Kamailio (OpenSER) 3.0.0

A SIP-Router.org Project

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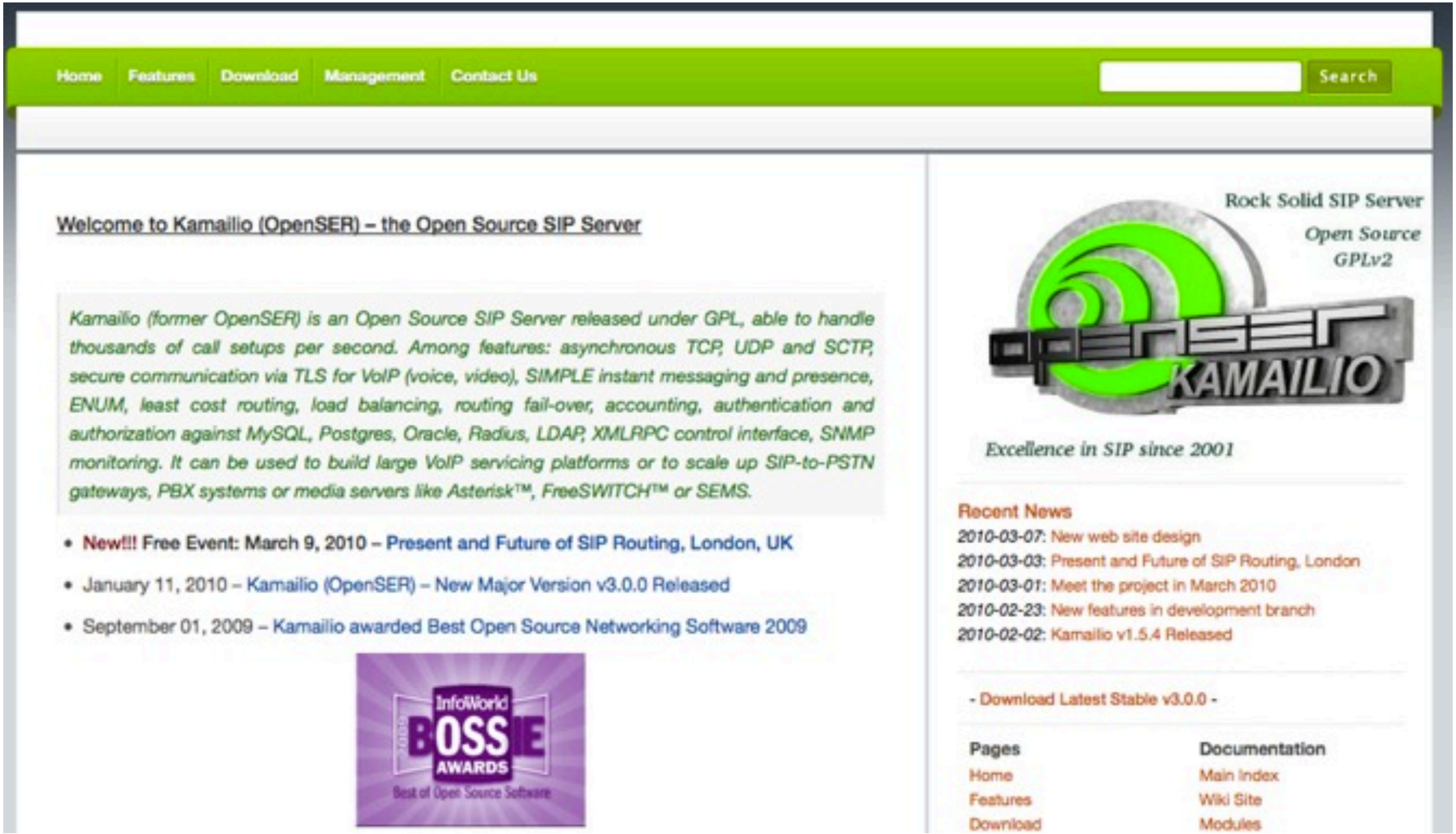
Elena-Ramona Modroiu

