Cisco Voice over Frame Relay, ATM, and IP

Course Summary

Description

CVOICE lays the foundation for gaining hand-on skills and significant understanding of packet telephony by presenting the technologies that are common for both Enterprise and Service Provider students. The course also teaches students how to use the available Cisco tools to find the information needed to accomplish their everyday tasks. Since no two networks are alike, this approach enables a student to apply the knowledge gained in this course to their specific needs.

Objectives

At the end of this course, students will be able to:

- Identify the components, processes, and features of traditional telephony networks that provide end-to-end call functionality
- Describe two methods of call control used on voice and data networks and provide one example of a protocol for each
- List at least five components or capabilities that are required to provide integrated voice and data services in campus LAN, enterprise, and service provider environments
- Select the appropriate analog voice connection to a Cisco device given the types of analog connections and their susceptibility to line quality problems
- Choose a voice compression scheme that best suits your needs given the fundamentals of digital voice encoding
- Describe the appropriate signaling method to deploy in a telephony system given the type of signaling: between PBXs; between PBXs and central offices; or specialized, such as ISDN
- Implement an effective method of transporting fax and modem traffic over a Voice over IP network given the standard implementations of fax and the methods used to transport modem traffic

Topics

- Introducing Voice Over IP
- Voip Network Technologies
- VoIP Network Architectures
- Building Scalable Dial Plans
- Calculating Bandwidth Requirements
- Allocating Bandwidth for Voice and Data Traffic
- Considering Security in VoIP Networks
- Configuring Voice Networks
- Configuring Voice Ports
- Adjusting Voice Interface Settings
- Configuring Dial Peers
- Configuring Voice Port Connections
- VoIP Signaling and Call Control
- Introducing Signaling and Call Control
- Introducing H.323
- Deploying and Configuring H.323
- Configuring SIP
- Configuring MGCP
- Comparing Call Control Models
- Improving and Maintaining Voice Quality
- Designing for Optimal Voice Quality
- Implementing CAC

Prerequisites

Before attending this course students should have:

- Prior experience and knowledge of traditional PSTN operations, requirements of Voice over IP, and a basic understanding of VoIP benefits. In addition, to fully comprehend the concepts and technologies taught in this course, a working knowledge of LANs, WANs, and IP switching and routing is essential.
- Basic internetworking skills taught in the Interconnecting Cisco Network Devices training course, or equivalent knowledge, is considered the minimum knowledge needed for this course.

Duration

Four days