VOICE OVER IP FOUNDATIONS

Course Code: 3277

Discover how and why Voice over IP works and gain hands-on experience with the latest VoIP software.

In this course, you will learn core concepts of how the Internet Protocol (IP) carries a Voice over IP (VoIP) packet. You will learn the fundamentals of Session Initiation Protocol (SIP) architecture, SIP-related IP services, the advantages and disadvantages of SIP Trunking as well as Quality of Service (QoS)-Related Protocol.

In the Classroom Live version of this course, you will gain proficiency with some of the most popular VoIP software and hardware, such as Wireshark, Asterisk PBX, Kamailio SIP Proxy, Linksys Ethernet phone, and SIP-based ATA in a hands-on labs. You will also cover Cisco QoS policy administration and demonstrate successful VoIP calls in high data traffic conditions.

What You’ll Learn

• Core concepts of how Internet Protocol (IP) carries a VoIP packet
• Advantages and disadvantages of SIP Trunking
• Configure DHCP and DNS to support IP telephony
• Real-Time Transport Protocol (RTP)
• Session Initiation Protocol (SIP) - Call set up, Instant Messaging, Presence
• Session Description Protocol (SDP)
• SIP proxy, Session Border Controller (SBC), and SIP softswitch
• Media Gateway Control Protocol (MGCP) analysis
• MGCP architecture
• How to implement QoS to ensure the highest voice quality over your IP networks
• The impact of jitter, latency, and packet loss on VoIP networks
• How to use Wireshark to decode and troubleshoot RTP, SIP, and MGCP call flows
• Configure the trixbox Softswitch and SIP proxy
• Configure SIP gateways and softphones

Who Needs to Attend

This class is for people who need to understand VoIP technology. IT managers, technical sales/marketing personnel, consultants, network designers and engineers, product design engineers developing integrated-services products, telecom technicians and managers integrating PBX services within data networks, and
systems administrators who will manage a converged network would benefit from this course.
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Classroom Live Outline

1. Packetizing Voice
   - Telephony Architecture
     - Introduction to the VoIP Standards
   - Connecting VoIP to PSTN
     - Traffic Engineering
     - PSTN to VoIP Using Magic
   - Voice Digitization
     - Companding Mu-Law vs. A-Law
   - Time Division Circuit Switching
   - Voice Packet
     - The 20-Millisecond Voice Packet
     - The 60-Millisecond Voice Packet
     - The Voice Packet Header
     - Other Voice Packet Sample Sizes
     - Voice Packet Analysis
     - Voice Packet Analysis: Other Voice Packet Sample Sizes
   - QoS Overview
     - Latency
     - Packet Loss
     - Jitter
   - Controlling Delay
     - Sources of Delay
     - The First Voice Packet
     - The Second Voice Packet
     - The Third Voice Packet
     - Jitter Buffer Under Perfect Conditions
     - An Adaptive Jitter Buffer

2. SIP Trunking
• The Legacy Circuit Switch
• VoIP Phases
  ◦ VoIP Phase 1: LAN Connect the Line Side
  ◦ VoIP Phase 2: Decompose the Switch Cabinet
  ◦ VoIP Phase 3: Shrink the MGs and Add Survivability
  ◦ VoIP Phase 4: Add SIP Trunking
  ◦ VoIP Phase 5: Eliminate the Old MGs
  ◦ VoIP Phase 6: Add EMUN
  ◦ VoIP Phase 7: Mass Acceptance of SIP Trunking with ENUM?
• SIP Trunking Costs
• Other Means of Connection
• The "Old PBX can do SIP Trunking if the Vendor Offers the Software
• SIP Trunking Protocols
  ◦ Peer-to-Peer RTP
  ◦ Hairpin RTP
• Disadvantages and Advantages of SIP Trunking
  ◦ Disadvantages
  ◦ Advantages
• ITSPs
• SIP Trunking Examples
  ◦ SIP Trunk Outbound Call
  ◦ Public VoIP

3. VoIP in the LAN
• IP and Ethernet
  ◦ A Sample Ethernet Switched Network
• MAC Addresses
• IP MAC Address Learning
  ◦ Unknown Destination MAC Addresses
  ◦ Flood the Broadcast
  ◦ Response to Flooded Packet
  ◦ Learning Port Information
  ◦ Switching
• MAC Table Aging
• Ethernet Communications Limits
• Virtual LANs
  ◦ VLAN Trunk
  ◦ VLAN Tags
  ◦ Untagged Frames
• Port-Based VLANs
  ◦ Broadcast Frame in VLAN 10
• VLAN Trunking for VoIP Phones
• IEEE 802.3af Device Detection
  ◦ IEEE 802.3af Power Classifications
  ◦ QoS at Layer 2
  ◦ VLAN Tagging Process
4. IP Networking

