CIPT2
revised by gosh

- Rewrote most explanations with references from Cisco CIPT2 Book (Implementing Cisco Unified Communications Manager, Part 2 (CIPT2) Foundation Learning Guide CCNP Voice CIPT2 642-457)
- Fixed the drag and drop
- Fixed spelling errors and misc errors in questions
- Changed 5 answers. Each answer I changed I put why in explanation notes.
- removed 8 questions and put them in Exam B since they are unlikely to be on exam. Study exam A. Briefly study Exam B. I didn't get any questions from Exam B on the exam.
- I made very minor changes to some questions to match what you will see on the test
- No one has gotten a perfect score for this exam. I think it's because 1 or 2 questions are wrong on the real Cisco exam.

If you make any changes to this dump document exactly what you did

Exam Resources:

- http://voicetut.com/cipt2-v8-0-642-457/share-your-cipt2-v8-0-experience
- http://quickref.4shared.com/ (password:gosh)

Good luck!
Exam A

QUESTION 1
Which method can be used to address variable-length dial plans?

A. Overlap sending and receiving.
B. Add a prefix for all calls that are longer than 10-digits long
C. Use nested translation patterns to eliminate inter-digit timeout
D. Use the @macro on the route pattern
E. Use MGCP gateways, which support variable-length dial plans

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
"The three ways of detecting the end of dialing in variable-length numbering plans are as follows:
■ Interdigit timeout
■ Use of # key
■ Use of overlap sending and overlap receiving"

since only overlap sending and receiving is mentioned that has to be the answer.

"The third way to indicate end of dialing is the use of overlap send and overlap receive. If overlap is supported end to end, the digits that are dialed by the end user are sent one by one over the signaling path. Then, the receiving end system can inform the calling device after it receives enough digits to route the call (number complete). Overlap send and receive is common in some European countries, such as Germany and Austria. From a dial plan implementation perspective, overlap send and receive is difficult to implement when different PSTN calling privileges are desired. In this case, you have to collect enough digits locally (for example, in CUCM or Cisco IOS Software) to be able to decide to permit or deny the call. Only then can you start passing digits on to the PSTN one by one using overlap. For the end user, however, overlap send and receive is comfortable because each call is processed as soon as enough digits have been dialed. The number of digits that are sufficient varies per dialed PSTN number. For example, one local PSTN destination might be reachable by a seven-digit number, whereas another local number might be uniquely identified only after receiving nine digits."

CIPT2 book page 12-13
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QUESTION 2
Refer to the exhibit.
Which trunk would be most suitable for Connection Y?

A. an H.225 trunk (gatekeeper-controlled)
B. intercluster trunk (gatekeeper-controlled)
C. a SIP trunk on the U.S. cluster and an intercluster trunk on the remote cluster
D. intercluster trunk (nongatekeeper-controlled)
E. No extra connections are required. As long as IP connectivity exists, you need only configure a route pattern for each site. The Cisco Unified Communications Manager will automatically forward the calls over the WAN if the destination directory number is not registered locally.

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:
There is no gatekeeper so A and B are invalid.

A SIP trunk would have to go to the CUBE then the ITSP so you couldnt go to the remote cluster
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QUESTION 3
Which two features require or may require configuring a SIP trunk? (Choose two.)

A. SIP gateway
B. Call Control Discovery between a Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
C. Cisco Device Mobility
D. Cisco Unified Mobility
E. registering a SIP phone

Correct Answer: AB
A CUCM SIP trunk can connect to Cisco IOS gateways, a CUBE, other CUCM clusters, or a SIP implementation with network servers (such as a SIP proxy).

CIPT2 book page 60-61

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QUESTION 4
A Cisco 3825 needs to be added in Cisco Unified Communications Manager as an H.323 gateway. What should the gateway type be?

A. H.323 gateway
B. Cisco 3825
C. Cisco 3800 series router. The specific model will be selected after the Cisco 3800 is selected.
D. The gateway can be added either as an H.323 gateway or Cisco 3800 series router.
E. The gateway can be added either as an H.323 gateway or Cisco 3825 series router.

Correct Answer: A

QUESTION 5
Which statement best describes globalized call routing in Cisco Unified Communications Manager?

A. All incoming calling numbers on the phones are displayed as an E.164 with the + prefix.
B. Call routing is based on numbers represented as an E.164 with the + prefix format.
C. All called numbers sent out to the PSTN are in E.164 with the + prefix format.
D. The CSS of all phones contain partitions assigned to route patterns that are in global format.
E. All phone directory numbers are configured as an E.164 with the + prefix.

Correct Answer: B

QUESTION 6
When an incoming PSTN call arrives at an MGCP gateway, how does the calling number get normalized to a global E.164 number with the + prefix in Cisco Unified Communications Manager?

A. Normalization is done using translation patterns.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
 **QUESTION 7**
When an incoming PSTN call arrives at an MGCP gateway, how does the called number get normalized to an internal directory number in Cisco Unified Communications Manager?

A. Normalization is done by configuring the significant digits for inbound calls on the MGCP gateway.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
D. Normalization is done using the gateway incoming calling party prefixes based on number type.
E. Normalization is achieved by local route group that is assigned to the MGCP gateway.

**Correct Answer:** A

**Explanation/Reference:**
Called Number:
- Significant Digits
- Translation Patterns
- Incoming Called Party Settings (GW, DP, SP)

CIPT2 book page 106

"The called number can be normalized by significant digits that are configured at the gateway (applicable only if no calls to other external destinations are permitted and a fixed-length number plan is used), or by translation patterns, or by incoming called-party settings (if available at the ingress device). In Figure 4-15, the gateway is configured with four significant digits."

Page 107

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**QUESTION 8**
Which process can localize a global E.164 with + prefix calling numbers for inbound calls to an IP phone so that users see the calling number in a local format?

http://www.gratisexam.com/
QUESTION 9
Refer to the exhibit.

The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. To match the US-TEHO pattern \+1, how should the translation pattern be configured?

A. 9001.4085551234 with the Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: +

B. 9.0014085551234 with the Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: +1

C. 900.14085551234 with the Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: +1
D. 900.14085551234 with the Called Party Transformation:
   Discard Digits PostDot
   Prefix Digits Outgoing Calls: +
E. 001.4085551234 with the Called Party Transformation:
   Prefix Digits Outgoing Calls: +

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: ABC
The PSTN access code for the UK is 9, International call code is 001, The international escape character, +,
signifies the international access code in a complete E.164 number format Link:

QUESTION 10
Refer to the exhibit.

The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO
calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international
calls to the U.S. What should the TEHO-US route list configuration consist of?

A. First route group should point only to the U.K. gateway. The second route group should point to the U.S.
gateway.
B. First route group should be only the local route group. The second route group should point to the U.S.
gateway.
C. First route group should point only to the U.S. gateway. The second route group should be the local route
group.
D. The TEHO-US route list should contain only the local route group. The globalized configuration means that
the appropriate gateway will be selected automatically.
E. The \+! route pattern should point directly to the U.S. gateway.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

"If TEHO was configured, there would be a TEHO route pattern in E.164 format with a leading + sign. The TEHO pattern would refer to a site-specific route list in order to select the correct gateway for PSTN egress. The backup gateway would then again be selected by the local route group feature."

CIPT2 Book, page 309-310

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QUESTION 11
Refer to the exhibit.

The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. Assuming the PSTN does not accept globalized numbers with + prefix. What should the Called Party Transformation Pattern at the U.S. gateway be configured as?

A. \+.! with the following Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: +

B. \+1.! with the following Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: None
C. +408! with the following Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: 1
D. +1408! with the following Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: None
E. +1.408! with the following Called Party Transformation:
   Discard Digits PreDot
   Prefix Digits Outgoing Calls: None

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation:

QUESTION 12
Refer to the exhibit.

```
router eigrp SAF
   !
service-family ipv4 autonomous-system 1
   !
sf-interface FastEthernet0/0
   topology base
   exit-sf-topology
   exit-service-family
```

Which configuration elements must match for adjacent neighbors to establish a SAF neighbor relationship?

A. the label name specified in the router eigrp command
B. the autonomous-system number specified in the service-family ipv4 autonomous-system command
C. the sf-interface configuration
D. the topology base configurations
E. the label name specified in the router eigrp command and the autonomous-system number

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
"In this configuration, a SAF forwarder is configured with autonomous system 1. All SAF forwarders that will exchange information with each other have to be in the same autonomous system."

CIPT2 book page 366
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QUESTION 13
Which statement about Service Advertisement Framework is true?

A. SAF requires that the EIGRP be configured on all routers, including non-SAF routers.
B. SAF requires that the EIGRP be configured only on SAF routers. Non-SAF routers act as an IP cloud.
C. SAF has no dependency on the underlying routing protocol, as long as it is a dynamic routing protocol. Static routes are not supported.

D. SAF operates on any dynamic or static IP routing configuration. SAF is totally independent of the underlying routing protocol.

Correct Answer: D

Section: (none)
Explanation

Explanation/Reference:

Because Cisco SAF is independent of IP routing and uses underlying Cisco routing technology to distribute service advertisements in a reliable and efficient manner, Cisco SAF will run in networks over any routing protocol they may have in place such as Enhanced Interior Gateway Routing Protocol (EIGRP), Open Shortest Path First (OSPF), Exterior Border Gateway Protocol (EBGP) over an MPLS service, or static routing (Figure 2).


QUESTION 14
Assume that the Cisco IOS SAF Forwarder is configured correctly. Which minimum configurations on Cisco Unified Communications Manager are needed for the SAF registration to take place?

A. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Advertising Service
B. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Requesting Service
C. SAF Trunk, SAF Security Profile, SAF Forwarder, CCD Requesting Service, and CCD Advertising Service
D. SAF Trunk, SAF Security Profile, and SAF Forwarder
E. SAF Trunk, CCD Requesting Service, and CCD Advertising Service

Correct Answer: B

Section: (none)
Explanation

Explanation/Reference:

Explanation:

QUESTION 15
Refer to the exhibit.
What should the destination IP address be configured as on the HQ and BR1 SIP trunks?

A. The HQ SIP trunk destination IP address should be 10.1.6.10. The BR1 SIP trunk destination IP address should be 10.1.5.10.
B. The destination IP address is not configurable. Each cluster will learn about the remote trunk IP address through SAF learned routes.
C. The destination IP address will be learned automatically and configured on the SIP trunks after the Cisco Unified Communications Managers discover themselves.
D. The HQ SIP trunk destination IP address should be the HQ SAF Forwarder IP address. The BR1 SIP trunk destination IP address should be the BR1 SAF Forwarder IP address.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
CIPT2 book page 353

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QUESTION 16
When a SIP trunk is added for Call Control Discovery, which statement is true?

A. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Enable SAF check box should be selected.
B. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Trunk Service Type should be Call Control Discovery.
C. The SIP trunk is added by selecting Call Control Discovery Trunk and then selecting SIP as the protocol to be used.
D. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The destination IP address field is
QUESTION 17
When an H.323 trunk is added for Call Control Discovery, which statement is true?

A. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Enable SAF check box should be selected in the trunk configuration.
B. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Trunk Service Type should be Call Control Discovery.
C. The H.323 trunk is added by selecting Call Control Discovery Trunk and then selecting H.323 as the protocol to be used.
D. The H.323 trunk is added by selecting H.323 Trunk, and selecting Inter-Cluster Trunk as the Device Protocol. The destination IP address field is configured as `SAF` to indicate that this trunk is used for SAF.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
"You can configure one SAF-enabled SIP trunk, as shown in Figure 12-18, or one SAF-enabled H.323 trunk, but you need to create only one of each trunk type. With a SAF-enabled H.323 trunk, you have to first add a standard nongatekeeper-controlled ICT and check the Enable SAF checkbox. After the check box is checked, the IP address field is disabled. The reason is that the configured trunk does not refer to a particular destination IP address but instead acts as a template for a dynamically created trunk once a SAF call is placed. The destination IP address is then taken from the learned SAF service data."

CIPT2 book page 369
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QUESTION 18
Refer to the exhibit.
What must be configured on the HQ Cisco Unified Communications Manager to allow HQ users to dial the SAF learned directory number pattern 3XXX?

A. Route pattern 3XXX should be configured and made available to HQ users through the phone CSS.
B. Route pattern 3XXX should be configured and made available to HQ phone users through the phone AAR CSS.
C. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone CSS.
D. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone AAR CSS.
E. The SAF directory number pattern 3XXX will be made available to all users automatically as soon as the SAF partition is selected.

Correct Answer: C

Section: (none)

Explanation/Reference:
In CUCM select Call Routing > Call Control Discovery > Requesting Service. Select "Route Partition"

From CUCM Help:

"From the drop-down list box, choose the partition where you want the learned patterns to belong. The Route Partition field only supports the call control discovery feature; that is, all learned patterns automatically belong to the partition that you choose. This route partition gets used exclusively by the CCD requesting service to ensure that all learned patterns get placed in digit analysis under the route partition.

If you choose a partition besides None from the drop-down list box, the partition that you
choose must belong to a calling search space that the devices can use for calling the learned patterns. In this case, if you do not assign a calling search space that contains the partition to the device, the device cannot call the learned patterns. **Tip:** Cisco strongly recommends that you configure a unique partition and assign it to the CCD requesting service. If you choose None from the Route Partition drop-down list box, all devices can call the learned patterns. **Tip:** Updating the Learned Pattern Prefix field or Route Partition field may impact system performance because the digit-analysis master routing table automatically gets updated when these fields are changed. To avoid system performance issues, Cisco recommends that you update these fields during off-peak hours.

CIPT2 book page 372
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**QUESTION 19**
Refer to the exhibit.

```plaintext
dial-peer hunt 2
voice service saf
    profile trunk-route 1
        session protocol sip interface Loopback1 transport tcp port 5060
    profile dn-block 1 alias-prefix 1972555
        pattern 1 type extension 4xxx
    profile dn-block 2
        pattern 1 type global 14087071222
    profile callcontrol 1
        dn-service
            trunk-route 1
            dn-block 1
            dn-block 2
    channel 1 vrouter SAF asystem 1
        subscribe callcontrol wildcarded
        publish callcontrol 1
```
QUESTION 20
Refer to the exhibit.

```plaintext
dial-peer hunt 2
voice service saf
    profile trunk-route 1
        session protocol sip interface Loopback1 transport tcp port 5060
    profile dn-block 1 alias-prefix 1972555
        pattern 1 type extension 4XXX
    profile dn-block 2
        pattern 1 type global 14087071222
    profile callcontrol 1
        dn-service
            trunk-route 1
                dn-block 1
                dn-block 2
        channel 1 vrouter SAF asystem 1
        subscribe callcontrol wildcarded
        publish callcontrol 1
```

How does the Cisco Unified Communications Manager advertise dn-block 1?

A. 4XXX and the ToDID will 0:
B. 4XXX and the ToDID will 0:1972555
C. 4XXX
D. 4XXX and the ToDID will 0:+1972555
E. 19725554XXX

Correct Answer: B

QUESTION 21
Refer to the exhibit.
How does the Cisco Unified Communications Manager advertise dn-block 2?

A. 14087071222 with number type international
B. +14087071222 with number type international
C. +14087071222
D. 14087071222

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:
"Neither the ToDID-prefix argument nor the pattern argument support the use of the +
sign. If you want to advertise a number with a + sign, you must use the command pattern
tag type global pattern. Again, you cannot enter the + sign in the pattern argument;
however, because of the type global, a + sign is prefixed to the configured pattern. The
ToDID of global patterns is always unset."

CIPT2 book page 377

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QUESTION 22

Refer to the exhibit.
Which statement about the configuration between the Default and BR regions is true?

A. Calls between the two regions can use either 64 kbps or 8 kbps.
B. Calls between the two regions can use only the G.729 codec.
C. Only 64 kbps will be used between the two regions because the link is "lossy".
D. Both codecs can be used depending on the loss statistics of the link. When lossy conditions are high, the G.711 codec will be used.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
The default region shows it is configured to use G.729 to BR, therefore the answer is "call between the two regions can only use G.729"

"When a call is placed between two devices, the codec is determined based on the regions of the two devices and on the devices’ capabilities. The devices use the codec that is best supported by both devices and that does not exceed the bandwidth requirements of the codec permitted for the region or regions that are involved in the call. If the two devices cannot agree on a codec when a region configuration allows a G.729 codec but a device on the other end supports only G.711 or G.722, a transcoder is invoked if available. The loss type of a link can also be configured. On links that are configured to be lossy, codecs that are less sensitive to packet loss are preferred over codecs that result in higher quality degradation."

CIPT2 Book, page 203 under section Review of CUCM Codecs

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QUESTION 23
Refer to the exhibit.
When a call between two HQ users is being conferenced with a remote user at BR, which configuration is needed?

A. The BR_MRG must contain the transcoder device. The BR_MRGL must be assigned to the BR phones.
B. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the HQ phones.
C. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.
D. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the software conference bridge.
E. Enable the software conference bridge to support the G.711 and G.729 codecs in Cisco Unified Communications Manager Service Parameters.

Correct Answer: B
Section: (none)
Explanation/Reference:
HQ_MRG must contain the transcoder device because HQ users are initiating the conference. You cannot assign a MRGL to a software conference bridge, you can only assign a MRGL to users.

Original answer was D. Changed to B: The HQ_MRG must contain the transcoder device. The MQ_MRGL must be assigned to the HQ phones.

Page 207 of CIPT2 book provides exactly this scenario and says:
"HQ_HW-MRG is the first entry of the media resource group list called HQ_MRGL, and HQ_SW-MRG is the next entry. **Main site phones are configured with HQ_MRGL.** Because media resource groups are used in a prioritized way, main site phones that invoke a conference first use the available hardware conference resources. When all of them are
in use, the software conference resources are accessed. Software conference resources run on a CUCM server and use the server CPU to mix the calls. Note that the remote site does not have a software conference bridge because it does not have CUCM servers.

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**QUESTION 24**
Refer to the exhibit.

![Diagram](image.png)

All HQ phones are configured to use HQ_MRGL and all BR phones are configured to use BR_MRGL. For the HQ phones always to use the hardware conference bridge as a first choice, which configuration should be implemented?

A. Ensure that both the hardware and software conference bridges are listed in the HQ_MRGL. Ensure that the instance ID for the hardware conference bridge is 0.

B. Ensure that both the hardware and software conference bridges are listed in the HQ_MRGL. The hardware conference bridge must be configured first.

C. Assign the hardware conference bridge to HQ_MRGL. Configure a second HQ_MRGL_2 and assign the software conference bridge to it. Add both the HQ_MRGL and HQ_MRGL_2 to the HQ_MRGL and list the HQ_MRGL first.

D. Assign the hardware conference bridge to HQ_MRGL. Configure a second HQ_MRGL_2 and assign the software conference bridge to it. Configure an additional HQ_MRGL_2. Add the HQ_MRGL to HQ_MRGL_2. Add HQ_MRGL_2 to HQ_MRGL_2. The HQ_MRGL should be assigned to the HQ phones. The HQ_MRGL_2 should be assigned to the HQ device pool.

**Correct Answer:** C

**Section:** (none)

**Explanation**

**Explanation/Reference:**
"HQ_HW-MRG is the first entry of the media resource group list called HQ_MRGL, and HQ_SW-MRG is the next entry. Main site phones are configured with HQ_MRGL. Because media resource groups are used in a prioritized way, main site phones that invoke a conference first use the available hardware conference resources. When all of them are in use, the software conference resources are accessed. Software conference resources run on a CUCM server and use the server CPU to mix the calls. Note that the remote site does not have a software conference bridge because it does not have CUCM servers."

CIPT2 book page 207

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QUESTION 25
Refer to the exhibit.

<table>
<thead>
<tr>
<th>Location Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name: BR</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Audio Calls Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio Bandwidth: 96 kbps</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Video Calls Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video Bandwidth:</td>
</tr>
</tbody>
</table>

Assuming the regions configuration to BR only permits G.729 codec, how many calls are allowed for the BR location?

A. Total of four calls; two incoming and two outgoing.
B. Total of two calls in either direction.
C. Total of four calls to the BR location. Outgoing calls are not impacted by the location configuration.
D. Total of four calls in either direction.
E. Two outgoing calls. Incoming calls are unlimited.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Locations are not aware of the topology, they are logical only. 24kbps * 4 = 96 kbps

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QUESTION 26
Refer to the exhibit.
How many calls are permitted by the RSVP configuration?

A. one G.711 call
B. two G.729 calls
C. one G.729 call and one G.711 call
D. eight G.729 calls
E. four G.729 calls

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Locations use the following to calculate bandwidth:

- G.711 call uses 80 kb/s.
- G.722 call uses 80 kb/s.
- G.723 call uses 24 kb/s.
- G.728 call uses 26.66 kb/s.
- G.729 call uses 24 kb/s.

CIPT2 book page 252:

Note The bandwidth reserved for a call depends on the codec that is used. As with standard non-RSVP-enabled CUCM locations, it is 80 kbps for G.711 and 24 kbps for G.729. During call setup, however, the RSVP agent always requests an additional 16 kbps, which is released immediately after the RSVP reservation is successful. Therefore, the interface bandwidth has to be configured in such a way that it can accommodate the desired number of calls, considering the codec that will be used plus the extra 16 kbps. For example, if two G.729 calls are permitted on the interface, 64 kbps must be configured. For two G.711 calls, 176 kbps is required. In Example 9-1, only one G.729 call is permitted.

QUESTION 27
Refer to the exhibit.
To permit three G.729 calls, what should the bandwidth value be for the `ip rsvp bandwidth` command?

A. 32  
B. 48  
C. 64  
D. 88  
E. 128

**Correct Answer:** D  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**  
G.729 call = 24 kbps  
Add 16 kbps for overhead  
3 * 24 = 72 + 16 = 88 kbps

---

**QUESTION 28**  
How is a SIP trunk in Cisco Unified Communications Manager configured for SIP precondition?

A. The configuration is done by selecting a SIP precondition trunk for trunk type.  
B. The configuration is automatically selected when RSVP is enabled for the location assigned to the trunk.  
C. SIP precondition is configured by selecting E2E for RSVP over SIP on the default SIP profile assigned to the SIP trunk.  
D. SIP precondition is configured by configuring a new SIP profile and selecting E2E for RSVP over SIP. The new SIP profile must then be assigned to the SIP trunk.

**Correct Answer:** D  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**  
Explanation:
QUESTION 29
Which statement about SIP precondition is most correct?

