



# Large Unified Communication Platforms

ClueCon 2010, Chicago

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Co-Founder Kamailio  
<http://www.asipto.com>



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## Welcome to Kamailio (OpenSER) – the Open Source SIP Server

Kamailio (former OpenSER) is an Open Source SIP Server released under GPL, able to handle thousands of call setups per second. Among features: asynchronous TCP, UDP and SCTP, secure communication via TLS for VoIP (voice, video), SIMPLE instant messaging and presence, ENUM, least cost routing, load balancing, routing fall-over, accounting, authentication and authorization against MySQL, Postgres, Oracle, Radius, LDAP, XMLRPC control interface, SNMP monitoring. It can be used to build large VoIP servicing platforms or to scale up SIP-to-PSTN gateways, PBX systems or media servers like Asterisk™, FreeSWITCH™ or SEMS.

- Kamailio SIP Router at Google Summer of Code 2010
- SIP Router Devel Meeting, Berlin, June 8, 2010
- Listen VoIP User Conference – The SIP Router Project
- Remarks About v3.0.x Strong Stability
- January 11, 2010 – Kamailio (OpenSER) – New Major Version v3.0.0 Released
- September 01, 2009 – Kamailio awarded Best Open Source Networking Software 2009

Excellence in SIP since 2001

### Recent News

2010-06-03: Kamailio Booth at LinuxTag 2010  
2010-06-02: Kamailio Presentation at LinuxTag 2010  
2010-06-07: VoIPToday Kamailio Interview  
2010-05-29: Kamailio and Freeswitch Integration, Jun 2, 2010  
2010-05-28: Kamailio at Arnocon 2010

- Download Latest Stable v3.0.2 -

Pages	Documentation
<a href="#">Home</a>	<a href="#">Main Index</a>
<a href="#">Features</a>	<a href="#">Wiki Site</a>
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<a href="#">About</a>	<a href="#">SIP Router Wiki</a>
<a href="#">Old Site</a>	<a href="#">Devel Guide</a>
	<a href="#">Doxygen</a>

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# A bit of history



2002 Jun 2005 Jul 2008 Aug 2008 Nov 2008 Oct 2009 Jan 2010 Autumn 2010

SIP Express Router (SER)

SIP Router Project

Integration  
Completed

v3.0.0 v3.1.0

v1.5.0

OpenSER

Kamailio

Other Forks...

# Development portals



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

The screenshot shows two side-by-side web pages. On the left is the main 'SIP-ROUTER.ORG' website, which has a blue header and navigation links for Menu, About, Answers, Benefits, Download, Releases, History, How to Contribute, Licensing, Management, Links, Bug Tracker, Docbook Docs, Doxygen Docs, GIT Repository, and Mailing Lists. It features a search bar and a sidebar with 'Recent news:' and a list of events from May 2010. A central column contains a brief description of the project and its goals. On the right is the 'SourceForge' project page for 'Kamailio (OpenSER) SIP server'. The SourceForge header includes 'FIND AND DEVELOP OPEN SOURCE SOFTWARE' and navigation links for Find Software, Develop, Create Project, Community, Site Support, and About. Below the header, the project name 'Kamailio (OpenSER) SIP server' is displayed along with developer names and links for Summary, Files, Support, and Develop. A large green button at the bottom encourages users to 'Download Now!' with a file link: 'kamailio-1.5.2-tls\_src.tar.gz (3.6 MB)'. The overall layout is clean and professional, designed for developer collaboration and software distribution.



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

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

**Plug in module  
interface  
Perl programming  
interface**

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

**Multi-domain support  
LDAP/H.350 support**

**Instant Messaging  
Offline message  
service  
Presence server**

**ENUM lookup support  
Advanced routing  
(dispatching and LCR)  
Dialing support  
aliases and speeddial**



# Features

OSPF 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



## MODULES\_K

acc  
auth  
auth\_db  
dispatcher  
dialog  
Kex  
msilo  
presence  
pua  
registrar  
rr  
sl  
usrloc  
xlog  
...

(over 80 modules)

## MODULES

TM  
app\_lua  
auth\_identity  
avpops  
db\_mysql  
db\_postgres  
debugger  
dialplan  
enum  
geoip  
lcr  
sanity  
tls  
topoh  
...

(over 25 modules)

## MODULES\_S

acc\_db  
auth  
auth\_db  
avp  
dispatcher  
dialog  
exec  
msilo  
pdt  
registrar  
rr  
sl  
usrloc  
xlog  
...