Co-Founders Kamailio

Asipto

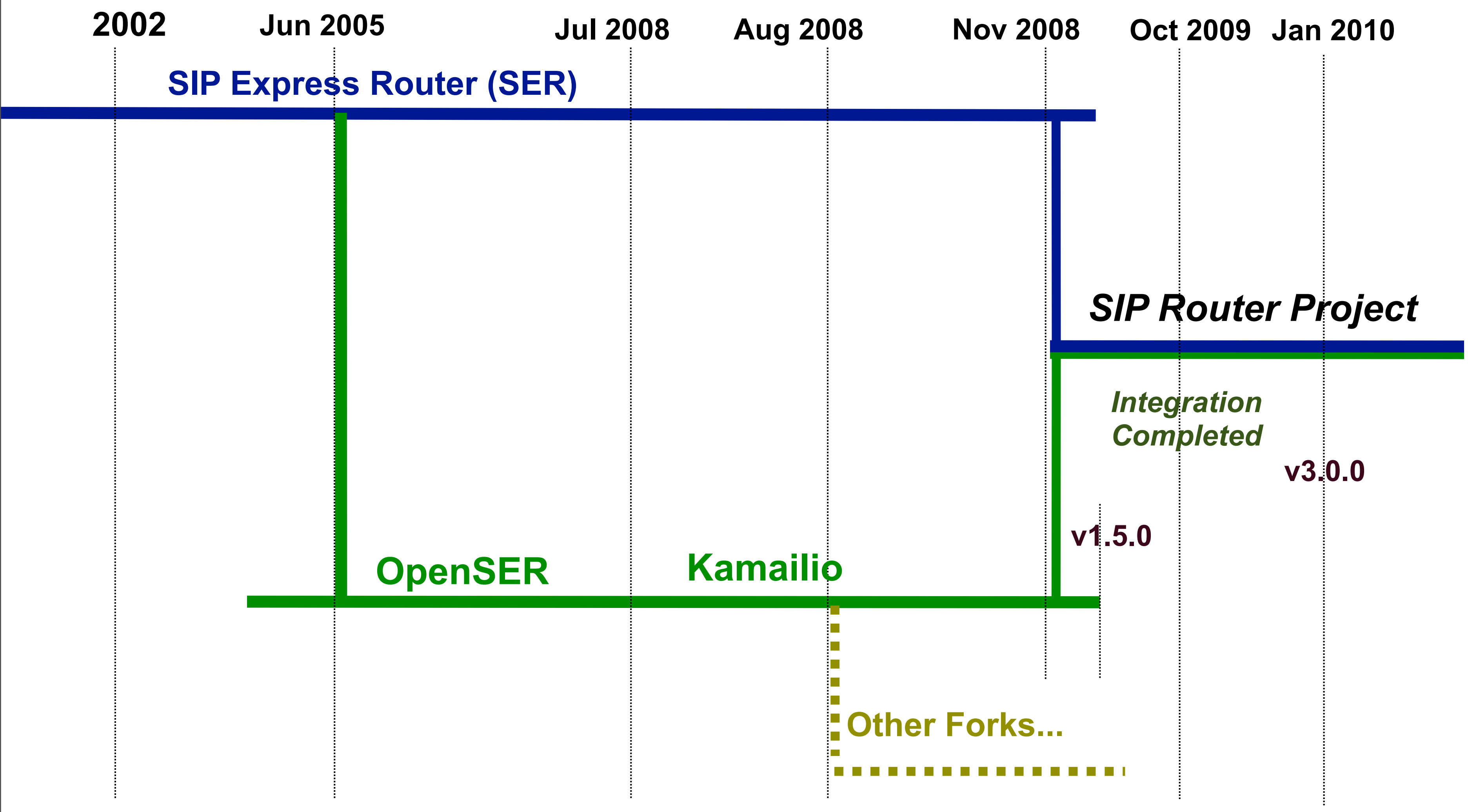
www.kamailio.org



About

- Berlin, quite some time ago
  - FhG Fokus Institute
    - where the SIP was born
    - core developer SIP Express Router (SER) project
      - 2001 - internal project
      - 2002 - released as GPL
- co-founder OpenSER
  - 2005 - started from SER
  - willingness for a more Open Source project
    - more than an Open Source application
- Kamailio
  - 2008 - trademark issues
    - still same SourceForge.net project: openser

History



2002
Jun 2005
Jul 2008
Aug 2008
Nov 2008
Oct 2009
Jan 2010
SIP Express Router (SER)
SIP Router Project
Integration Completed
v3.0.0
v1.5.0
OpenSER
Kamailio
Other Forks...

Kamailio (OpenSER) SIP Server

- Open source, GPL SIP server – IETF RFC3261
  - Web: http://www.kamailio.org
  - SourceForge Project: http://sourceforge.net/projects/openser/
  - SIP Router Project: http://sip-router.org

Kamailio (OpenSER) SIP Server

- Development environment:
  - Linux (ported to many unix-like platforms)

  - Written in standard C
    - dedicated code for memory management, synchronization, …
  - Built with free and well-known tools: .
    - gcc, lex, yacc, sed …

Project Management

- Community oriented
  - open source project
  - over 50 registered developers
  - large community
  - many companies involved
    - service providers
    - integrators
    - vendors
- Management team
  - conflict resolution
  - volunteering individuals
  - different companies and countries

New Management Team

- Individuals
  - 11 people as of January 25, 2010
    - Daniel-Constantin Mierla, Elena-Ramona Modoiu, Henning Westerholt, Juha Heinanen, Klaus Darilion
    - Alex Balashov, Andreas Granig, Carsten Bock, Inaki Baz Castillo, Jesus Rodriguez, Marcus Hunger
- Companies
    - Asipto, 1&1, TutPro, Enum.at, Evaristesys, Sipwise, Ngn-ims.com, Sipdoc.net, Voztelecom and Sipgate
- Countries
    - Austria, Germania, Finland, Spain, Romania and USA

Awarded

Best Open Source Networking Software 2009

By InfoWorld

World wide usage

- Some of biggest VoIP deployments world wide
  - 1&1 (members of the management board)
    - Over 3 millions subscribers
    - Over 1.5 billion minutes per month
  - Sipgate
  - Freenet
    - 0.8 million subscribers
    - Hundreds of millions of minutes per month

World wide usage

- Telio
  - Norway, Switzerland, ...
  - over 150 000 lines in 2008
- Voztele
  - Spain, Latin America
- UPC
  - Austria, world wide

http://www.kamailio.org/w/references/

SIP Server

- SIP registrar
  - record registry for users
- SIP location server
  - counterpart for registrar
- SIP proxy server (router)
  - best at this job
  - initial target for the project
    - achieved long time ago
- **SIP application server**
  - **rich telephony services**
  - **integration with third party applications**
  - **integration with web 2.0 and social networking**

Functionality

- It is not
  - SIP phone
  - SIP media server
  - SIP back-to-back user agent

- **It handles only signaling**

Features

Plug in module interface
Perl programming interface

SIP proxy, redirect and registrar server
user registration with
digest authorization

Customizable routing policy
User location service
IPv4-IPv6
UDP/TCP/TLS/SCTP
SIP translator

Offline message service
Presence server
ENUM lookup support
Advanced routing (dispatching and LCR)
Dialing support
aliases and speeddial

