



Cisco Voice Over IP

Length
5 days

Format
Lecture/lab

Version
6.0

Course Description

CVOICE is the first course in the Cisco Voice Professional Curriculum. CVOICE teaches 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 Cisco tools to find the information needed to accomplish their everyday tasks.

Who Should Attend

CVOICE is designed for Cisco customers and Channel Partners who are working toward CCVP certification.

Recommended Prerequisites

- Successful completion of Cisco Voice Fundamentals or prior experience and knowledge of traditional PSTN operations, requirements of Voice over IP, and a basic understanding of VoIP benefits
- working knowledge of LANs, WANs, and IP switching and routing is essential
- Basic internetworking skills taught in the Interconnecting Cisco Network Devices (ICND) course, or equivalent knowledge

Related Courses

- Cisco IP Telephony Part 1 (CIPT1)
- Quality of Service (QOS)
- IP Telephony Troubleshooting (IPTT)
- Implementing Cisco Voice Gateways and Gatekeepers (GWGK)

CVOICE

Learning Objectives

After completing this course, you 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
- 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 for a Cisco device given the types of analog connections and their susceptibility to line quality problems
- Choose a voice compression scheme that best meets 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



Learning
Solutions



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Course Outline

Module 1: Introduction to VoIP

Lesson 1: Introducing VoIP Network Technologies

- Business CASE for VoIP
- VoIP Functions
- Components of a VoIP Network
- VoIP Signaling Protocols
- VoIP Service Considerations
- RTP and RTCP

Lesson 2: Introducing VoIP Network Architectures

- Centralized Network Architectures
- Distributed Network Architectures Using H.323
- Distributed Network Architectures Using SIP
- Comparing Network Architectures
- Interconnecting VoIP Protocols
- Understanding Gateways
- Guidelines for Selecting the Correct Gateway
- Enterprise Central and Remote Site Gateway
- Interconnection Requirements
- Service Provider Gateway Interconnection Requirements

Lesson 3: Building Scalable Dial Plans

- Numbering and Dial Plans
- Internal Numbering and Public Numbering Plan Integration
- Scalable Dial Plans
- Practice Item 1: Span Engineering Dial Plan Worksheet
- Scalable Dial Plan Attributes
- Enhancing and Extending an Existing Numbering Plan to Accommodate VoIP
- Accounting for Caller Mobility for 911 Services
- Practice Item 2: Numbering Plan for Span Engineering
- Chicago Airport Location
- London Location

Lesson 4: Calculating Bandwidth Requirements

- Codec Bandwidths
- Impact of Voice Samples and Packet Size on Bandwidth
- Data-Link Overhead
- Security and Tunneling Overhead
- Calculating the Total Bandwidth for a VoIP Call
- Effects of VAD on Bandwidth
- Practice Item: Span Engineering Voice Bandwidth Requirement

Lesson 5: Allocating Bandwidth for Voice and Data Traffic

- Sources of Traffic Statistics
- Network Objectives for Voice and Data
- Meeting the Current Network Objective
- Traffic Theory
- Busy Hour
- Erlangs
- Traffic Probability Assumptions
- Traffic Calculations
- Call Density Matrix
- Bandwidth Calculations
- Determining IP Bandwidth
- Practice Item: Bandwidth Calculation

Lesson 6: Considering Security Implications of VoIP Networks

- Security Policies for VoIP Networks
- Threats to VoIP
- Secure LAN Design
- Communicating Through a Firewall
- Delivering VoIP over a VPN
- International Issues
- Bandwidth Overhead Associated with a VPN
- Practice Item: Span Engineering VoIP Network Security Components

Module 2: Configuring Voice Networks

Lesson 1: Configuring Router Voice Ports

- Voice Port Applications
- FXS Ports
- FXO Ports
- E&M Ports
- Timers and Timing
- Digital Voice Ports
- ISDN
- CCS Options
- Monitoring and Troubleshooting

Lesson 2: Adjusting Voice Interface Settings

- Factors Affecting Voice Quality
- Setting Input and Output Power Levels
- Baselining Input and Output Power Levels
- Voice Quality Tuning
- Configuration Parameters
- Echo Cancellation Commands



Learning Solutions



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Course Outline

Module 2: Configuring Voice Networks (Cont.)

