Large Unified Communication Platforms

ClueCon 2010, Chicago

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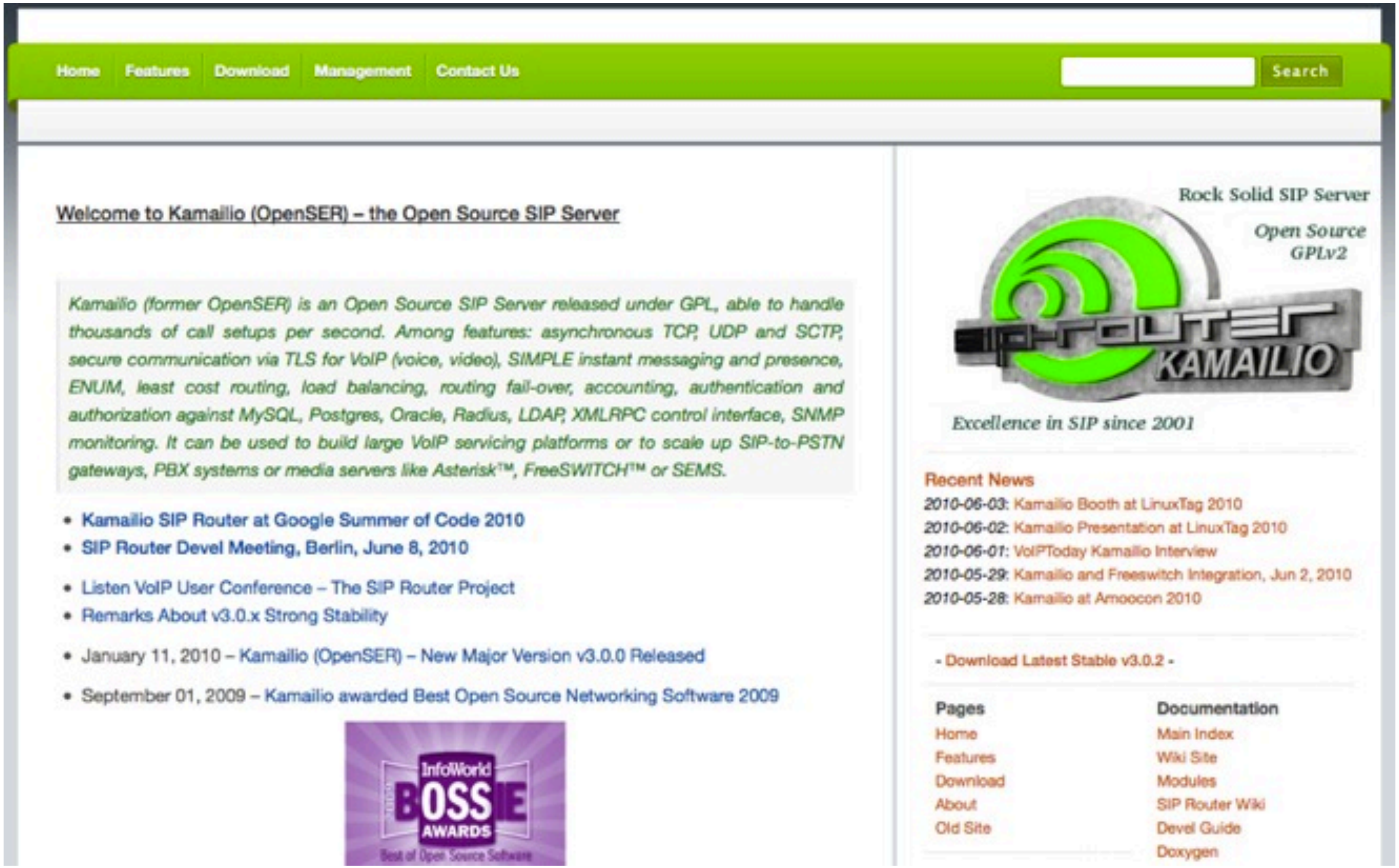
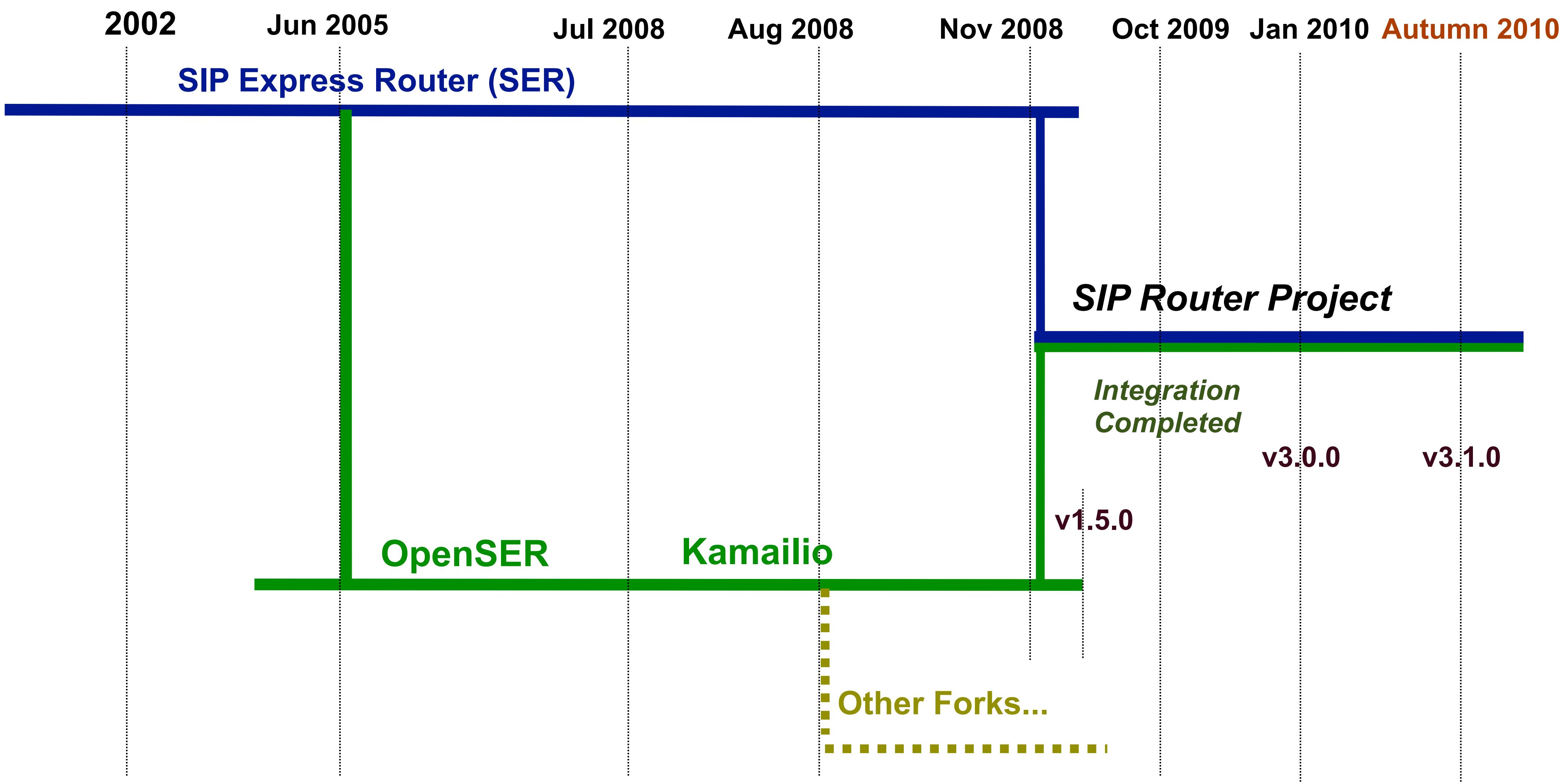
http://www.asipto.com