Multi-domain support
LDAP/H.350 support

Features

OSP support for peering
Java SIP Servlet programming interface

Link any application to Kamailio using FIFO/UNIXSOCK/DATAGRAM/XMLRPC interfaces

NAT traversal
Security
permissions
anti-DOS attacks
User call preferences
Call Processing Language

Database API
MySQL
PostgreSQL
UNIXODBC
BERKELEYDB
ORACLE
Text files
RADIUS

Gateway

SMS
XMPP

Accounting through log file, database or Radius/DIAMETER servers

New in 3.0.0

Released January 11, 2010

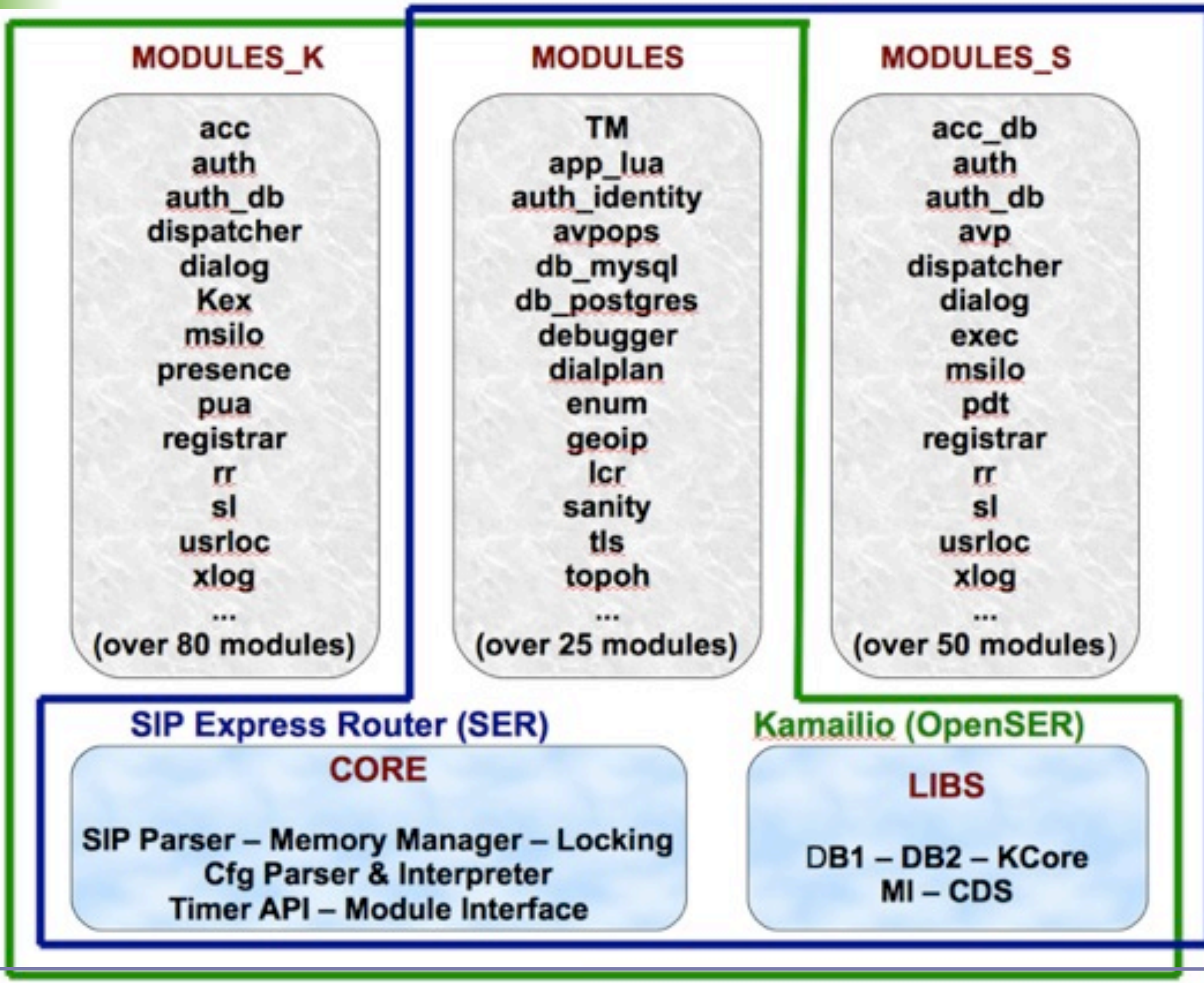
The Linux - Kernel Approach

Work to build rock-solid SIP server:

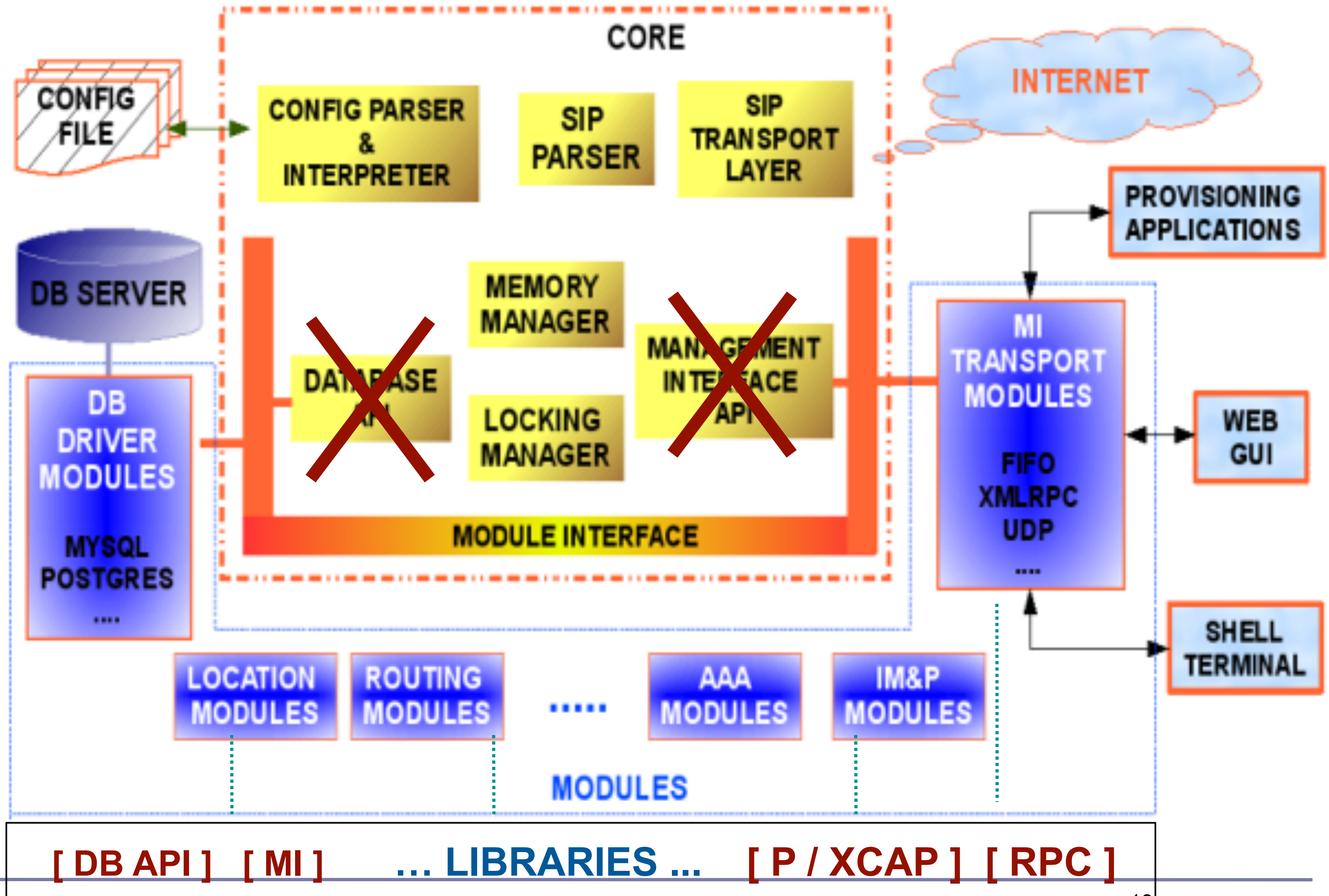
- SIP parser
- Share extensions development
- Memory manager
- Transport layer: TCP, TCP, TLS, SCTP
- SIP transaction management
- Synchronization and locking system

