Vendor: Cisco

Exam Code: 300-075

Exam Name: Implementing Cisco IP Telephony and Video, Part 2 (CIPTV2)

Version: Demo

QUESTION 1

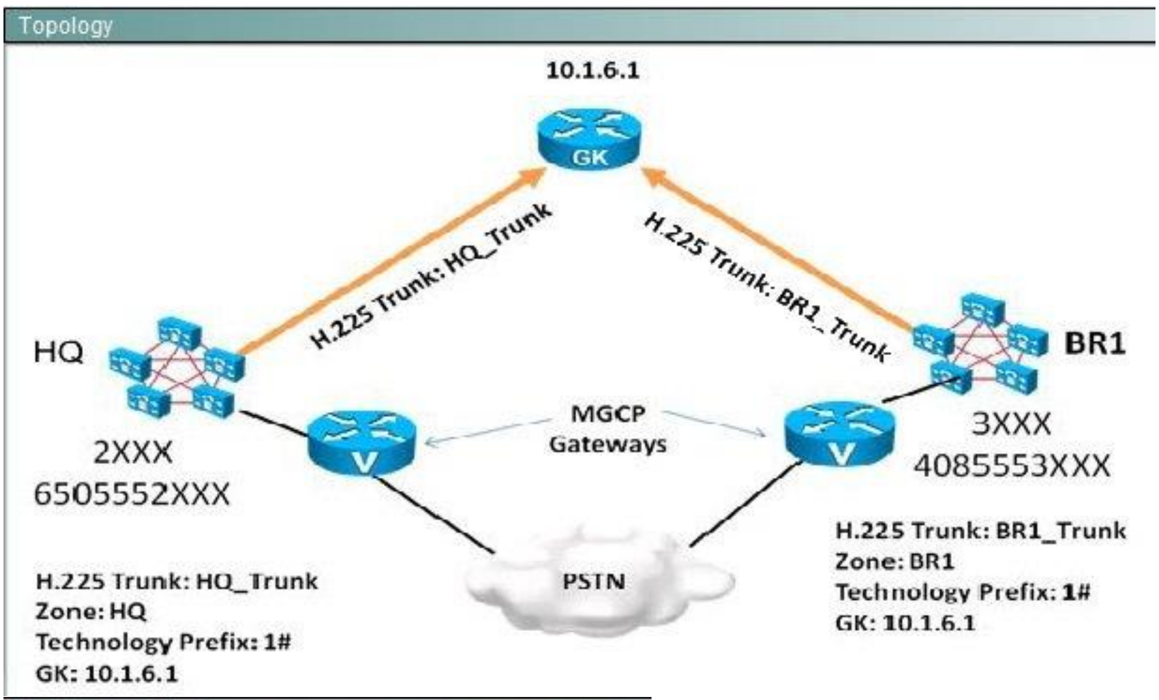
How is a SIP trunk in Cisco Unified Communications Manager configured for SIP precondition?

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

QUESTION 2

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?



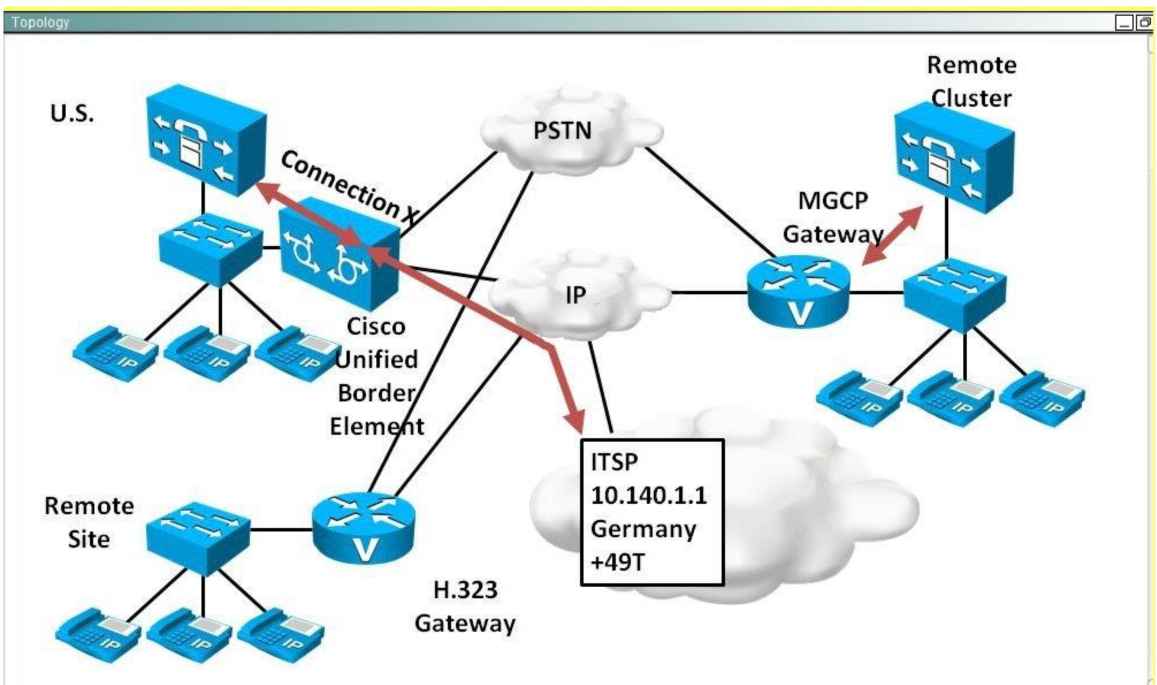
Gatekeeper Config

```
gatekeeper
 zone local HQ cisco.com 10.1.6.1
 zone local BR1 cisco.com
 zone prefix HQ 2...
 zone prefix BR1 3...
 gw-type-prefix 1#* default-technology
 no shutdown
```

