Asterisk Voice-Over-IP PBX Training Course Datasheet (Tentative)

The Institute for Open Systems Technologies Pty Ltd

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This 3-day course covers the Asterisk open source PBX. This class teaches you how to install, configure, integrate, provision, troubleshoot and manage the Asterisk software. **Installation topics** *include setting up SIP and IAX phone handsets as well as the base software on Linux systems.* **Configuration topics** *include setting up dialplans, call centres, transfers, agent logins, on-hold music, call parking, call recording, billing and chargeback.* **Integration topics** *include Jabber (for caller ID details), the wider PSTN/ISDN telephony networks, ENUM and some simple AGI scripts for interfacing with arbitrary customer systems.* **Provisioning topics** *include options with LDAP and Radius and other template creaton tools, as well as developing an understanding of Asterisk's performance issues.* **Troubleshooting topics** *include deciphering SIP protocol messages, and a variety of network and performance measurement tools.* **Management topics** *include using astman, supporting failover configurations, SNMP agents and techniques for pre-emptively alerting about problems.*

Audience

- System and Network Administrators
- Network Architects
- Telephony Engineers, Designers and Operators

Pre-requisites

- Familiarity with networking
- Knowledge of Linux will be helpful

Benefits to You

- Understand the features and functions of Asterisk.
- Get to the forefront of voice-over-IP technology
- Develop skills across telephony, open source solutions and advanced networking.
- Practical hands-on lab exercises.

Course Outline

Installation

- RPMs, DEBs and compiling from source
- SIP phone handsets
- Software phones
- Creating entries in sip.conf

Basic Dialplans

- 1-1
- Ringing several handsets
- Contexts
- Calling another system
- IAX
- SIP outgoing calls
- Dangerous things to avoid

Troubleshooting

- Astman, gastman, asterisk console
- SIP protocol messages
- Debugging using asterisk, linphone and x-list
- Ettercap, ethereal and stak
- Latency and jitter – iperf
- Proactive monitoring

Featureful Dialplans

- Call parking
- Recording calls
- Attended call transfers
- Unattended call transfers
- Answering calls automatically
- Speech synthesis and existing sound files

Call Centres

- Call queues
- Agent logins
- Call backs
- Orderly queues
- On-hold music

Financials

- Call records (CDR)
- Database interfaces
- Yada
- Calling card applications
- Least-cost routing

Sophisticated Dialplans

- Caller-ID messages and Jabber Integration
- AGI scripts
- ENUM and DUNDI
- Zaptel cards, PSTN and ISDN

Provisioning

- Simple approaches
- Setting up LDAP
- LDAP lookup in a dialplan
- LDAP lookup for phones and agents
- Radius integration

Large Deployments

- Clustering and failover
- SIP Reinvites
- Dialplan options that cause Asterisk to stay in the Media Path?
- SER as a front-end
- SNMP management
- Scalability issues