

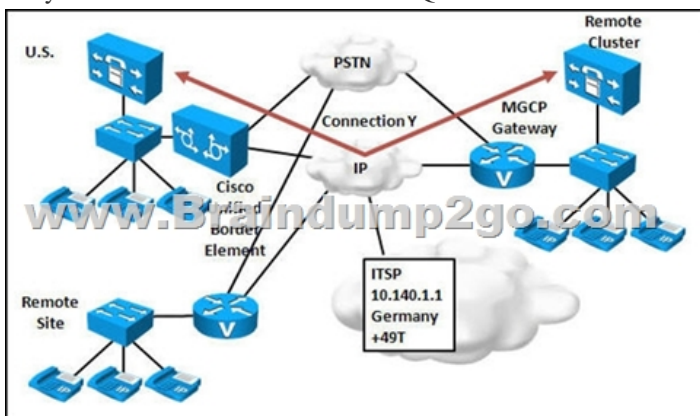
## [Dec-2017 Exam Pass 100% ! Braindump2go 300-075 Dumps 391Q Instant Download [Q187-Q197]

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**QUESTION 187** What component acts as a user agent for both ends of a SIP call while Cisco Unified SIP SRST is providing failover during a WAN outage?  
A. B2BUA  
B. SIP server  
C. SIP proxy  
D. SIP SRST router  
E. SIP registrar  
Answer: A  
**QUESTION 188** Which two configurations are needed to implement SRST in Cisco Unified Communications Manager? (Choose two.)  
A. SRST Gateway setting in Cisco Unified Communications Manager  
B. SRST Reference configured in Cisco Unified Communications Manager  
C. Device Pool SRST Reference setting  
D. Call Manager Group setting  
E. Cisco Unified Communications Locations setting  
Answer: BC  
**QUESTION 189** Which of the following are two functions that ensure that the telephony capabilities stay operational in the remote location Cisco Unified SRST router? (Choose two.)  
A. Automatically detecting a failure in the network.  
B. Initiating a process to provide call-processing backup redundancy.  
C. Notifying the administrator of an issue for manual intervention.  
D. Proactively repairing issues in the voice network.  
Answer: AB  
**QUESTION 190** Which three of the following are steps in configuring MGCP fallback and Cisco Unified SRST? (Choose three.)  
A. Define the SRST reference for phones in the Device Pool configuration  
B. Enable and configure the MGCP fallback and Cisco Unified SRST features on the IOS gateways.  
C. Implement a simplified SRST dial plan on the remote-site-gateways to ensure connectivity for remote-site phones in SRST mode.  
D. Enable SIP trunking between both remote and hub sites to provide mesh coverage.  
E. Define the SRST reference in the configuration on the IP Phones.  
F. Enable and configure the MGCP fallback on the IOS gateway but not Cisco Unified SRST since it is enabled automatically.  
Answer: ABC  
**QUESTION 191** 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: 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.  
**QUESTION 192** 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

**QUESTION 193** 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: AB  
Explanation: 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.

QUESTION 194 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

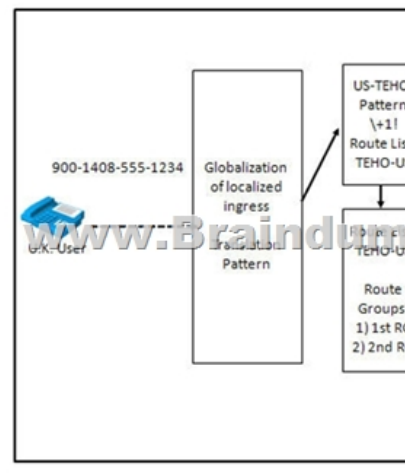
QUESTION 67 When an incoming PSTN call arrives at an MGCP gateway, how does the calling number get normalized to a global E.164 number with the + prefix in Cisco Unified Communications Manager?  
A. Normalization is done using translation patterns.  
B. Normalization is done using route patterns.  
C. 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.  
Answer: D

Explanation: Configuring calling party normalization alleviates issues with toll bypass where the call is routed to multiple locations over the IP WAN. In addition, it allows Cisco Unified Communications Manager to distinguish the origin of the call to globalize or localize the calling party number for the phone user.

QUESTION 195 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.  
Answer: A

QUESTION 196 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?  
A. Calling number localization is done using translation patterns.  
B. Calling number localization is done using route patterns.  
C. Calling number localization is done by configuring a calling party transformation CSS at the phone.  
D. Calling number localization is done by configuring a calling party transformation CSS at the gateway.  
E. Calling number localization is done by configuring the phone directory number in a localized format.  
Answer: C

QUESTION 197 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 +!, how should the translation pattern be configured?



- A. 9001.4085551234 with the Called Party Transformation: Discard Digits PreDotPrefix Digits Outgoing Calls: +B.
  - 9.0014085551234 with the Called Party Transformation: Discard Digits PreDotPrefix Digits Outgoing Calls: +1C.
  - 900.14085551234 with the Called Party Transformation: Discard Digits PreDotPrefix Digits Outgoing Calls: +1D.
  - 900.14085551234 with the Called Party Transformation: Discard Digits PreDotPrefix Digits Outgoing Calls: +E. 001.4085551234 with the Called Party Transformation: Prefix Digits Outgoing Calls: +
- Answer: D
- Explanation: 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.
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