OpenSIPS Bootcamp

The ***OpenSIPS Bootcamp*** is a full 5 day (40 hours) intensive training providing in depth coverage of OpenSIPS Installation, configuration and administration. The students will learn how to download, compile and install OpenSIPS. After the installation, you will start to learn step by step how to configure OpenSIPS to authenticate users, install a GUI to help with daily administration, forward calls to the PSTN, integrate Asterisk and Voice Mail, Presence agent, Load Balancing, Traverse Nat for SIP and generate CDR records to a Radius Server. At the end, you will learn how to use troubleshooting tools to solve end user problems.

All the knowledge that is transferred to you will be strongly backed-up by practice sessions where you will get hands-on experience in handling OpenSIPS SIP Server. The training is structured to be offer 50% - 50% between the theoretical and practical sessions.

Optionally, an certification exam, to proof the knowledge consolidation during the training, can be sustained on request at the end of the course.

1. Key objectives
2. Audience
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.1 Key Objectives

- Explain what SIP, what it does and its architecture
- Explain Open SIP Express Router
- Install OpenSIPS on a Linux Machine
- Routing basics and the default configuration
- OpenSIPS authentication using MySQL
- Install SerMyAdmin for Administration
- Connect to the PSTN using a gateway
- Implement call forward to voicemail
- Implement presence agent
- Understand important aspects of load balancing and high availability
- Implement SIP NAT traversal using RTPProxy
- Account Calls to MySQL and Radius
- How to use test and monitoring tools to check your configuration

.2 Audience

- VoIP providers seeking "Open Source" platforms to enhance their businesses
- Anyone seeking proficiency in OpenSIPS
- Network Consultants and VARs who need a jump start in the technology
- Developers who want to use OpenSIPS to create new telephony applications and appliances

.3 Instructors

Bogdan-Andrei Iancu – OpenSIPS founder and main developer. Also CEO of Voice System, an "know-how" OpenSIPS company.

Flávio E. Goncalves – CEO of V.Office Networks, writer of the book, Building Telephony Systems with OpenSER.

.4 Prerequisites

- Basic Linux knowledge
- Basic text edition
- Basic SIP protocol
- Programming logic knowledge (you won't need to program, but you to understand logical concepts applied to the dial-plan)

.5 Syllabus

1. Introduction to OpenSIPS
    a. What is Open SIP Express Router
    b. Main characteristics
    c. Usage scenarios
    d. OpenSIPS architecture
    e. Sessions, Dialogs and Transactions
    f. Message Processing according to the RFC3261
    g. Strict Routing and Loose Routing

h. SIP and RTP

2. OpenSIPS installation
    a. Hardware requirements
    b. Software requirements
    c. LAB 2.1 – Installing Linux for OpenSIPS (previously installed DVD)
    d. LAB 2.2 – Download, compile and install OpenSIPS
    e. LAB 2.3 – Running OpenSIPS at the Linux Boot
    f. OpenSIPS directory structure and log files
    g. OpenSIPS startup options
    h. Starting OpenSIPS with default configuration script
3. Routing Basics and the Standard Configuration
    a. Scripting Basics
    b. Routing Basics
    c. Analyzing the standard configuration files
    d. LAB 3.1 Connecting two phones to OpenSIPS
    e. LAB 3.2 Running stateful with record routing (packet capture)
    f. LAB 3.3 Running stateless with record routing (packet capture)
    g. LAB 3.4 Running stateless with no record-routing.
4. Adding authentication with MySQL
    a. The Auth_DB modules
    b. Register authentication sequence
    c. Invite authentication sequence
    d. Digest authentication
    e. QOP – Quality of protection
    f. Plaintext or hash passwords
    g. LAB 4-1 Installing MySQL Support
    h. The OpenSIPsctl shell utility
    i. The OpenSIPsCTL resource file
    j. Checking From and TO tags
    k. Multidomain support
    l. Inter-domain and intra-domain routing
    m. LAB4-2 Enhancing the script
5. Building a user portal with SerMyAdmin
    a. Introduction to SerMyadmin
    b. LAB 5-1 Installing SerMyAdmin
    c. Basic tasks
    d. Registering a new user
    e. Domain administration
    f. User administration
    g. Interface customization
6. Connectivity to the PSTN
    a. Introduction to PSTN routing
    b. Accepting calls from the PSTN
    c. The permissions module and the allow_trusted() function
    d. Routing a call to the PSTN