A. When configuring SIP precondition, the SIP trunk must have access to an RSVP agent.
B. When configuring SIP precondition, the IP phones must have access to an RSVP agent.
C. When configuring SIP precondition, the IP phones and SIP trunk must have access to an RSVP agent.
D. RSVP agents are only required for the IP phones. SIP trunks require RSVP agents only when fall back to local RSVP is configured.
E. SIP trunk will always require RSVP agents regardless of what RSVP type is configured.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
"The RSVP agent that is associated with the IP Phone is used for the call leg to the far-end SIP device. If QoS fallback is not enabled, the SIP trunk will never allocate an RSVP agent. If QoS fallback mode is enabled, two local RSVP agents are required in a fallback scenario: one for the IP Phone and one for the SIP trunk. Therefore, the MRGL at the SIP trunk is only required for QoS fallback mode or for when the SIP trunk is not configured for SIP Preconditions at all but is configured to use local QoS."

CIPT2 book page 271

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QUESTION 30
Refer to the exhibit.

The HQ Cisco Unified Communications Manager has been configured for end-to-end RSVP. The BR Cisco Unified Communications Manager has been configured for local RSVP.

RSVP between the locations assigned to the IP phones and SIP trunks at each site are configured with mandatory RSVP. When a call is placed from the IP phone at HQ to the BR phone at the BR site, which statement is true?

A. The Cisco Unified Communications Manager at HQ will fall back to local RSVP and place the call. No RSVP end-to-end will occur.
B. RSVP end-to-end will occur.
C. The Cisco Unified Communications Manager at HQ will use end-to-end RSVP. The BR Cisco Unified Communications Manager will use local RSVP.
D. The call will fail.
E. The call will proceed as a normal call with no RSVP reservation.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
- No Reservation: No RSVP reservations get made between any two locations.
- Optional (Video Desired): A call can proceed as a best-effort, audio-only call if failure to obtain reservations for both audio and video streams occurs. The RSVP agent continues to attempt an RSVP reservation for audio and informs CUCM if the reservation succeeds.
- Mandatory: CUCM does not ring the terminating device until RSVP reservation succeeds for the audio stream and, if the call is a video call, for the video stream.
- Mandatory (Video Desired): A video call can proceed as an audio-only call if a reservation for the audio stream succeeds but the reservation for the video stream does not succeed.

Because the RSVP is set to mandatory the call will fail.

CIPT2 book page 247

gosh

QUESTION 31
Which statement about H.323 Gatekeeper Call Admission Control is true?

A. Gatekeeper Call Admission Control applies to centralized Cisco Unified Communications deployments that use locations based Call Admission Control.
B. Gatekeeper Call Admission Control applies to distributed Cisco Unified Communications deployments.
C. Gatekeeper Call Admission Control applies only to distributed Cisco Unified Communications Express deployments.
D. Gatekeeper Call Admission Control setting is configured in Cisco Unified Communications Manager.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Gatekeeper CAC applies to distributed environments. The gatekeeper gets the bandwidth configuration from IOS not from call manager locations, therefore option A is invalid.

gosh

QUESTION 32
Refer to the exhibit.
How many calls can be placed to Cluster B?

A. three G.729 calls  
B. one G.711 call  
C. one G.711 and three G.729 calls  
D. There is no limit for incoming calls to Cluster B. Outgoing calls are limited to one G.711 and three G.729 calls.

Correct Answer: A  
Section: (none)  
Explanation

Explanation/Reference:  
Explanation:

key part:  bandwidth interzone zone ClusterB 48

CUCM locations limit g.729 calls to 24k, but cisco IOS limits g.729 calls bandwidth to 16k.  16*3=48

CIPT2 Book page 278-279

gosh

QUESTION 33  
Refer to the exhibit.
The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number. Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

What should the AAR group prefix be?

A. 9  
B. 91  
C. none  
D. +  
E. +1

Correct Answer: C  
Section: (none)  
Explanation

Explanation/Reference:  
"If the AAR destination mask is entered in the globalized form, and if every AAR CSS is able to route calls to destinations in the globalized form, system administrators can forego the configuration of AAR groups, because their sole function is to determine which digits to prefix based on the local requirements of the PSTN access of the calling phone to reach the specific destination."

Since globalized call routing is used aar groups are not used.

CIPT2 book page 258
QUESTION 34
Refer to the exhibit.

The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number.

Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

Which partition should be configured in the AAR CSS applied at the phones?

A. PSTN partition
B. LD partition
C. The HQ AAR CSS must include a partition assigned to route pattern 91408XXXXXXX. The BR1 AAR CSS must include a partition assigned to route pattern 91650XXXXXXX.
D. AAR CSS must contain translation pattern 9.1[2-9][X][2-9][XXX for each site that must be globalized. Otherwise the called numbers will not be localized at the egress gateway.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
see graph in CIPT2 book page 258. Only a single route partition with \+! is configured with globalization.
Refer to the exhibit.

The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number.

Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

How many route lists and route groups should be configured for AAR at a minimum?

A. a single route list with a local route group for each site
B. two route lists and two route groups for each site
C. a single route list and four route groups for each site
D. None. The AAR CSS can point directly to the route pattern.

Correct Answer: A

Explanation/Reference:

QUESTION 36
Refer to the exhibit.
The remote site needs to run multicast MOH from flash. Which statement about the MOH server configuration in Cisco Unified Communications Manager is true?

A. The MOH server must be enabled for G.729 in the Cisco IP Voice Media Streaming Application service parameters.
B. Multicast MOH can use only G.711. So you must configure the command codec G711ulaw under the call-manager-fallback configuration at the remote site router.
C. The MOH for the remote site is a standalone configuration. No extra configuration is required on the Cisco Unified Communications Manager MOH server.
D. Configure a separate region for the MOH server. The codec between the MOH region and all other regions should be specified as G.711. Apply the MOH region through a device pool at the MOH server configuration page.
E. Configure the location setting for the MOH server to 80 Kbps. This configuration forces the MOH server to use G.711 for the remote site.

Correct Answer: D

Explanation

**Explanation/Reference:**
Explanation:

**QUESTION 37**
Assume that local route groups are configured. When an IP phone moves from one device mobility group to another, which two configuration components are not changed? (Choose two.)

A. IP subnet
B. user settings
C. SRST reference
D. region
E. phone button settings

Correct Answer: BE
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: ACD
Although the phone may have moved from one subnet to another, the physical location and associated services
have not changed.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_1_2/ccmfeat/fsdevmob.html#wp 1137460

QUESTION 38
Where do you specify the device mobility group and physical location after they have been configured?

A. phones
B. DMI
C. device mobility CSS
D. device pool
E. MRGL
F. locale

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: ABCE
Before you configure a device pool, you must configure the following items if you want to choose them for the
device pool, Cisco Unified Communications Manager group (required), Date/time group (required), Region
(required), SRST reference (optional), Media resource group list (optional), Calling search space for auto-
registration (optional), Reverted call focus priority (optional), Device mobility group (optional), Device mobility
calling search space, Physical location (optional), Location, AAR group, AAR calling search space.

QUESTION 39
Which two options for a Device Mobility-enabled IP phone are true? (Choose two.)

A. The phone configuration is not modified.
B. The roaming-sensitive parameters of the current (that is, the roaming) device pool are applied.
C. The user-specific settings determine which location-specific settings are downloaded from the Cisco Unified
Communications Manager device pool.
D. If the DMGs are the same, the Device Mobility-related settings are also applied.

Correct Answer: BD
Section: (none)
Explanation

Explanation/Reference:
The current device pool is chosen as follows:
■ If the DMI is associated with the phone’s main or home device pool, the phone
is considered to be in its home location. Therefore, Device Mobility does not
reconfigure the phone.
If the DMI is associated with one or more device pools other than the phone’s main or home device pool, one of the associated device pools is chosen based on a round-robin load-sharing algorithm.

If the current device pool is different from the home device pool, the following checks are performed:
- If the physical locations are not different, the phone’s configuration is not modified.
- If the physical locations are different, the roaming-sensitive parameters of the current roaming device pool are applied.
- If the Device Mobility Groups are the same, in addition to different physical locations, the Device Mobility-related settings are also applied, along with the roaming-sensitive parameters.

CIPT2 book page 297

gosh

QUESTION 40
What impact do roaming-sensitive settings and Device Mobility settings have on call routing?

A. Device Mobility settings have no impact on call routing, but roaming-sensitive settings modify the AAR group, AAR CSS, and device CSS.
B. Device Mobility settings modify the device CSS and the roaming-sensitive settings modify the AAR group and AAR CSS.
C. Device Mobility settings modify the AAR group and the AAR CSS, the roaming-sensitive settings modify the device CSS.
D. Roaming-sensitive settings are settings that do not have an impact on call routing. Device Mobility settings, on the other hand, may have an impact on call routing because they modify the device CSS, AAR group, and AAR CSS.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Roaming-Sensitive Settings:
- Date/Time Group
- Region
- Location
- Connection Monitor Duration
- Network Locale
- SRST Reference
- Media Resource Group List (MRGL)
- Physical Location
- Device Mobility Group
- Local Route Group

Device Mobility-Related Settings:
- Device Mobility CSS
- AAR CSS
- AAR Group
- Calling Party Transformation CSS

"Roaming-sensitive settings are settings that do not have an impact on call routing. Device Mobility-related settings, on the other hand, have a direct impact on call routing, because they modify the device CSS, AAR group, and AAR CSS. Depending on the implementation of Device Mobility, roaming-sensitive settings only, or both roaming-sensitive settings and Device Mobility-related settings, can be applied to a roaming phone."
QUESTION 41
In what Cisco solution is Simple Network-Enabled Auto Provision technology used?

A. Cisco Unified Gateway Duplication
B. Cisco Unified CallManager Redundancy
C. Cisco Unified SRST
D. Cisco Unified Call Survivability

Correct Answer: C

Explanation/Reference:
Explanation: Incorrect answer: ABD
When the system automatically detects a failure, Cisco Unified SRST uses Simple Network Auto Provisioning (SNAP) technology to auto-configure a branch office router to provide call processing for the Cisco Unified IP phones that are registered with the router Link:

QUESTION 42
While configuring Call Survivability in Cisco Unified Communications Manager, what step is mandatory to reach remote sites while in SRST mode?

A. Enable Cisco Remote Site Reachability.
B. Configure CFUR.
C. Enable the SRST checkbox in the MGCP gateway.
D. Configure the H.323 gateway for SRST in Cisco Unified Communications Manager.
E. Enable the Failover Service parameter.

Correct Answer: B

Explanation/Reference:
"When the DN is unregistered, calls can be rerouted to the voice mail that is associated with the extension or to a DN that is used to reach the phone through the PSTN. The latter approach is preferable when a phone is located within a site whose WAN link is down. If the site is equipped with SRST, the phone (and its co-located PSTN gateway) reregisters with the co-located SRST router. The phone then can receive calls placed to its PSTN direct inward dialing (DID) number."

CIPT2 book page 161

gosh

QUESTION 43
While operating in SRST, what is needed to route calls outside of the remote site location to the PSTN?

A. SIP trunk
B. CallManager route patterns
C. translation patterns
D. POTS dial peers
E. VOIP dial peers
Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:
Since there is no communication with CUCM there must be dial peers on the voice gateway to route calls

gosh

QUESTION 44
When using Cisco Unified Communications Manager Express in SRST mode, how many multicast music on hold streams can be utilized by the system at any given time?