www.kamailio.org



A bit of history



Development portals

- SourceForge Project: http://sourceforge.net/projects/openser/
- SIP Router Project: http://sip-router.org

Awarded

Best Open Source Networking Software 2009

By InfoWorld

Sample deployments

- Some of biggest VoIP deployments world wide
  - 1&1 (members of the management board)
    - Over 4 millions subscribers
    - Over 1.5 billion minutes per month
  - Sipgate
  - Freenet
    - 0.8 million subscribers
    - Hundreds of millions of minutes per month

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

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

Instant Messaging
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

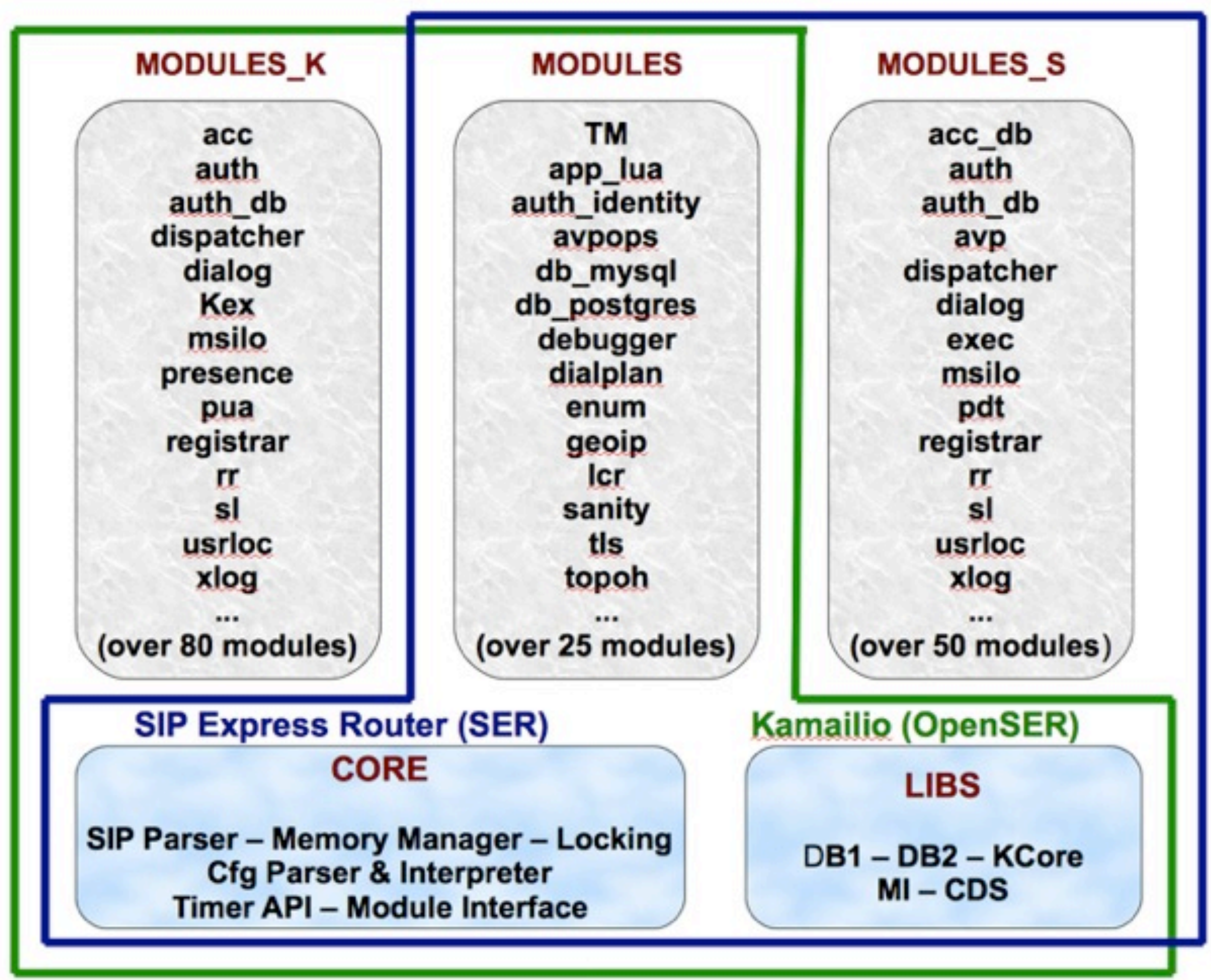
SMS
XMPP

Accounting through log file,
database or Radius/DIAMETER
servers

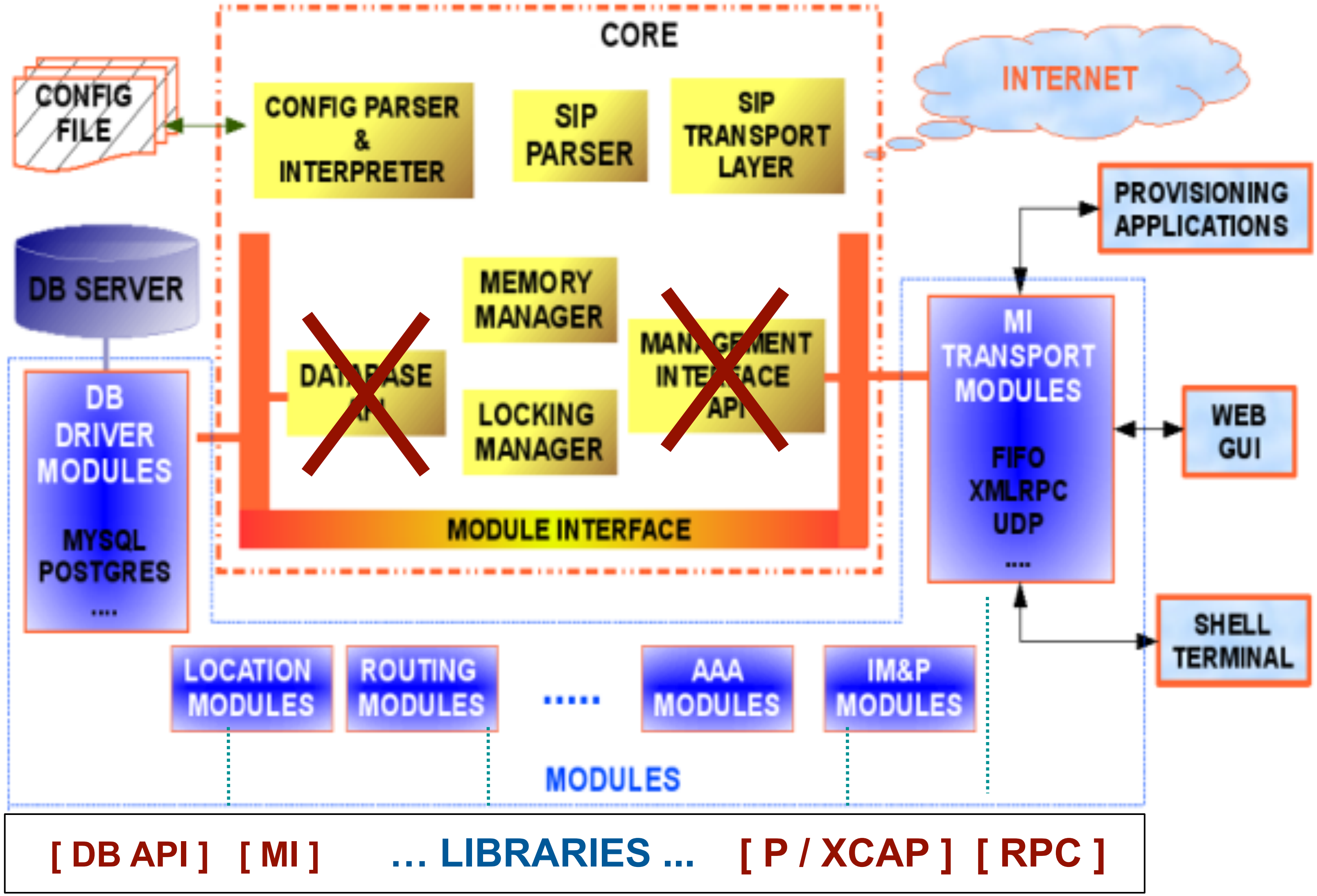
Features (1.5.x)

- DialogInfo Presence support
- Configuration file shared cache system
- Fast SQL operations
- XMPP/MSN/... interconnection via PURPLE library
- HTTP query
- Timer-based route execution
- Perl-like regular expression support
- Least cost routing re-shaping
- Fine control and access of request/reply during transaction processing
- Initiate SIP requests from config file
- Fine access to user location records
- Priority based load balancing destinations
- Option to store and manage only one location record per subscriber
- Message body handling dedicated functions

3.0.0 Releases



Improved 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
load balancing

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

suspend - process - resume

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

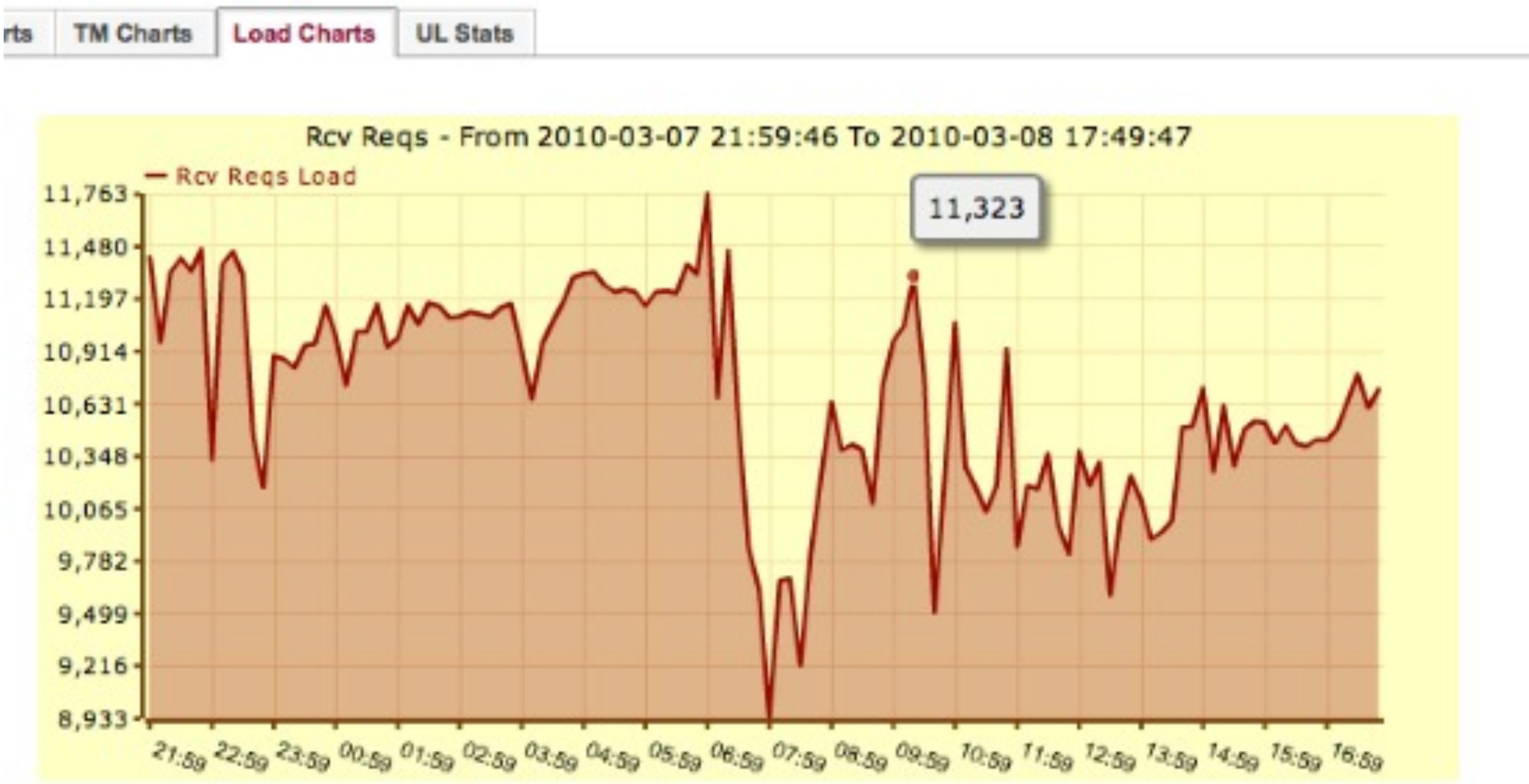
Accounting Details | Call Details Records