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
Explanation:
http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/b03rtgrp.html#wpxr
ef46617

QUESTION 3
Refer to the following exhibits. 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?



CUBE Config

```
!
hostname HQ_gateway
!
card type e1 0 0
enable password 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 GigabitEthernet0/0.5
 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
 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
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.1.5.10
!
mgcp
```

```
ngcp call-agent 10.1.5.10 service-type mgcp version 0.1
ngcp rtp unreachable timeout 1000 action notify
ngcp modem passthrough voip mode nse
ngcp package-capability rtp-package
ngcp package-capability sst-package
ngcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
ngcp sdp simple
ngcp fax t38 ecm
ngcp rtp payload-type g726r16 static
ngcp behavior g729-variants static-pt

ngcp profile default

lial-peer voice 1111 voip
 session protocol sipv2
 incoming called-number .

lial-peer voice 222 voip
 session protocol sipv2
 destination-pattern +49T
 session target ipv4:10.140.1.1


gateway

gatekeeper
 shutdown

  !
  line con 0
  line aux 0
  line vty 0 4
  !
  end
```

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

Explanation:
Incorrect answer: A, B, D
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 code is needed to route the call to the PSTN, but the PSTN switch does not expect that access code.
Link: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a03rp.html

QUESTION 4
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

QUESTION 5
Refer to the exhibit. The H.323 Gateway is showing status "unknown". Which statement is true?


Gateway Status

	Device Name	Description	Device Pool	Calling Search Space	Device Type
H.323	10.1.120.1		Default		H.323 Gateway
3825	HQ	HQ			Cisco 3825

Gateway Config

```
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname BR1
!
boot-start-marker
boot system flash:c2800nm-ipvoice_ivs-mz.124-22.T.bin
boot-end-marker
!
card type t1 0 0
logging message-counter syslog
enable password cisco123
!
no aaa 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 multicast-routing
no ipv6 cef
multilink bundle-name authenticated
!
!
!
isdn switch-type primary-ni
!
!
!
voice service voip
 allow-connections h323 to h323
!
!
!
!
!
voice class h323 1
  h225 timeout tcp establish 3
!
!
!
!
!
!
!
!
!
!
voice translation-rule 1
```

```
 rule 1 /^710(....$)/ /  /
 rule 2 /^212710(....$)/ /  /
!
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...$/ /212710&/
!
!
voice translation-profile pstn-in
 translate called 1
!
voice translation-profile srst
 translate calling 3
 translate called 2
!
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
!
!
!
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
 description BR1 Phone1
 switchport access vlan 20
 switchport voice vlan 120
 spanning-tree portfast
```

```
!
interface FastEthernet0/1/1
 description BR1 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
 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 10.1.5.2
 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
 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 3000 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 intlPt
 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 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 0/0/0:23
 prefix 866
!
dial-peer voice 877 pots
 destination-pattern 91877.......
 port 0/0/0:23
 prefix 877
```

```
 prefix 877
!
dial-peer voice 888 pots
 destination-pattern 91888.......
 port 0/0/0:23
 prefix 888
!
dial-peer voice 11 pots
 corlist outgoing ldPt
 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
 max-dn 8 dual-line
 after-hours block pattern 1 91900 7-24
 voicemail 916506032000
 call-forward busy 916506032000
 call-forward noan 916506032000 timeout 7
 moh music-on-hold.au
 multicast moh 239.1.1.1 port 16384 route 10.1.20.1 10.1.120.1
!
!
line con 0
 location CIEV2-GA-LAB04, SJ
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password cisco123
 login
!
scheduler allocate 20000 1000
end
```

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.323 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:
Incorrect answer: A, B, C, E
After a gateway is registered with Cisco Unified Communications Manager, gateway registration status may display in Cisco Unified Communications Manager Administration as unknown.
Link:
http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b06gtway.html

QUESTION 6
Which device is needed to integrate H.320 into the Cisco video solution?