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

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
"Before Cisco Unified SRST v8.x, only a single MOH file was supported by Cisco Unified SRST, CUCM Express in SRST mode, and CUCM Express in standalone mode. Cisco Unified SRST v8.x allows you to configure up to five additional MOH sources by configuring MOH groups. Only SCCP IP Phones support these newly introduced MOH groups. You can configure each MOH group with an individual MOH file that is located in the flash memory of the router, and you can enable multicast MOH for each MOH group. Each MOH group is configured with the DN ranges that should use the corresponding MOH group when callers are put on hold, as described in Chapter 7, “Implementing Cisco Unified Communications Manager Express (CUCME) in SRST Mode.”

I think the question is asking how many MOH Files can be used, not just streams, because more than 6 streams can exist at a time.

CIPT2 book page 139
gosh

QUESTION 45
Refer to the exhibit.
What music on hold audio source will be heard if a user at extension 1372 places the user at extension 3041 on hold?

A. moh2.au  
B. moh1.wav  
C. default.wav  
D. moh2.wav  
E. moh1.au

**Correct Answer: E**  
**Section: (none)**  
**Explanation**

**Explanation/Reference:**  
**Explanation:**  
Extension 1372 is in voice moh-group 1 and its moh that it uses to place people on hold is customer services moh moh1.au

**QUESTION 46**  
To preserve analog calls in an MGCP switchback event, which three commands must be configured in the MGCP fallback router? (Choose three.)

A. h323  
B. mgcp-switchback-graceful  
C. voice service voip  
D. mgcp-graceful  
E. preserve-h323  
F. no h225 timeout keepalive

**Correct Answer: ACF**  
**Section: (none)**  
**Explanation**

**Explanation/Reference:**  
A fallback is when connection is lost to CUCM and the gateways fallback to MGCP mode

Page 126:

“MGCP gateway fallback preserves active calls from remote-site IP Phones to the PSTN when analog or CAS protocols are used. For ISDN protocols, call preservation is impossible
because Layer 3 of the ISDN stack is disconnected from the MGCP call agent and is restarted on the local Cisco IOS gateway. This means that for active ISDN calls, all callstate information is lost in cases of switchover to fallback operation."

A switchback is when connection is restore to CUCM:

page 134:

"The MGCP-gateway-fallback feature provides the rehome functionality to switch back to MGCP mode. As shown in Figure 5-9, the switchback or rehome mechanism is triggered by the reestablishment of the TCP connection between CUCM and the Cisco MGCP gateway. Rehome function in gateway-fallback mode detects the restoration of a WAN TCP connection to any CUCM server. When the fallback mode is in effect, the affected MGCP gateway repeatedly tries to open a TCP connection to a CUCM server that is included in the prioritized list of call agents. This process continues until a CUCM server in the prioritized list responds. The TCP open request from the MGCP gateway is honored, and the gateway reverts to MGCP mode. The gateway sends a RestartInProgress (RSIP) message to begin registration with the responding CUCM."

The commands needed are:

page 159:

RemoteSite# configure terminal
RemoteSite(config)# voice service voip
RemoteSite(conf-voi-serv)# h323
RemoteSite(conf-serv-h323)# call preserve

Note: the correct answer should include call preserve, however this isn't an option on the test. The test answers are wrong. Therefore study the answers as marked, but also look out for "call preserve" as an option. Cisco may eventually fix this question on the test.

gosh

QUESTION 47
Which two locations are the best locations that an end user can use to determine if an IP phone is working in SRST mode? (Choose two.)

A. Cisco Unified Communications Manager Administration
B. IP phone display
C. Cisco Unified SRST Router
D. Cisco Unified MGCP Fallback Router
E. physical IP phone settings

Correct Answer: BE

Explanation/Reference:
Explanation: Incorrect answer: ACD
IP Phone display and Physical phone IP settings are two locations were an end user can determine if an IP phone is working in SRST mode.

QUESTION 48
What is the fastest way for an engineer to test the implementation of SRST in a production environment?

A. Shut down the Cisco Unified Communications Manager Servers.
B. Shut down the switch ports connected to the Cisco Unified Communications Manager Servers.
C. Add a null route to the publisher Cisco Unified Communications Manager at the remote router. Remove the null route when the operation is verified.
D. Unplug the IP phones from their switch ports.
E. Verification is not needed.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
All the other options would result in dropped calls. The null route would only affect your testing.

gosh

QUESTION 49
When Cisco Extension Mobility is implemented, which CSS is used for calling privileges?

A. The user device profile line CSS combined with the device CSS of the physical phone used to log in the extension mobility user.
B. The user device profile device CSS combined with the line CSS of the physical phone used to log in the extension mobility user.
C. Only the user device profile device CSS is used.
D. The combined line/device CSS of the physical phone is used to log in the extension mobility user.
E. The combined line/device CSS of the user device profile.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Cisco Extension Mobility and CSSs

Cisco Extension Mobility does not modify the device CSS or the Automated Alternate Routing (AAR) CSS (both of which are configured at the device level). Cisco Extension Mobility replaces the line CSS/CSSs that are configured at the phone with the line CSS/CSSs that are configured at the device profile of the logged-in user.

Thus, in an implementation that uses the line/device approach, the following applies:

- The line CSS of the login device is updated with the line CSS of the user. This update is used to enforce the same class of service (CoS) settings for the user, independent of the physical device to which the user is logged in.
- The device CSS of the login device is not updated, and the same gateways (those that were initially configured at the phone before the user logged in) are used for external route patterns. Because the phone did not physically move, the same local gateways should be used for PSTN calls, even when a different user is currently logged into the device.

If the traditional approach is used to implement partitions and CSS, the following applies:

- If only device CSSs are used, the CSS is not updated, and no user-specific privileges can be applied. The user inherits the privileges that are configured at the device that is used for logging in.
- If only line CSSs are used, the line CSS that is configured at the device profile of the user replaces the line CSS of the login device. In a multisite environment, this configuration can cause problems in terms of gateway choice because the same gateway is always used for external calls. To avoid gateway selection problems in such an environment, you should use local route groups.
QUESTION 50
When multiple Cisco Extension Mobility profiles exist, which actions take place when a user tries to log in to Cisco Extension Mobility?

A. The login will fail because only a single Cisco Extension Mobility profile is allowed.
B. The user must select the desired profile.
C. The user must login to both profiles in the order they are presented.
D. The user may login to both profiles in any order.
E. Login will only be allowed to multiple profiles if the service parameter Allow Multiple Logins is enabled.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
CUCM Extension Mobility allows users to log in to any phone and get their individual user-specific phone configuration applied to that phone. Thus, users can be reached at their personal directory number, regardless of their location or the physical phone they are using. Extension Mobility is implemented as a phone service and works by default within a single CUCM cluster. With CUCM v8.x or later, Extension Mobility Cross Cluster (EMCC) can be enabled. EMCC is not covered in this book.

The user-specific configuration is stored in device profiles. After successful login, the phone is reconfigured with user-specific parameters, and other device-specific parameters remain the same. If a user is associated with multiple device profiles, he must choose the device profile to be used.

If a user logs in with a user ID that is still logged in at another device, one of the following options can be configured:

- Allow multiple logins: When this method is configured, the user profile is applied to the phone where the user is logging in, and the same configuration remains active at the device where the user logged in before. The line number or numbers become shared lines because they are active on multiple devices.
- Deny login: In this case, the user gets an error message. Login is successful only after the user logs out at the other device where she logged in before.
- Auto-logout: Like the preceding option, this option ensures that a user can be logged in on only one device at a time. However, it allows the new login by automatically logging out the user at the other device.

On a phone configured for Extension Mobility, either another device profile that is a logout device profile can be applied, or the parameters as configured at the phone are applied. The logout itself can be triggered by the user or enforced by the system after expiration of a maximum login time.

QUESTION 51
When Cisco Extension Mobility is implemented, how is the audio source for the MOH selected?

A. The audio source that is configured at the user device profile is selected.
B. The audio source that is configured at the home phone of the user is selected.
C. The audio source that is configured at the physical phone used for the Cisco Extension Mobility login is selected.
D. The audio source that is configured in the IP Voice Media Streaming parameters is selected.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
"The device profile is configured with all user-specific settings that are found at the device level of an IP Phone, such as user MOH audio source, phone button templates, softkey templates, user locales, DND and privacy settings, phone service subscriptions, and all phone buttons, including lines and speed dials. One or more device profiles are applied to an end user in the End User Configuration window."

CIPT2 book page 321

gosh

QUESTION 52
Refer to the exhibit.

Assume a centralized Cisco Unified Communications deployment with the headquarters in the U.K. and remote site in RTP. All route patterns are assigned a route list that points to the local route group. Local route groups have been configured on the U.K and RTP device pools. A U.K. user logs onto an RTP phone using the Cisco Extension Mobility feature and places an emergency call to 0000. Which statement about the emergency call is true?

A. The call will match the U.K_Emergency route pattern partition and will egress at the RTP gateway.
B. The call will match the U.K_Emergency route pattern partition and will egress at the U.K. gateway.
C. The call will match the RTP_Emergency route pattern partition and will egress at the U.K.gateway.
D. The call will match the RTP_Emergency route pattern partition and will egress at the RTP gateway.
E. The call will fail.
Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
"The device CSS of the login device is not updated, and the same gateways (those that were initially configured at the phone before the user logged in) are used for external route patterns. Because the phone did not physically move, the same local gateways should be used for PSTN calls, even when a different user is currently logged into the device."

CIPT2 book page 328

gosh

QUESTION 53
Which Cisco IOS command is used to verify that a SAF Forwarder that is registered with Cisco Unified Communications Manager has established neighbor relations with an adjacent SAF Forwarder?

A. show eigrp service-family ipv4 neighbors
B. show eigrp address-family ipv4 neighbors
C. show voice saf dndball
D. show saf neighbors
E. show ip saf neighbors

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: BCDE

Router# show eigrp service-family ipv4 4453 neighbors


QUESTION 54
Which Cisco IOS command is used to verify that the Cisco Unified Communications Manager Express has registered with the SAF Forwarder?

A. show eigrp service-family ipv4 clients
B. show eigrp address-family ipv4 clients
C. show voice saf dndb all
D. show saf registration
E. show ip saf registration

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: BCDE
show eigrp service-family ipv4 clients Displays information from the EIGRP IPv4 service-family results.

QUESTION 55
DRAG DROP
Select and Place:

Click and drag the minimum Cisco Unified SRST configuration steps on the left to the spaces on the right.

<table>
<thead>
<tr>
<th>ccm-manager fallback-mgcp</th>
</tr>
</thead>
<tbody>
<tr>
<td>call-manager-fallback</td>
</tr>
<tr>
<td>application</td>
</tr>
<tr>
<td>ip source-address 10.5.1.1 port 2000</td>
</tr>
<tr>
<td>service alternate default</td>
</tr>
<tr>
<td>max-ephones 3</td>
</tr>
<tr>
<td>max-dn 6</td>
</tr>
<tr>
<td>limit-dn 7965 2</td>
</tr>
<tr>
<td>keepalive 20</td>
</tr>
</tbody>
</table>
Correct Answer:

Click and drag the minimum Cisco Unified SRST configuration steps on the left to the spaces on the right are used.

- ccm-manager fallback-mgcp

- application

- service alternate default

- limit-dn 7965 2

- keepalive 20

Section: (none)
Explanation

Explanation/Reference:
Explanation:
Call-manager-fallback
Ip source-address 10.5.1.1 port 2000
Max-dn 6
Max-ephones 3

QUESTION 56
Refer to the exhibit.
To stream multicast MOH to the remote site across the WAN, what should the minimum value for the Max Hops be configured as?

