



VoIP & Telephony

Advanced Deployment of Voice over IP

Design secure, high-performance VoIP networks.

Successful VoIP implementation means you must build knowledge spanning a broad range of technology, including VoIP certification testing, security issues, firewalls, NATs, VLAN, QoS, signaling protocols, and how to use trace tools to debug VoIP signaling problems.

Using real-world equipment in an extensive hands-on classroom environment, you will go through the process of testing an existing network, implementing several vendors' products, traversing NATs, interoperating with firewalls, and ultimately building a working Voice over IP system in spite of the roadblocks. You need to test these skills in a classroom environment first, not on your own network.

This class is for people who MUST make VoIP work.

Course Outline

1. Converging Voice and Data on the LAN

- Telephony architecture
- VoIP architecture
- ENUM
- DNS and ENUM
- Review of Ethernet switching essentials
- VLAN 802.1Q
- Building the Voice VLAN
 - Power over Ethernet cabling
 - Classifying Voice at the VLAN level
 - Securing Voice in the LAN
 - VoIP access lists
 - Deploying QoS in LANs, how to utilize IEEE 802.1p
- Multilayer switching essentials

2. RTP (Real-Time Transport Protocol)

- Payload types
- RTP Timestamps
- Silence Suppression
- Supporting Real-time Traffic
- The First Voice Packet
- The Second Voice Packet
- The Third Voice Packet
- A Jitter Buffer operation
- The Adaptive Jitter Buffer example
- RTP Sequence #
- Synchronization Source

Course Code: 3285

On-Site Learning

4 Day Course

Who Needs to Attend

This advanced course will help telecom technical staff gain the data-oriented skills needed to support newly implemented VoIP systems. This course will also help datacom technical staff deal with issues such as network certification, security, QoS, and signaling issues. Students who are planning to take Cisco VoIP product training would benefit from the skills obtained in this advanced technology course.

Prerequisites

Attendees need a solid understanding of TCP/IP and telecommunications. Voice over IP Foundations is highly recommended.

[Voice over IP Foundations](#)

Follow-On Courses

[CIPT1 - Cisco IP Telephony Part 1 v4.1](#)

- Contributing Source (CSRC)
- DTMF (RFC 2833)
- Telephony Tones & Signals (RFC 2833)
- RTP Header Compression (RFC 2508)
- RTP Packet - Using Ethereal, find all these RTP fields
- RTCP - (RTP Control Protocol)
 - Source descriptions (SDES)
 - Sender reports (SENDER)
 - Receiver reports (RECEIVER)
 - RTCP Calculates Round Trip
 - RTCP BYE Report

3. Build an IP Core Network

- Configure Cisco 2600 and 2600XM routers running current IOS
- Configure Real T1 interfaces
 - CSU/DSU
 - Fractional T1
- Configure V.35 serial interfaces
- Compare VLAN and Non-VLAN configurations and determine which is best
 - Interface routers and Ethernet switches
 - Configure VLAN on Fast Ethernet interfaces to take advantage of VLAN trunks
 - Configure 10Base-T Ethernet interfaces using switched access links
- Connect to the real Internet using NAT (not available at some locations)
- Configure routing on a Cisco 3550 multilayer switch
- Implement a TFTP server
- Implement a syslog server
- DHCP best practices
 - Configure DHCP for Cisco and Nortel IP telephones
 - Configure PC hosts to use DHCP

4. CallManager Express

- 100% Hands-on
- SIP protocol analysis
- How to use SIP trunking to avoid expensive media gateways

5. Session Initiation Protocol (SIP)

- SIP Methods
 - INVITE
 - ACK
 - OPTIONS
 - CANCEL
 - BYE
 - REGISTER
- SIP Method Extensions
 - SUBSCRIBE/NOTIFY "SIP Events RFC 3265"
 - REFER "SIP REFER Method"
 - MESSAGE "SIP Extensions for Instant Messaging"
 - PRACK "Reliability of Provisional Responses in SIP RFC 3262"
 - UPDATE Early media and preconditions, the "SIP UPDATE Method"
 - The SIP INFO Method RFC 2976
 - COMET - Packetcable QoS
- SIP Responses
 - 1xx Informational
 - 2xx Final
 - 3xx Redirection
 - 4xx Client Error
 - 5xx Server Error
 - 6xx Global Failure
- SIP Uniform Resource Indicators (URIs)
 - Generic URI information (RFC 2396)
 - Direct or Proxy

