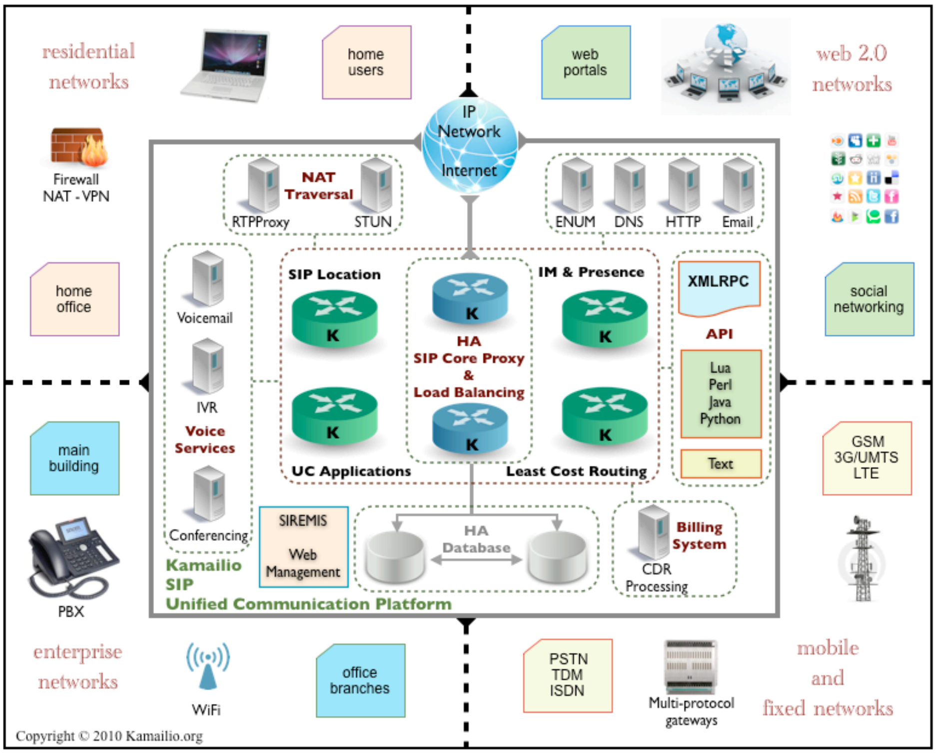
SIP ROUTER MASTERCLASS

November 8-12, 2010
Berlin, Germany



Complete Kamailio SIP Server Training - http://www.kamailio.org

SIP Router Masterclass is a five days of training Kamailio SIP Server and integration with Asterisk Media Server. Starting with ground level, the concept and design of a SIP server, the course focuses on building a complete telephony system.

Kamailio (former OpenSER) and SER are the leading open source SIP servers, routing billions of minutes and handling millions of active VoIP users each month. Well known for its stability and flexibility the SIP Express Router (SER) family of SIP servers is continuously increasing the adoption on the market. With the launch of SIP-Router.org project, the SIP server secured a reliable development team, backed up by strong business community.

Very popular among VoIP and Internet Telephony service providers, Kamailio went beyond voice, offering complete communication service: Voice/ Video, Instant Messaging and Presence. More over, the application can serve you implement any session-based communication, like gaming or peer-to-peer data exchange.

Learning to configure the SIP server is not easy, but is the key for a successful and secure business. The flexibility of Kamailio allows you to implement in no time innovative services, doing it right will save time and money.

Target attendees

- administrators of VoIP/Internet Telephony systems using SER-family of SIP Servers: Kamailio, SIP Express Router or other variants
- people looking to create or use cost effective Internet Telephony solutions
- network operators needing to scale up their SIP infrastructure

Kamailio was awarded Best of Open Source Networking Software by InfoWorld magazine 2009

Venue Details

Time: The training starts at 10:00 AM on Monday, November 8, 2010, and ends 3:00 PM on Friday, November 12, 2010. Tuesday to Thursday class start 9:00 AM and ends 5:00 PM.

Location: Berlin, Germany. Address of our training center and hotel recommendations will be provided at registration time.

Course Structure

- Each student will have access to a desktop or laptop computer and a SIP phone, therefore is no need to bring anything with you. Servers and students workstations are provided on site
- A print out of the training materials will be handed over upon arrival
- However, feel free to bring your own laptop and/or SIP equipment to the training. Make sure that all equipment is insured by you or your company. It is a good place where you can test your devices or applications and an unique opportunity to work together with the people that created these technologies.

Monday	Tuesday	Wednesday
Welcome - introduction - presentation of course structure - lab environment	Helper tools: - kamctl - siremis web interface Kamailio Internal design - SIP routing concept	Media Services - Asterisk Overview - design - Asterisk vs Kamailio - RTP and RTCP
Session Initiation Protocol - RFC3261 - protocol design - SIP routing - SIP extensions	Troubleshooting - debug config file - SIP traffic monitoring - watchdog your SIP server and benchmarking LAB: Troubleshooting	Asterisk and SIP channel - realtime - b2bua - dialplan - IVR LAB: Install Asterisk
SIP Continuation - SDP Kamailio overview LAB: Setting up Kamailio	Database Integration - Authentication - Authorization, ACL Transport protocols TLS secure communication	Integration of Kamailio and Asterisk - voicemail - announcement services - conferencing
SIP Continuation - SIP vs. Kamailio config - transactions, dialogs Kamailio, SER and the SIP-Router.org project	Core cookbook - elements of config file - routing blocks LAB: Kamailio with Database support & AAA	LAB: Kamailio + Asterisk – complete VoIP solution