Independent management and release policy

Kamailio 3.0 Release



Improving architecture



New in 3.0.0

Asynchronous TCP

several ten thousands
of TCP connections

Number Portability

customizable policies
caching system

SCTP

multi-homing
multi-streaming
statistics

Topology hiding

fast and secure
no session dependency

Web & CLI Tools
SIREMIS
SERCMD

New in 3.0.0

DNS Caching System

fast failover
blacklisting
avoid blocking

CFG Reload Framework

update global parameters
at runtime

XMLRPC

scalable control
interface
secure

Memcached connector

data distribution across
many instances

CFG Optimizations

operations - operators
switch - break - while

New in 3.0.0

CFG Directives

include
define

Transport layer

UDP MTU fallback
scalable TLS

Auth Identity

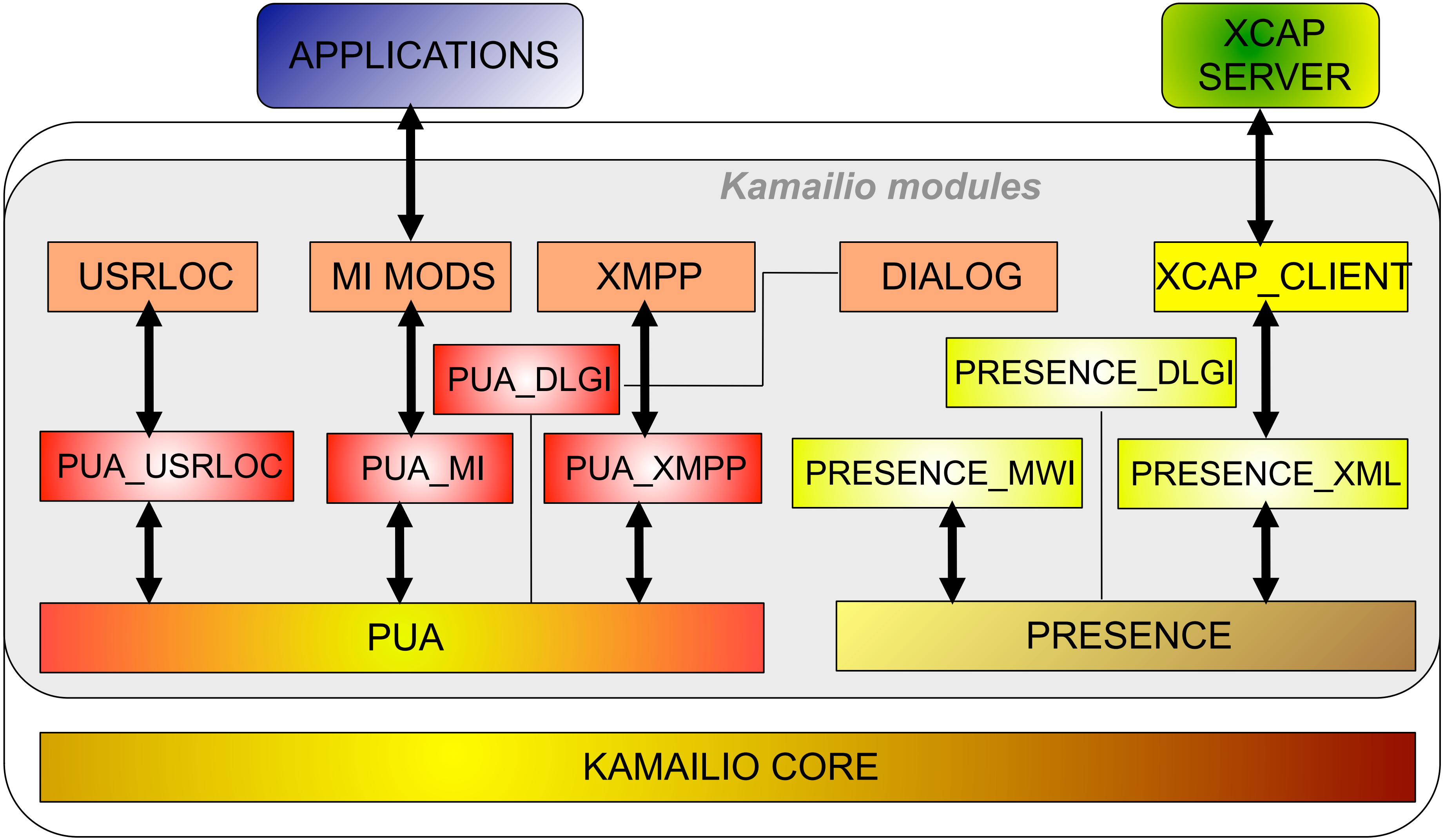
RFC4474

New routing blocks

onsend route
event route

Async SIP Message Processing

park - process - resume


APPLICATIONS
XCAP SERVER
Kamailio modules
USRLOC
MI MODS
XMPP
DIALOG
XCAP_CLIENT
PUA_DLGI
PRESENCE_DLGI
PUA_USRLOC
PUA_MI
PUA_XMPP
PRESENCE_MWI
PRESENCE_XML
PUA
PRESENCE
KAMAILIO CORE

New in 3.0.0

http://www.kamailio.org/dokuwiki/doku.php/features:new-in-3.0.x

http://www.kamailio.org/w/kamailio-openser-v3.0.0-release-notes/

http://by-miconda.blogspot.com/2010/01/best-of-new-in-kamailio-300-toc.html

New in Development
3.1.0

ETA 4-5 months

New in devel

- for 3.1.0
  - embedded LUA
  - geoip call tracing
  - tree indexed caching system for config
  - SIP registration to remote servers
  - trunk-based traffic limitation policies

New in devel

- for 3.1.0
  - embedded Python
  - config message queuing system
  - load balancing
    - weight based distribution
    - call load distribution
  - More modules using cfg reload framework
  - Reason header extension for canceled branches

New in devel

- for 3.1.0
  - xlog enhancements
    - print config line number
    - print config file name (good for include file cases)
  - config file debugger
    - print config execution trace for SIP message
    - step-by-step execution of each action in config
    - gdb-like debugging
    - print values of pseudo-variables in each step
  - more at:
    - http://sip-router.org/wiki/features/new-in-devel

Administration Tools

SIREMIS

SERCMD

- Web Management Interface
  - Uses PHP OpenBiz framework
    - straightforward database management
    - web 2.0 - ajax web page navigation
  - Code development
    - MI communication with SIP server via UDP
    - RPC communication with SIP server via XMLRPC
    - TCP communication with FreeSWITCH
    - Build and display charts
  - Flexibility
    - Developed on linux but should be OS independent
    - Support for many DB backends
    - Model-View-Controller (MVC) architecture