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

Correct Answer: B  
Section: (none)  
Explanation

Explanation/Reference:
Explanation: The Max Hops field in the Music On Hold (MOH) Server Configuration window indicates the maximum number of routers that an audio source is allowed to cross. If max hops is set to zero, the audio source must remain in its own subnet. If max hops is set to one, the audio source can cross up to one router to the next subnet. Cisco recommends setting max hops to two. Link:http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsMoh.html

QUESTION 57
When the user of a phone registered to the Cisco Unified Communications Manager places a call to 3001 when the SAF network is down, what happens?

A. The call fails.
B. The call is rerouted to the PSTN with the constructed PSTN number as +442288223001
C. The call is rerouted to the PSTN with the constructed PSTN number as 442288223001
D. The call is rerouted to the PSTN with the constructed PSTN number as 0002288223001
E. The call is rerouted to the PSTN with the constructed PSTN number as +0002288223001

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Original answer: A) The call fails
Answer changed to C) the call is rerouted to 442288223001
When the learned pattern is unreachable the ToDID field is used to route the call to the PSTN.

CIPT2 book page 355:

"Sometimes, the IP path for a learned route might not be available, and a ToDID rule might have been advertised with the hosted directory number. In that situation, a call to the transformed number (a ToDID rule that is applied to the advertised pattern) is placed with the Automated Alternate Routing (AAR) CSS of the calling device."

page 361:

"If the learned pattern was removed when the IP path became unavailable, the originating site would not know what PSTN number to use for the backup call. By default, a route is completely removed only if it has not advertised for 48 hours."

Don't confuse CCD with AAR. AAR will deny a call if the network is down:

page 256:

"AAR is a fallback mechanism for calls that are denied by locations-based CAC or RSVP-enabled locations-based CAC. It does not apply to voice calls that are denied by gateways because they exceed the available or administratively permitted number of channels. It also does not apply to calls that have been rejected on trunks, such as gatekeeper-controlled H.225 or intercluster trunks. If such calls fail for any reason, fallback mechanisms are provided by route lists and route groups. AAR is invoked only when the locations-based CAC denies the call because of a lack of network bandwidth. AAR is not invoked when the IP WAN is unavailable or other connectivity issues cause the called device to become unregistered with CUCM. In such cases, the calls are redirected to the target specified in the Call Forward No Answer field of the called device."

CCD will not deny a call if the network is down (unlike AAR)

page 355:

"PSTN backup for CCD is completely independent from AAR. AAR places PSTN backup calls for cluster-internal destinations when the IP path cannot be used because of insufficient bandwidth as indicated by call admission control (CAC). Only the AAR CSS is reused for CCD PSTN backup. Otherwise, CCD PSTN backup does not interact with AAR at all. For example, CCD PSTN backup works even when AAR is globally disabled by the corresponding Cisco CallManager service parameter."

gosh

QUESTION 58
Refer to the exhibit.
Assume that NANP is being used and 9 is used for PSTN access code. Long distance national calls are preceded with 1.

How should the HQ Cisco Unified Communications Manager be configured for calls to 3XXX to be sent to the gatekeeper at 1.0.1.6.1 with PSTN backups?

A. Configure a route pattern for 3XXX. Assign this route pattern to a route list that points to two route groups. The first route group contains the H 225 trunk. The second route group contains the MGCP gateway with prefix digits 1 408555 for the outgoing called number.

B. Configure a route pattern for 1#3XXX. Assign this route pattern to a route list that points to a route group that lists the H 225 trunk as first choice and the MGCP gateway as a second choice.

C. Configure a route pattern for 4085543XXX. Assign this route pattern to a route list that points to two route groups. The first route group contains the H 226 trunk. The second route group contains MGCP gateway.
D. Configure a route pattern for 3XXX Assign this route pattern to a route list that points to two route groups. The first route group contains the H.225 trunk. The second route group contains MGCP gateway with prefix digits 91 408554 for the called number.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
see CIPT2 book page 281-282

"For a PSTN backup, you need to perform digit manipulation in such a way that the calling number and (more importantly) the called number are transformed to always suit the needs of the device that is actually used. This transformation can be done at the route list, where digit manipulation can be configured per route group. In the example, the called number, 9 1 511 555-1234, has to be changed to a ten-digit number for the H.225 trunk, because the gatekeeper is configured with area code prefixes without the long distance 1. The called number must also be changed to an 11-digit number if rerouting the call to the PSTN gateway is necessary. A better solution would be using global transformations at the egress devices (H.225 trunk and PSTN gateways). In a large multisite environment or in an international deployment, the implementation of globalized call routing would be the best solution."

A route pattern of 3XXX has to be assigned to a route list with 2 route groups. First route group points to the trunk. Second route group points to the MGCP gateway and must prefix 1408555 to the called number. You can't include a 9 because the PSTN does not understand leading 9's.

original answer D
Changed answer to A because the called number has to be 10 digits, not 11.

gosh

**QUESTION 59**
Refer to the following exhibit.

Which Cisco IOS SAF Forwarder configuration is correct?

A. Exhibit A
Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: BCD
Summary steps to configure IOS SAF forwarder is given below
1. enable
2. configure terminal
3. router eigrp virtual-instance-name
4. service-family {ipv4 | ipv6} [vrfvrf-name] autonomous-system autonomous-system-number
5. topology base
6. external-client client_label
7. exit-sf-topology
8. exit-service-family
9. exit
10. service-family external-client listen {ipv4 | ipv6} tcp_port_number
11. external-client client-label
12. username user-name
13. password password-name
14. keepalive number
15. exit

Link:

QUESTION 60
Refer to the exhibit.

Which pattern will be advertised try the Cisco Unified Communications Manager?

A. 3XXX and the ToDID will be 0:.
B. 3XXX and the TnOID will be 0:44228822.
C. 3XXX and the ToDID will be 44228822.
D. 3XXX and the ToDID will be 0:+44228822.
E. 3XXX and the ToDID will be 0:+

Correct Answer: A

Section: (none)

Explanation

"Select the Use Hosted DN as PSTN Failover check box to create a ToDID rule of 0:. As a result, the number that is to be used for PSTN backup is identical to the internally used number. Usually, this result occurs only when tail-end hop-off (TEHO) patterns are advertised."
Because the Hosted DN Pattern has "use HostedDN as PSTN Failover" checked, that means the todid will be 0.

CIPT2 book page 370
gosh

**QUESTION 61**
Refer to the exhibit.
Locations-based CAC has been configured between HQ and the BR site. Assume that the priority queue has been provisioned correctly for three G.729 calls. What happens when the fourth call is placed from HO to BR?

A. The call will get through via the WAN, but it will experience poor audio quality.
B. The call will fail.
C. The call will be queued until one of the existing calls drop.
D. The call will get through without any issues.

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:
Explanation: Incorrect answer: BCD
QOS policy is not set to drop the packet.

QUESTION 62
Refer to the exhibit.

What media resource should be configured in Cisco Unified Communications Manager?

A. Cisco Media Termination Point Hardware
B. Cisco IOS Enhanced Software Media Termination Termination Point
C. Cisco IOS Media Termination Point
D. Cisco Media Termination Point Hardware (WS-SVC-CMM)

Correct Answer: B

Section: (none)

Explanation
Explanation/Reference:
"In the Media Termination Point Configuration window, choose the type of the media termination point. Currently the only option is Cisco IOS Enhanced Software Media Termination Point. Enter a name and description, and then choose the device pool that should be used."

"Note The name of the media termination point has to match the name that was configured at the Cisco IOS router with the associate profile id register command entered in SCCP CCM group id configuration mode. The name is case-sensitive."

CIPT2 book page 252-253

gosh

QUESTION 63
Refer to the exhibit

The exhibit shows a SAF Forwarder configuration attached to a Cisco Unified Communications Manager.

Which minimum configuration for a Cisco Unified Communications Manager Express SAF Forwarder is needed to establish a SAF neighbor relationship with this SAF Forwarder?

A. router eigrp SAF
   | service-family ipv4 autonomous-system 1
   | topology base
   | exit-sf-topology
   | exit-service-family
   | voice service saf
   | profile trunkroute 1
   | session protocol sip interface Loopback1 transport tcp port 5060 !

B. router eigrp SAF
   | service-family ipv4 autonomous-system 1
   | topology base
   | exit-sf-topology
   | exit-service-family
   | voice service saf
   | profile trunk-route 1
session protocol sip interface Loopback1 transport tcp port 5060
profile dn-block 1 alias-prefix 1972555
pattern 1 type extension 4xxx

profile callcontrol 1
dn-service
trunk-route 1
dn-block 1
dn-block 2

channel 1 vrouter SAF asystem 1
subscribe callcontrol wildcarded
publish callcontrol 1

C. router eigrp SAF

service-family ipv4 autonomous-system 1

topology base
exit-sf-topology
exit-service-family

D. None of above configurations contain sufficient information.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Interface Loopback1
IP Address 10.1.1.1 255.255.255.255

router eigrp SAF

service-family ipv4 autonomous-system 1

sf-interface Loopback1
topology base
exit-sf-topology
exit-service-family

CIPT2 book page 365-366
gosh

QUESTION 64
Refer to the following exhibits.
hostname HQ_gateway

card type el 0 0
cable cisco123

no aaa new-model

network-clock-participate wic 0

ip source-route
ip cef

isdn switch-type primary-net5

voice-card 0

voice service voip
  allow-connections sip to sip

controller E1 0/0/0
pri-group timeslots 1-12,16 service mgcp

interface Loopback0
ip address 10.1.111.1 255.255.255.0

interface GigabitEthernet0/0
no ip address
ip pim sparse-dense-mode
duplex auto
speed auto
media-type rj45
interface GigabitEthernet0/0
  no ip address
  ip pim sparse-dense-mode
duplex auto
  speed auto
media-type rj45

interface GigabitEthernet0/0.5
  encapsulation dot1q 5
  ip address 10.1.5.1 255.255.255.0
  ip pim sparse-dense-mode

interface GigabitEthernet0/0.10
  encapsulation dot1q 10
  ip address 10.1.10.1 255.255.255.0
  ip pim sparse-dense-mode

interface GigabitEthernet0/0.110
  encapsulation dot1q 110
  ip address 10.1.110.1 255.255.255.0
  ip helper-address 10.1.5.10
  ip pim sparse-dense-mode

interface GigabitEthernet0/1
  ip address 10.140.1.2 255.255.255.0
duplex auto
  speed auto
media-type rj45

interface Serial0/0/0:15
  no ip address
interface Serial0/0/0:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable

interface Serial0/1/0
  no ip address
  ip pim sparse-dense-mode
  encapsulation frame-relay IETF

interface Serial0/1/0.101 point-to-point
  ip address 10.12.1.1 255.255.255.0
  ip pim sparse-dense-mode
  snmp trap link-status
  frame-relay interface-dlci 101

interface Serial0/1/0.102 point-to-point
  ip address 10.13.1.1 255.255.255.0
  snmp trap link-status
  frame-relay interface-dlci 102

router eigrp 10
  network 10.0.0.0

  ip forward-protocol nd

voice-port 0/0/0:15
voice-port 0/0/0:15

ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.1.5.10

mgcp
mgcp call-agent 10.1.5.10 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp nodes passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm
mgcp rtp payload-type g726 g726 static
mgcp behavior g729-variants static-pt

mgcp profile default

dial-peer voice 1111 voip
    session protocol sipv2
    incoming called-number

dial-peer voice 222 voip
    session protocol sipv2
    destination-pattern +49
Users in the U.S dial Germany by calling 9011 49 followed by the remaining digits. What would be the most suitable configuration for Connection X?