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



rts
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

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

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
}
```

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

More

- http://www.kamailio.org/w/features/
- http://www.kamailio.org/dokuwiki/doku.php/features:new-in-1.5.x

Upcoming 3.1.0

Flexibility

- Embedded Lua
- Embedded Python
- Extended preprocessor directive
  - #!define
  - #!subst
- New variables

Maintenance

- Interactive config debugger
  - step-by-step execution
  - execution trace
- xlog enhan's
  - print cfg line
- k&s modules integration

Performance

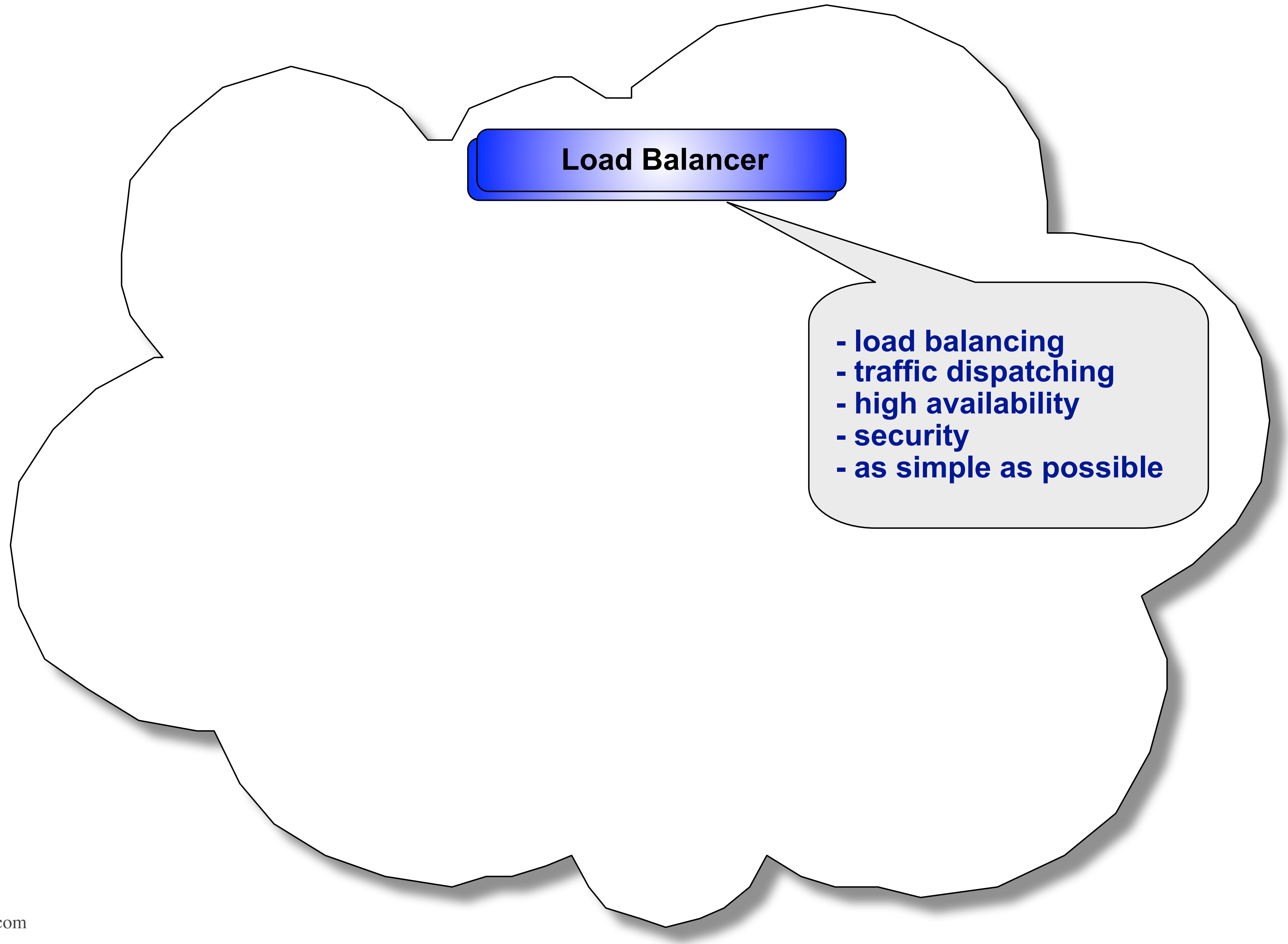
- Asynchronous TLS
- UDP raw sockets
- Multi-homed improvements
- Load balancing
  - weight
  - call load