- One-Way vs. Both-Way Routing
- Static Routing
  - Subnet Masks and Routing
  - Routing and Switching
- Routing Protocols
  - Distance Vector Routing
  - Link-State Routing

5. TCP/IP Review

- Transmission Control Protocol (TCP) vs. User Datagram Protocol (UDP)
  - Connection-Oriented Protocol (TCP)
  - TCP/IP Packet Format and Operation
  - Connectionless Protocols (UDP)
  - UDP Packet
- DNS
  - Basic Method of DNS

6. Dial Plan Essentials

- Dial Plan Example
- Digit Map
- Enbloc vs. Overlap
- Common Modifications to REGEX
- Symbols
  - Regular Expressions
  - Metacharacters
- Matching
- Normalization Examples

7. SIP-Related IP Services

- DHCP Option for SIP
  - DHCP Discover
  - DHCP Offer
- Root-Level Domain Registration
- Basic Method of DNS
  - Why Start with ENUM?
- ENUM: NAPTR Query
  - ENUM: NAPTR Response
- Locating SIP Servers: An Example
  - NAPTR Response
  - SRV Query
  - SRV Response
  - A Record Query
- Regular Expressions
  - The Metacharacters
8. Voice Compression
   • Voice Compression Hardware
     □ ASICs
     □ DSPs
   • Mean Opinion Scores
   • Codecs
     □ G.711, G.723.1, G.726
     □ G.728 and G.729
   • Voice Compression
     □ Formants
     □ The Predictor
     □ PCM Sampling
   • Voice Compression Algorithms
     □ ADPCM Compression
     □ Vocoder
     □ G.729 Example
   • Codec Comparison Exercise
     □ Zero Packet Loss
     □ Ten Percent Packet Loss
     □ Twenty Percent Packet Loss
   • T.38 Fax Spoofing
     □ Call Setup
     □ Discovering the Fax Tone
     □ T.30 Negotiation
     □ Shifting to 9.6 Kbps
     □ T.38 Phase

9. Real-Time Transport Protocol (RTP)
   • RTP Architecture
     □ RTP and RTP Control Protocol
     □ Encapsulating the Voice Packet
     □ RTP Ports
   • RTP Profile
     □ Payload Types
     □ Mapping Payload Type to Codec Type
     □ How H.323 Identifies the Payload Type
     □ NTP vs. RTP Timestamp
     □ RTP Timestamps
     □ RTP Timestamps and Silence Suppression
     □ RTP Timestamps and Jitter Calculation
   • Controlling Jitter
     □ Jitter Buffer Delay
   • Mixers
     □ Synchronization Source
     □ Conference Bridge Adds CSRC
   • RTP Header
UDP Packet with RTP Header and Voice

Required Fields

- Version
- Padding Bit
- Extension Bit
- CSRC
- Market Bit
- Payload Type
- Sequence Number
- Timestamp
- SSRC

- The Format-Specific Parameter (fmt) Attribute

- RFC 2833 Example: A Dialing Event
  - Transmitter Processing
  - Receiver Processing

- Controlling Serialization Delay
  - Perfect Candidate for LFI and RTP Header Compression

- RTP Header Compression Process (RFC 2508)
  - RTP Header Compression Format

- RTCP
  - RTCP QoS: Round-Trip Delay Calculation
  - Sender Reports
  - Receiver Reports
  - Source Descriptions
  - Source Description Items
  - Other RTCP Packets

10. SIP Architecture

- SIP User Agents
  - SIP Requests (Methods)
  - SIP Response Codes

- SIP Proxy
  - SIP Back-to-Back UA
  - Session Border Controller
  - Forking Proxy
  - SIP Redirect Proxy

- Global SIP Architecture
  - Overview of Operation
  - Classic SIP Trapezoid
  - INVITE Request
  - Session Description Protocol
  - Proxy Function
  - 180 Response
  - 200 Final Response
  - BYE
  - INVITE and ACK
• SIP Functional Stack
• SIP Core Documents and Extensions

11. SIP Call Flow Examples
• SIP Call Analysis
  ◦ SIP Registration with Authentication
  ◦ SIP Call without INVITE Authentication
  ◦ The 100rel Process
  ◦ Busy Number
  ◦ Abandoned Call (Cancel)
  ◦ SIP Redirect (Call Forward)
  ◦ Call Transfer
• E&M Tie Trunk
  ◦ See a Problem?
  ◦ Solution: SIP 183 Response

