Voice over IP – 4 days

CONTENTS

Embark on a 4-day VoIP journey, exploring core concepts, SIP architecture, SDP headers, and RTP in-depth. Learn to connect VoIP to PSTN, navigate SIP trunking, and optimize voice quality through QoS. This course blends theory and hands-on labs for a comprehensive understanding of VoIP essentials.

- Core concepts of how Internet Protocol (IP) carries a VoIP packet
- DHCP and DNS to support IP telephony
- Session Initiation Protocol (SIP) - Call set up, Instant Messaging, Presence
- SIP Trunking
- Session Description Protocol (SDP)
- SIP proxy, Session Border Controller (SBC)
- Media Gateway Control Protocol (MGCP)
- Real-Time Transport Protocol (RTP)
- QoS to have high voice quality in IP networks
- The impact of jitter, latency, and packet loss on VoIP networks
- Wireshark to analyze and troubleshoot VoIP-traffic

NOTE: This course is not delivered with the FoldOut methodology.

The TCP/IP architecture review

- The Internet protocol review, addressing, NAT/PAT
- Transmission Control Protocol (TCP) -. User Datagram Protocol (UDP)
    - Basic functions of TCP and UDP
    - TCP/IP Packet Format and Operation
    - The RTP protocol
- Applications like DNS, DHCP, TFTP, SIP & Codec
    - Basic Method of DHCP and DNS

LAN, Ethernet technology

- IP and Ethernet
    - A Sample Ethernet Switched Network
- MAC Addresses
- IP MAC Address Learning
    - Unknown MAC Addresses, ARP
    - Switching
    - Learning Port Information
- MAC Table
- Ethernet Communications Limits
- Virtual LANs
    - VLAN Tags Trunk
    - Untagged Frames
- Port-Based VLANs
    - Broadcast in VLAN
    - QoS at Layer 2
    - VLAN Tagging Process
    - IEEE 802.1q Frame Tagging

IP Networking

- Static or dynamic Routing
    - Subnet Masks and Routing
    - Routing and Switching
- Routing Protocols
    - Distance Vector Routing
    - Link-State Routing

Basic Method of DNS

- Root-Level Domain Registration
- Locating SIP Servers:
    - NAPTR Response
    - SRV Query
    - SRV Response
    - A Record Query
- Regular Expressions

SIP Architecture

- SIP documents, RFC's
- SIP equipment, User Agents, Proxy, Redirect Proxy, Location Service
    - SIP Requests (Methods), Response Codes
    - SIP Back-to-Back UA, Session Border Controller (SBC)
    - Classic SIP Trapezoid
    - SIP Operations, functions and levels
    - Proxy Functions, call signaling
    - SIP Registration
    - Basic methods, INVITE, ACK, BYE, CANCEL, OPTIONS, NOTIFY, UPDATE
    - SIP responses
    - SIP Call Flow
    - Forking and routing calls
    - SIP error responses,
    - SIP Redirect (Call Forward)
    - Call Transfer
    - Presence

Session Description Protocol (SDP)

- SDP-headers
- Offer/Answer Model
    - Put a call on hold, SDP SEND and RECV
    - Media Direction and RTCP
    - How RTCP Works

SIP-Related IP Services

- DHCP Option for SIP
    - DHCP Discover, offer, request, ack

Media transport

Packetizing Voice

- Telephony Architecture
    - VoIP Standards

Real-Time Transport Protocol (RTP)

- RTP Architecture
    - RTP and RTP Control Protocol (RTCP)
    - RTP Ports
    - RTP Profile
        - Payload Types
        - Mapping Payload Type to Codec Type
        - RTP Timestamps

RTP Header

- UDP Packet with RTP Header and Voice
- Required Fields
- Version
- Payload Type
- Sequence Number
- Timestamp
- SSRC
    - Transmitter, Receiver Processing

Security

- Secure SIP and SRTP
- IPSEC, TLS, SSL
- S/MIME
- Authentication
- DTMF

Connecting VoIP to PSTN

- PSTN to VoIP
- Voice Digitization
- Time Division Circuit Switching
- Voice Packet
    - The 20-Millisecond Voice Packet
    - The Voice Packet Header

SIP to PSTN

- The Legacy Circuit Switch
- Why Start with ENUM?
- VoIP Phases
- SIP Trunking
- ENUM
- SIP Trunking with ENUM?
    - SIP Trunking Costs
    - SIP Trunking Protocols
        - Peer-to-Peer RTP
        - Hairpin RTP
    - SIP Trunking, Pros & Cons
        - Disadvantages
        - Advantages

Queuing

- CoS vs. QoS

- Buffer handling
- First In, First Out
- Type Classification
- Dequeuing

QoS Concept

- Latency
- Packet Loss
- Jitter
- Sources of Delay
- Jitter Buffer
  - Controlling Jitter
  - Jitter Buffer Delay
- An Adaptive Jitter Buffer

QoS-Related Protocol

- Sources of Delay
  - Packetization Delay
  - Queuing Delay
  - Serialization Delay
- QoS Technology Differentiated Services (DiffServ)
  - ToS Field
  - DiffServ Edge Router, Core

SIP NAT Traversal

- SIP NAT Traversal
  - Symetric and full Cone NAT
  - IP Address Restricted NAT
  - Port Restricted NAT
  - Simple Traversal of UDP through NATs
  - Traversal Using Relay NAT

Theoretical and practical trainings/labs