

Installing and Configuring Cisco Cube/Gateways (ICCC)

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In this 3-day Cisco Course, students will learn how to deploy Voice Gateways/CUBE and set up Cisco Unified Communication Manager (CUCM) to deploy SIP Trunking. This course offers a comprehensive, hands-on exploration of Cisco Unified Border Element (CUBE) technologies, architecture, and deployment in real-world enterprise and cloud environments. The curriculum includes configuration practices, signaling protocol deep-dives, high availability, advanced routing, Webex Calling integration, and robust troubleshooting methodologies. The course is intended for network engineers, collaboration specialists, and enterprise voice architects.

How you'll benefit

This class will help you:

- Configure Cisco Unified Communication Manager 11.x
- Implement Cisco Unified Border Element
- Monitor and Troubleshoot on Cisco Unified Border Element

Why Attend with Current Technologies CLC

- Our Instructors are in the top 10% rated by Cisco
- Our Lab has a dedicated 1 Gig Fiber Connection for our Labs
- Our Labs run up to Date Code for all our courses

Who Should Attend

The primary audience for this course is as follows:

- Network Video Engineer
- Voice / UC / Collaboration / Communications Engineer
- Collaboration Tools Engineer
- Collaboration Sales / Systems Engineer

OUTLINE

Module 0: Introduction

Introduces learners to the Cisco CUBE training course by outlining the course structure, goals, and expectations. It includes brief participant and instructor introductions, an overview of the key learning objectives such as SIP trunking, CUBE integration, and troubleshooting, and ensures everyone is comfortable using Webex Meetings for interactive sessions. The module also covers basic administrative details like lab

Course Duration

3 days

Course Price

\$3,895.00 or 40 CLCs

Methods of Delivery

- Instructor Led
- Virtual ILT
- On-Site

Module 1: Cisco CUBE Overview

This foundational module introduces Cisco Unified Communications architecture and the critical role of CUBE as a Session Border Controller (SBC). Students will learn the difference between TDM gateways and IP-based border elements, understand VoIP signaling protocols like SIP and H.323, and identify use cases where CUBE acts as an intermediary for call control, security, and interoperability.

Key topics include:

- TDM vs. IP-based voice gateways
- VoIP signaling protocols and transport
- SBC roles and adoption trends
- CUBE in contact centers, Webex Calling, and SIP trunking environments
- Strategic deployment models (centralized, distributed, hybrid)

Module 2: CUBE Interconnections – SIP Gateways & Trunks

This module provides in-depth instruction on configuring Cisco Unified Communications Manager (CUCM) to interoperate with CUBE using SIP trunks. Students will explore region settings, device pools, media resources (like MTP and MoH), and security profiles required for a secure and functional trunking interface.

Key topics include:

- CUCM audio codec preferences
- Device pool and region configuration
- Media resource group creation
- SIP trunk profiles, security settings, and route patterns
- Trunk path selection logic

Module 3: Gateway Dial Plans

Students will develop a deep understanding of call leg structure, dial peer logic, and SIP signaling flow. This module breaks down SIP addressing, codec negotiation, and RTP media security. Learners will also configure basic SIP call flows and explore advanced SDP scenarios such as early offer and delayed offer negotiation.

Key topics include:

- RTP and SRTP transport
- SIP message analysis (INVITE, 200 OK, BYE)
- SDP offer/answer mechanisms
- SIP UA timers, address resolution, and digit matching
- Gateway debugging, show commands, and performance tuning

Module 4: Configuring Cisco CUBE

This module is hands-on and configuration-heavy, guiding learners through building a full CUBE deployment. It includes dial-peer creation, provisioning policies, media manipulation, NAT traversal, DNS SRV resolution, and integration with the Webex Control Hub for gateway registration and management.

Key topics include:

- CUBE LAN/WAN call flow
- Dial-peer provisioning policies (From/To match)
- Media forking and SIPREC for call recording
- Call admission control strategies
- Webex Control Hub gateway registration
- Using DNS SRV records for load-balanced routing

Module 5: Configuring High Availability

CUBE High Availability is vital for enterprise-grade uptime and fault tolerance. This module focuses on HA deployment models, box-to-box redundancy with keepalives, SIP OPTIONS monitoring, and resilient media/call handling. Learners will configure and test CUBE HA for survivability and session preservation during failover.

Key topics include:

- HA design strategies and prerequisites
- B2B (box-to-box) redundancy with crossover/PoE
- OOD (Out-of-dialog) SIP OPTIONS for monitoring
- Protected mode for ASR platforms
- Use of DNS SRV for failover

Module 6: CUBE Security Settings

This module covers security considerations for SIP gateways, focusing on protecting signaling, media, and control planes. Students will implement TLS with SRTP, configure ACLs and firewall rules, enable SIP digest authentication, and isolate VRF-based call routing for multi-tenant or segmented designs.

Key topics include:

- Five-layer CUBE security model
- ACL and firewall deployment strategies
- SIP TLS and SRTP passthrough
- PKI Trustpoint and certificate operations
- Toll fraud prevention
- Multi-VRF call routing

Module 7: Webex Calling PSTN Gateway Troubleshooting

Troubleshooting hybrid Webex Calling with CUBE as a Local Gateway requires deep visibility across both control and media planes. This advanced module presents real-world scenarios, diagnostic tools, and workflows to troubleshoot registration failures, SIP misconfiguration, call drops, and media quality issues using logs, debugs, and analyzers.

Key topics include:

- Local Gateway onboarding and registration
- Control Hub diagnostics and config validation
- Collaboration Solutions Analyzer (CSA)
- Debugs and trace interpretation (VoIP Trace, Wireshark, TranslatorX)
- STUN/ICE signaling verification
- Call quality metrics and packet loss analysis