- Traffic shaping

Features

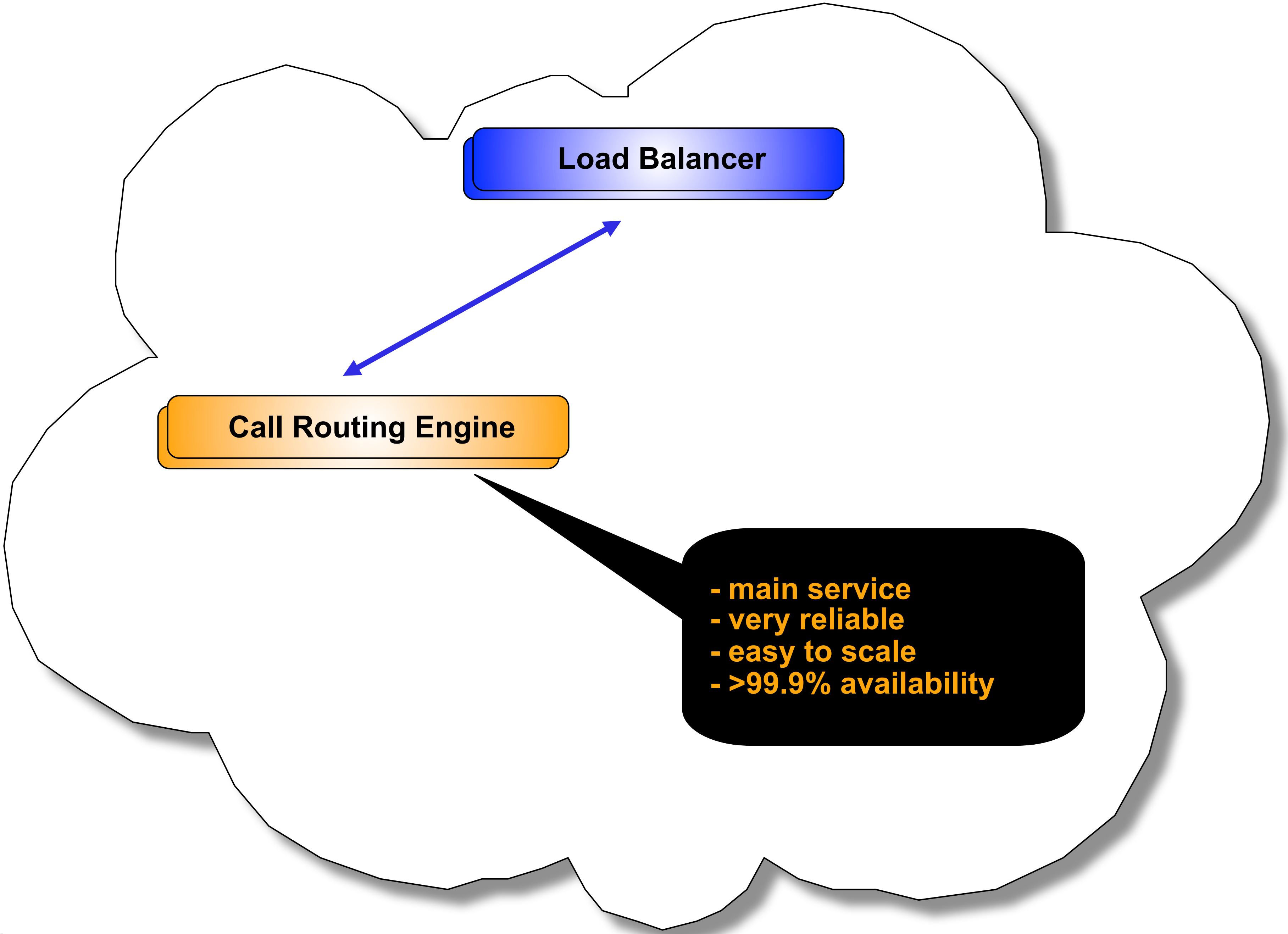
- GeoIP API
- Registration to remote servers
- Reason header for Cancel
- Embedded HTTP & XCAP servers
- Cfg tree cashing & message queue systems
- XML ops

http://sip-router.org/wiki/features/new-in-devel

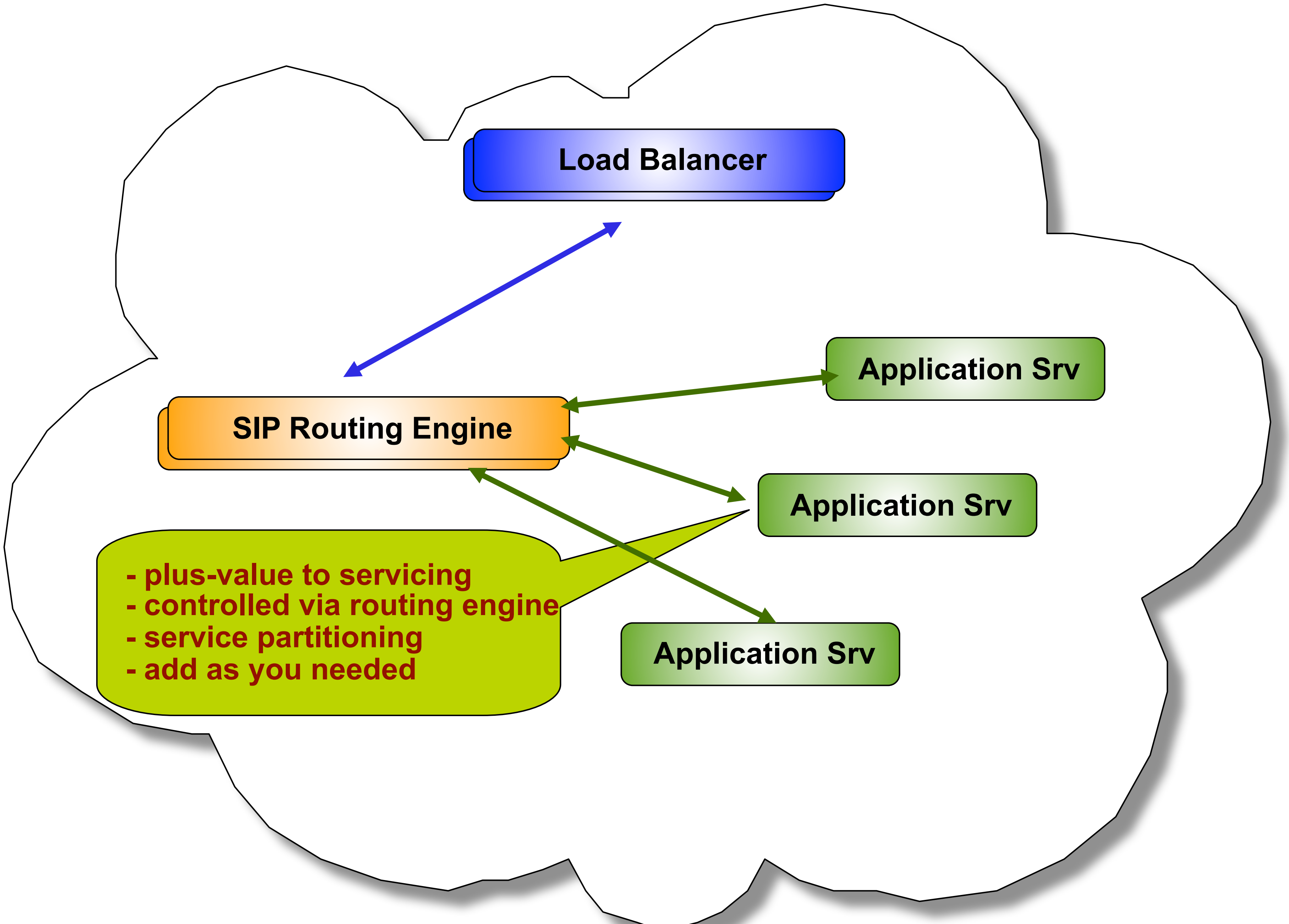
Load balancing



Load balancing



Load balancing



Using dispatcher module

- dedicated load balancing module - **dispatcher** - developed since 2004
- can load destination set from flat file or database
- flexibility to define address of targets
- many dispatching algorithms: hashing, priority, round robin, random, weight, call load (upcoming 3.1)
- detection for unavailable targets and ability to mark them inactive
- periodically ping (SIP OPTIONS) to inactive targets and can automatically mark them active

