Cisco 642-457 Exam Questions & Answers

Number: 642-457
Passing Score: 800
Time Limit: 120 min
File Version: 35.5

Cisco 642-457 Exam Questions & Answers

Exam Name: CIPT2 v8.0 Implementing Cisco Unified Communications Manager, Part 2 v8.0

Sections

1. 1-18
2. 19-36
3. 37-54
4. 55-72

Actualcerts

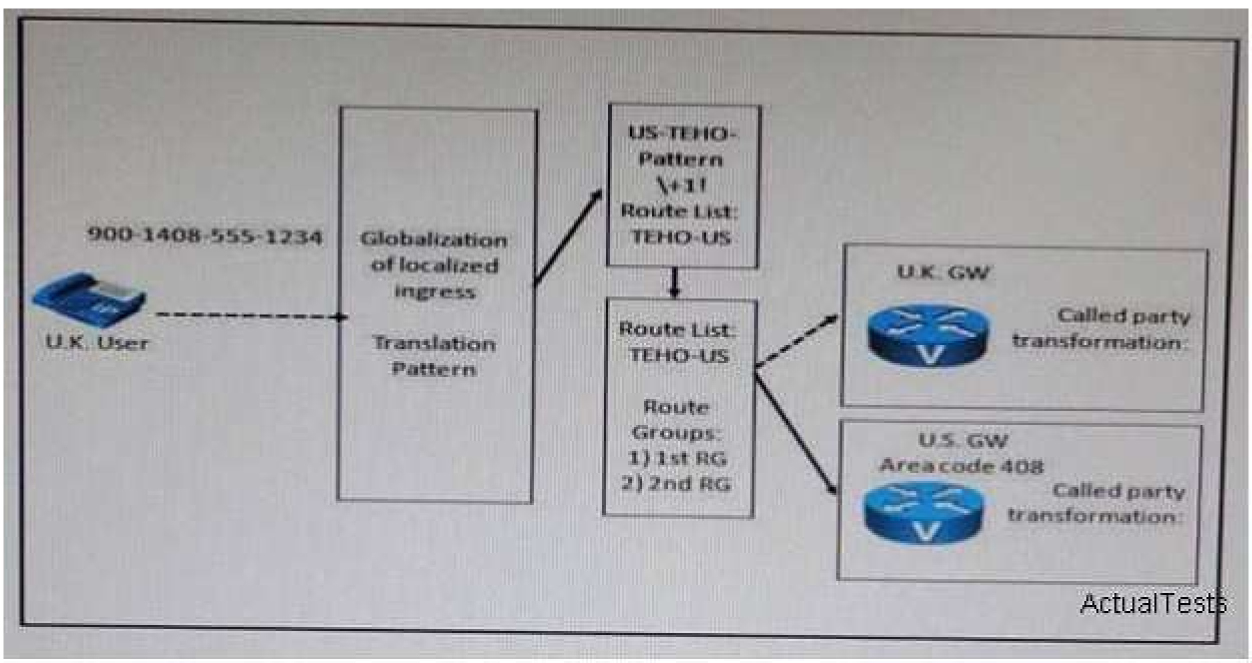
QUESTION 1
Which two features require or may require configuring a SIP trunk? (Choose two.)

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: AD
Section: 1-18
Explanation

Explanation/Reference:

QUESTION 2
Refer to the exhibit.



The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code J08 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 elected automatically.

E. The \+!route pattern should point directly to the U.S gateway.

Correct Answer: C
Section: 1-18
Explanation

Explanation/Reference:

QUESTION 3
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 supportvariable-length dial plans

Correct Answer: D
Section: 1-18
Explanation

Explanation/Reference:

QUESTION 4
Refer to the exhibit.