   e. DID redirection using Aliases
   f. ACL and Group permissions
   g. Introduction to LCR
   h. LCR tables
   i. OpenSIPsctl lcr commands
   j. LAB 6-1 Using LCR to route calls to the PSTN
   k. Inter-domain Peering
7. Call Forwarding and Voicemail
   a. Introduction to Call Forwarding
   b. Pseudo-variables and AVPs
   c. AVP functions
   d. Implementing Blind Call Forwarding
   e. Busy or Unanswered forwarding to Voice Mail
   f. LAB 7-1 Testing the Call Forwarding feature
8. Using Presence
   a. SIP presence overview
   b. Presence Agent setup
   c. Publishing Presence from non-SIP devices
   d. Registration-to-Presence conversion (old SIP devices)
   e. Scalability of the presence model
   f. Aggregation of the presence information
   g. LAB 8-1 Implementing presence aggregation
   h. LAB 8-2 Publishing non-SIP Presence
9. Load Balancing and High Availability
   a. OpenSIPS High Availability
      i. Active/Active and Active/Backup setups
      ii. SIP and Data Replication
   b. OpenSIPS Load balancing/Dispatching Capabilities
      i. Balancing algorithms
      ii. Balancing and failover
      iii. Multiple groups of balancing
   c. LAB 9-1 Load balancing & failover foran Asterisk Cluster
10. SIP NAT Traversal
   a. NAT Types
   b. Solving the NAT traversal challenge
   c. Implementing a far end NAT solution
   d. RFC3581 and forc_rport() function
   e. Solving the traversal of RTP packets
   f. Handling Register Requests
   g. Detecting clients Behind NAT
   h. Handling Invite requests behind NAT
   i. RTPProxy installation and configuration
   j. LAB 9-1 Usind RTPProxy for NAT traversal
   k. STUN – Simple Traversal of UDP NAT
11. OpenSIPS accounting and Billing

a. Authentication, Accounting and Authorization
b. LAB 10-1 Accounting to a MySQL database
c. Accouting using a RADIUS server
d. LAB 10-2 Accounting to a Radius Server

12. Troubleshooting Tools
    a. Built in tools
    b. Packet Capture and Trace Tools
    c. The SIPTRACE module
    d. Predefined and Custom Statistics
    e. Stress Testing Tools
    f. LAB 12-1 (If time permits) using sipp to stress test OpenSIPS

.6 Agenda: 40 hours

	Mon	Tue	Wed	Thu	Fri
08:30-10:00	Arrivals	Routing Basics LABs 3.1-3.4	Admin LABs – Manag.users and domains	8. Presence and LAB 8-1	NAT Traversal LABs 10.1 or 10.2
10:00-10:15	Coffee Break	Coffee Break	Coffee Break	Coffee Break	Coffee Break
10:15-12:30	1. Introduction to OpenSIPS	4. Adding Authentication with MySQL	6. Connectivity to the public network	9. Load Balancing and High-Availability	10. Accounting with Freeradius
12:30-13:30	Lunch Break	Lunch Break	Lunch Break	Lunch Break	Lunch Break
13:30-15:30	2. OpenSIPS Installation LAB	Auth LABs 4.1-4.3	PSTN Labs.6.1	9. LAB 9.1 – Load Balancing	LAB 10.1 – Accounting using freeradius
15:30-15:45	Coffee Break	Coffee Break	Coffee Break	Coffee Break	Coffee Break
15:45-18:30	3.Routing and Scripting Basics.	5. Managing OpenSIPS with SerMyAdmin	7. Call forwarding and LABs	10. SIP NAT Traversal	11. Troubleshooting tools and LABs

.7 Certification

Optional, the students can choose to additionally sustain a certification exam to confirm the knowledge consolidation during the course.

The certification exam (with time limit) will be supervised by our teachers and it will consist in random questions (from a larger set) that cover all OpenSIPS relevant areas. If approved, Voice-Systems and V.Office Networks will issue a certificate of OpenSIPS Accredited Professional.

The certificates are signed by both our teachers as representative players in the OpenSIPS project.

.8 Handouts / Materials

1. Book – Building Telephony Systems with OpenSER

2. DVD containing:
    a. PDF copy of the Overhead transparencies
    b. DVD with the Virtual machine containing Linux with Debian pre-installed and all the scripts used in the training
3. Headset
4. IP Phone: Polycom IP320
5 . T-Shirt: OpenSIPS official T-Shirt (male or female version)