```
mysql> select * from dispatcher;
+----+-------+-----------------------------+-------+----------+------------------+
| id | setid | destination                 | flags | priority | description      |
+----+-------+-----------------------------+-------+----------+------------------+
|  1 |   100 | sip:10.1.1.100              |     0 |        0 | Freeswitch One   |
|  2 |   100 | sip:10.1.1.101:5080         |     0 |        0 | Freeswitch Two   |
|  3 |   100 | sip:10.1.1.102;transport=tcp |     0 |        0 | Freeswitch Three |
+----+-------+-----------------------------+-------+----------+------------------+
```

Using dispatcher module

- load the module
- set path to file or URL to database for loading destination sets
- adapt parameter values as you need
- documentation - in source tree at modules_k/dispatcher/README or online at http://www.kamailio.org

```
loadmodule "dispatcher.so"

# ----- dispatcher params -----
modparam("dispatcher", "db_url",
    "mysql://openser:openserrw@localhost/openser")
modparam("dispatcher", "table_name", "dispatcher")
modparam("dispatcher", "flags", 2)
modparam("dispatcher", "dst_avp", "$avp(AVP_DST)")
modparam("dispatcher", "grp_avp", "$avp(AVP_GRP)")
modparam("dispatcher", "cnt_avp", "$avp(AVP_CNT)")
modparam("dispatcher", "ds_append_branch", 0)
```

Using dispatcher module

- when you have done with initial processing (sanity checks, authentication, authorization, routing of within-dialog SIP request, a.s.o.) use in your config ***ds_select_dst(setid, algorithm)***
- arm a failure route to re-route to next available target
- relay SIP request

```
# Dispatch requests
route[DISPATCH] {
    # round robin dispatching
    if(!ds_select_dst("100", "4"))
    {
        send_reply("404", "No destination");
        exit;
    }
    xlog("L_DBG", "--- SCRIPT: going to <$ru> via <$du>\n");
    t_on_failure("RTF_DISPATCH");

    if (!t_relay()) {
        sl_reply_error();
    }
    exit;
}
```

Using dispatcher module

- in case of failure, re-route only for 500 reply from downstream and local timeout
- use ***ds_next_dst()*** to select next available target
- re-arm the failure route
- create the branch and relay again