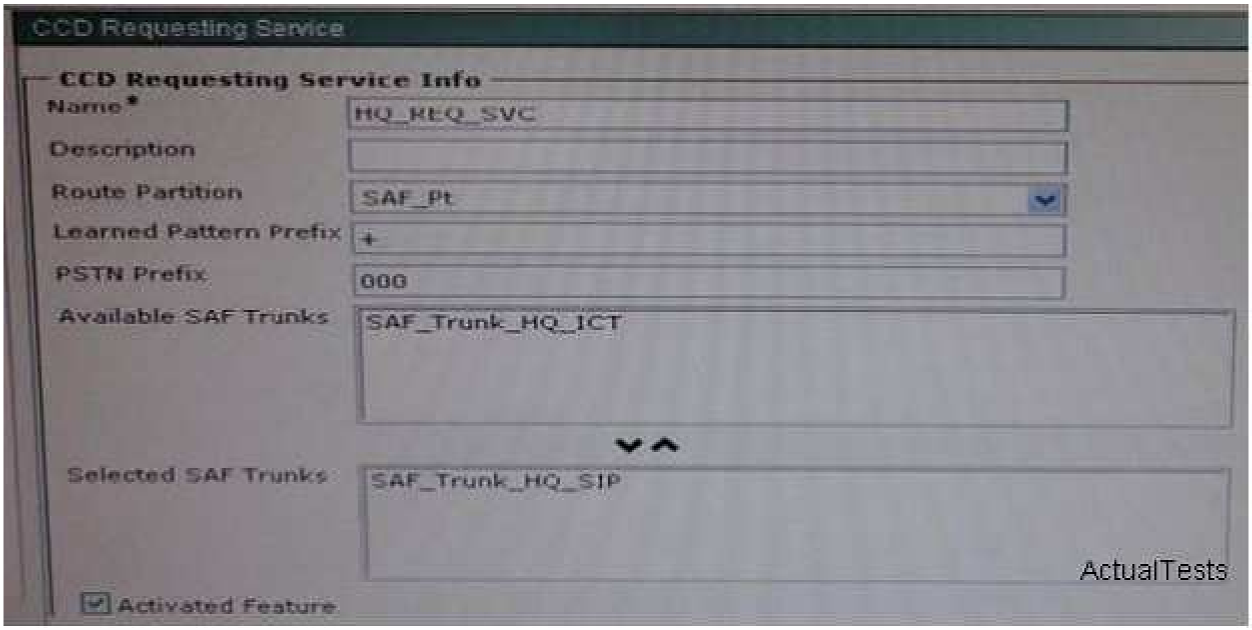
Which statement about the configuration between the Default and BK regions is true?

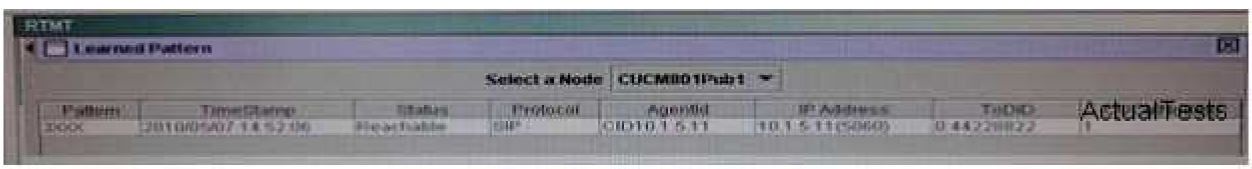
A. Calls between the two regions can use either 64 kbps or 8 kbps.
B. Calls between the two regions can use only the G 729 codec
C. Only 64 kbps will be used between the two regions because the link is "lossy"
D. Bothcodecs can be used depending on the loss statistics of the link, when lossy conditions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

Correct Answer: A
Section: 1-18
Explanation

Explanation/Reference:

QUESTION 5
Refer to the exhibit.





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: B
Section: 1-18
Explanation

Explanation/Reference:

