

OpenSIPS Bootcamp 1.5

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

The new training will employ a new exclusive hybrid methodology combining online and Instructor-led Training. The idea is to level the training students using eLearning concepts. The videos and exercises about SIP and the introduction to OpenSIPS will be available in our LMS platform. So when you come to the training we will go to the advanced stuff instead of starting from the very basics. In our opinion this adds value to the training and enhances the student satisfaction.

1. Key objectives
2. Audience
3. Instructors
4. Prerequisites
5. Syllabus
6. Agenda
7. Certification
8. Hand Outs/Materials

.1 Key Objectives

- Install OpenSIPS on a Linux Machine
- Routing basics and the default configuration
- OpenSIPS authentication using MySQL and Memcache
- Install OpenSIPS control Panel. SerMyAdmin demonstration
- Connect to the PSTN using Dialplan and Dynamic Routing
- Voicemail integration using Call Forward and AVPs
- Implement a presence agent
- Understand important aspects of load balancing and high availability
- Implement SIP NAT traversal using RTPProxy
- Account Calls to MySQL
- 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 (Free online learning)
- Basic OpenSIPS Knowledge (Free online learning)
- Programming logic knowledge (you won't need to program, but you to understand logical concepts applied to the dial-plan)

.5 Syllabus

1. 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
- 2. 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.
- 3. 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 / osipconsole shell utility
 - i. The OpenSIPsCTL / osipsconsole resource file
 - j. Checking From and TO tags
 - k. Multidomain support
 - l. Inter-domain and intra-domain routing
 - m. LAB4-2 Enhancing the script
 - n. Using memcache to speed up queries
- 4. Graphical User Interfaces
 - a. Introduction to the OpenSIPS Control Panel
 - b. Installing the OpenSIPS Control Panel
 - c. Basics of the OpenSIPS Control Panel
- 5. 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 Dialplan and Drouting
 - h. DROUTING Tables
 - i. LAB 6-1 Using drouting to route calls to the PSTN
 - j. Inter-domain Peering
- 6. 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

7. 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
8. Load Balancing and High Availability
 - a. OpenSIPS High Availability
 - i. Active/Active and Active/Backup setups
 - ii. SIP and Data Replication
 - b. OpenSIPS Dispatching Capabilities
 - i. Dispatching algorithms
 - ii. Dispatching and failover
 - iii. Multiple groups of Dispatching
 - c. Real Load Balancing
 - i. Resource management and load
 - ii. Balancing algorithm
 - iii. Balancing and failover
 - d. LAB 9-1 Load balancing & failover for an Asterisk Cluster
9. 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
10. OpenSIPS accounting
 - a. Authentication, Accounting and Authorization
 - b. LAB 10-1 Accounting to a MySQL database
11. 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	3. Adding Authentication with MySQL	5. Connectivity to the public network	8. Load Balancing and High-Availability	10. Accounting
10:00-10:15	Coffee Break	Coffee Break	Coffee Break	Coffee Break	Coffee Break
10:15-12:30	1. OpenSIPS Installation LAB	Auth LABs 3.1-3.3	PSTN Labs.5.1	LAB 8.1 – Load Balancing	11. Troubleshooting tools
12:30-13:30	Lunch Break	Lunch Break	Lunch Break	Lunch Break	Lunch Break
13:30-15:30	2. Routing and Scripting Basics	4. Managing OpenSIPS with the Control Panel	6. Call forwarding and LABs	9. SIP NAT Traversal	Spare Time
15:30-15:45	Coffee Break	Coffee Break	Coffee Break	Coffee Break	Coffee Break
15:45-18:30	LABs 2.1-2.4.	Control Panel Labs	7. Presence and LAB 8-1	NAT Traversal LABs 9.1 and 9.2	Spare Time

.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