```
# Re-route failed calls
failure_route[RTF_DISPATCH] {
    if (t_is_canceled()) {
        exit;
    }
    # next DST - only for 500 and local timeout
    if (t_check_status("500")
            || (t_branch_timeout() && !t_branch_replied()))
    {
        if(ds_next_dst())
        {
            t_on_failure("RTF_DISPATCH");
            append_branch();
            t_relay();
            exit;
        }
    }
}
```

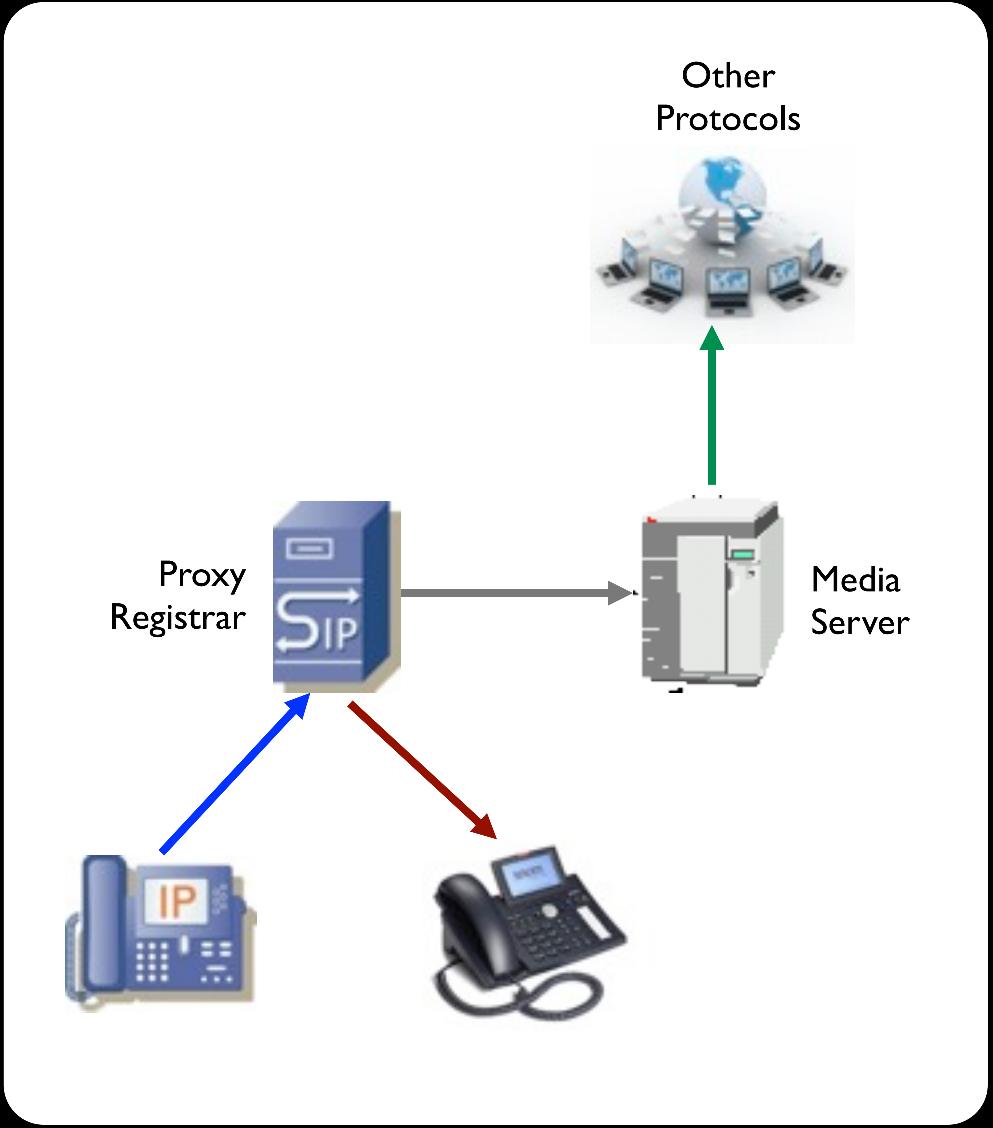
FreeSWITCH as media server

Adding

- voicemail
- announcements
- conferencing
- troubleshooting audio and video
- gateway to other protocols

SIP routing logic

- Kamailio: do initial sanity checks and caller-related processing (e.g., authentication)
- Kamailio: if for a special service or other protocol then relay to FS (can be via load balancing)
- Freeswitch: answer the call with appropriate application or gateway to other protocols