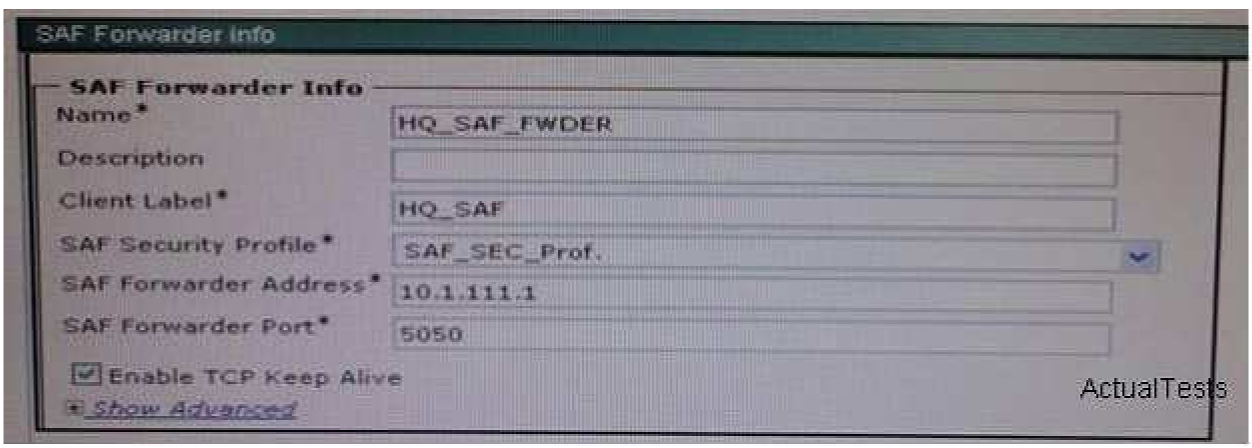
QUESTION 6
When an incoming PSTN call arrives at an MGCP gateway, how does the calling number get normalized to a global E.164 number with the + prefix in Cisco Unified Communications Manager?

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.
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: D
Section: 1-18
Explanation

Explanation/Reference:

QUESTION 7
Refer to the following exhibit.



Which Cisco IOS SAF Forwarder configuration is correct?

A. Answer
B.
C.
D.

Correct Answer: A
Section: 1-18
Explanation

Explanation/Reference:

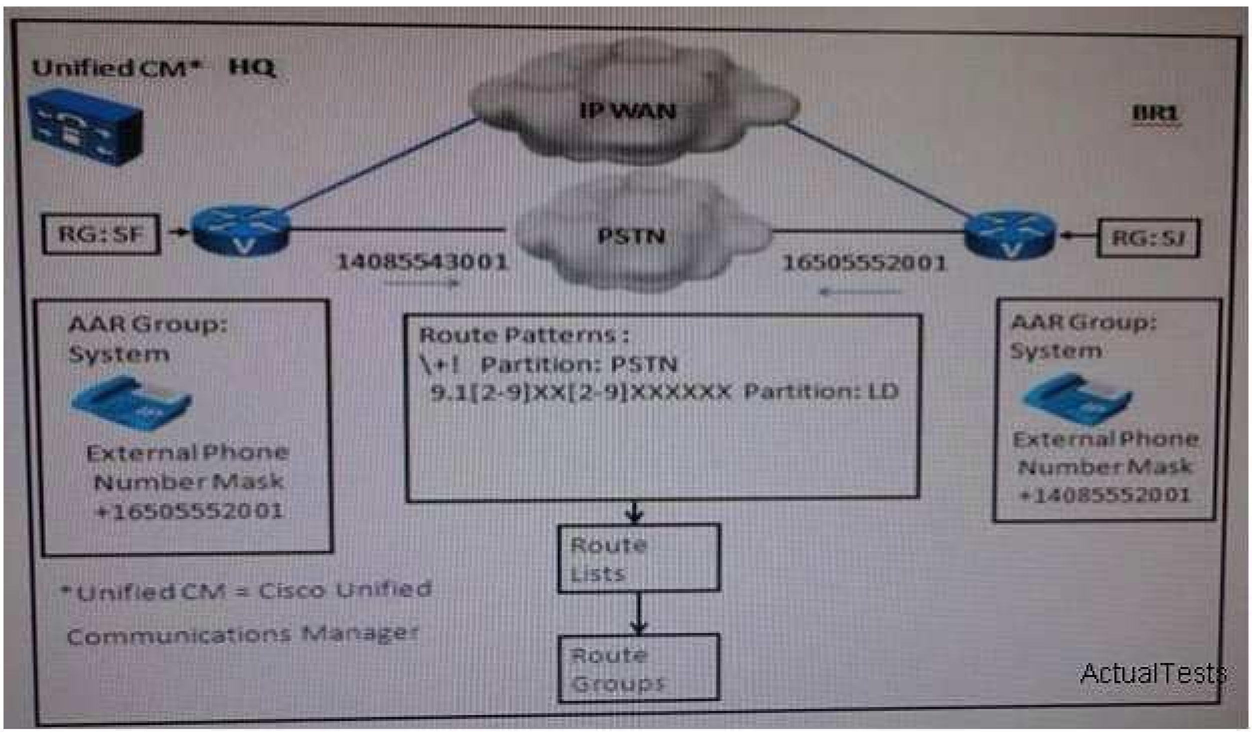
QUESTION 8
Which statement best describes globalized call routing in Cisco Unified Communications Manager?