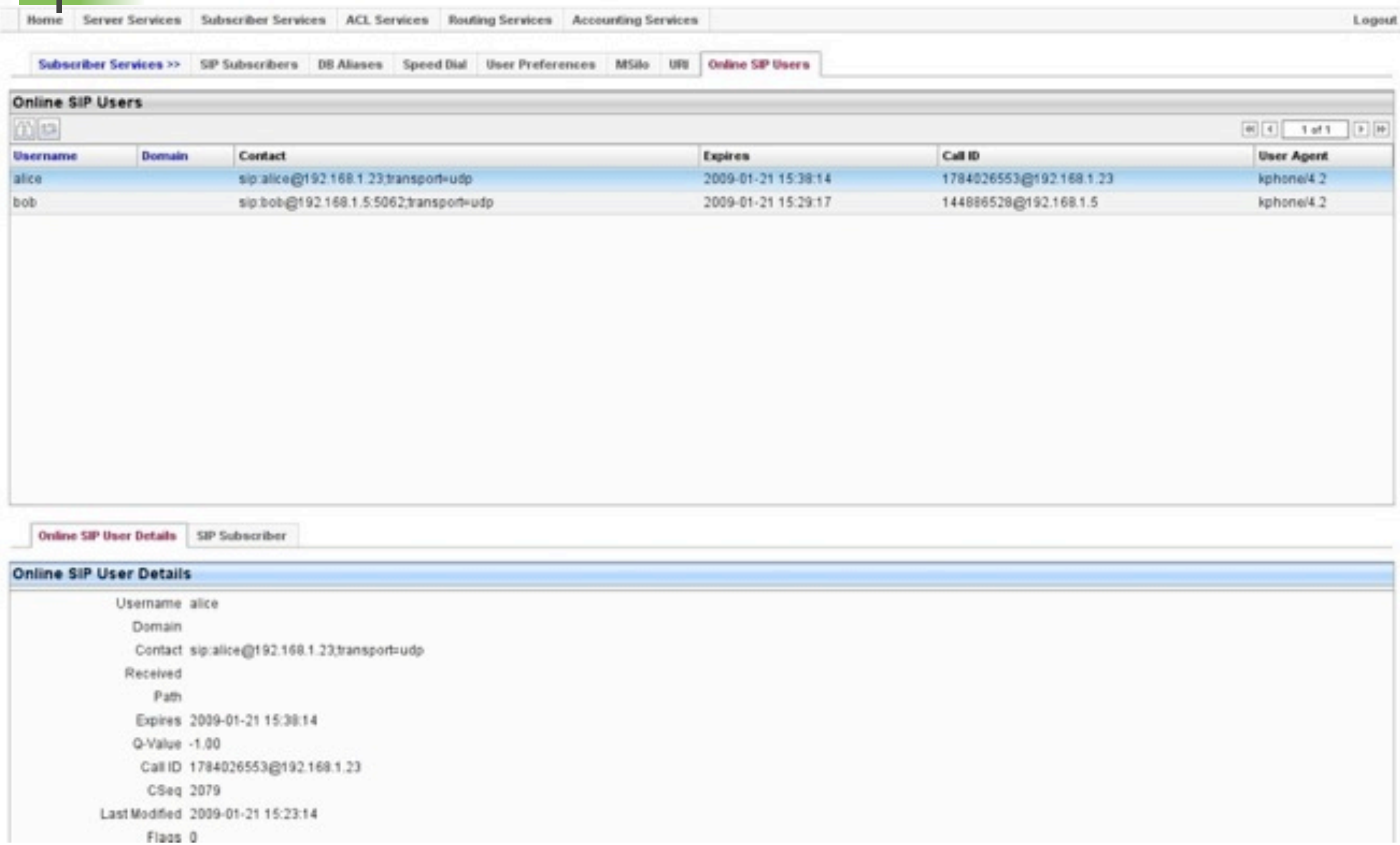
SIREMIS

- subscriber, database aliases and speed dial management
- location table view
- dispatcher (load balancing), prefix-domain translation and least cost routing (lcr) management
- access control lists (user groups) and permissions management
- accounting records and missed calls vies
- manage call data records (generated from acc records)
- hash table, dial plan table and user preferences table management
- offline message storage, presence service and sip trace views

SIREMIS

- communication with Kamailio via MI UDP sockets
- communication with Kamailio 3.0.0 and SIP-Router.org via XMLRPC
- communication with FreeSWITCH via event socket
- create and display charts from statistic data stored by Kamailio, for example
  - shared memory usage
  - SIP traffic load
  - online users and phones
- user location statistics charts

SIREMIS


Home
Server Services
Subscriber Services
ACL Services
Routing Services
Accounting Services
Logout
Subscriber Services >>
SIP Subscribers
DB Aliases
Speed Dial
User Preferences
MSilo
URI
Online SIP Users
Online SIP Users
1 of 1
Username
Domain
Contact
Expires
Call ID
User Agent
alice
sip:alice@192.168.1.23;transport=udp
2009-01-21 15:38:14
1784026553@192.168.1.23
kphone/4.2
bob
sip:bob@192.168.1.5:5062;transport=udp
2009-01-21 15:29:17
144886528@192.168.1.5
kphone/4.2
Online SIP User Details
SIP Subscriber
Online SIP User Details
Username alice
Domain
Contact sip:alice@192.168.1.23;transport=udp
Received
Path
Expires 2009-01-21 15:38:14
Q-Value -1.00
Call ID 1784026553@192.168.1.23
CSeq 2079
Last Modified 2009-01-21 15:23:14
Flags 0

SIREMIS

Home | Server Services | Subscriber Services | ACL Services | Routing Services | Accounting Services | Logout

Accounting Services >> | Accounting | Call Detail Records | Missed Calls

Accounting Table

1 of 1

Time	SIP Method	Src Username	Src Domain	Dst Username	Dst Domain	SIP Call-ID	SIP Reply Code
2009-01-21 14:47:36	INVITE	bob	asipto.com	alice	192.168.1.23	684598749@192.168.1.5	200
2009-01-21 14:55:41	INVITE	alice	asipto.com	bob	192.168.1.5	1142159929@192.168.1.23	200
2009-01-21 14:47:56	BYE	alice	asipto.com	bob	192.168.1.5	684598749@192.168.1.5	200
2009-01-21 14:55:41	ACK	alice	asipto.com	bob	192.168.1.5	1142159929@192.168.1.23	200
2009-01-21 14:56:03	BYE	alice	asipto.com	bob	192.168.1.5	1142159929@192.168.1.23	200
2009-01-21 15:29:38	INVITE	alice	asipto.com	bob	192.168.1.5	1553149770@192.168.1.23	200
2009-01-21 15:29:38	ACK	alice	asipto.com	bob	192.168.1.5	1553149770@192.168.1.23	200
2009-01-21 15:31:40	BYE	bob	asipto.com	alice	192.168.1.23	1553149770@192.168.1.23	200