- Kamailio: if not for a specials service or other protocon then do callee-related processing (e.g., user location, least cost routing, load balancing)



http://kb.asipto.com

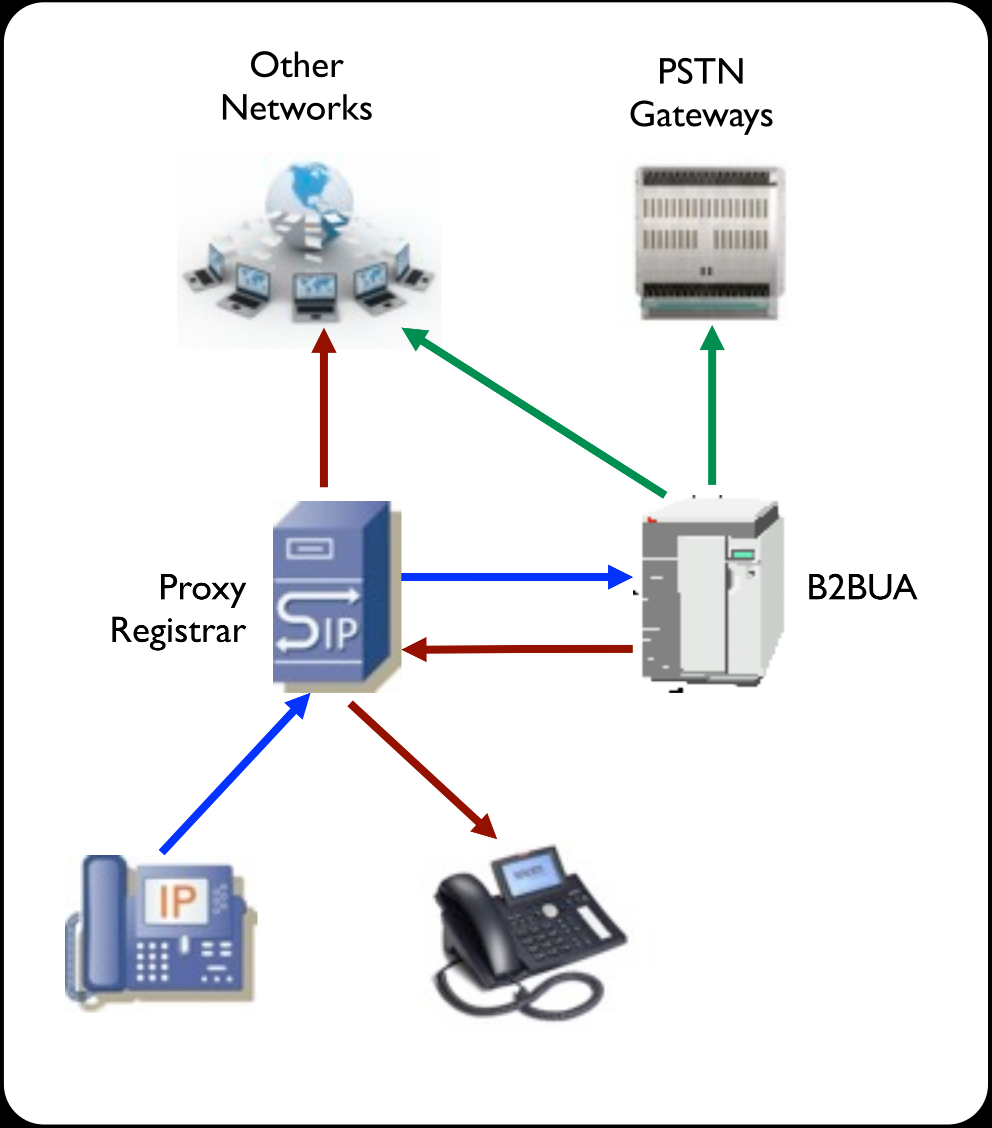
FreeSWITCH as back-to-back user agent

Adding

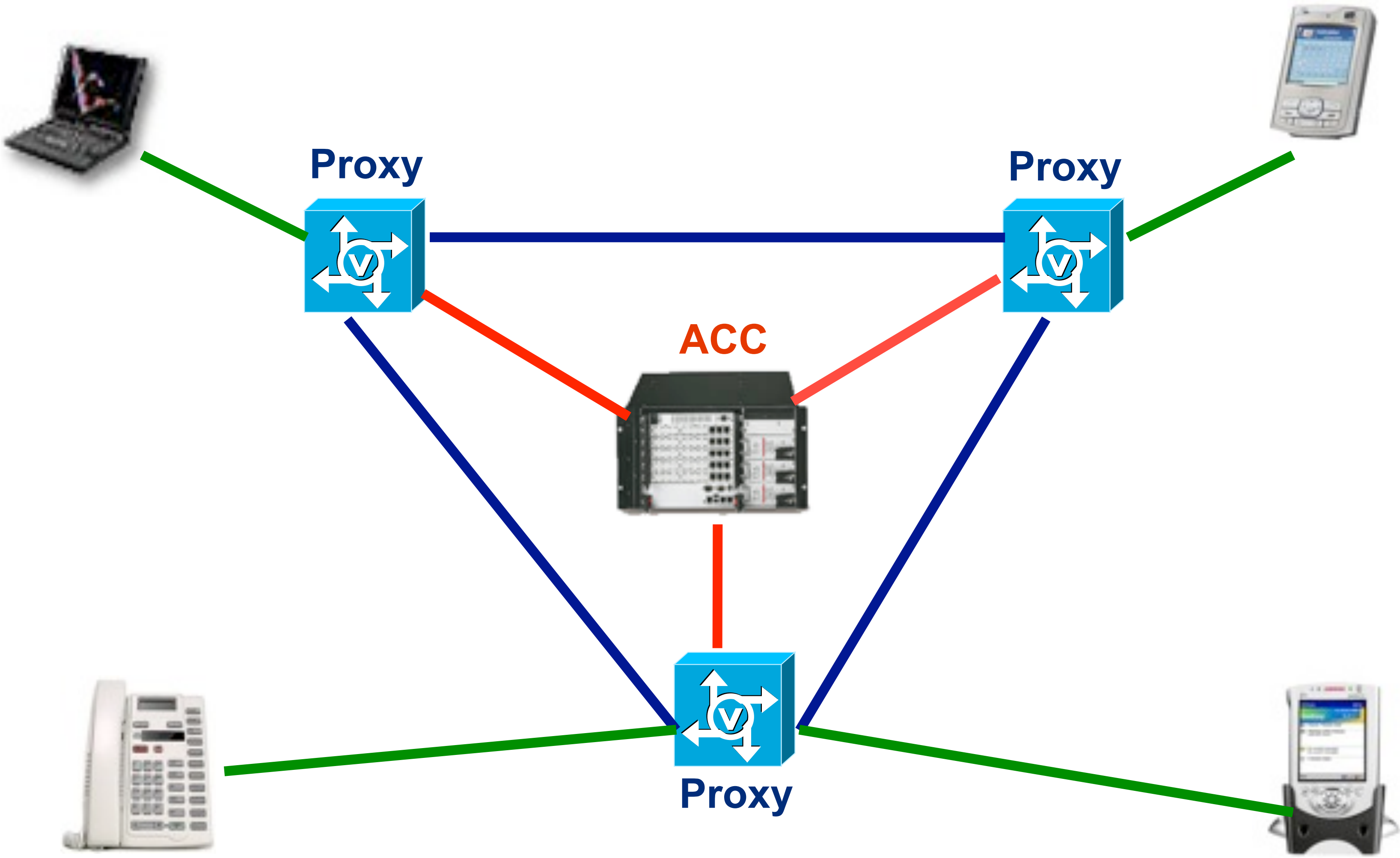
- session control (e.g., rtp timeout)
- topology hiding
- one-to-one SIP hop
- prepaid billing
- calling cards, call queues
- call recording

SIP routing logic

- Kamailio: do initial sanity checks
- Kamailio: if not coming from FS do caller-related processing (e.g., authentication) and then relay to FS (can be via load balancing)
- Freeswitch: do the call processing and relay back to Kamailio or to other destination
- Kamailio: if coming from FS do callee-related processing (e.g., user location, least cost routing, load balancing)



Extensibility: accounting server



Extensibility: accounting server