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
Section: 1-18
Explanation

Explanation/Reference:

QUESTION 9
Refer to the exhibit.



The HO 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 HO 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]XX[2-9]XXXXXX for each site that must be globalized. Otherwise the called numbers will not be localized at the egrees gateway.

Correct Answer: A
Section: 19-36

Explanation

Explanation/Reference:

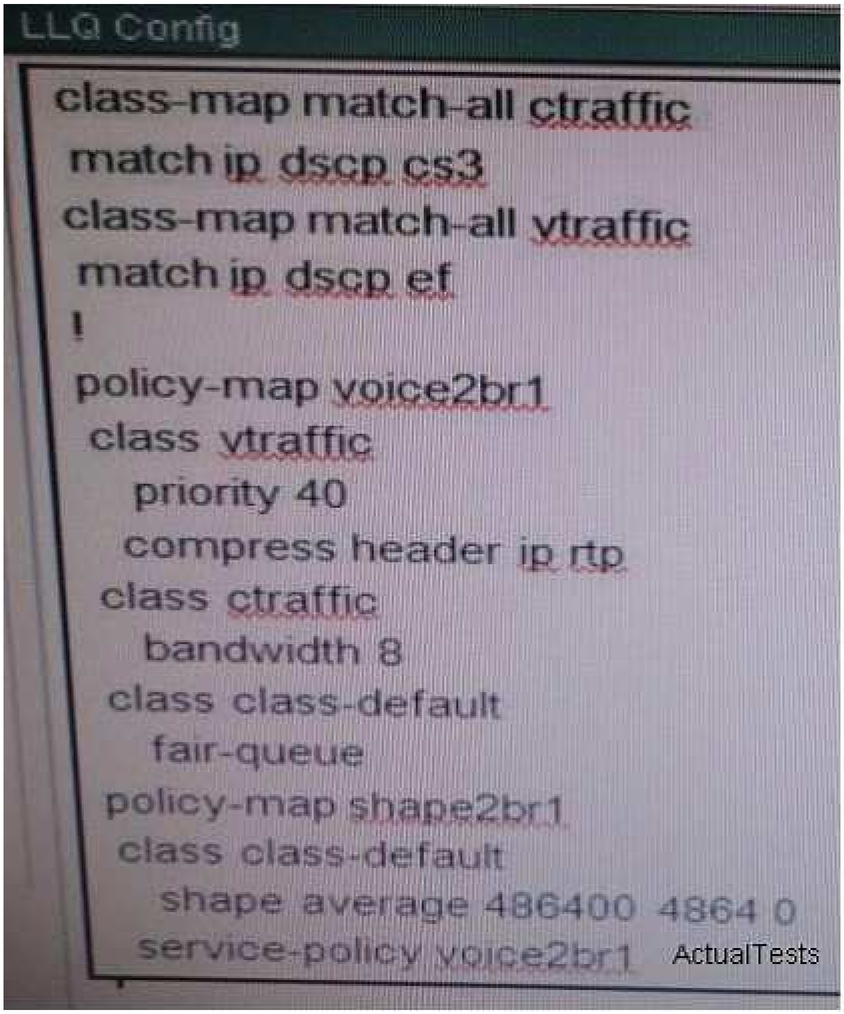
QUESTION 10
Which statement about H.323 Gatekeeper Call Admission Control is true?

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.

Correct Answer: B
Section: 19-36
Explanation

Explanation/Reference:

QUESTION 11
Refer to the exhibit.



