bracket The trigger pattern 813510[012345] matches ([]) characters all numbers in the range 8135100 through 8135105

8135105.
values.

- The hyphen (-) The trigger pattern 813510[0-5] matches all character, used with numbers in the range 8135100 through the square brackets, 8135105.
denotes a range of
values.

^ The circumflex (^) The trigger pattern 813510[^0-5] matches all character, used with numbers in the range 8135106
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.

QUESTION 265

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: Cisco IOS UC Applications and Features

Explanation

Explanation/Reference:

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

QUESTION 266

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: Cisco IOS UC Applications and Features

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 entry for any route to reach.

QUESTION 267

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: Cisco Unity Connection

Explanation**Explanation/Reference:**

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 268

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: Cisco Collaboration Infrastructure

Explanation**Explanation/Reference:**

Reference: <http://angryciscoguy.com/jello/cisco-answer-file-generator-to-the-rescue/>

QUESTION 269

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 rule-based 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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Reference: http://docwiki.cisco.com/wiki/Unified_Communications_VMware_Requirements

QUESTION 270

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: Cisco Collaboration Infrastructure

Explanation

Explanation/Reference:

Explanation/Reference:

http://docwiki.cisco.com/wiki/UC_Virtualization_Supported_Hardware#UC_on_UCS_Testing_Reference_Configurations