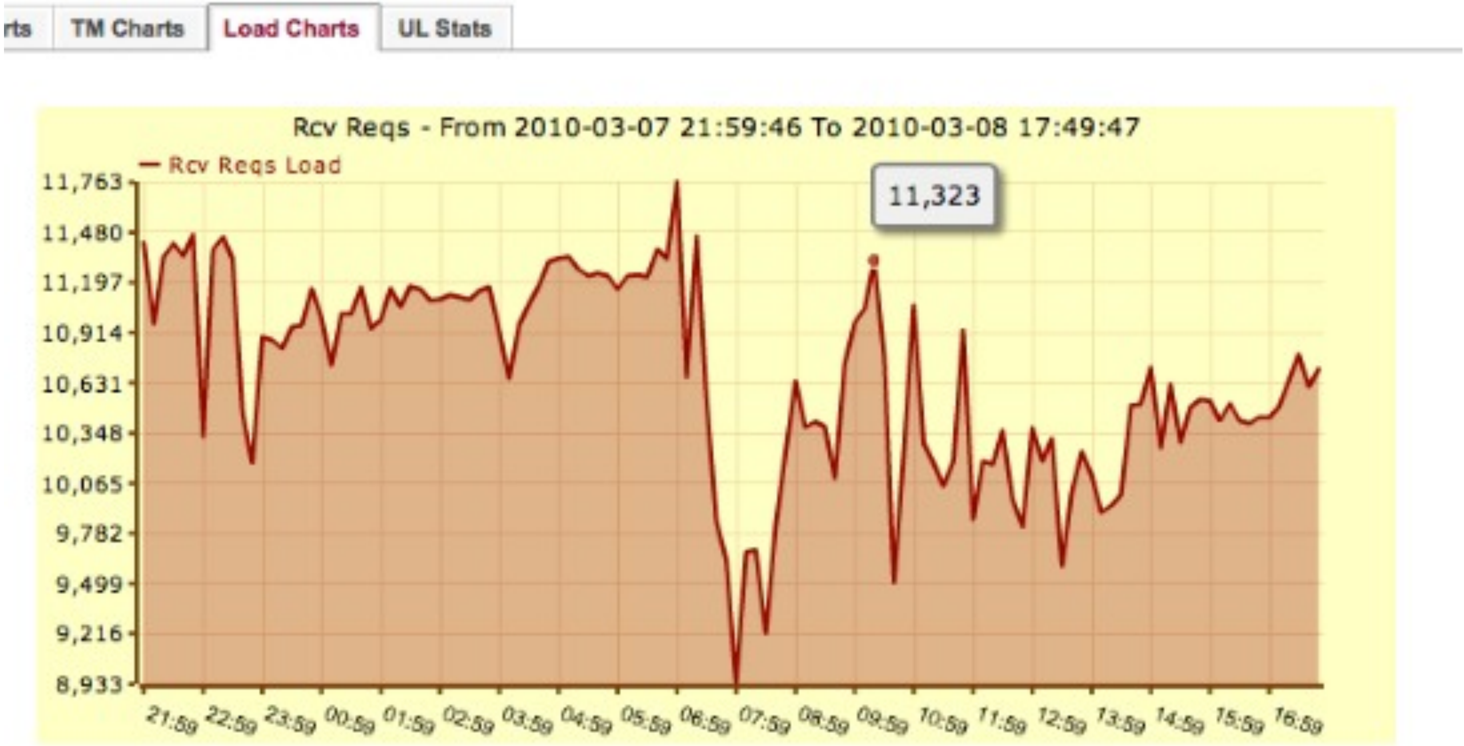
Accounting Details | Call Details Records

CDRS

1 of 1

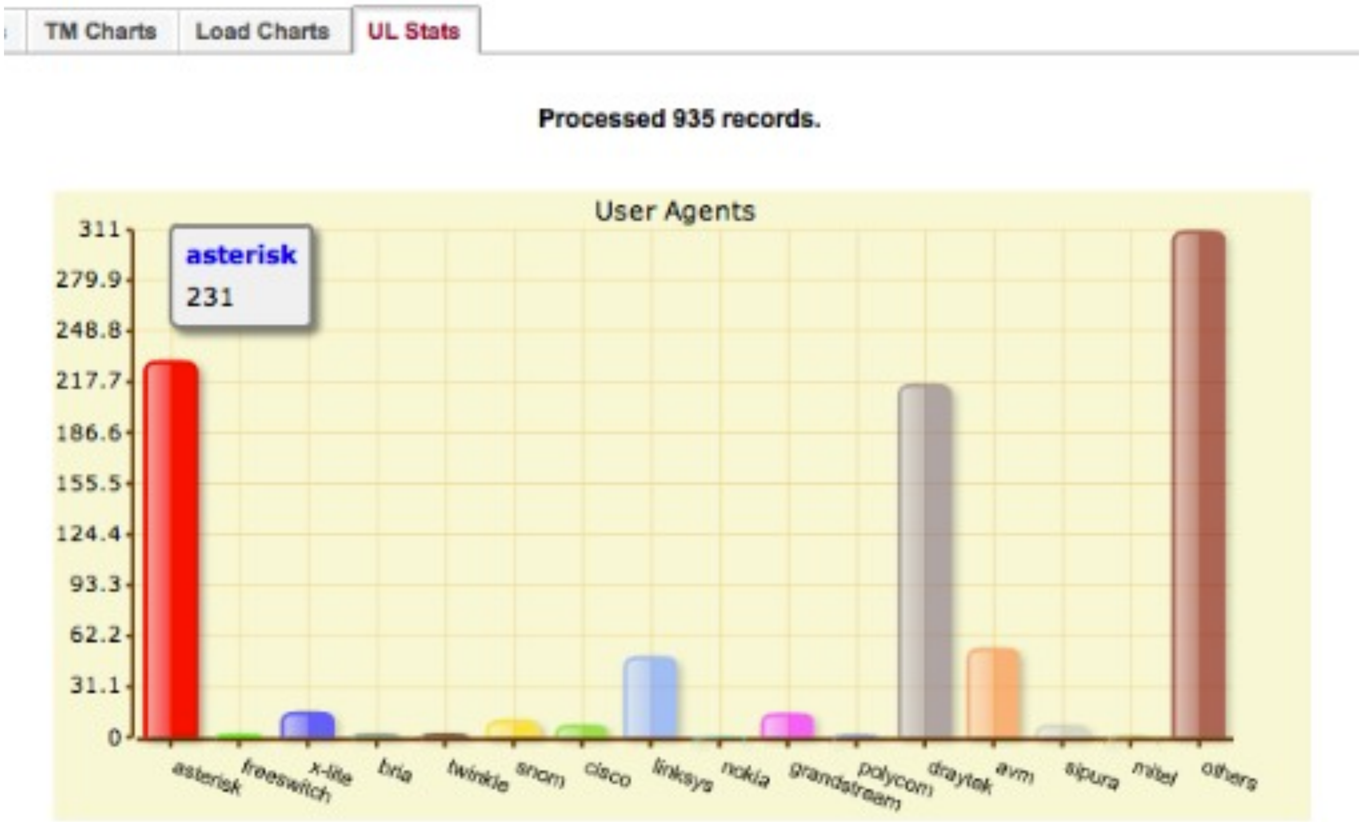
Id	Src Username	Src Domain	Dst Username	Dst Domain	Call Start Time	Duration
2	alice	asipto.com	bob	192.168.1.5	2009-01-21 14:55:41	22