A. Configure a SIP trunk to 10.140.1.1 and a SIP route pattern +49T in Cisco Unified Communications Manager.

B. Configure a SIP trunk to the Cisco Unified Border Element and route pattern +49T in Cisco Unified Communications Manager.

C. Configure a SIP trunk to the Cisco Unified Border Element. Configure a translation pattern for 9011.49T using DDI predot prefix + and CSS to point to a route pattern partition \+! which uses the SIP trunk.

D. Configure a SIP trunk to the ITSP. Configure a translation pattern for 9011.49T using DDI predot prefix + and CSS to point to a route pattern partition \+! which uses the SIP trunk.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:
Explanation: Incorrect answer: ABD
SIP trunks for public switched telephone network (PSTN) access are an important new access method for business collaboration. Service providers throughout the world offer SIP trunking as an alternative to traditional TDM (T1/E1) connections.
A discard digits instruction (DDI) removes a portion of the dialed digit string before passing the number on to the adjacent system. A DDI must remove portions of the digit string, for example, when an external access
A PSTN call arrived at the MGCP gateway. The calling number was received as 14087071222 with number set to type international. The HQ_clng__pty_CSS contains the HQ_clng__pty__Pt partition. Which caller ID is displayed on the IP phone?

A. +087071222
B. 14087071222
C. 087071222
D. 4087071222
E. 14087071222

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
The gateway transforms the number to +087071222, which matches the transform pattern \\+\., which removes everything before the dot, which in this case is a +. Therefore the output calling number is 08707122.

gosh

QUESTION 66
Refer to the exhibit.
## Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registered with</td>
<td>Cisco Unified Communications Manager 10.1.5.10</td>
</tr>
<tr>
<td>IP Address</td>
<td>10.1.5.10</td>
</tr>
<tr>
<td>Music On Hold Server Name</td>
<td>MOH_2</td>
</tr>
<tr>
<td>Description</td>
<td>MOH_CUCM881Pub1</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>Maximum Half Duplex Streams</td>
<td>Default</td>
</tr>
<tr>
<td>Maximum Multi-cast Connections</td>
<td>250</td>
</tr>
<tr>
<td>Fixed Audio Source Device</td>
<td></td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Off</td>
</tr>
<tr>
<td>Run Flag</td>
<td>Yes</td>
</tr>
</tbody>
</table>

### Multi-cast Audio Source Information

- **Enable Multi-cast Audio Sources on this MOH Server**: On

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Base Multi-cast IP Address</td>
<td>229.1.1.1</td>
</tr>
<tr>
<td>Base Multi-cast Port Number</td>
<td>16384</td>
</tr>
<tr>
<td>Increment Multi-cast on</td>
<td>(Even numbers only)</td>
</tr>
</tbody>
</table>

### Selected Multi-cast Audio Sources

<table>
<thead>
<tr>
<th>No.</th>
<th>Audio Source Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>SampleAudioSource</td>
</tr>
</tbody>
</table>
Multicast MOH needs to be run from flash at the remote site. Which Cisco IOS configuration at the SRST router is correct?

A. ```
! 
call-manager-fallback
ip source-address 10.1.5.102 port 2000
max-ephones 2
max-dn 8 dual-line
moh music-on-hold.au
multicast moh 239.1.1.1 port 16384
```  
B. ```
! 
call-manager-fallback
ip source-address 10.1.5.102 port 2000
max-ephones 2
max-dn 8 dual-line
moh music-on-hold.au
multicast moh 239.2.1.1 port 16384
```  
C. ```
! 
call-manager-fallback
ip source-address 10.1.120.1 port 2000
max-ephones 2
max-dn 8
moh music-on-hold.au
multicast moh 239.1.1.1 port 32767
```  
D. ```
! 
call-manager-fallback
ip source-address 10.1.5.102 port 2000
max-ephones 2
```
max-dn 8 dual-line
moh music-on-hold.au
multicast moh 239.1.1.1 port 32767

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
call-manager-fallback
max-ephones 1
max-dn 1
ip source-address 10.1.5.102
moh moh-file.au
multicast moh 239.1.1.1 port 16384

CIPT2 book page 219 "Implementing Multicast MOH from Remote Site Router Flash"
gosh

QUESTION 67

Refer to the exhibit.
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
hostname BRI
!
boot-start-marker
boot system flash:c2800nm-ipvoice_iws-mz.124-22.T.bin
boot-end-marker
!
card type tl 0 0
logging message-counter syslog
enable password cisco123
!
no asa new-model
network-clock-participate wic 0
!
ip source-route
!
ip cef
ip dhcp excluded-address 10.1.20.1 10.1.20.9
ip dhcp excluded-address 10.1.20.21 10.1.20.254
!
ip dhcp pool Data
    network 10.1.20.0 255.255.255.0
    default-router 10.1.20.1
!
no ip domain-lookup
ip dhcp excluded-address 10.1.20.21 10.1.20.254

ip dhcp pool Data
    network 10.1.20.0 255.255.255.0
    default-router 10.1.20.1

no ip domain lookup
ip multicast-routing
no ipv6 cef
multilink bundle-name authenticated

isdn switch-type primary-mi

voice service voip
    allow-connections h323 to h323

voice class h323 1
    h225 timeout tcp establish 3

voice translation-rule 1
    rule 1 /^\d{3,4}$/ /D/
    rule 2 /^2\d{7,8}$/ /D/

voice translation-rule 2
    rule 1 /^2/ /6506032/ type any national
    rule 2 /^4/ /4989531214/ type any international
    rule 3 /^9011/ // type any international

voice translation-rule 3
    rule 1 /^3...$/ /212710x/

voice translation-profile pstn-in
    translate called 1

voice translation-profile sisl
    translate calling 3
    translate called 2

voice-card 0
dspfarm
dsp services dspfarm

wtp mode transparent
archive
Gateway Config

vtp mode transparent
archive
  log config
    hidekeys
  ;
controller T1 0/0/0
cablelength short 110
pri-group timeslots 1-12,24

  vlan 20
    name BR1-Data

  vlan 120
    name BR1-Voice

  interface GigabitEthernet0/0
    no ip address
    shutdown
duplex auto
speed auto

  interface GigabitEthernet0/1
    no ip address
    shutdown
duplex auto
speed auto

  interface FastEthernet0/1/0
    no ip address
    shutdown
duplex auto
speed auto
interface GigabitEthernet0/0
no ip address
shutdown
duplex auto
speed auto

interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto

interface FastEthernet0/1/0
description BRI Phone1
switchport access vlan 20
switchport voice vlan 120
spanning-tree portfast

interface FastEthernet0/1/1
description BRI Phone2
switchport access vlan 20
switchport voice vlan 120
spanning-tree portfast

interface FastEthernet0/1/2

interface FastEthernet0/1/3
description Trunk for L2TPv3 - Do Not Modify
switchport mode trunk

interface Serial0/0/0:23
interface FastEthernet0/1/3
  description Trunk for L2TPv3 - Do Not Modify
  switchport mode trunk

interface Serial0/0/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-ni
  isdn incoming-voice voice
  isdn bchan-number-order ascending
  no cdp enable

interface Serial0/2/0
  no ip address
  encapsulation frame-relay IETF

interface Serial0/2/0.101 point-to-point
  ip address 10.12.1.2 255.255.255.0
  ip pim sparse-dense-mode
  snmp trap link-status
  frame-relay interface-dlci 101

interface Vlan1
  no ip address
  shutdown

interface Vlan20
  ip address 10.1.20.1 255.255.255.0

interface Vlan120
  ip address 10.1.120.1 255.255.255.0
  ip helper-address 10.1.1.5
interface Vlan120
ip address 10.1.120.1 255.255.255.0
ip helper-address 10.1.5.2
ip pim sparse-dense-mode
h323-gateway voip bind srcaddr 10.1.120.1

router eigrp 10
network 10.0.0.0
no auto-summary

ip forward-protocol nd

no ip http server

control-plane

voice-port 0/0/0:23
  translation-profile incoming pstn-in
  translation-profile outgoing srst

ccm-manager fax protocol cisco

mgcp fax t38 ecm

dial-peer voice 911 pots
destination-pattern 911
dial-peer voice 911 pots
  destination-pattern 911
  port 0/0/0:23
  forward-digits all

! dial-peer voice 9911 pots
  destination-pattern 9911
  port 0/0/0:23
  forward-digits all

! dial-peer voice 123 pots
  incoming called-number .
  direct-inward-dial

! dial-peer voice 5000 voip
  destination-pattern 3... 
  voice-class h323 1
  session target ipv4:10.1.5.3
  dtmf-relay h245-alphanumeric
  no vad

! dial-peer voice 9011 pots
  corlist outgoing intPt
  destination-pattern 9011T
  port 0/0/0:23

! dial-peer voice 7 pots
  corlist outgoing localPt
  destination-pattern 9[2-9].....
  port 0/0/0:23
dial-peer voice 7 pots
corlist outgoing localPt
destination-pattern 9[2-9]......
port 0/0/0:23

!  
dial-peer voice 24000 pots
destination-pattern [24]...
port 0/0/0:23

!  
dial-peer voice 30001 voip
  preference 1
  destination-pattern 3...
  voice-class h323 1
  session target ipv4:10.1.5.2
dtmf-relay h245-alphanumeric
  no vad

!  
dial-peer voice 800 pots
destination-pattern 91800......
port 0/0/0:23
prefix 800

!  
dial-peer voice 866 pots
  destination-pattern 91866......
  port 0/9/0:23
prefix 866

!  
dial-peer voice 877 pots
  destination-pattern 91877......
  port 0/0/0:23
prefix 877
dial-peer voice 888 pots
    destination-pattern 91888........
    port 0/0/0:23
    prefix 888

dial-peer voice 11 pots
    corlist outgoing IdPt
    destination-pattern 91[2-9]..[2-9].....
    port 0/0/0:23

dial-peer voice 3900 voip
    destination-pattern 3500
    session target ipv4:10.1.5.3
    dtmf-relay h245-alphanumeric
    no vad

gateway
    timer receive-rtp 1200

gatekeeper
    shutdown

call-manager-fallback
    max-conferences 8 gain -6
    transfer-system full-consult
    ip source-address 10.1.120.1 port 2000
    max-ephones 4
The H.323 Gateway is showing status "unknown". Which statement is true?

A. The gateway must be reset in Cisco Unified Communications Manager.
B. The no gateway command followed by the gateway command must be issued in Cisco IOS.
C. The mgcp commands must be removed.
D. H.23 gateways do not register with Cisco Unified Communications Manager. H.323 gateways always show status "Unknown".
E. VUAN 1 20 may be down and so the H.323 gateway appears offline to the Cisco Unified Communications Manager

Correct Answer: D

Explanation/Reference:
H.323 gateways will always show status unknown because they don't register with CUCM. Only MGCP gateways register with CUCM.

gosh

QUESTION 68
Which Cisco IOS command is used for internal SAF Clients to check SAF learned routes?
A. show eigrp address-family ipv4 saf
B. show voice saf routes
C. show voice saf detail
D. show eigrp service-family ipv4 saf
E. show voice saf dndb all

Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:
"Example 12-7 shows the output of the SAF-learned routes in Cisco IOS Software configured as CUCME with the command **show voice saf dndp all**. Note that only a call agent can interpret SAF service data; SAF forwarders cannot interpret SAF service data. Therefore, this command works only on Cisco IOS routers that are internal SAF clients and SAF forwarders.
The output shows two types of patterns: extensions (with a ToDID) that were learned from a device other than a Cisco IOS device; and global patterns, which include a + sign and no ToDID information (most likely advertised by another Cisco IOS internal client)."

