

VoIP Fundamentals Workshop (VoIP-FDN, 4 days)

Course Description

In this course, you will gain essential knowledge on how VoIP works, how VoIP can be implemented and how to support it.

With roughly 50% of class time spent on practical labs, you will gain proficiency with many of the most popular VoIP software and hardware, such as Wireshark, Asterisk soft PBX, Brekeke SIP Proxy Servers, Ethernet VOIP phones and more.

Who Can Benefit

- Network and Systems Administrators
- LAN Designers and Infrastructure Integrators and Consultants
- Network, Infrastructure and general IT managers

Prerequisites

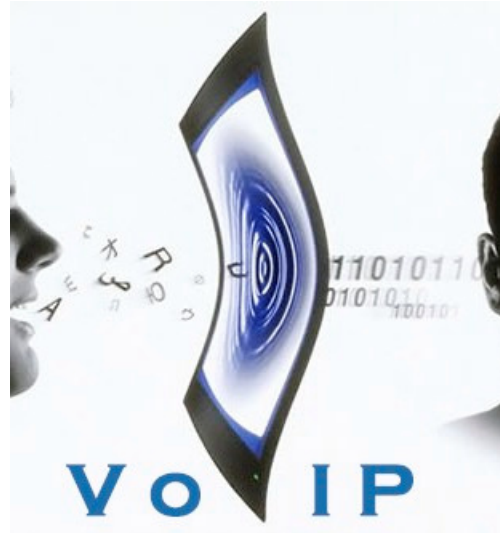
To succeed fully in this course, participants should have:

- Basic understanding of, and experience in, LAN technologies

Skills Gained

Upon completion of this course, participants should understand:

- Packetizing Voice
- Voice Compression and Decompression (CODECs)
- IP Infrastructure for VoIP
- Session Initiation Protocol and H.323 Protocols
- Real-time Transport Protocol
- Session Description Protocol
- QoS (Performance) Mechanisms and concerns
- Monitoring and Troubleshooting call flows



REGISTRATION AND INFORMATION

education@ecs.com.sg
www.ecs.com.sg/training

VoIP Fundamentals Workshop (VoIP-FDN, 4 days)

Course Content

Module 1 - Digitizing Voice

- PSTN Architecture
- VOIP Standards
- Traffic Engineering
- VOIP Trucking
- VOIP Connection
- Voice Digitization
- Companding
- TDMA
- Voice Packet Analysis
- QoS Overview
- Controlling Delay

Module 2 - Voice Compression

- Voice Compression
- Dedicated Hardware
- PCM Sampling
- Codecs
- G.711, G.723.1, and G.726
- G.728 and G.729
- Mean Opinion Scores
- Formants and Fricatives
- The Predictor
- Voice Compression Algorithms
- ADPCM Compression
- Vocoder

Module 3 - VoIP LAN Transmission

- ARP, Hardware and Logical Addresses
- Hubs and Switches
- Flooding, Broadcast and Multicast
- Switching and Port Information
- Virtual LAN Review
- Port-Based VLANs for VoIP
- VLAN Trunking for VOIP
- IEEE 802.1q Frame Tagging for CoS
- IEEE 802.1p Frame Prioritizing for QoS

Module 4 - VoIP WAN Transmission

- Interior Routing Protocols
- Static Routing
- Subnet Masks and Routing
- Routing and Switching
- Transmission Control Protocol
- TCP/IP Packet Format and Operation
- User Datagram Protocol
- UDP Packet Format and Operation

Module 5 - Supporting IP Services

- Dynamic Host Configuration Protocol
- Domain Name Services for VoIP
- NAPTR Enumeration
- SRV Server Location
- Replacement Rules for NAPTR

Module 6 - Real-Time Transport Protocol

- Encapsulating the Voice Packet
- Real-Time Transport Protocol Architecture
- RTP Timestamps
- RTP Control Protocol and Profiles
- RTP Payload Types and Codec Mapping
- Multi-stream Mixers
- SSRC and CSRC
- UDP / RTP Packet and Payload
- Dialing Events and Format-Specific Parameter Controlling Jitter
- Controlling Serialization Delay
- RTP Header Compression
- RTCP Round Trip Delay Calculation

Module 7 - SIP Components

- SIP Proxy Types and Functions
- SIP Proxy
- SIP Back-to-Back UA
- Session Border Controller
- SIP Trapezoid Configuration
- RFC 2396 URI Structure and Parameters
- SIP User Agent
- SIP Methods (Requests)
- SIP Responses (Codes)

Module 8 - VoIP Call Flow Analysis

- SIP Call Analysis
- SIP Registration with Authentication
- SIP 100 Call without INVITE Authentication
- SIP 100rel Process
- SIP 200 Response
- SIP Busy Number
- SIP Call Cancel
- SIP Call Forward
- SIP Call Transfer

Module 9 - SIP Header Analysis

- Request Message Syntax
- Response Message Syntax
- Content-Type Header
- Supported Header
- Content-Length Header
- Allow Header
- The Start Line
- Via Header
- SIP Dialog
- From Header
- To Header
- Call-ID Header
- Dialog State
- CSeq Header
- Max-Forwards Header
- Proxy-Authenticate Header
- Contact Header, Expires Header, User-Agent Header

REGISTRATION AND INFORMATION

education@ecs.com.sg
www.ecs.com.sg/training



VoIP Fundamentals Workshop (VoIP-FDN, 4 days)

Module 10 - Session Description Protocol Analysis

- Session Description Protocol
- Offer/Answer Model
- SDP Offer/Answer Rules
- UPDATE Method
- RTP SEND and RECV Defined
- Media Direction and RTCP
- How RTCP Works
- Call Hold, Forward and Transfer Issues

Module 11 - NAT and Firewall Issues

- 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 12 - Media Gateway Control Protocol

- MGCP Call Model
- MGCP Command Syntax
- Defined Endpoints
- Return Codes
- Parameter Lines
- DTMF Package
- Line Package
- Digit Maps
- MGCP Trace Procedure

Module 13 - H.323

- H.323 Components
- H.323 Terminal Equipment
- H.323 Gateway
- H.323 MCU Components
- ASN1 Encoding
- H.323 Protocols
- Registration: GRQ / RRQ
- Signaling: H.225 and Q.931
- Negotiation: H.245
- Media Transfer: H.323 and RTP
- MCU Controlled Transcoding
- Terminal Type Identifiers for H.323 Master/Slave Determination
- Gatekeeper Control
- Gatekeeper Discovery Multicast Method
- Gatekeeper Multicast Filtering
- Gatekeeper Direct Endpoint Routing
- Gatekeeper Routed Trace

Module 14 - QoS-Related Protocol

- CoS vs. QoS
- Service Differentiation
- Session ID Classification (Fair Queuing)
- Type Classification
- Leaky Bucket
- FIFO
- Dequeuing
- Packetization Delay
- Algorithmic Delay
- Coder Processing Delay
- Queuing Delay
- Serialization Delay
- VOIP with DiffServ
- Calculating ERLANGs and Bandwidth
- Planning to meet your VoIP Grade of Service Agreement

REGISTRATION AND INFORMATION

education@ecs.com.sg
www.ecs.com.sg/training