```
onreply_route[OK] {
  if(status!="200")
       return;
   $uac_req(method)="ACCOUNTING";
   $uac_req(ruri)="sip:store@accounting.kamailio.org;transport=sctp";
   $uac_req(furi)="sip:server@server1.kamailio.org";
   $uac_req(hdrs)="Content-Type: text/accounting-csv\r\n";
   pv_printf($uac_req(body), "$TS,$ci,$ft,$tt,$T_req($fu),$T_req($ru)");
   uac_send_req();
 }
```

SIP Proxy Config

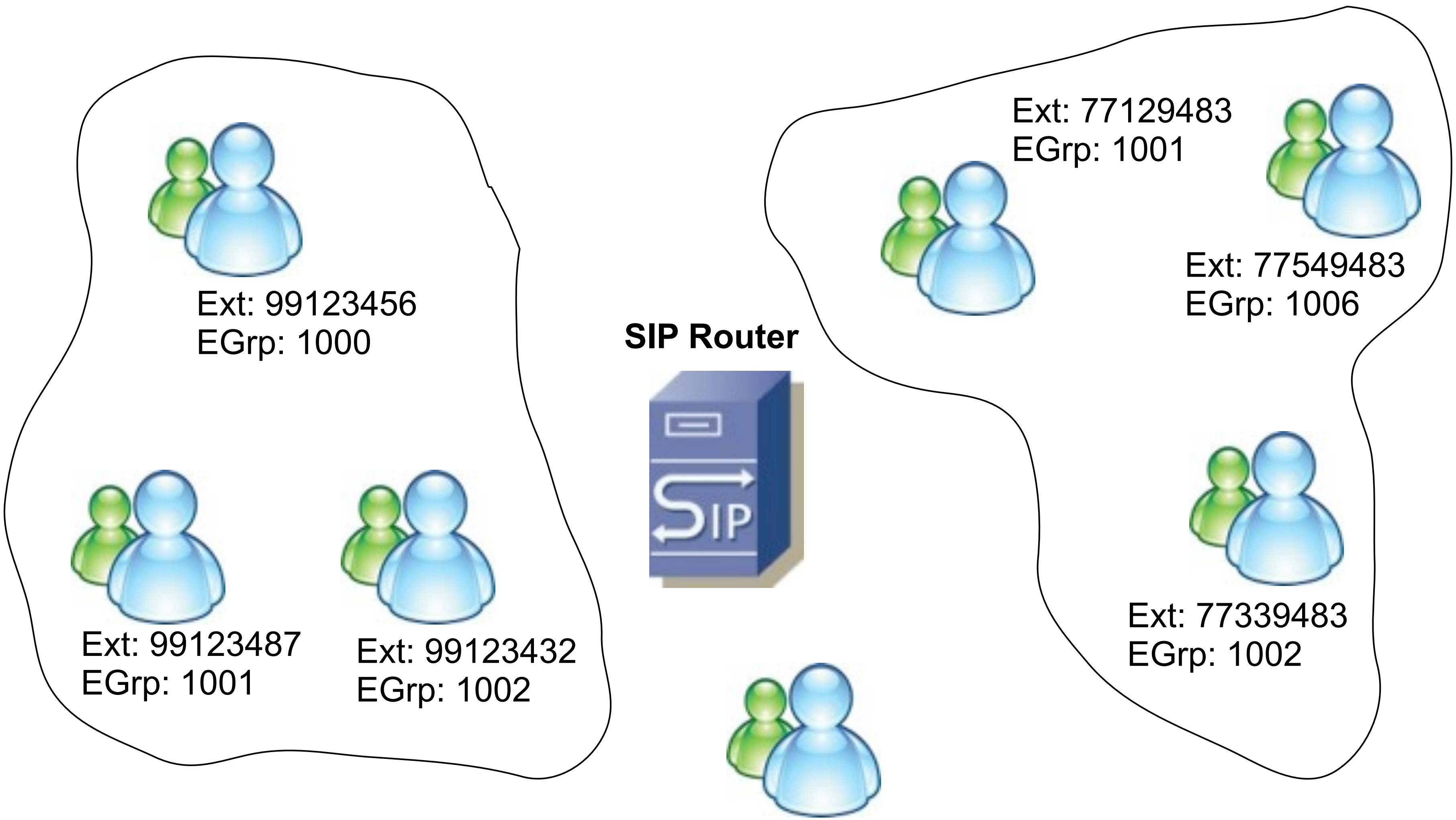
ACC Server Config

```
route {
 if(method=="ACCOUNTING" && $rU="store")
 {
     sql_query("ca",
          "insert into accounting
                (timeval,callid,ftag,ttag,src,dst)
                 values ('$(rb{s.select,0,,})',
                 '$(rb{s.select,1,,})', '$(rb{s.select,2,,})',
                 '$(rb{s.select,3,,})', '$(rb{s.select,4,,})',
                 '$(rb{s.select,5,,})' )",
          "ra");
    send_reply("200", "Stored");
 }
```

Extensibility: within-group dialing

- Centrex like feature
  - Users grouped by affiliation, friendship a.s.o.
  - Inside the group each member has a short extension assigned
- What it takes
  - For many: new module and big waves
  - For the rest:
    - One table definition plus one more attribute per user
    - 5 lines of configuration

Extensibility: within-group dialing



Extensibility: within-group dialing

```
ALTER TABLE subscriber ADD COLUMN pbxgroupid INT NOT NULL DEFAULT 0;

CREATE TABLE pbxgroups (
  id INT(10) UNSIGNED AUTO_INCREMENT PRIMARY KEY NOT NULL,
  pbxgroupid INT DEFAULT 0 NOT NULL,
  shortdial VARCHAR(16) DEFAULT '' NOT NULL,
  extension VARCHAR(64) DEFAULT '' NOT NULL,
  CONSTRAINT pg_u UNIQUE (groupid, shortdial)
) ENGINE=MyISAM;
```

```
□ modparam("auth_db", "load_credentials", "$avp(s:pbxgroupid)=pbxgroupid")
....
□ sql_query("ca"
    "select extension from pbxgroups where pbxgroupid=$avp(s:pbxgroupid)
    and shortdial='$(rU{s.escape.common})'",
    "ra");
□ if($dbr(ra=>rows)>0) $rU = $dbr(ra=>[0,0]);
...
```

Thank you!
Questions?

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http://www.asipto.com

OpenIMSCore