Implementing Cisco Advanced Call Control and Mobility Services v1.0 (300-815)

Exam Description: Implementing Cisco Advanced Call Control and Mobility Services v1.0 (CLACCM 300-815) is a 90-minute exam associated with the CCNP Collaboration Certification. This exam tests 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 Unified CM 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 H.323 protocol elements
  - 1.2.a DTMF
  - 1.2.b Call set up and tear down
- 1.3 Troubleshoot media establishment

10% 2.0 CME/SRST Gateway Technologies

- 2.1 Configure Cisco Unified Communications Manager Express for SIP phone registration
- 2.2 Configure Cisco Unified CME dial plans
- 2.3 Implement toll fraud prevention
- 2.4 Configure these advanced Cisco Unified CME features
  - 2.4.a Hunt groups
  - 2.4.b Call park
  - 2.4.c Paging
- 2.5 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 Header and SDP manipulation with SIP profiles
3.1.f Signaling and media bindings

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 Header and SDP manipulation with SIP profiles
3.2.f Signaling and media bindings

25% 4.0 Call Control and Dial Planning

4.1 Configure these globalized call routing elements in Cisco Unified Communications Manager
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 Unified Communications Manager
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

10% 6.0 Mobility

6.1 Configure Cisco Unified Communications Manager Mobility
6.1.a Unified Mobility
6.1.b Extension Mobility

6.1.c Device Mobility

6.2 Troubleshoot Cisco Unified Communications Manager Mobility

6.2.a Unified Mobility

6.2.b Extension Mobility

6.2.c Device Mobility