Cisco SIP, CUBEs and Gateways

Durée: 5 Jours   Réf de cours: CSCGW   Méthodes d'apprentissage: Classe à distance

Résumé:

In this course, you will focus on the legacy gateway and router portions of IP Telephony. You will gain extensive experience with the configuration of legacy analog telephony technologies such as Foreign Exchange Station (FXS), Foreign Exchange Office (FXO), Primary Rate Interface (PRI). In addition to legacy technologies you will gain hands on experience with CUBE and SIP protocols. You will build a working Cisco Unified Communications Manager which will support all major gateway protocols such as MGCP, H.323, and SIP. Troubleshooting will be addressed as a gateway level including common debug techniques and commands.
You'll gain an understanding of converged voice and data networks as it relates to gateway design and deployment. You will gain comprehensive hands-on experience configuring and deploying Gateways, CUBEs, Quality of Service, and troubleshooting in VoIP networks.
In addition to the knowledge and skills required to integrate gateways into an enterprise VoIP network, you'll learn how to build and test sophisticated IP telephony dial plans that use both CUCM Dial Plan and Dial Peers at an IOS level which can be used as a template for a real deployment.
The course includes a comprehensive study of Quality of Service (QoS), in which you'll learn to configure QoS to support real-time traffic.
Looking for **CVOICE - Implementing Cisco Unified Communications Voice over IP and QoS v8.0** ? This has been retired by Cisco and **CSCGW - Cisco SIP, CUBEs and Gateways** is the recommended replacement.
A Global Knowledge Exclusive:You Get...
Enhanced content that exceeds standard authorized Cisco content
Only course dedicated to specific Gateway technologies and Quality of Service
World-class Certified Cisco Systems Instructors

Virtual Learning

This interactive training can be taken from any location, your office or home and is delivered by a trainer. This training does not have any delegates in the class with the instructor, since all delegates are virtually connected. Virtual delegates do not travel to this course, Global Knowledge will send you all the information needed before the start of the course and you can test the logins.

Public visé:

Network engineers, architects, and support staff who: Maintain and configure voice and data network devices: Are considering various methodologies to implement VoIP: Require a fundamental understanding of the issues and solutions related to implementation: Require a fundamental understanding of packet telephony technologies that are common for both enterprise and service provider applications.

Objectifs pédagogiques:

- **After completing this course you should have an understanding of:**
- VoIP, components of a VoIP network, VoIP protocols, special requirements for VoIP calls, and Codecs
- Configure gateway interconnections to support VoIP and PSTN calls
- Basic signaling protocols used on voice gateways
- Configure a gateway to support calls using different call control and signaling protocols
- Define a dial plan, describing the purpose of each dial plan component, and implement a dial plan on a voice gateway
- Implement a Cisco Unified Border Element (CUBE) gateway to connect to an Internet Telephony Service Provider
- Investigate the use of various traditional telephony connections, such as FXS, FXO, E&M, T1 (CAS and PRI), and E1 (CAS and PRI)
- Configure and troubleshoot Cisco's new ISR routers and explore their DSP configuration (PVDM3 cards)
- Configure H.323 gateways and review their functions and operation
- Configure Session Initiation Protocol (SIP) and Media Gateway Control Protocol (MGCP)
- Experience G.711, and G.729 voice coding schemes
- Configure Call Admission Control three different ways
- Configure proper Caller ID
- Experience real-world connections to PBXs, and the PSTN
- Configure your router/gateway equipment to connect to our public dial plan network using different call control protocols and procedures

Pré-requis:

Attendees should meet the following prerequisites:

- Working knowledge of networking fundamentals, including LANs, WANs, and IP switching and routing
- Ability to configure and operate Cisco routers and switches and to enable VLANs and DHCP
- Knowledge of traditional PSTN operations and technologies
- Attendance of CICD or CIPTV1 is recommended.
- CLFNDU - Comprendre les bases des solutions Cisco Collaboration

Test et certification

Recommended as preaparation for the following exams:

- There are no exams currently aligned to this course.

Contenu:

Introduction to Voice Gateways

- Cisco UC Networks and the Role of Gateways
- Gateway Call Routing and Call Legs
- Gateway Voice Ports Configuration
- DSP Functionality, Codecs, and Codec Complexity

VoIP Call Legs

- VoIP Call Leg Characteristics
- VoIP Media Transmission
- H.323 Signaling Protocol
- SIP Signaling Protocol
- MGCP Signaling Protocol
- Requirements for VoIP Call Legs
- VoIP Call Legs Configuration

Dial Plan Implementation

- Call Routing and Dial Plans
- Digit Manipulation
- Path Selection Configuration
- Calling Privileges Configuration

Gatekeeper and CUBE Implementation

- Fundamentals of Gatekeepers
- Cisco Unified Border Element

QoS

- QoS Mechanisms and Models
- Classification, Marking, and Link Efficiency Mechanisms
- Managing Congestion and Rate Limiting
- Cisco AutoQoS

Labs - Part 1

- Lab 1: Remote Labs Connectivity
- Lab 2: Topology and Deployment Walkthrough
- Lab 3: CUCM Disaster Recovery
- Lab 4: MGCP Gateways
- Lab 5: Route Groups and Route Lists
- Lab 6: CUCM Dial Plan
- Lab 7: IP Phone Registration
- Lab 8: 9951 Registration
- Lab 9: Unified FX
- Lab 10: Traditional Route Patterns and Dial Plan Testing with MGCP
- Lab 11: CUBE and SIP Trunks
- Lab 12: Traditional Route Patterns and Dial Plan Testing with SIP
- Lab 13: H.323 Gateways
- Lab 14: Traditional Route Patterns and Dial Plan Testing with H.323

Labs Part 2

- Lab 15: Analog FXO
- Lab 16: Traditional Route Patterns and Dial Plan Testing with FXO
- Lab 17: Analog FXS
- Lab 18: Traditional Route Patterns and Dial Plan Testing with FXS
- Lab 19. PRI and T1-CAS
- Lab 20. Traditional Route Patterns and Dial Plan Testing with PRI and T1-CAS
- Lab 21: Deep Dive - VoIP Dial Peers
- Lab 22: Deep Dive - PSTN Dial Peers
- Lab 23: Deep Dive – Dial Peer Digit Manipulation
- Lab 23: IOS Conference Bridges
- Lab 24: IOS Transcoding
- Lab 25: IOS Media Termination Points
- Lab 26: IOS Gatekeepers
- Lab 27: Call Admission Control
- Lab 28: Configuring AutoQoS
- Lab 29: Configuring WAN QoS Policies
- Lab 30: Configuring LAN QoS Policies

Autres moyens pédagogiques et de suivi:

• Compétence du formateur : Les experts qui animent la formation sont des spécialistes des matières abordées et ont au minimum cinq ans d'expérience d'animation. Nos équipes ont validé à la fois leurs connaissances techniques (certifications le cas échéant) ainsi que leur compétence pédagogique.
• Suivi d'exécution : Une feuille d'émargement par demi-journée de présence est signée par tous les participants et le formateur.
• En fin de formation, le participant est invité à s'auto-évaluer sur l'atteinte des objectifs énoncés, et à répondre à un questionnaire de satisfaction qui sera ensuite étudié par nos équipes pédagogiques en vue de maintenir et d'améliorer la qualité de nos prestations.

Délais d'inscription :

• Vous pouvez vous inscrire sur l'une de nos sessions planifiées en inter-entreprises jusqu'à 5 jours ouvrés avant le début de la formation sous réserve de disponibilité de places et de labs le cas échéant.