Knowledge Transfer

With a step by step approach and many hands on labs, you will learn about:

- SIP – introduction and the relation with SIP server
- Kamailio, SER and the SIP-Router.org project
- Troubleshooting SIP and test tools
- Configure authentication and authorization
- Create accurate accounting and call data records
- Go through NAT and firewall
- Secure communication with SIP and TLS
- SIP beyond voice: IM, Presence, Video
- Introduction to Asterisk media server
- Asterisk real-time integration with SIP server
- Prefix and Least Cost Routing
- Load balancing and traffic dispatching
- Scalability, redundancy and failover
- Building a SIP network with Kamailio and Asterisk

At the end of the course you will be able to design and deploy a complete SIP platform, with proper network infrastructure dimensioning, service and user partitioning, security and failover.

The content of the course will be adapted to meet the interest of the attendees as long as it follows the generic content and the goal of the class.

Thursday	Friday
Accounting - design, call data records - postpaid vs. postpaid - multi-leg accounting	Buffer zone LAB Building a failover SIP network
NAT traversal - solutions, optimizations Topology hiding Privacy LAB: Accounting & NAT	Buffer zone LAB Building a failover SIP network
SIP Presence - server user agent Advanced routing - least cost routing - load balancing	LAB Building a failover SIP network
- ENUM, traffic shaping Service reliability - replication - failover LAB: IM and Presence	LAB Building a failover SIP network

TEACHERS

Daniel-Constantin Mierla

Co-founder of Kamailio (OpenSER) project in 2005, focusing on technical aspects and quality assurance as core developer but also deeply involved in building the community around the project, he had a major contribution to turn into a great success the open source sip server now used in many VoIP platforms around the world.

Starting in 2002 as researcher at Fokus Institute, Berlin, Germany and core developer of SIP Express Router, along the last 8 years he has designed, developed and deployed close to hundred VoIP platforms witnessing the evolution of VoIP services world wide.

With a strong background in SIP and VoIP, he is proud to share his knowledge helping tech people and team leaders implement in the shortest time scalable VoIP platforms.

Olle E. Johansson

Active Asterisk developer with many years of experience of running Asterisk in enterprise and service provider networks, he has been teaching Asterisk since early 2005.

Previously taught many classes in networking, IP, IP security, LDAP, XML and other topics proving highly skilled in sharing the knowledge to the audience.

His vast experience in teaching is a key ingredient to a high standard training on deploying VoIP platforms based on Kamailio open source sip server.

Olle currently works on enhancing the Asterisk SIP channel and have participated in two international SIP interoperability test events with Asterisk. He's also a member of the Digium Asterisk Advisory Board.

WHAT MAKES THE COURSE SPECIAL?

learn from people deep involved in these technologies - the ones that have driven the projects to become successful – founders and core developers of OpenSER, Kamailio, SIP Express Router and Asterisk

experienced teachers – Daniel-Constantin Mierla and Olle E. Johansson have by now a long collaboration relationship in providing VoIP and SIP related courses. The expertise they accumulated helps to adapt to the needs of students without losing the substance of the course.

entire network infrastructure at your hands – the lab is fully equipped by us. You don't need to worry you would screw up something, students have the freedom to use preferred Linux distribution, install tools they feel comfortable with. Some highlights:

you get access to: dozens of servers, SIP phones, network switches and hubs. It is very rare when somebody gets access to such testbed and it is in the company of other experienced people. It is very easy to simulate heavy traffic, large deployments and failover scenarios

you can plug your devices in the network, therefore you have the chance to test and integrate it in a SIP platform during the course

focus on quality – we provide to students everything is needed for the course so they can use the time strictly to learning, labs and testing. The price of the course is set by the quality of the content.

usable results – the labs are planned carefully so you just keep adding new features as you learn to a SIP platform that can be used in real world. It is not a mosaic listing functionalities and snippets of configuration file. The Friday is the buffer zone and allocates lot of time to build a scalable SIP platform, with redundancy and failover, that includes the features you have learned about during the previous days.

team work – the class works as a team. Labs are discussed and planned together, students will group to design and implement components, supervised by teachers.

open discussions – the course is structured to allow breaks for coffee, cookies, beverages and open discussions, time that can be used to talk about specific needs or technologies

networking – not ultimately, the SIP Router Masterclass is a networking opportunity. You will meet there people with expertise in different areas of interest, sharing the experience and learning about various tools may easy your job, give new ideas and enrich your knowledge

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Contact us for registration
or to ask more details at:

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REGISTRATION PRICE

- before Sep 24: 2750 EUR + 19%VAT
- before Oct 20: 2950 EUR + 19%VAT
- before Nov 06: 3450 EUR + 19%VAT

The payment must be done in advance by bank wire (You will get an invoice with instructions). The price includes training documentation, lunches and refreshments.

We do not give refunds on late cancellations or no-shows. Cancellation has to be sent to us at least two weeks in advance.

Companies sending groups of at least three persons have a discount of 10%. If you are interested in reselling the course, please contact us to discuss terms and conditions.