Example 12-7 Monitoring Learned Routes in CUCME
BR2# **show voice saf dndp all**
—
Last successful DB update @ 2010:03:12 16:03:27:838
******* Private Dialplan Partition *******
Pattern - 2XXX

CIPT2 Book page 382
go

QUESTION 69
Refer to the following exhibit.
The MGCP gateway has the following configurations:

called party transformation CSS HQ_cld_pty CSS (partition=HQ_cld_pty.Pt)
call-ng party transformation CSS HQ_clng_pty CSS (partition=HQ_clng_pty Pt)

All translation patterns have the check box "Use Calling Party's External Phone Number Mask" enabled.

When the IP phone at extension 3001 places a call to 9011 49403021 56001# what is the resulting called and calling number that is sent to the PSTN?

A. The called number is 01 1 49403021 56001. The calling number will be 5553001 and number type set to subscriber.
B. The called number is 011 49403021 56001. The calling number will be 5215553001 and number type set to national.
C. The called number is 4940302156001 with number type set to international. The calling number will be 5215553001 and number type set to national.
D. The called number is +49403021 56001 with number type set to international. The calling number will be 5215553001 and number type set to subscriber.

Correct Answer: A

Explanation

This is a tricky question! You have to look at this reversed. First, get the calling number.

Calling number: Since the question says "use calling party's external phone number mask", that means the calling number will be 15215553001. Look at the 2nd exhibit. That number will match the last one: clngPrtyTP: +1521.xxxxxx. The reason why the 2nd option is not correct is because the calling CSS is HQ_clng_pty, which includes HQ_clng_pty, therefore the partition is wrong for the second option. Therefore the output calling number is 5553001, with a TON of subscriber. The only possible answer then is A, as the other answers have incorrect calling number.

Called number: The called number is 9011 49403021 56001#. If you look at the first exhibit, it matches transformation pattern 4: TP: 9011.##.HQ_Int_PT. The output will be +49403021 56001. Therefore, the called number is +49403021 56001. However, answer A says the called number is 011 49403021 56001, which is wrong. Either I am missing something, or this question is wrong on the test.
When the Cisco Unified Communications Manager advertises the Hosted DN Pattern, which pattern would be advertised?

A. 2XXX and the TODID will be 0:+498950555
B. 2XXX and the TODID will be 0:+498953121
C. 4989S05552XXX and the ToDiD will be 0:
D. +4989631 21 2XXX and the ToDiD will be 0:
E. Both +4989505552XXX and +4989531 21 2XXX will be advertised with ToDID of 0:

Correct Answer: A

Section: (none)

Explanation

Hosted DN Pattern: Directory number or directory-number range to be advertised. Configured with PSTN failover strip digits and PSTN failover prepend digits; if not configured, hosted DN group configuration is used. Applied to hosted DN group.

Hosted DN Group Info: Configured with PSTN failover strip digits and PSTN failover prepend digits. Refers to hosted DN patterns.

The hosted DN Group is IGNORED if the Hosted DN Pattern has any configuration, therefore answer is 2xxx and todid 0:+498950555

CIPT2 book, page 362, 370
QUESTION 71
If your IP telephony administrator asks you to configure a local zone for your dial plan to control the volume of calls between two end points in a centralized multisite environment, which two types of Call Admission Control can be implemented? (Choose two.)

A. locations based
B. automated alternate routing
C. gatekeeper based
D. SRST
E. Cisco Unified Communications Manager based

Correct Answer: AC
Section: (none)
Explanation

Explanation/Reference:
The question asks how to limit calls. Locations based CAC and a gatekeeper can control network access. AAR does not restrict anything, it's only invoked after a call has been rejected for bandwidth (by a gatekeeper or locations based CAC).

Gatekeepers are centralized and use zones, both of which are asked in the question.

Locations based CAC is topology unaware and works best in hub and spoke topology. (page 236 CIPT2 book)

Original answer: A) locations based and B) automated alternate routing
changed answer to A) locations based and C) gatekeeper based

QUESTION 72
Refer to the exhibit.
It shows an H.323 gateway configuration in a Cisco Unified Communications Manager environment. An inbound PSTN call to this H.323 gateway fails to connect to IP phone extension 2001. The PSTN user hears a reorder tone. Debug isdn q931 on the H.323 gateway shows the correct called-party number as 5015552001. Which two configuration changes can correct this issue? (Choose two.)

A. Add port 1/0:23 to dial-peer voice 123 pots.
B. Ensure that the Significant Digits for inbound calls on the H.323 gateway configuration is 4.
C. Add a voice translation profile to truncate the number from 10 digits to 4 digits. Apply the voice translation profile to the Voice-port. The configuration field “Significant Digits for inbound calls” is left at default (All).
D. Add the command h323-gateway voip id on interface vlan120.
E. Change the destination-pattern on the dial-peer voice 23000 VoIP to 501501? and change the Significant Digits for inbound calls to 4.

Correct Answer: AE

Explanation/Reference:
Original answer was B and E. Changed based on feedback. You need a port command under the pots dial peer, and you need the significant digits to 4. Changed answer to A and E
QUESTION 73
Which two have to be defined in the Forward All field? (Choose two.) (Source: Preventing Toll Fraud)

A. calling search space
B. destination
C. partition
D. hunt list

Correct Answer: AB

Explanation/Reference:
Explanation: Incorrect answer: CD
Destination--This setting indicates the directory number to which all calls are forwarded. Use any dialable phone number, including an outside destination. Calling Search Space--This setting applies to all devices that are using this directory number.

QUESTION 74
Refer to the exhibit. With the Mobile Connect feature configured, when the PSTN phone calls the Enterprise user at extension 3001, the Enterprise user’s mobile phone does not ring. Which CSS is responsible for ensuring that the correct partitions are accessed when calls are sent to the Enterprise user’s mobile phone?

A. the gateway CSS
B. the Phone Device CSS
C. the Remote Destination Profile CSS
D. the Remote Destination Profile Rerouting CSS
E. the Phone Line (DN)CSS

Correct Answer: D
Explanation/Reference:
Explanation: Incorrect answer: ABCE
Ensure that the gateway that is configured for routing mobile calls is assigned to the partition that belongs to the Rerouting Calling Search Space. Cisco Unified Communications Manager determines how to route calls based on the remote destination number and the Rerouting Calling Search Space.

QUESTION 75
Refer to the exhibit. You have configured transcoder resources in both an IOS router and a Cisco Unified Communications Manager. When you review the configurations in both devices the IP addresses and transcoder names are correct, but the transcoder is failing to register with the Cisco Unified Communications Manager. Which command needs to be edited to allow the transcoder to register properly?

A. The associate profile and dsp farm profile numbers need to match associate ccm 2 command.
B. The associate ccm 2 priority 1 command needs to be changed so the ccm value matches identifier 1 in the sccp ccm 10.1.1.1 command.
C. The sccp ccm group number needs to match the associate ccm 2 command.
D. The maximum sessions command must match the number of codecs configured under the dsp farm profile.
E. The sccp ccm group number must match the voice-card number.

Correct Answer: B
"associate ccm: To associate a CUCM with a CUCM group and establish its priority within the group, use the associate ccm command in SCCP CUCM configuration mode."

gosh

QUESTION 76
Which statement is correct about AAR?

A. The end users sees, "Network Congestion Rerouting?" but AAR is otherwise transparent to the end user and works without user intervention.
B. AAR will display "not enough bandwidth" on the IP phone while it reroutes the call.
C. AAR allows calls to be rerouted because of insufficient Cisco Unified Border Element controlled bandwidth to an ITSP.
D. AAR allows calls to be rerouted due to insufficient gatekeeper controlled IP WAN bandwidth.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
"Without AAR, the user gets a reorder tone, and the IP Phone displays “Not enough bandwidth.” - CIPT2 book page 255, therefore that option is invalid

"With the AAR Network Congestion Rerouting Text parameter, you specify the text that is displayed on an IP Phone when AAR reroutes a call." - CIPT2 book page 261

gosh

QUESTION 77
The relationship between a Region and a Location is that the Region codec parameter is used between a Region and its configured Locations.

A. TRUE
B. FALSE

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation: Locations work in conjunction with regions to define the characteristics of a network link. Regions define the type of compression (G.711, G.722, G.723, G.729, GSM, or wideband) that is used on the link, and locations define the amount of available bandwidth for the link Link:http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#w p1033331

QUESTION 78
Refer to the exhibit. A user in RTP calls a phone in San Jose during congestion with Call Forward No Bandwidth (CFNB) configured to reach cell phone 4085550150. The user in RTP sees the message "Not Enough Bandwidth" on their phone and hears a fast busy tone. Which two conditions can correct this issue? (Choose two.)

A. The calling phone (RTP) needs to have AAR Group value of AAR under the AAR Settings.
B. The called phone (San Jose) needs to have AAR Group value of AAR under the AAR Settings.
C. The calling phone (RTP) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.
D. The calling phone (RTP) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.
E. The called phone (San Jose) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.
F. The called phone (San Jose) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.

Correct Answer: BF

Section: (none)

Explanation

Explanation/Reference:
"not enough bandwidth" means AAR is not in use, so the question is asking how to enable AAR.

page 255:
"Without AAR, the user gets a reorder tone, and the IP Phone displays “Not enough bandwidth.”

page 44:
"If a call over the IP WAN to another IP Phone in the same cluster is not admitted by CAC, the call can be rerouted over the PSTN using AAR, as shown in Figure 2-17. The AAR feature includes a CFNB option that allows the alternative number to be set for each IP Phone. In the example, because the remote site does not have PSTN access, the call is not rerouted to the IP Phone over the PSTN (instead of over the IP WAN). It is alternatively rerouted to the cell phone of the affected user. AAR and CFNB improve availability in multisite environments by providing the ability to reroute on-net calls that failed CAC over the PSTN."
QUESTION 79
Which two statements describe RSVP-enabled locations-based CAC? (Choose two.)

A. RSVP can be enabled selectively between pairs of locations.
B. Using RSVP for CAC simply allows admitting or denying calls based on a logical configuration that is ignoring the physical topology.
C. RSVP is topology aware, but only works with full mesh networks.
D. An RSVP agent is a Media Termination Point that the call has to flow through.
E. RSVP and RTP are used between the two endpoints.

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect Answer: BC
The RSVP policy that is configured for a location pair overrides the default interlocation RSVP policy that configure in the Service Parameter Configuration window. RSVP supports audio, video, and data pass-through. Video data pass-through allows video and data packets to flow through RSVP agent and media termination point devices.
Link: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02rsvp.html#wp1070214

QUESTION 80
You are the Cisco Unified Communications Manager in Certpaper.com. You use a remote site MGCP gateway to provide redundancy when connectivity to the central Cisco Unified Communications Manager cluster is lost. How to enable IP phones to establish calls to the PSTN when they have registered with the gateway? (Choose three.)