12. Session Description Protocol
• Session Description Protocol
  ◦ v= Header
  ◦ o= Header
  ◦ s= Header
  ◦ c= Header
  ◦ t= Header
  ◦ m= Header
  ◦ a= Header
• Offer/Answer Model
  ◦ Offer/Answer: Example 1
  ◦ Offer/Answer: Example 2
  ◦ SDP Offer/Answer Rules
  ◦ UPDATE Method
  ◦ RTP SEND and RECV Defined
  ◦ Media Direction and RTCP
  ◦ How RTCP Works
  ◦ Placing a Call on HOLD

13. SIP NAT Traversal
• SIP NAT Traversal
  ◦ One-Way Voice Results
  ◦ Full Cone NAT
  ◦ IP Address Restricted NAT
  ◦ Port Restricted NAT
  ◦ Symmetric NAT
  ◦ Simple Traversal of UDP through NATs
  ◦ Traversal Using Relay NAT
  ◦ NAT with Embedded SIP Proxy
  ◦ Public VoIP Example
14. Media Gateway Control Protocol (MGCP)
   • Protocol Comparison
   • MGCP Call Model
     ■ Hairpin Call Example
     ■ Defined Endpoints
   • MGCP Commands
     ■ MGCP Syntax Example
     ■ Return Codes
     ■ Return Code Table
     ■ Parameter Lines
     ■ DTMF Package
     ■ Line Package
   • Digit Maps
   • MGCP Trace Procedure
     ■ MGCP Trace (Steps 1-8)
     ■ MGCP Trace (Steps 9-14)
     ■ MGCP Trace (Steps 15-22)
     ■ MGCP Trace (Steps 23-28)
   • MGCP Established Call
     ■ MGCP Trace (Steps 29-36)
     ■ MGCP Trace (Steps 37-40)

15. Queuing
   • CoS vs. QoS
     ■ Leaky Bucket
     ■ First In, First Out
     ■ Type Classification
     ■ Session ID Classification (Fair Queuing)
     ■ Dequeueing

16. QoS-Related Protocol
   • Sources of Delay
     ■ Packetization Delay
     ■ Algorithmic Delay (Look Ahead)
     ■ Coder Processing Delay (Think Time)
     ■ Queuing Delay
     ■ Serialization Delay
   • Low-Speed Link
     ■ How 56-Kbps Links Cause Jitter
     ■ Upgrade to T1/E1 and Prioritize Voice
   • QoS Technology Solutions: Differentiated Services (DiffServ)
     ■ Supporting a VoIP Call with DiffServ
     ■ ToS Field
     ■ DiffServ Process at the Edge Router
     ■ DiffServ Process in the Core
     ■ DiffServ Highlights
• Traffic Engineering: An Art Form
  ▪ Measuring Engineering
  ▪ Grade of Service

Appendix A: Glossary
Appendix B: H.323

Classroom Live Labs
Lab 1: Install the Network Hardware
Lab 2: Configure PC, Router, and Switch
Lab 3: Calling without a SIP Proxy
Lab 4: Registering with a CORE Proxy
Lab 5: VoIP Islands: Part 1
  ▪ Set Up Your Own Proxy
  ▪ Add the Linksys Phone
Lab 6: VoIP Islands: Part 2
  ▪ Add the ATA Phone
Lab 7: Create a Line-Side Dial Plan
Lab 8: Create a Trunk-Side Dial Plan
Lab 9: Configure a SIP Softphone
Lab 10: How to use Wireshark
Lab 11: Codec MOS Testing
  ▪ Configure Various Codecs
  ▪ Complete Test Calls to Compare Voice Quality (G.711, G.729, and G.723.1)
Lab 12: Packet Interval
  ▪ Reduce Bandwidth Consumption by 50% or More
  ▪ Modify Packet Intervals and Witness the QoS Tradeoff
Lab 13: Codec Bandwidth Testing
Lab 14: Silence Suppression
Lab 15: Codec Negotiation (OFFER/ANSWER)
Lab 16: DTMF RFC 2833 and SIP INFO
Lab 17: SIP Call Flow Analysis
  ▪ Normal Call with Authentication
  ▪ Busy Call
  ▪ Vacant Call
  ▪ Abandoned Call
• Call Forward Immediate
Lab 18: Configure Diff-Serv on a Gateway
Lab 19: Queuing
Lab 20: Final Exam: Configure a Complete VoIP System
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Mar 23 - 27, 2020 | 12:00 - 5:00 PM EST
May 18 - 22, 2020 | 12:00 - 5:00 PM EST
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PRIVATE GROUP TRAINING  5 days

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