- PSTN number (RFC 2808)
- Instant messaging
- Presence
- In registrations
- Session Description Protocol (SDP)
 - Session parameters
 - SDP format
 - Extending SDP
 - Media negotiation
 - Changing session parameters
- SIP Architecture
 - DNS
 - SIP elements
 - UA - UAC/UAS
 - SIP servers
 - Registrar
 - Back-to-back user agent
 - Proxy
 - Redirect
 - Local architecture
 - Global architecture
- SIP Fields
 - Via
 - Branch
 - Max-Forwards
 - Dialog (formerly call leg)
 - Includes To, From, and tag = fields
 - CSeq
 - Call-ID
 - Contact
 - SIP reliability
- Registration
 - DHCP
 - User registration
 - Registration expiration
 - Lookup and delete
 - Finding a local registrar
- Call Routing
 - Direct call
 - Proxied call
 - Forking
 - Loops and spirals
 - Response path
 - Creation of via-path
 - Response merging
 - Record route
 - Heterogeneous Error Response Forking Problem (HERFP)
- Control models
 - Third party
 - Multi-party
- ENUM
 - Application
 - Architecture
- Call Flow Examples
 - Call attempt - unsuccessful
 - Presence subscription
 - Registration
 - Presence notification
 - Instant Message Exchange
 - Call setup - successful
 - Call hold
 - Call transfer
 - Unified messaging

6. Deploying VoIP Gateways

- Datacom professionals who are new to telecommunications are required

to implement a local battery, hand generator subset straight from the 1920s and observe how it works. The pure physics behind loop start signaling has not changed since then. Clearly understand the importance of the hybrid, ring battery, local battery, the loop, balanced pair, and loop signaling.

- Learn how to measure the gateway's input gain and output attenuation. Properly developed skills will allow you to configure systems that are never "too loud" or "too soft."
- Deploy echo cancellation
- Measure echo cancellation effectiveness
- Configure an Avaya key system and install Avaya Partner Key Telephones
- Connect a Cisco gateway to an Avaya Partner Key system and make VoIP calls using legacy phones
- Configure ground start interfaces and compare with loop start
- Configure dial peers to make test calls
- Using a Gateway with Firewalls and NATs
 - Firewall essentials
 - Fixup protocol function
 - Example of a PIX firewall configuration for VoIP
 - How to keep it simple and bulletproof
- Supporting fax and modem services

7. The Soft Switch

- Deploy Cisco CallManager Express
 - Install CallManager Express on a 2600XM router
 - Install Cisco 2950 IP telephones
 - Configure VLAN for the 2950 phones
 - Configure DHCP for i2004
 - Configure Buttons on the phones
 - Test the simplicity of doing moves and changes
- Configure Microsoft Live Communications Server

8. Media Gateway Control Protocol (MGCP)

- IETF RFC 2705 (See appendix G)
- PacketCable residential and trunk gateways profile for DOCSIS networks
- Simple Gateway Control Interface
 - endpoints
 - call identifiers
 - connection identifiers
 - digit maps
 - SDP
- Gateway Control Functions
 - RQNT - Notification Request
 - NTFY - Notifications
 - CRCX - CreateConnection
 - MDCX - ModifyConnection
 - DLCX - DeleteConnection
 - Audit Endpoint
 - Audit Connection
- Digit maps
- Return Codes
- Verbs
- Parameter lines
- Connection parameters (RTCP)
- Connection mode
- Event names
- Requested events
- Wildcarding structure
- Packages
 - Generic
 - DTMF
 - MF
 - Trunk
 - Line

- Handset
- RTP
- NAS
- Announcement
- script
- Mandatory and Optional extension parameters
- How MGCP handles ringing

