Cisco Voice Over IP

Price: $2,995

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Length: 35 Hours (5 days)

Introduction: Securing the Cisco Voice Over IP (CVOICE) lays the foundation for gaining hand-on skills and significant understanding of packet telephony by presenting the technologies that are common for Small to Medium, 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. Students will benefit from hands-on experience enabling analog FXS and Digital T1 configuration to the PSTN. H.323, SIP, and MGCP are explained and configured in detailed lab scenarios. Gatekeepers and the Unified Border elements are also explained in this CVOICE class.

Prerequisites: Students taking this course should have the following skills:

- Working knowledge of fundamental terms and concepts of computer networking to include LANs, WANs, and IP
- switching and routing
- Basic internetworking skills taught in Interconnecting Cisco Network Devices (ICND), or equivalent knowledge
- Ability to configure and operate Cisco routers and switches and to enable VLANs and DHCP
- Knowledge of traditional public switched telephone network (PSTN) operations and technologies

Course Materials: Students will be provided with the following software for use in the classroom:

- Certification lesson guides, workbooks

The above mentioned materials are yours to keep.

Objectives: After completing this course, students will be able to:

- Describe VoIP, voice gateways, special requirements for VoIP calls, codecs and codec complexity, and how DSPs are
- used as media resources on a voice gateway
- Configure gateway interconnections to support VoIP and PSTN calls and to integrate with a PSTN and PBX
- Describe the basic signaling protocols that are used on voice gateways and configure a gateway to support calls using
- the various signaling protocols
- Define a dial plan, describe the purpose of each dial plan component, and implement a dial plan on a voice gateway
- Implement gatekeepers and directory gatekeepers, and identify redundancy options for gatekeepers
- Implement a Cisco UBE gateway to connect to an Internet telephony service provider

Course Outline

I. Introduction to VoIP

A. Introducing VoIP
1. Describe the components of the Cisco Unified Communications architecture
2. Describe VoIP and the basic components of a VoIP network
3. Describe the major VoIP signaling protocols
4. Describe the differences between the gateway signaling protocols
5. Describe issues that can affect voice service in the IP network
6. Describe the characteristics of the protocols that are used for media transmission.

B. Introducing Voice Gateways
1. Describe the functionality of gateways and their role of connecting VoIP to traditional PSTN and telephony equipment
2. Describe the different Cisco gateway platforms
3. Identify supported IP telephony deployment models
4. Identify the major characteristics and design guidelines of a single-site IP telephony deployment model
5. Identify the major characteristics and design guidelines of a multisite centralized IP telephony deployment model
6. Identify the major characteristics and design guidelines of a multisite distributed IP telephony deployment model
7. Identify the characteristics, limitations, and advantages of clustering over the IP WAN

C. Specifying Requirements for VoIP Calls
   1. Describe the factors that are present in IP networks that affect audio clarity
   2. Describe MOS and PSQM and how they are used to measure audio quality
   3. Describe QOS features as they relate to a VoIP network and the features of Cisco IOS software that deliver QoS throughout the network
   4. Describe the challenge of transporting modulated data, including fax and modem calls, over IP networks
   5. Describe how fax and modem pass-through, relay, and store and forward are implemented using Cisco IOS gateways
   6. Describe how T.38 and pass-through are supported by H.323, SIP, and MGCP
   7. Describe DTMF relay and how it is supported in MGCP, H.323 and SIP
D. Understanding Codecs, Codec Complexity, and DSP Functionality
   1. Describe various codecs and their bandwidth requirements
   2. Describe how the number of voice samples that are encapsulated impacts bandwidth requirements
   3. Calculate the overhead for Layer 2 and other protocols on a VoIP call
   4. Use a formula to calculate the total bandwidth that is required for a VoIP call with and without VAD
   5. Describe various types of DSPs, DSP functions and how DSPs are used as media resources
   6. Describe codec complexity and where and how to configure it
   7. Describe the DSP requirements for various media resources and show how calculate the actual number of required DSPs
   8. Describe DSP farms, DSP farm profiles and how to configure conferencing and transcoding on a voice gateway
   9. Describe the commands that are required to configure DSP farms on Cisco IOS gateways for enhanced media resources
   10. Describe how to verify the correct operation of available media resources