LLQ Config

```
class-map match-all ctraffic
 match ip dscp cs3
class-map match-all vtraffic
 match ip dscp ef
!
policy-map voice2br1
 class vtraffic
   priority 40
   compress header ip rtp
 class ctraffic
   bandwidth 8
 class class-default
   fair-queue
policy-map shape2br1
 class class-default
   shape average 486400 4864 0
   service-policy voice2br1
```


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: B
Section: 19-36
Explanation

Explanation/Reference:

QUESTION 12
Which Cisco IOS command is used to verify that the Cisco Unified Communications Manager Express has registered with the SAF forwarder?

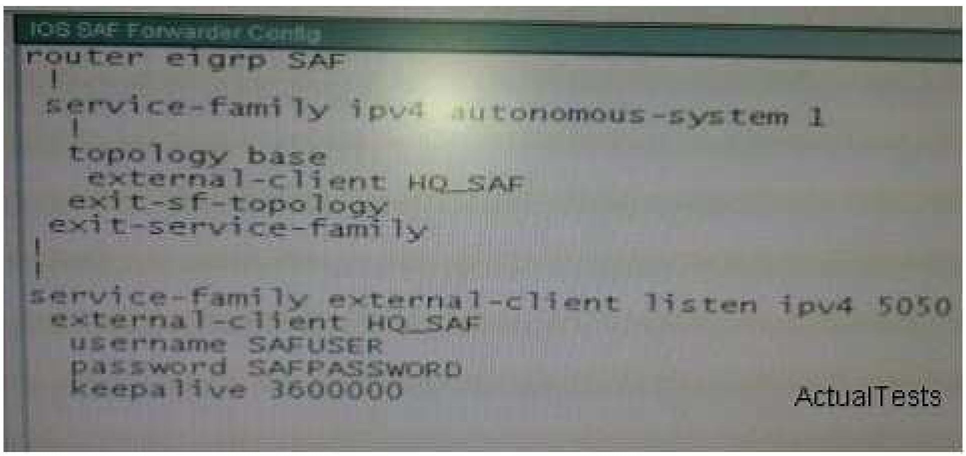
A. show eigrp service-family ipv4 clients
B. show eigrp address-family ipv4 clients
C. show voice saf dndball

D. show saf registration
E. show ip saf registration

Correct Answer: A
Section: 19-36
Explanation

Explanation/Reference:

QUESTION 13
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
   i
   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: 19-36
Explanation

Explanation/Reference:

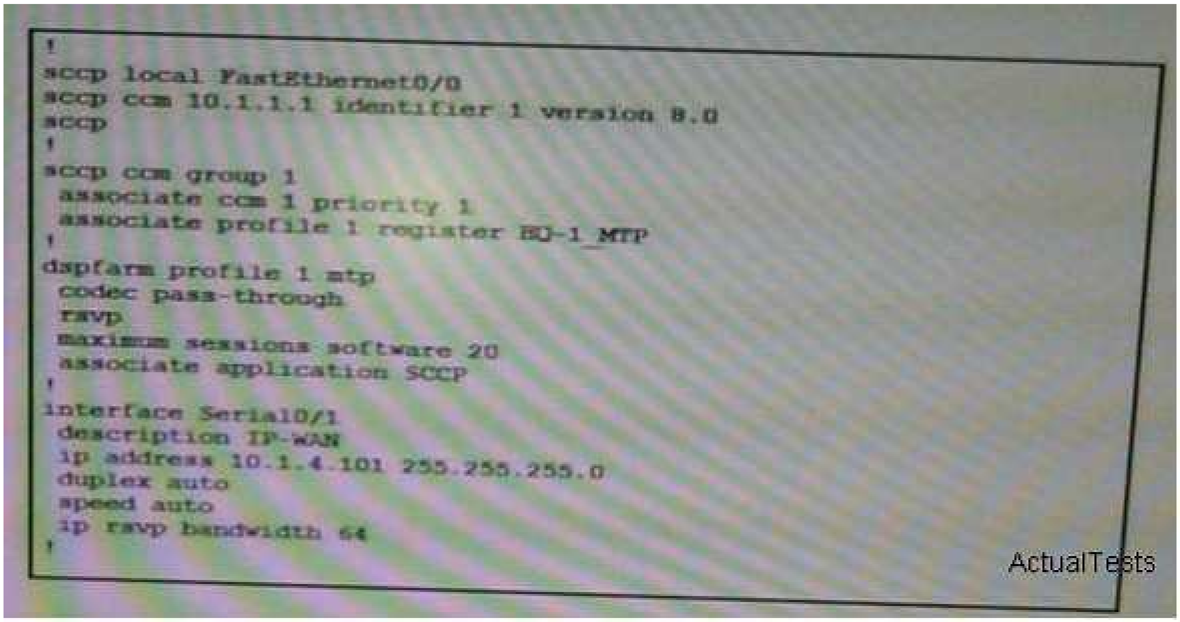
QUESTION 14
While operating in SRST, what is needed to route calls outside of the remote site location to the PSTN?