9. How to Deploy QoS

- Voice is more forgiving than you may think; learn how far you can push the envelope before you hear complaints
- Learn how to calculate the delay budget
- Lost packets and Gap Fill technology
- Learn how to design networks for bounded jitter, not zero jitter (overkill)
- Voice is not Constant Bit Rate. Voice is bursty, just like data, so learn how to treat it that way.
- Learn how to deploy proper queuing algorithms to avoid starving data queues while maintaining high quality VoIP QoS
- Learn how to write access lists that enforce QoS
- Learn how to write Firewall rules that protect both QoS and security
- Deploy differentiated services on the WAN
- Interoperate 802.1q/p and Diff-Serv
- How to bound jitter on low bandwidth links
 - Why simply setting an MTU may be a bad idea
 - FRF.12
 - Multilink PPP
 - Which makes the most sense?
- Recognize when QoS is not necessary or the effort will yield minor results
- Frame relay on the way out? Perhaps, so learn how to use metro Ethernet services to achieve unparalleled QoS services from your service provider.
- How to implement the stress testing on a live network
- How to avoid creating a disaster while testing the network
- How SNMP is used in testing
 - The community string
 - SNMP queries synchronized with traffic load
- Compare Cisco 2500 and 2600 router performance
- How bandwidth, packets per second, and packet size can affect network performance

10. VoIP Security

- Firewall Config
- SIP Security Overview
- Hop-by-hop Encryption of SIP Messages
- IPSEC
- SIP over TLS (Transport Layer Security)
- End-to-End Encryption of SIP Messages
- Media Privacy
- SIP Authentication
- Authentication for Proxies
- Firewalls and SIP
- Network Address Translators
- Application Layer Gateways
- NAT Transversal for SIP
- Full Cone NAT
- Restricted Cone NAT

11. Performance Testing

- Data problems VoIP will uncover
- Produce a network utilization report
- How to read the Erlang B chart
- Stress test the network

- Avoiding stress test disasters
- QoS management system

Labs

Lab 1: Configure your PCs

Lab 2: Cisco IOS command line interface overview

Lab 3: Configure a Layer 2 switch

Lab 4: Configure 802.1Q/P VLAN Access

Lab 5: Add VLAN Trunking to a Cisco 7960 phone

Lab 6: Add VLAN Trunking to a Nortel i2004 phone

Lab 7: Implement a TFTP server on your PC

Lab 8: Implement Syslogging

Lab 9: Port Scanning

Lab 10: How to use Ethereal

- MGCP Trace
- SIP Trace

Lab 11: VLAN Trunking - Switch side

Lab 12: VLAN Trunking - Router side

Lab 13: Implement DHCP

Lab 14: Configure the Core network interfaces

Lab 15: Configure IP Routing

Lab 16: Configure NTP services

Lab 17: Install SolarWinds SNMP Management

Lab 18: Install Cisco CallManager Express (CCME)

Lab 19: Navigating the CCME

Lab 20: Install a Cisco ATA

Lab 21: PBX Essentials, FXS/FXO interfaces

Lab 22: Configure a KSU (Key Service Unit)

Lab 23: Configure a SOHO VoIP System

Lab 24: Configure VoIP dial-peers

Lab 25: Digit Translation

Lab 26: Show Call Active Voice

Lab 27: Configure MGCP on the Cisco CallManager

Lab 28: Configure MGCP on a Media Gateway

Lab 29: Configure a phone on a SoftSwitch

Lab 30: Capture SCCP (skinny client control protocol)

Lab 31: Install a T1

Lab 32: Link Fragmentation and Interleave

Lab 33: Configure Low Latency Queuing (LLQ) on interfaces

- Apply Diff-Serv to Dial-peers
- Write a QoS access list
- Test QoS controls by competing data and voice traffic

Lab 34: RTP Header Compression

Lab 35: QoS over frame relay using vc-bundling

Lab 36: Analyze a SIP Capture

Lab 37: Install a Microsoft Live Action Server

Lab 38: Install a Home Server (SIP Server)

Lab 39: Add a user to Live Action Server

Lab 40: Join a domain

Lab 41: Install MS Messenger

Lab 42: VoIP Firewall Configuration with NAT

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