II. Voice Port Configuration

A. Understanding Call Types
   1. List the seven call types in a VoIP network
   2. Describe the local call type
   3. Describe the on-net call type
   4. Describe the off-net call type
   5. Describe the PLAR call type
   6. Describe the PBX-to-PBX call type
   7. Describe the intercluster trunk call type
   8. Describe the on-net to off-net call type
B. Configuring Analog Voice Ports
   1. Describe the various types of voice port interfaces and where they are used
   2. Describe the various types of analog interfaces and their characteristics
   3. Describe how to configure three types of analog voice ports
   4. Describe CAMA how to configure a voice port for CAMA
   5. Describe how to configure voice ports for DID service
   6. Describe timing configuration parameters on voice ports
   7. Explain how to use show, test, and debug commands to verify analog voice port operation
C. Understanding Dial Peers
   1. Describe the functions of POTS and VoIP dial peers and call legs in relations to a simple VoIP network
   2. Describe how gateways interpret call legs to establish end-to-end calls
   3. Describe the functions of the POTS, VoIP, and default dial peers
   4. Describe how to configure POTS dial peers
   5. Describe how to configure VoIP dial peers
   6. Explain how to use destination-pattern options to associate a telephone number with a given dial peer
   7. Describe how the router matches inbound dial peers
   8. Describe the default dial peer
   9. Describe how the router matches outbound dial peers
D. Configuring Digital Voice Ports
   1. Describe the various types of digital voice ports
   2. Describe T1 CAS trunks and associated signaling
   3. Describe E1R2 CAS trunks and associated signaling
   4. Describe ISDN
   5. Describe ISDN signaling
   6. Configure a T1 CAS trunk to the PSTN
   7. Configure an E1 CAS trunk to the PSTN
   8. Configure and verify BRI and PRI trunks to the PSTN
   9. Verify digital voice port connections

E. Understanding QSIG
   1. Describe QSIG and its associated features
   2. Describe how to configure QSIG support on Cisco IOS gateways
   3. Describe how to verify QSIG trunks

III. VoIP Gateway Implementation

A. Implementing H.323 Gateways
   1. Describe the functions that are performed by a typical H.323 gateway
   2. Describe the advantages of H.323 as a voice gateway protocol
   3. Describe the functional components that make up an H.323 environment
   4. Describe the H.323 call establishment and maintenance process
   5. Describe H.323 call signaling
   6. Describe the types of multipoint conferences that are supported by H.323
   7. Describe how to configure an H.323 gateway
   8. Describe how to configure a single codec or codec negotiation on an H.323 gateway
   9. Describe how to tune some H.323 timers
   10. Configure fax pass-through and relay on H.323 gateways
   11. Describe how to configure H.323 DTMF relay on an H.323 gateway
   12. Describe how to verify the status of an H.323 gateway

B. Implementing MGCP Gateways
   1. Describe MGCP and its associated standards
   2. Describe the advantages of MGCP as a voice gateway protocol
   3. Describe the basic components of MGCP and their roles
   4. Describe the basic concepts of MGCP
   5. Describe the interactions between an MGCP call agent and its associated gateways
   6. Configure an MGCP residential and trunk gateway on a Cisco router
   7. Describe the commands that are used to verify an MGCP configuration

C. Implementing SIP Gateways
   1. Describe SIP and its related standards
   2. Describe the advantages of SIP as a voice gateway protocol
   3. Describe the functional and physical components of a SIP network
   4. Describe three models for SIP call setup and disconnects: direct, using a proxy server, and using a redirect server
   5. Describe the types, use, and structure of SIP messages
   6. Describe SIP address formats, address registration, and address resolution
   7. Describe special considerations for dealing with DTMF tones in a SIP environment
   8. Configure SIP functionality on Cisco IOS gateways
   9. Verify and troubleshoot a SIP gateway

IV. Dial Plan Implementation on Voice Gateways

A. Understanding Dial Plans
   1. Describe the characteristics and components of a dial plan
   2. Describe the concept of endpoint addressing, including overlapping directory number issues
   3. Describe the characteristics of call routing and the importance of path selection
   4. Describe the characteristics of digit manipulation
   5. Describe the characteristics of calling privileges
   6. Describe the characteristics of call coverage
   7. Describe the characteristics of a scalable dial plan in a VoIP network
   8. Describe the requirements for PSTN dial plans in Cisco IOS environments and explain which dial plan components are important
   9. Describe the special requirements for ISDN in Cisco IOS gateway deployments
   10. Configure a PSTN dial plan on Cisco IOS gateways for inbound and outbound calls, including proper DNIS and ANI modification
   11. Verify a PSTN dial plan on Cisco IOS gateways

B. Implementing Numbering Plans
   1. Describe the basic characteristics of a numbering plan
   2. Describe the different types of numbering plans
   3. Describe the attributes of a scalable numbering plan
   4. Describe overlapping numbering plans and strategies to address the issue
   5. Describe how to integrate internal and external PSTN numbering plans
   6. Describe how to integrate existing dial plans into a VoIP network
   7. Describe how VoIP operators can provide the location and telephone number of mobile callers to 911 operators
   8. Implement a numbering plan