A. SIP trunk
B. CallManager route patterns
C. translation patterns
D. POTS dial peers
E. VOIP dial peers

Correct Answer: D
Section: 19-36
Explanation

Explanation/Reference:

QUESTION 15
Refer to the exhibit.



```
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 8.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ-1_MTP
!
dspfarm profile 1 mtp
 codec pass-through
 rsvp
 maximum sessions software 20
 associate application SCCP
!
interface Serial0/1
 description IP-WAN
 ip address 10.1.4.101 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth 64
!
```

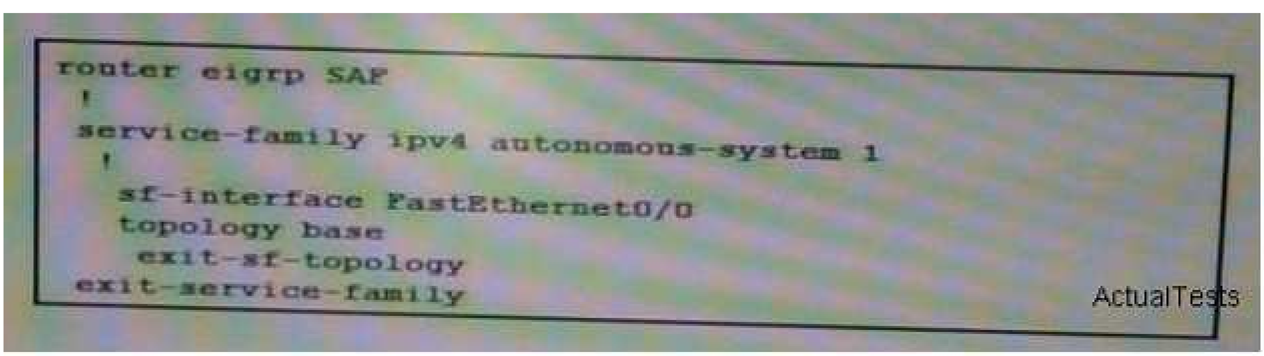

How many calls are permitted by the RSVP configuration?

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: A
Section: 37-54
Explanation

Explanation/Reference:

QUESTION 16
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: E
Section: 37-54
Explanation

Explanation/Reference:

QUESTION 17
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 timeoutkeepalive

Correct Answer: BCD
Section: 37-54
Explanation

Explanation/Reference:

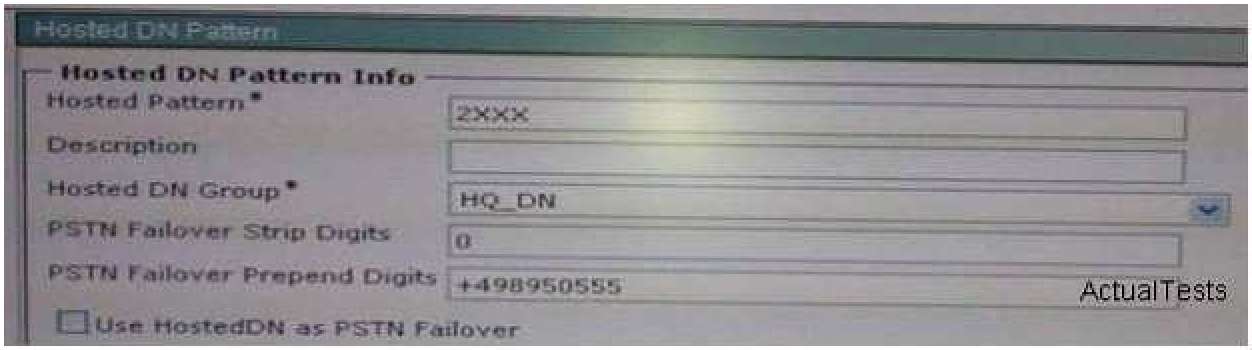
QUESTION 18
Which Cisco IOS command is used for internal SAF Clients to check SAF learned routes?

