

VOIP & IP tel

Course 1135 – 40 Hours

Overview

The students will study introduction to PSTN Telephony and VoIP, about Interworking between PSTN protocols and IP protocols (SIGTRAN), protocols IP: some versions H323, SIP, SIP-T, MGCP, MEGACO/H248, IP Telephony systems, design issues, security aspects and QOS

Who Should Attend

- R&D and Marketing staff from companies supplying telecom equipment
- Marketing and maintenance staff from service providers
- Communications and plant managers of companies using telecom equipment and services
- Communications consultants

Prerequisites

- Knowledge of Telephony network, PSTN, SS#7 and ISDN and basic knowledge of data networks and TCP/IP

Course Contents

- Networking recapitulation (separate frontal lecture on customer site)
 - Ethernet frame format
 - IP Architecture
 - Address scheme
 - Configuration
 - Utilities
 - Protocol format
 - ARP
 - ICMP
 - DHCP
 - DNS
 - TCP
 - UDP
 - Wireshark
- VoIP Introduction
 - Introduction to telephony
 - PSTN architecture and functionality
 - Analog Line
 - E1 & PRI
 - SS#7
- Codecs
 - Analog to digital voice
 - PCM G.711
 - Quality of voice, MOS

- Compression G.723, G.726, G.729, etc.
- IP telephony
 - Traditional telephony
 - Moving towards IP
 - Hardware saving - comparing tdm network with packet network
 - Traffic management and trunk forecasting
 - Circuit vs packet switched networks
 - Services and key terms (ip pbx/ pure IP, hybrid, enabled, centrex , softphone and more)
 - Architecture
 - Terminals
 - Gateways
 - Call Control Servers
 - Application Servers
 - Media Servers
 - Management devices
 - Work process highlights
- Media transport
 - RTP
 - RTCP
 - cRTP
 - Hands on:
 - Wireshark practice with TCP/IP, DHCP, ARP, DNS, HTTP, etc.
 - Capture, analysis and playing of RTP session live traffic
- SIP
 - SIP properties
 - Functional components
 - SIP message format
 - SIP requests
 - SIP responses
 - PSTN Cause Code and SIP Event Mappings
 - SIP transactions
 - SIP procedures
 - Address resolution
 - SDP
 - SIP extensions and enhancements
 - Call flows
 - Interworking
 - SIP design framework
 - Hands on:
 - Wireshark practice with SIP, SDP and correlation with RTP/RTCP live traffic

- H.323
- H.323 - overview (voice / video / data over ip)
 - H.323 - what does it do?
 - What is h.323?
 - Key benefits
- H.323 component descriptions
 - H.323 components and building blocks
 - Endpoints, terminals
 - Gatekeeper properties and gateway properties
 - Mcu , mc , mp characteristics
 - H.323 zone
- H.323 protocol descriptions
- Call signaling models
- Hands on:
- Wireshark practice with H.323 captures
- Industry perspective
- WebRTC
 - Hands on:
 - Wireshark trace of WebRTC live traffic
- SIGTRAN and SIP-T
 - SS#7
 - VoIP and SS#7
 - SCTP
 - SIGTRAN
 - SIP-T
 - Hands on:
 - Wireshark practice with SCTP, SIGTRAN and SIP-T captures
- Gateways
 - Motivations
 - History
 - Architecture
 - Gateway Control Protocol Features
 - MGCP
 - Architecture
 - Connection Model
 - Commands
 - Protocol Issues
 - Packages
 - Example of Message Flows
 - Hands on:
 - Wireshark practice with MGCP captures
- H.248/Megaco

- Architecture
- Connection Model
- Commands
- Protocol Issues
- Packages
- Example of Message Flows
- Hands on:
 - Wireshark practice with MEGACO captures
- Industry Perspective
- QoS
 - What is QoS?
 - QoS over IP:
 - Static / predictive
 - Intserv / RSVP / MPLS / ATM
 - Diffserv
 - Hands on:
 - Wireshark practice with DHCP captures
 -
 - QoS for VoIP
 - H.323
 - SIP
 - Network management and billing
 - Security
 - Security model
 - Threats
 - Telephony
 - IPTel
 - Application
 - Security strategies
 - IP Network
 - FW
 - NAT traversal (STUN, TURN, ICE)
 - Application
 - SBC
 - AAA - Authentication, Authorization, Accounting
 - LI - Lawful interception
 - Best practices
 - Presence
 - Hands on:
 - Wireshark practice with presence live traffic
 - SIP-IM
 - Hands on:

- Asterisk
- Trends
 - NGN
 - FMC
 - 3GPP
 - IMS
- Wireshark practice with SIP-IM live traffic