[2017 New 300-075 New Questions For Passing The 300-075 Certification Exam (176-200)

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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.Answer: ABC QUESTION 177Which two configurations provide the best SIP trunk redundancy with Cisco Unified Communications Manager? (Choose two.) A. Configure all SIP trunks with DNS SRVB. Configure all SIP trunks with Cisco Unified Border ElementC. Configure all SIP trunks to point to a SIP gatewayD. Configure SIP trunks to be members of route groups and route listsE. Configure all SIP trunks to allow TCP ports 5060F. Configure all SIP trunks to point to a gatekeeper through SIP to H.323 gateway Answer: ADExplanation: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 178When you configure Cisco Unified Communications Manager, you need to configure the router for Survivable Remote Site Telephony in case the Cisco Unified Communications Manger stops working. On which two factors would the number of IP phones and Directory Numbers that can register to the SRST router depend? (Choose two.) A. The protocol that is used in Cisco Unified Communications ManagerB. Cisco Unified Communications Manager versionC. Cisco IOS Software versionD. WAN link bandwidthE. capacity of the Cisco Media Convergence ServerF. router platform Answer: CF QUESTION 179Which remote-site redundancy technology fails over to POTS dial peers from the Cisco Unified Communications Manager dial plan during a WAN failure? A. MGCP fallbackB. H.323 fallbackC. SCCP fallbackD. SIP fallback Answer: A QUESTION 180How does the system intelligently shift call processing upon restoration of WAN connectivity? A. automatically back to the primary Cisco Unified Communications Manager clusterB. manually back to the primary Cisco Unified Communications Manager clusterC. automatically back to the secondary Cisco Unified Communications Manager clusterD. manually back to the secondary Cisco Unified Communications Manager cluster Answer: A QUESTION 181Which option configures call preservation for H.323-based SRST mode? A. voice service voip h323 call preserveB. call preservation not possible with H.323C. call-manager-fallback preserve-callD. dial-peer voice 1 voip call preserve Answer: A QUESTION 182Which configuration command disables the secondary dial tone on the branch office ISR for users calling from the PSTN into the branch office during a WAN failure? A. direct-inward-dialB. voice translation-ruleC. incoming called-numberD. application Answer: A QUESTION 183A Cisco Unified Communications Manager cluster is installed in headquarters only.How can international calls be blocked while national calls are allowed for branch office Cisco IP Phones during a WAN failure? A. Configure CSS and partitions in Cisco Unified Communications Manager and apply the CSS and partitions to the SRST ISR.B. Configure CSS and partitions in the SRST ISR.C. Configure COR in the SRST ISR.D. Configure voice translations in the SRST ISR. Answer: C QUESTION 184Which command can be used to manually send the MGCP gateway to register with the secondary Cisco Unified Communications Manager server? A. ccm-manager switchover-to-backupB. mgcp use backupC. ccm-manager register backupD. not supported Answer: A QUESTION 185This is the configuration on the voice gateway: telephony-servicemax-ephones 30max-dn 60 preference 0srst mode auto-provision allsrst dn line-mode dualsrst dn template 3srst ephone description srst fallback auto-provision phonesrst ephone template 5 Which ephone-dn would be expected upon activation of SRST? A. ephone-dn 1 dual-linenumber 7001description 7001name 7001ephone-dn-template 5This DN is learned from srst fallback ephonesB. ephone-dn 1 dual-linenumber 7001description 7001name 7001ephone-dn-template 3This DN is learned from srst fallback ephonesC. ephone-dn 1number 7001description 7001name 7001ephone-dn-template 5This DN is learned from srst fallback ephonesD. ephone-dn 1number 7001description 7001name 7001ephone-dn-template 3This DN is learned from srst fallback ephones Answer: A QUESTION 186Which ability does the Survivable Remote Site Telephony feature provide? A. a means to allow the local site to continue to send and receive calls in the event of a WAN failureB. a means to route calls on-net through other sites during high utilization periodsC. a method that allows for backup calls in the event that your gateway failsD. the ability to force a call out of a certain trunk when the Cisco Unified Communications Manager is being upgraded Answer: A QUESTION 187What 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. B2BUAB. SIP serverC. SIP proxyD. SIP SRST routerE. SIP registrar Answer: A QUESTION 188Which two configurations are needed to implement SRST in Cisco Unified Communications Manager? (Choose two.) A. SRST Gateway setting in Cisco Unified Communications ManagerB. SRST Reference configured in Cisco Unified Communications ManagerC. Device Pool SRST Reference settingD. Call Manager Group settingE. Cisco Unified Communications Locations setting Answer: BC QUESTION 189Which 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 190Which 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 configurationB. 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 191Which 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 longC. Use nested translation patterns to eliminate inter-digit timeoutD. Use the @macro on the route patternE. Use MGCP gateways, which support variable-length dial plans Answer: AExplanation: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 192Refer 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 clusterD. 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 193Which two features require or may require configuring a SIP trunk? (Choose two.) A. SIP gatewayB. Call Control Discovery between a Cisco Unified Communications Manager and Cisco Unified Communications Manager ExpressC. Cisco Device MobilityD. Cisco Unified MobilityE. registering a SIP phone Answer: ABExplanation: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 194A 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 gatewayB. Cisco 3825C. 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 67When 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: DExplanation: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 195When 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 196Which 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 197Refer 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: DExplanation: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. QUESTION 198Refer 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. Answer: CExplanation:The route group points to one or more gateways and can choose the gateways for call routing based on preference. The route group can serve as a trunk group by directing all calls to the primary device and then using the secondary devices when the primary is unavailable. One or more route lists can point to the same route group. QUESTION 199Refer 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 PreDotPrefix Digits Outgoing Calls: +B. +1.! with the following Called Party Transformation:Discard Digits PreDotPrefix Digits Outgoing Calls: NoneC. +408.! with the following Called Party Transformation:Discard Digits PreDotPrefix Digits Outgoing Calls: 1D. +1408.! with the following Called Party Transformation:Discard Digits PreDotPrefix Digits Outgoing Calls: NoneE. +1.408! with the following Called Party Transformation:Discard Digits PreDotPrefix Digits Outgoing Calls: None Answer: D QUESTION 200Refer 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. 9B. 91C. noneD. +E. +1 Answer: C Your focus should be getting the best dumps to prepare for 300-075 exam. That is where Lead2pass comes in. We have collected an extensive library of exam dumps from Cisco certification. 300-075 new questions on Google Drive: https://drive.google.com/open?id=0B3Syig5i8gpDVEF4X2NxM3FVclk 2017 Cisco 300-075 exam dumps (All 356 Q&As) from Lead2pass: http://www.lead2pass.com/300-075.html [100% Exam Pass Guaranteed]