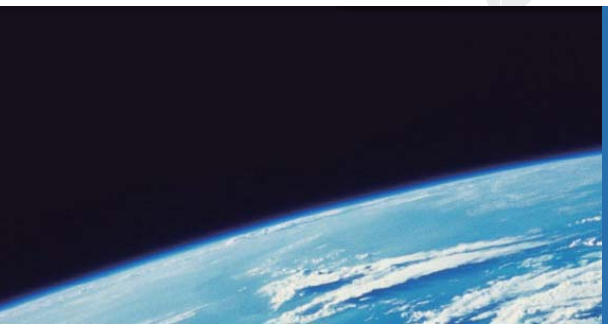

ITTEST

QUESTION & ANSWER

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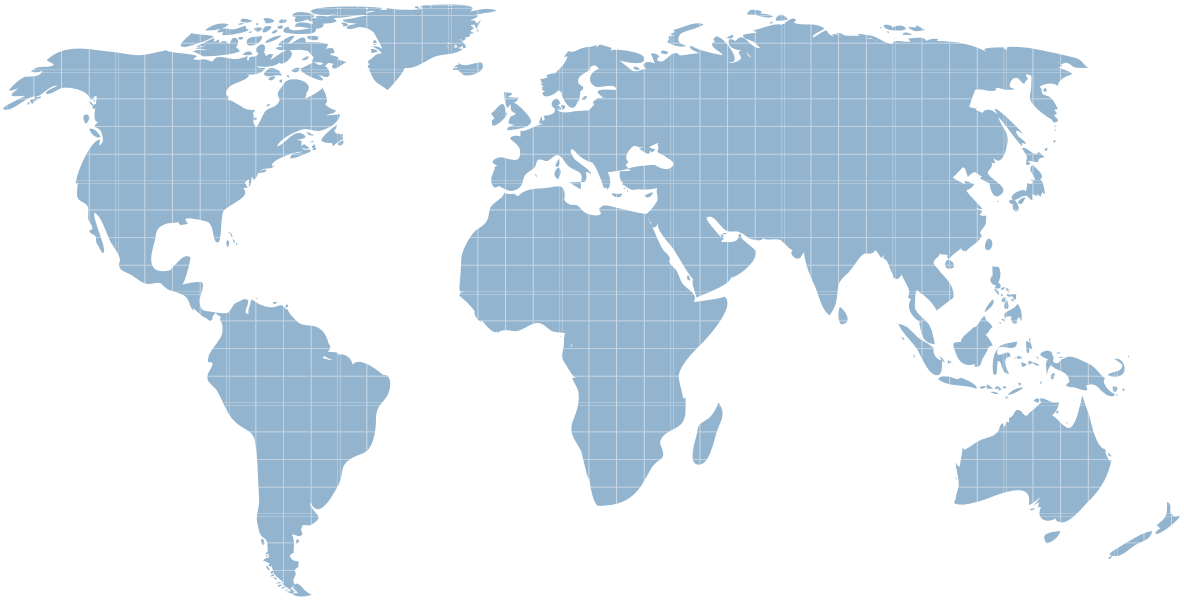

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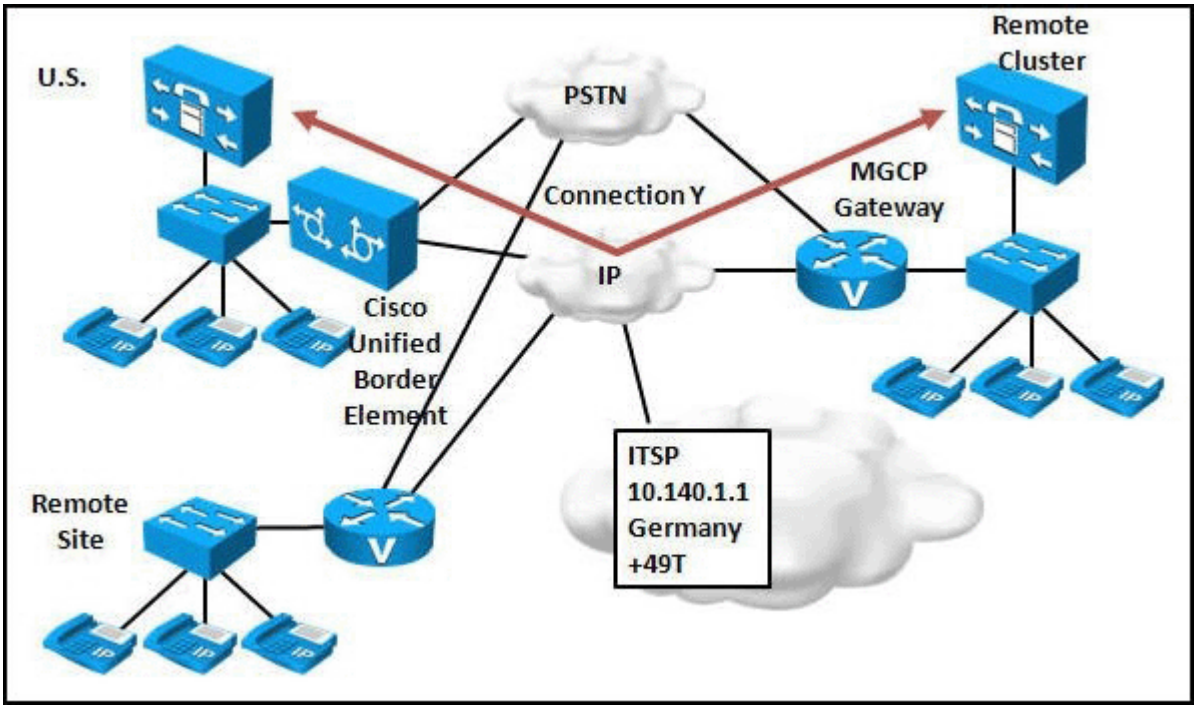
Exam : **642-457**

Title : Implementing Cisco Unified Communications

Version : DEMO

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
Answer: A
Explanation:
Incorrect answer: BCDE
If the dial plan contains overlapping patterns, Cisco Unified Communications Manager does not route the call until the interdigit timer expires (even if it is possible to dial a sequence of digits to choose a current match). Check this check box to interrupt interdigit timing when Cisco Unified Communications Manager must route a call immediately. By default, the Urgent Priority check box displays as checked. Unless your dial plan contains overlapping patterns or variable length patterns that contain!, Cisco recommends that you do not uncheck the check box.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsintrcm.html

2.Refer to the exhibit.



Which trunks 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.
Answer: D

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
Answer: A,B
Explanation:
Incorrect answer: CDE
All protocols require that either a signaling interface (trunk) or a gateway be created to accept and originate calls. Device mobility allows Cisco Unified Communications Manager to determine whether the phone is at its home location or at a roaming location. Cisco Unified Mobility gives users the ability to redirect incoming IP calls from Cisco Unified Communications Manager to different designated phones, such as cellular phones.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08sip.html#wpxref

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.
Answer: A

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.
Answer: B
Explanation:
Incorrect answer: ACDE
For the destination to be represented in a global form common to all cases, we must adopt a global form of the destination number from which all local forms can be derived. The + sign is the mechanism used by the ITU's E.164 recommendation to represent any PSTN number in a global, unique way. This form is sometimes referred to as a fully qualified PSTN number.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/7x/dialplan.html#wp1153205