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# Cisco SIP, CUBEs and Gateways

**Course#:CSCGW**
**Duration:5 Days**
**Price:0.00**

## Course Description

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), and 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.

Youll 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, youll 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 youll learn to configure QoS to support real-time traffic.

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

Why Take CSCGW from Global Knowledge?
Every pod has internal and external phones, and just like in a real network, the same simulated public switched telephone network (PSTN) is accessible through all clusters providing failover scenarios for bandwidth and connectivity problems. To more accurately reflect real-world scenarios, you will configure the gateway connections to simulated PBX systems.

We have set ourselves apart from other Cisco training providers by enhancing our CSCGW hands-on labs to include a real dial plan and Class of Service for calling out to the PSTN. Our voice network labs use the latest hardware and software so you will gain experience with the recent stable IOS release (15.X IOS M currently). Plus, each pod contains the following gateway cards for student configuration: 2xFXS, 2xFXO, and 2xT1 ports (PRI and T1-CAS) as well as serial ports for WAN connectivity. All of our IP telephony courses provide a realistic simulated PSTN accessible through both PRI and FXO ports. You will build and test a real dial plan including:

911
three-digit service codes: 411, 511, and so forth
seven-digit local numbers
10-digit local numbers
11-digit long distance numbers
International numbers
Configure and test all dial peers as appropriate

## Objectives

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, EM, T1 (CAS and PRI), and E1 (CAS and PRI)
Configure and troubleshoot Ciscos 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

## Audience

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

## 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

## Content

Classroom Live Outline

1.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

2.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

3.Dial Plan Implementation

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

4.Gatekeeper and CUBE Implementation

Fundamentals of Gatekeepers
Cisco Unified Border Element

5.QoS

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

Classroom Live Labs

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
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