SIREMIS



TM Charts
Load Charts
UL Stats
Rcv Reqs - From 2010-03-07 21:59:46 To 2010-03-08 17:49:47
Rcv Reqs Load
11,323
11,763
11,480
11,197
10,914
10,631
10,348
10,065
9,782
9,499
9,216
8,933
21:59
22:59
23:59
00:59
01:59
02:59
03:59
04:59
05:59
06:59
07:59
08:59
09:59
10:59
11:59
12:59
13:59
14:59
15:59
16:59

SIREMIS



TM Charts
Load Charts
UL Stats
Processed 935 records.
User Agents
asterisk
231
311
279.9
248.8
217.7
186.6
155.5
124.4
93.3
62.2
31.1
0
asterisk
freeswitch
x-lite
bria
twinkle
snom
cisco
linksys
nokia
grandstream
polycom
draytek
avm
sipura
mitel
others

SERCMD

- Command Line Interface
  - Written in C
    - included in source tree
    - re-using code from SIP server
  - Features
    - connection or batch mode
    - execute RPC commands
    - support to connect via UDP, TCP, FIFO file
    - connect from remote locations
  - Flexibility
    - command history per session
    - command tab completion and help messages
    - output format can be customized

SERCMD

```
Terminal — ssh — 80×25
?                          ctl.who                    dst_blacklist_mem_info
cfg.commit                 dns.add_a                  exit
cfg.diff                   dns.add_aaaa               help
cfg.get                    dns.add_srv                license
cfg.help                   dns.debug                  list
cfg.list                   dns.debug_all              listen
cfg.rollback               dns.delete_a               ls
cfg.set_delayed_int        dns.delete_aaaa            mi
cfg.set_delayed_string     dns.delete_all             mi_dg
cfg.set_now_int            dns.delete_cname           mi_fifo
cfg.set_now_string         dns.delete_ebl             mi_xmlrpc
core.arg                   dns.delete_naptr           ps
core.echo                  dns.delete_ptr             quit
core.kill                  dns.delete_srv             server
core.printi                dns.delete_txt             serversion
core.prints                dns.lookup                 system.listMethods
core.ps                    dns.mem_info               system.methodHelp
core.pwd                   dns.view                   system.methodSignature
core.sctp_info             dns_debug                  tm.cancel
core.sctp_options          dns_debug_all              tm.hash_stats
core.shmmem                dns_mem_info               tm.reply
core.tcp_info              dst_blacklist.add          tm.stats
core.tcp_options           dst_blacklist.debug        tm.t_uac_start
core.uptime                dst_blacklist.delete_all   tm.t_uac_wait
--More--
```

SERCMD

```
sercmd>
sercmd> mi ul_dump brief
Domain:: location table=512 records=3 max_slot=1
AOR:: 104
AOR:: 102
AOR:: 101
sercmd>
sercmd>
```

SERCMD

Terminal — ssh — 80×25

```
sercmd> core.tcp_options
{
        connect_timeout: 10
        send_timeout: 10
        connection_lifetime: 120
        max_connections(soft): 2048
        no_connect: 0
        fd_cache: 1
        async: 1
        connect_wait: 1
        conn_wq_max: 32768
        wq_max: 10485760
        defer_accept: 0
        delayed_ack: 1
        syncnt: 0
        linger2: 0
        keepalive: 1
        keepidle: 0
        keepintvl: 0
        keepcnt: 0
        crlf_ping: 1
        accept_aliases: 0
        alias_flags: 1
        new_conn_alias_flags: 2
}
```

Asipto

SIP Solutions

Services

ASIPTO

SIP & VOIP SOLUTIONS

- deliver simply functional servicing solutions

Training

SIP Router Masterclass
March 22-26, Berlin, Germany
- Kamailio SIP Server professional training
http://www.asipto.com/index.php/sip-router-masterclass/

SIP Introduction
SIP Advanced

Training on demand
- SIP & VoIP
- Open Source Technologies

eLearning

Consultancy

High availability

SIP & VoIP Service Design

Security analysis

Development

Research

Scalability

Technical support

VoIP Solutions

IP Telephony Platform

full featured service
admin interface
modular
extensible

Number Portability

customizable policies
caching system

Prepaid System

scalability
calling card
carrier grade
flexibility

Load Balancer

large set of algorithms
protocol level dispatching

Contact:

http://www.asipto.com
http://www.kamailio.org
http://sip-router.org