Implementing Cisco Advanced Call Control and Mobility Services v1.1 (300-815)

Exam Description: Implementing Cisco Advanced Call Control and Mobility Services v1.1 (CLACCM 300-815) is a 90-minute exam associated with the CCNP Collaboration Certification. This exam certifies a candidate's knowledge of advanced call control and mobility services, including signaling and media protocols, CME/SRST gateway technologies, Cisco Unified Board Element, call control and dial planning, Cisco UCM Call Control, and mobility. The course, Implementing Cisco Advanced Call Control and Mobility Services, helps candidates to prepare for this exam.

The following topics are general guidelines for the content likely to be included on the exam. However, other related topics may also appear on any specific delivery of the exam. To better reflect the contents of the exam and for clarity purposes, the guidelines below may change at any time without notice.

20% 1.0 Signaling and Media Protocols

- 1.1 Troubleshoot these elements of a SIP conversation
  - 1.1.a Early media
  - 1.1.b PRACK
  - 1.1.c Mid-call signaling (hold/resume, call transfer, conferencing)
  - 1.1.d Session timers
  - 1.1.e UPDATE
- 1.2 Troubleshoot these SIP protocol elements
  - 1.2.a DTMF
  - 1.2.b Call set up and tear down
- 1.3 Troubleshoot media establishment

10% 2.0 Gateway Technologies

- 2.1 Configure Cisco UCME for SIP phone registration
- 2.2 Configure Cisco UCME dial plans
- 2.3 Configure advanced SRST features
- 2.4 Configure SIP SRST gateway

15% 3.0 Cisco Unified Border Element

- 3.1 Configure these Cisco Unified Border Element dial plan elements
  - 3.1.a DTMF
  - 3.1.b Voice translation rules and profiles
  - 3.1.c Codec preference list
  - 3.1.d Dial peers
  - 3.1.e SIP and SDP header manipulation with SIP profiles
  - 3.1.f Signaling and media bindings
  - 3.1.g Toll fraud prevention
  - 3.1.h Multiple trunks using tenants

3.1.i Registration-based SIP trunks (local gateway)
3.1.j SIP TLS and SRTP

3.2 Troubleshoot these Cisco Unified Border Element dial plan elements
3.2.a DTMF
3.2.b Voice translation rules and profiles
3.2.c Codec preference list
3.2.d Dial peers
3.2.e SIP and SDP header manipulation with SIP profiles
3.2.f Signaling and media bindings

3.3 Describe call recording options
3.3.a Network-based recording
3.3.b SIPREC

3.4 Describe VoIP Trace for troubleshooting voice calls

25% 4.0 Call Control and Dial Planning
4.1 Configure these globalized call routing elements in Cisco UCM
4.1.a Translation patterns
4.1.b Route patterns
4.1.c SIP route patterns
4.1.d Transformation patterns
4.1.e Standard local route group
4.1.f TEHO
4.1.g SIP trunking

4.2 Troubleshoot these globalized call routing elements in Cisco UCM
4.2.a Translation patterns
4.2.b Route patterns
4.2.c SIP route patterns
4.2.d Transformation patterns
4.2.e Standard local route group
4.2.f TEHO
4.2.g SIP trunking

20% 5.0 Cisco Unified CM Call Control Features
5.1 Troubleshoot Call Admission Control (exclude RSVP)
5.2 Configure ILS, URI synchronization, and GDPR
5.3 Configure hunt groups
5.4 Configure call queuing
5.5 Configure time of day routing
5.6 Configure supplementary functions
5.6.a Call park
5.6.b Meet-me
5.6.c Call pick-up

5.7 Describe call recording options

5.7.a Built-in-bridge
5.7.b Gateway
5.7.c SPAN-based

10% 6.0 Mobility

6.1 Configure Cisco UCM Mobility
- 6.1.a Unified Mobility
- 6.1.b Extension Mobility

6.2 Troubleshoot Cisco UCM Mobility
- 6.2.a Unified Mobility
- 6.2.b Extension Mobility

6.3 Describe Cloud Mobility