A. video gateway
B. MGCP gateway

C. H.323 gatekeeper
D. MCU

Correct Answer: C
Explanation:
Incorrect answer: A, B, D
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

QUESTION 7
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.)

```
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-6,24

interface Serial1/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
 no cdp enable

interface Vlan120
 ip address 10.2.120.1 255.255.255.0
 h323-gateway voip bind srcaddr 10.2.120.1

voice-port 1/0:23

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

dial-peer voice 23000 voip
 destination-pattern 2...
 session target ipv4:10.1.5.11
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
```

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: BE
Explanation:
Incorrect answer: A, C, D
Choose the number of significant digits to collect, from 0 to 32. Cisco Unified Communications Manager counts significant digits from the right (last digit) of the number that is called.
Link: http://cisco.biz/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b06trunk.html

QUESTION 8
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?

```
dial-peer voice 7 pots
 destination-pattern 9[2-9]......
 port 1/0:23
!
dial-peer voice 11 pots
 destination-pattern 91[2-9]..[2-9]......
 prefix 1
 port 1/0:23
```

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
Explanation:
Incorrect answer: B, C, D, E
Destination pattern is 91, HQ office area code is 201 .

QUESTION 9
How many active gatekeepers can you define in a local zone?

A. 1
B. 2
C. 5
D. 10

E. 15
F. unlimited

Correct Answer: A

QUESTION 10
Which two statements about the functionality of a gatekeeper are true? (Choose two.)

A. Cisco Unified Communications Manager has gatekeeper functionality built in.
B. Cisco Unified Communications Manager registers with a gatekeeper via SIP.
C. Cisco Unified Communications Manager registers with a gatekeeper via H.323.
D. A gatekeeper can enable CAC and AAR.
E. A gatekeeper can enable CAC, but not AAR.

Correct Answer: CE

QUESTION 11
Which option describes a function of SIP preconditions?

A. SIP preconditions enable end-to-end RSVP over an SIP trunk.
B. SIP preconditions enable RSVP between Cisco IP Phones.
C. SIP preconditions can be enabled in a gatekeeper.
D. SIP preconditions enable end-to-end RSVP for calls through the PSTN.

Correct Answer: A

QUESTION 12
Which statement about the function of a gatekeeper is true?

A. A gatekeeper improves call routing between servers within a single Cisco Unified Communications Manager cluster.
B. A gatekeeper can replace the dial plan of a Cisco Unified Communications Manager cluster.
C. A gatekeeper can simplify the dial plan between many different Cisco Unified Communications Manager clusters.
D. Gatekeepers can be implemented to deploy RSVP-based CAC.

Correct Answer: C

QUESTION 13
For which VoIP protocol does a gatekeeper provide address translation and control access?

A. H.323
B. SIP
C. Skinny
D. H.248

Correct Answer: A

QUESTION 14
Which CAC configuration on a gatekeeper restricts to 10 G.711 audio calls?

A. Use the command bandwidth 10.
B. Use the command bandwidth 1280.
C. Use the command bandwidth 160.
D. Use the command bandwidth session 10.

Correct Answer: B

QUESTION 15
If the device pool in the phone record does not match the device pools in the matching subnet, what will the system consider the phone to be?

A. roaming
B. unregistered
C. unknown
D. new device

Correct Answer: A

QUESTION 16
What user profile is used to define the settings for a user on login?

A. Device Profile
B. Group Profile
C. Pool Profile
D. Specific Profile

Correct Answer: A

QUESTION 17
Which statement about technology implementation strategy is true?

A. Cisco Unified Communications Manager Express can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
B. Cisco Unified Communications Manager Express in SRST mode can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
C. SRST can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
D. SRST and MGCP fallback can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.

Correct Answer: A

QUESTION 18
Which commands are needed to configure Cisco Unified Communications Manager Express in SRST mode?

A. telephony-service and srst mode
B. telephony-service and moh
C. call-manager-fallback and srst mode
D. call-manager-fallback and voice-translation

Correct Answer: A

QUESTION 19
What command is used to map internal extensions to the corresponding E.164 PSTN number when using Cisco Unified Communications Manager Express in SRST mode?

A. ephone-dn
B. dialplan-pattern
C. number
D. number-e.164
E. ephone-transnumber

Correct Answer: B

QUESTION 20
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
Explanation:
Incorrect answer: B, C, D
The CAPF enrollment process is as follows:
1. The IP phone generates its public and private key pairs.
2. The IP phone downloads the certicate 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 certicate for the IP phone signed with its private key.
5. The CAPF sends the signed certicate to the IP phone.
Link: http://my.safaribooksonline.com/book/certification/cipt/9781587052613/understanding-cisco-ip-telephony-authentication-and-encryption-fundamentals/584.