Lesson 3: Configuring Dial Peers

- Dial Peers and Call Legs
- End-to-End Calls
- Types of Dial Peers
- Configuring POTS Dial Peers
- Practice Item 1: POTS Dial-Peer Configuration
- Configuring VoIP Dial Peers
- Practice Item 2: VoIP Dial-Peer Configuration
- Configuring Destination-Pattern Options
- Default Dial Peer
- Matching Inbound Dial Peers
- Practice Item 3: Matching Inbound Dial Peers
- Matching Outbound Dial Peers
- Configuring Hunt Groups
- Practice Item 4: Configuring Hunt Groups
- Digit Collection and Consumption
- Configuring Digit Manipulation
- Practice Item 5: Digit Manipulation

Lesson 4: Configuring Voice Port Network Connections

- Connection Types
- Using the connection Command
- Configuring PLAR Connections
- Configuring PLAR OPX Connections
- Configuring Trunk Connections
- Configuring Tie-Line Connection

Module 3: VoIP Signaling and Call Control

Lesson 1: Introducing Signaling and Call Control

- VoIP Signaling
- Call Control Models
- Translation Between Signaling and Call Control Models
- Call Setup
- Call Administration and Accounting
- Call Status and CDRs
- Address Management
- Admission Control

Lesson 2: Introducing H.323 H.323 and IP

- Functional Components of H.323
- H.323 Gateways
- IP-to-IP Gateways
- H.323 Gatekeepers
- Multipoint Conferences
- H.323 Call Establishment and Maintenance
- RAS Messages
- Call Flows Without a Gatekeeper
- H.323 Fast Connect Call Setup
- Call Flows with a Gatekeeper
- Call Setup with a Gatekeeper
- Gatekeeper-Routed Call Signaling
- Call Flows with Multiple Gatekeepers
- Call Setup with Multiple Gatekeepers
- Multipoint Conferences

Lesson 3: Deploying and Configuring H.323 Robust Design

- H.323 Proxy Server
- Cisco Implementation of H.323
- Configuring H.323 Gateways
- Configuring H.323 Gatekeepers
- Monitoring and Troubleshooting
- Selected debug Commands

Lesson 4: Configuring SIP

- Session Initiation Protocol
- Components of SIP
- SIP Messages
- Status Codes
- SIP Addressing
- Call Setup Models
- Robust Design
- Cisco Implementation of SIP
- Configuring SIP on a Cisco Router
- Monitoring and Troubleshooting



Learning Solutions



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Course Outline

Module 3: VoIP Signaling and Call Control (Cont.)

Lesson 5: Configuring MGCP

- MGCP and Its Associated Standards
- Basic MGCP Components
- MGCP Endpoints
- MGCP Gateways
- MGCP Call Agents
- Basic MGCP Concepts
- MGCP Calls and Connections
- MGCP Control Messages
- Call Flows
- Robust Design
- Cisco Implementation of MGCP
- Configuring MGCP
- Monitoring and Troubleshooting

Lesson 6: Comparing Call Control Models Feature Comparison Charts

- Standards Bodies (ITU-T vs. IETF)
- Architecture (Centralized vs. Distributed)
- Current Version
- Signaling Transport (TCP vs. UDP)
- Multimedia Capability (Yes or No)
- Call Control Encoding (ASN.1 vs. Text)
- Supplementary Services (Endpoint vs. Call Control)
- Strengths of H.323, SIP, and MGCP
- H.323
- Session Initiation Protocol
- Media Gateway Control Protocol
- Selecting Appropriate Call Control
- MGCP Call Control Model
- The H.323 Call Control Model
- SIP Call Control Model

Module 4: Improving and Maintaining Voice Quality

Lesson 1: Designing for Optimal Voice Quality

- IP Networking Overview
- Jitter
- Delay
- Acceptable Delay
- Packet Loss
- PESQ, MOS, and PSQM
- Mean Opinion Score
- Perceptual Speech Quality Measurement
- Perceptual Evaluation of Speech Quality
- Quality Measurement Comparison
- Objectives of QoS
- Using QoS to Improve Voice Quality
- Recognizing Common Design Faults
- Cisco AutoQoS Features

Lesson 2: Implementing CAC

- Effects of Oversubscribing Bandwidth
- CAC Operation
- Resource Reservation Protocol
- CAC Tools
- H.323 CAC
- SIP CAC
- Configuring SAA RTR Responder
- Configuring PSTN Fallback
- Configuring Resource Availability Check
- MGCP CAC
- Configuring SRC CAC
- Configuring RSVP CAC
- Configuring Cisco SAA CAC
- Cisco CallManager CAC



Learning Solutions



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Course Labs

Lab 2-1: Connecting a Voice-Enabled Router

Lab 2-2: Configuring Voice Interfaces

Lab 2-3: Configuring POTS Dial Peers

Lab 2-4: Configuring Special-Purpose Connections

Lab 2-5: Configuring Basic VoIP Network Connections

Lab 3-1: Configuring VoIP with H.323

Lab 3-2: Configuring VoIP with SIP

Lab 3-3: Configuring VoIP with MGCP

Lab 4-1: Implementing Cisco AutoQoS

Lab 4-2: Implementing CAC



Learning Solutions