C. Configuring Digit Manipulation
   1. Describe basic digit manipulation and why you would need to use it
   2. Describe digit stripping using a voice gateway
   3. Describe digit forwarding on a voice gateway
   4. Describe digit prefixing on a voice gateway
   5. Describe number expansion on a voice gateway
   6. Describe how a gateway collects and consumes digits and applies them to a dial peer
   7. Describe CLID manipulation
   8. Describe the capabilities of voice translation rules and profiles
   9. Contrast voice translation profiles with the dialplan-pattern command
   10. Create a dial peer with digit manipulation commands to divert calls that connect to a specified destination
D. Configuring Path Selection
   1. Describe how a call is routed and the correct path is selected
   2. Describe how a router matches dial peers to determine path selection
   3. Describe how the gateway matches information elements to dial peers
   4. Describe some routing and path selection best practices
   5. Describe various path selection strategies
   6. Describe the characteristics of site-code dialing and toll bypass
   7. Describe TEHO
   8. Configure site-code dialing and toll bypass
   9. Configure TEHO
E. Implementing Calling Privileges on Cisco IOS Gateways
   1. Describe calling privileges
   2. Describe how COR can be used to implement calling privileges on Cisco IOS gateways
   3. Describe how COR can be used in SRST and Cisco United Communications Manager Express environments
   4. Configure COR on a Cisco IOS gateway using Cisco Unified Communications Manager Express and SRST

V. H.323 Gatekeeper

A. Introducing Gatekeepers
   1. Describe the functionality of gatekeepers in an H.323 environment
   2. Define the hardware and software requirements for gatekeeper functionality
   3. Describe the signaling between gateways and gatekeepers
   4. Describe how directory gatekeepers enhance the scalability of a network
   5. Describe how gatekeeper zone prefixes are used for call routing
   6. Describe how gatekeeper technology prefixes are used for call routing
   7. Describe how gatekeepers perform address resolution and call routing in different scenarios
   8. Describe how GKTMP works
   9. Describe some commands that are used to verify H.323 gatekeeper operation
B. Configuring Basic Gatekeeper Functionality
   1. List the steps necessary to configure a multizone gatekeeper for local and remote zone call routing
   2. Configure local and remote zones on a gatekeeper
   3. Configure zone prefixes on a gatekeeper
   4. Configure technology prefixes on a gatekeeper
   5. Configure gateways to register with a gatekeeper
   6. Configure dial peers for gatekeepers
   7. Verify that H.323 endpoints are registered properly and calls are correctly routed across a single gatekeeper
C. Implementing Gatekeeper-Based CAC
   1. Describe bandwidth operation in a gatekeeper zone
   2. Describe zone bandwidth calculation in a gatekeeper network
   3. Configure zone bandwidth on a gatekeeper
   4. Verify zone bandwidth operation on gatekeepers
   5. Describe how RAI performs resource-availability in gatekeeper networks
   6. Configure RAI in a gatekeeper network
   7. Verify RAI operation in gatekeeper networks

VI. ITSP Connectivity

D. Understanding Special Requirements for External VoIP Connections
   1. Describe the functionality of a Cisco UBE
   2. Describe how Cisco UBEs can be utilized in enterprise VoIP environments
   3. Describe how protocol interworking is performed on Cisco UBEs
   4. Describe how Cisco UBEs handle media flows
   5. Describe how Cisco UBEs perform codec filtering
   6. Describe how Cisco UBEs can be used to perform RSVP-based CAC

7. Describe how Cisco UBEs can be integrated with gatekeeper networks
8. Describe call flows involving Cisco UBEs

E. B. Implementing a Cisco UBE

1. Describe the command that are used to enable protocol interworking
2. Configure H.323-to-H.323 interworking on a Cisco UBE
3. Configure H.323-to-SIP interworking on a Cisco UBE
4. Describe the commands that are used to configure media flow-around, media flow-through, and transparent codec pass-through
5. Configure transparent codec pass-through and media flow-around on a Cisco UBE
6. Configure a Cisco UBE to register with a via-zone gatekeeper
7. Verify Cisco UBE and via-zone gatekeeper operation
8. The lesson includes these topics:
9. Protocol Interworking Command
10. Configuring H.323-to-H.323 Interworking
11. Configuring H.323-to-SIP Interworking
12. Media Flow and Transparent Codec Commands
13. Configuring Transparent Codec Pass-Through and Media Flow-Around
14. Configuring Cisco UBEs and Via-Zone Gatekeepers
15. Verifying Cisco UBEs and Via-Zone Gatekeepers