A. POTS dial peers must be added to the gateway to route calls from the IP phones to the PSTN.
B. The default service must be enabled globally.
C. The command ccm-manager mgcp-fallback must be configured.
D. COR needs to be configured to disallow outbound calls.

Correct Answer: ABC
Section: (none)
Explanation

Explanation/Reference:
CIPT2 book page 158, 163

RemoteSite# configure terminal
RemoteSite(config)# ccm-manager fallback-mgcp
RemoteSite(config)# call application alternate Default
RemoteSite(config-app-global)# service alternate Default

At least one dial peer needs to be configured to enable calls to and from the PSTN. The destination pattern of that dial peer has to correspond to the PSTN access code (for example, 9T). The more elegant way is to configure several dedicated dial peers with destination patterns that match the number patterns in a closed numbering plan, such as 91. ...... (91 followed by ten dots).
Which two configurations provide the best SIP trunk redundancy with Cisco Unified Communications Manager? (Choose two.)

A. Configure all SIP trunks with DNS SRV
B. Configure all SIP trunks with Cisco Unified Border Element
C. Configure all SIP trunks to point to a SIP gateway
D. Configure SIP trunks to be members of route groups and route lists
E. Configure all SIP trunks to allow TCP ports 5060
F. Configure all SIP trunks to point to a gatekeeper through SIP to H.323 gateway

Correct Answer: AD

Section: (none)

Explanation

Explanation/Reference:
Explanation: Incorrect answer: BCEF
For SIP trunks, Cisco Unified Communications Manager supports up to 16 IP addresses for each DNS SRV and up to 10 IP addresses for each DNS host name. The order of the IP addresses depends on the DNS response and may be identical in each DNS query. The OPTIONS request may go to a different set of remote destinations each time if a DNS SRV record (configured on the SIP trunk) resolves to more than 16 IP addresses, or if a host name (configured on the SIP trunk) resolves to more than 10 IP addresses. Thus, the status of a SIP trunk may change because of a change in the way a DNS query gets resolved, not because of any change in the status of any of the remote destinations.


QUESTION 82
When an external call is placed from Ajax, they would like the ANI that is sent to the PSTN to be the main number, not the extension. For domestic calls, they would like 10 digits sent; for international calls, they would like to send the country code 1 and the 10 digits. How can this be accomplished?

A. Add a translation pattern to the dial peers in the gateway that adds the appropriate digits to the outgoing ANI.
B. In the external call route patterns, set the external phone number mask to the main number. Use 10 digits in the domestic route pattern and 1 followed by the main number digits in the international route patterns.
C. Use a calling party transform mask for each route group in the corresponding route list configuration. Set the explicit 10-digit main number for domestic calls and 1 followed by the main number for the international route patterns.
D. In the directory number configurations, set the prefix digits field to the country code and the 10 digits of the main number. This will be truncated to the 10-digit number for domestic calls and sent out in its entirety for international calls.

Correct Answer: C

Section: (none)

Explanation

Explanation/Reference:
Explanation: Incorrect answer: ABD
Calling party transformation mask value is Valid entries for the NANP include the digits 0 through 9; the wildcard characters X, asterisk (*), and octothorpe (#); and the international escape character +.


QUESTION 83
The relationship between a Region and a Location is that the Region codec parameter is combined with Location bandwidth when communicating with other Regions.

A. FALSE
B. TRUE

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation: Locations work in conjunction with regions to define the characteristics of a network link. Regions define the type of compression (G.711, G.722, G.723, G.729, GSM, or wideband) that is used on the link, and locations define the amount of available bandwidth for the link Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wp1033331

QUESTION 84
Video calls using 384 kbps need to be supported across a gatekeeper-controlled trunk. What value should be entered into the gatekeeper to support this bandwidth?

A. 768 kbps
B. 384 kbps
C. 512 kbps
D. 192 kbps

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Original answer: b) 384 kbps
This is wrong because an IOS gatekeeper requires 768kbps for a 384kbps video call. See CIPT2 Book page 278. The bandwidth reserved has to be TWICE the actual bandwidth.

The command to enter on the gatekeeper would be:

bandwidth interzone zone <zonename> 768

I changed answer from 384 to 768 per book.
gosh

QUESTION 85
The following exhibit shows configs for H.323 gateway. You have been asked to implement TEHO from a remote branch office with area code 301 to the HQ office with area code 201 using Cisco Unified Communications Manager. The remote office has an MGCP gateway and the HQ office has an H.323 gateway. Once the call arrives at the HQ, it should break out to the PSTN as a seven-digit local call. Which statement about the route pattern is true?
A. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot and Prefix 9
B. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot
C. route pattern should be 91201.[2-9]XXXXXX
D. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot
E. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot and Prefix 9

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
The number being dialed from the remote office will be 9-1-xxx-xxx-xxxx. You have to prefix a 9 so the POTS dial peer is matched. Because the number has to be 7 digits you must have the dot before the 7 digits to strip the 1 and area code.

gosh

QUESTION 86
Which media resource should be configured in CUCM?

Voice-card 0
dspfarm
dsp services dspfarm
!
sccp local loopback0
sccp ccm 10.1.1.1 identifier 1 version 6.0
sccp
!
sccp ccm group 1
associate ccm 1 priority 1
associate profile 1 register HQ-1_XCORDER
!
dspfarm profile 1 transcode
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
maximum session 2
associate application SCCP
no shut

A. Cisco Media Termination Point Hardware
B. Cisco IOS Enhanced Media Termination Point
C. Cisco IOS Media Termination Point
D. Cisco Media Terminatin Point Hardware (WS-SVC-CMM)

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
"The device name has to match the name listed at the Cisco IOS router that provides the media resource. The name is case-sensitive. If the transcoding resource is provided by Cisco IOS Enhanced Media Termination Point hardware, the name can be freely chosen. In all other cases, the name is MTP followed by the MAC address of the interface that is configured to be used for registering the media resource with CUCM."

Since the question matches page 212, which uses "Cisco IOS Enhanced Media Termination point", and since the name does not begin with MTP, therefore only option is "cisco ios enhanced media termination point"

Originally answer marked as "cisco media termination point hardware", changed to "cisco ios enhanced media termination point"

CIPT2 Book, page 212
gosh
Exam B

QUESTION 1
Which statement about enrollment in the IP telephony PKI is true? (Source: Understanding Cisco IP Telephony Authentication and Encryption Fundamentals)

A. CAPF enrollment supports the use of authentication strings.
B. The CAPF itself has to enroll with the Cisco CTL client.
C. LSCs are issued by the Cisco CTL client or by the CAPF.
D. MICs are issued by the CAPF itself or by an external CA.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: BCD
The CAPF enrollment process is as follows:
1. The IP phone generates its public and private key pairs.
2. The IP phone downloads the certificate of the CAPF and uses it to establish a TLS session with the CAPF.
3. The IP phone enrolls with the CAPF, sending its identity, its public key, and an optional authentication string.
4. The CAPF issues a certificate for the IP phone signed with its private key.

QUESTION 2
An update of the configuration using the Cisco CTL client not needed when ________.(Source: Configuring Cisco IP Telephony Authentication and Encryption)

A. a Cisco Unified CallManager has been removed
B. an LSC of the IP phone is upgraded
C. a security token is added to the system
D. an IP address of the Cisco TFTP server has been changed

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: ACD
The CTL file contains entries for the following servers or security tokens:
- System Administrator Security Token (SAST)
- Cisco CallManager and Cisco TFTP services that are running on the same server
- Certificate Authority Proxy Function ()
- TFTP server(s)
- ASA firewall

QUESTION 3
Which statement is not true about GARP? (Source: Hardening the IP Phone)

A. GARP attacks require access to the target LAN or VLAN.
B. GARP can be used for a man-in-the-middle attack.
C. GARP is normally used for HSRP.
D. GARP can be disabled at Cisco IP phones.

Correct Answer: C

Explanation

Explanation/Reference:
Explanation: Incorrect answer: ABD
GARP (Gratuitous ARP) announce the presence of IP Phone on the network. Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/security/4_0_1/secuphne.html

QUESTION 4
Enabling authentication and encryption for CTI, JTAPI, and TAPI applications requires which two tasks? (Choose two.) (Source: Configuring Cisco IP Telephony Authentication and Encryption)

A. Enter the encryption key into the application.
B. Set up an IPsec association between the application and Cisco Unified CallManager.
C. Configure related security parameters in the CTI, JTAPI, and TAPI application.
D. Add the application user or end users to the Standard CTI Secure Connection user group, Standard CTI Allow Reception of SRTP Key Material user group, and Standard CTI Enabled user group.

Correct Answer: CD

Explanation

Explanation/Reference:
Explanation: Incorrect answer: AB
You must also add the application users or the end users to the Standard CTI Secure Connection user group in Cisco Unified Communications Manager Administration to enable TLS for the application. After you add the user to this group and install the certificate, the application ensures that the user connects via the TLS port. Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/security/8_6_1/secugd/secucti.html#wp1166397

QUESTION 5
Which two statements about symmetric encryption are true? (Choose two.) (Source: Understanding Cryptographic Fundamentals)

A. With symmetric encryption, the encryption key equals the decryption key.
B. Symmetric encryption is commonly used to sign asymmetric keys.
C. Symmetric encryption is a good choice for real-time encryption of bulk data.
D. Symmetric encryption uses asymmetric keys.

Correct Answer: AC

Explanation

Explanation/Reference:
Explanation: Incorrect answer: BD
There are two basic techniques for encrypting information: symmetric encryption (also called secret key encryption) and asymmetric encryption (also called public key encryption.) In symmetric key As long as both sender and recipient know the secret key, they can encrypt and decrypt all messages that use this key. A public key is made freely available to anyone who might want to send you a message. A second, private key is kept
secret, so that only you know it.
Link: http://support.microsoft.com/kb/246071

**QUESTION 6**
What is the default value for the Drop Ad Hoc Conference service parameter? (Source: Preventing Toll Fraud)

A. Never  
B. When No On-Net Parties Remain in the Conference  
C. When No Off-Net Parties Remain in the Conference  
D. Drop Ad Hoc Conference When Creator Leaves

**Correct Answer: A**
**Section: (none)**
**Explanation**

**Explanation/Reference:**
This is a CIP1 question doubtful will be on the test

gosh

**QUESTION 7**
You are the Cisco Unified Communications Manager in Certpaper.com. After you add the space command, the application can____.

A. set aside memory for application variables  
B. access the data on an internal server  
C. access the data on an external server  
D. share parameters between different call applications

**Correct Answer: D**
**Section: (none)**
**Explanation**

**Explanation/Reference:**
Explanation: Incorrect answer: ABC

```application
service debitcard tftp://15.5.27.11/app_debitcard.2.0.2.8.tcl paramspace english index 1
paramspace english language en
paramspace english location tftp://15.5.27.11/prompts/en/ param pid-len 4
paramspace english prefix en
param uid-len 6
```


**QUESTION 8**
Which device is needed to integrate H.320 into the Cisco video solution? (Source: Introducing IP Video Telephony)

A. video gateway  
B. MGCP gateway  
C. H.323 gatekeeper  
D. MCU
Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:
Explanation: Incorrect answer: ABD
As with H.323 MCUs, H.320 gateways are provisioned in Cisco Unified CallManager as H.323 gateways, and then route patterns are configured to extend calls to these devices. Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42video.html#wp1046523

http://www.gratisexam.com/