Asterisk IP Telephony

This course will teach participants how to install, configure and maintain the popular Asterisk IP PBX. Asterisk is software that turns an ordinary computer into a PBX, VoIP Gateway, and conference server all rolled into one. At the end of the course, participants are expected to have an in-depth knowledge of Asterisk and its possible applications in the world of digital telephony. In order for the course to be beneficial, the trainees must have installed and played with Linux in the past and must be comfortable with the Linux CLI and basic text editing tools. This course also designed on VoIP, VSP, IGW, ICX, IIG ISP, IPTSP, Mobile & Telecom Company based VoIP infrastructure to prepare you for a successful VoIP professional.

Course Objectives

- Advance knowledge of VoIP
- Analogue and digital circuits, PBXs, trunk and tie lines
- Different types of voice Codec
- Signaling protocols H323, MGCP, SIP, SCCP and protocols, RTP and RTCP
- Dial peers and plans, voice gateways
- IPTSP provider circuits
- Messaging and call processing
- Understanding of Infrastructure and equipment required for voice and data at layer 2
- Install and configure Asterisk IP PBX System
- Billing System, Configuring dial plans

Course Duration

- 40 Hours, 10 Classes, 4 Hours per Class

Details Course Outlines

Lesson 01: VoIP Overview and IP Network Basics

- Fundamentals of IP Telephony and VoIP
- Analog Telephony and IP Telephony
- Open source Telecomm Technologies and Software
- IP, TCP, UDP, RTP, Mac Address, DNS, Routing, NAT, VPN
- Linux and Open Source Tools and Technologies

Lesson 02: Telephony and Signaling Systems

- Circuit Switching
- Packet Switching
- Jitter, Delay, Packet Loss
- Time Division Multiplexing
- Transports: E1/T1, STM, SDH, PDH, ISDN, PRI, BRI
- SS7 Signaling
- IP Telephony Protocols: H323, SIP, IAX2

Lesson 03: Asterisk System and Operations

- Asterisk Architecture
- Hardware
- Asterisk Modules and Configuration Files
- Audio/Video Codecs and Issues
- DAHDI Drivers
- Asterisk Installation
- Creating Extensions
- Creating Basic Dial plans

Lesson 04: Switching and Dial Plan

- Contexts
- Extensions
- Priorities
- Applications
- Building an Interactive Dial Plan
- Variables
- Pattern Matching

Lesson 05: Trunking

- Outside Connectivity
- PSTN Circuits
  - Traditional PSTN Trunks
  - Installing PSTN Trunks
- Coping with NAT
- PSTN Termination
- PSTN Origination
- Configuring VoIP Trunks (SIP/IAX2)

Lesson 06: Advanced PBX Features and Call Center System

- Parking, Paging and Conferencing
- IVR
- Queues
- ACD
- Agent
- FAX Integration

Lesson 07: Number Planning Schemes and ITSP Design and Implementation

- IPTSP Basics and Planning (NSPC,IGW,ICX)
- GOV Regulations and Issues
- DNS and SIP URI's
- ENUM and E.164

Lesson 08: System Monitoring and Logging

- Relational Database Integration
- Call Detail Records (CDR)
- CEL (Channel Event Logging)
- SNMP
- Monitoring Asterisk with OpenNMS
- SIP Message Debugging
- Low Level Debug and Tracing

Lesson 09: Security and Performance Issues

- QoS and Traffic Prioritization
- Authentication Weakness
- Voice Encryption (SRTP, ZRTP)
- Denial of Service Attacks
- Fail2Ban
- Firewall

Lesson 10: Web Integration, Monitoring, Reporting and Billing

- FreePBX
- Queue Status and Reporting
- Flash Operator Panels
- A2Billing

Lesson 11: Asterisk based Embedded Solutions, Call Centers and Predictive Dialer Solutions

- GoAutodial
- VICIDial
- Elastix
- Trixbox
- Askozia

Lesson 12: Asterisk Scalability, Clustering and High Availability Solutions

- SIP Express Router (SER)
- OpenSBC (Session Border Controller)
- Freeswitch
- GNUGK