VOICE OVER IP FOUNDATIONS

Course Code: 3277

Discover how and why Voice over IP works and gain an understanding of SIP.

In this lecture-only 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.

What You'll Learn

- Core concepts of how Internet Protocol (IP) carries a VoIP packet
- Advantages and disadvantages of SIP Trunking
- Understand how DHCP and DNS supports 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 Wireshark can decode and troubleshoot RTP, SIP, and MGCP call flows
- Discuss trixbox Softswitch and SIP proxy
- Discuss 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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VIRTUAL CLASSROOM LIVE	$2,595 USD	4 Day

Virtual Classroom Live Outline

Module 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

Module 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

Module 3 VoIP in the LAN

- IP and Ethernet
- A Sample Ethernet Switched Network
- MAC Addresses
- IP MAC Address Learning
- known 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

- IEEE 802.1q Frame Tagging

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

Module 4 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

Module 5 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

Module 6 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 (fmtp) 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

Module 7 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

Module 8 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

Module 9 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

Module 10 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

Module 11 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)

Module 12 Queuing

- CoS vs. QoS
- Leaky Bucket
- First In, First Out
- Type Classification
- Session ID Classification (Fair Queuing)
- Dequeuing
- 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

Jun 2 - 5, 2025 | 8:30 AM - 3:30 PM EDT

Aug 4 - 7, 2025 | 8:30 AM - 3:30 PM EDT

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PRIVATE GROUP TRAINING	4 Day

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