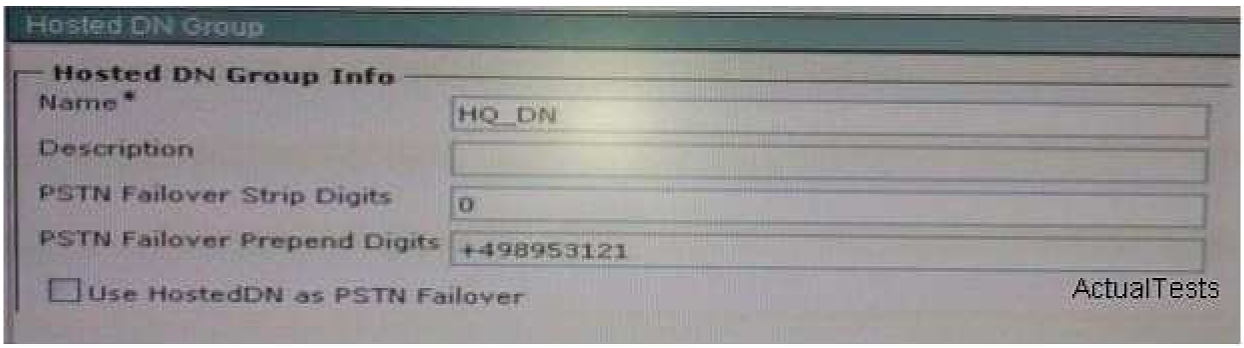
A. show eigrp address-family ipv4 saf
B. show voice saf routes
C. show voice saf detail
D. show eigrp service-family ipv4 saf
E. show voice saf dndb all

Correct Answer: E
Section: 37-54
Explanation

Explanation/Reference:

QUESTION 19
Refer to the exhibit





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. 4989505552XXX and the ToDiD will be 0:
D. +498963121 2XXX and the ToDiD will be 0:
E. Both +4989505552XXX and +4989531212XXX will be advertised with ToDID of 0:

Correct Answer: B
Section: 37-54
Explanation

Explanation/Reference:

QUESTION 20
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: 37-54
Explanation

Explanation/Reference:

QUESTION 21
Which statement about enrollment in the IP telephony PKI is true?

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: 37-54
Explanation

Explanation/Reference:
(Source: Understanding Cisco IP Telephony Authentication and Encryption Fundamentals)

QUESTION 22
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?

```
voice-card 0
 dspfarm
 dsp services dspfarm

sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 6.0
sccp

sccp ccm group 1
 associate ccm 2 priority 1
 associate profile 1 register HQ_XCODER

dsp farm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 2
 associate application SCCP
 no shutdown
```


A. The associate profile anddsp 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. Thesccp ccm group number needs to match the associate ccm 2 command.

D. The maximum sessions command must match the number ofcodecs configured under the dsp farm profile.
E. Thesccp ccm group number must match the voice-card number.

Correct Answer: B
Section: 55-72
Explanation

Explanation/Reference:

QUESTION 23
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: 55-72
Explanation

Explanation/Reference:

QUESTION 24
Which statement is not true about GARP?

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
Section: 55-72
Explanation

Explanation/Reference:
(Source: Hardening the IP Phone)

QUESTION 25
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: 55-72

Explanation

Explanation/Reference:

QUESTION 26
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: 55-72
Explanation

Explanation/Reference:

QUESTION 27
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: 55-72
Explanation

Explanation/Reference:

QUESTION 28
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: 55-72
Explanation

Explanation/Reference:
(Source: Configuring Cisco Unified Video Advantage)

QUESTION 29
What is the default value for the Drop Ad Hoc Conference service parameter?

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: 55-72
Explanation

Explanation/Reference:
(Source:Preventing Toll Fraud)