(over 50 modules)

## SIP Express Router (SER)

### CORE

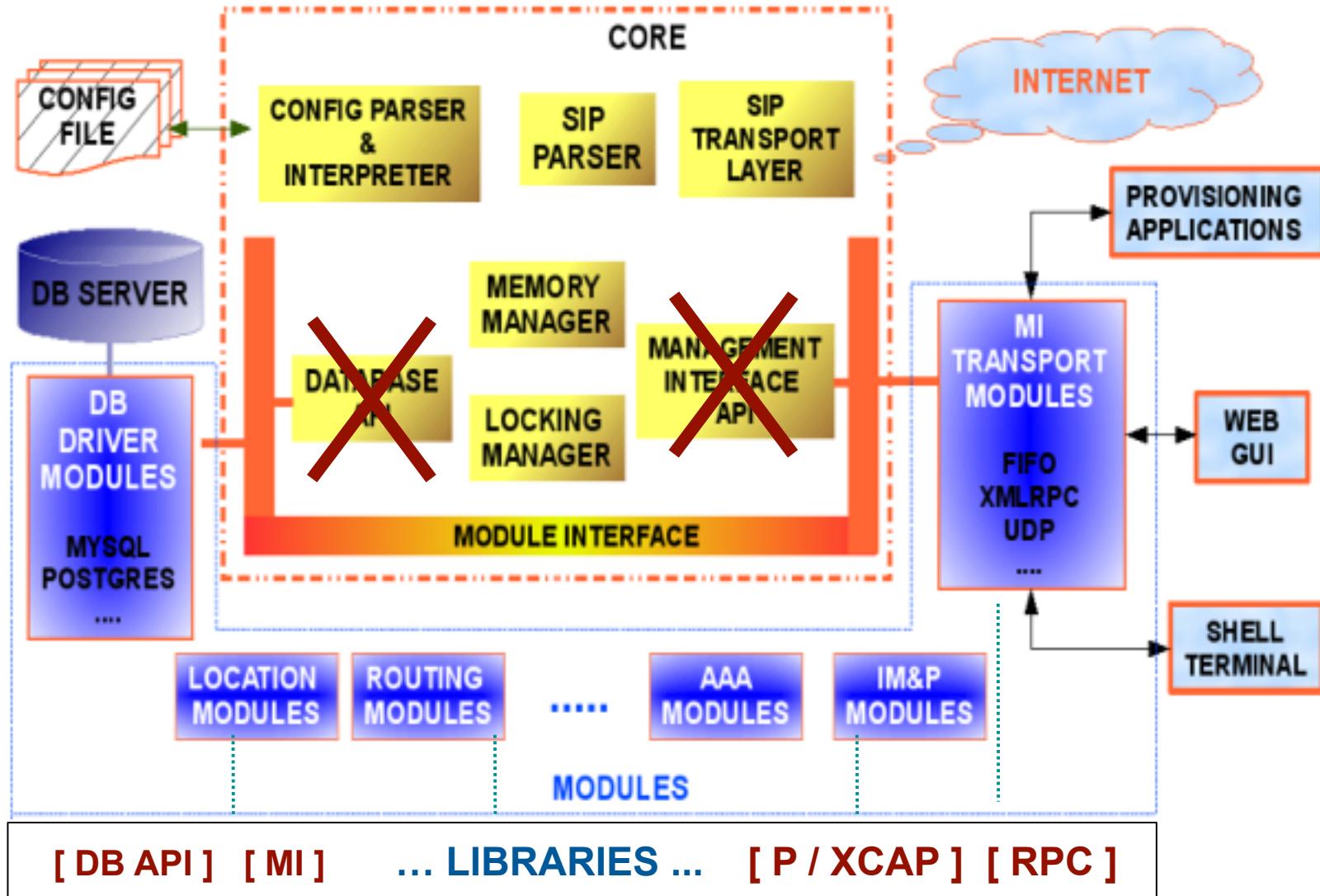
SIP Parser – Memory Manager – Locking  
Cfg Parser & Interpreter  
Timer API – Module Interface

## Kamailio (OpenSER)

### LIBS

DB1 – DB2 – KCore  
MI – CDS

# 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

## New routing blocks

onsend route  
event route

## Transport layer

UDP MTU fallback  
scalable TLS

## Auth Identity

RFC4474

## Async SIP Message Processing

suspend - process - resume

[Accounting Services >>](#) [Accounting](#) [Call Detail Records](#) [Missed Calls](#)
**Accounting Table**

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**

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

Powered by Asipto SRL - © 2009 Asipto SRL

RTS TM Charts Load Charts UL Stats



```
Terminal — ssh — 80x25

?
cfg.commit
cfg.diff
cfg.get
cfg.help
cfg.list
cfg.rollback
cfg.set_delayed_int
cfg.set_delayed_string
cfg.set_now_int
cfg.set_now_string
core.arg
core.echo
core.kill
core.printi
core.prints
core.ps
core.pwd
core.sctp_info
core.sctp_options
core.shmmem
core.tcp_info
core.tcp_options
core.uptime
--More--■

ctl.who
dns.add_a
dns.add_aaaa
dns.add_srv
dns.debug
dns.debug_all
dns.delete_a
dns.delete_aaaa
dns.delete_all
dns.delete_cname
dns.delete_ebl
dns.delete_naptr
dns.delete_ptr
dns.delete_srv
dns.delete_txt
dns.lookup
dns.mem_info
dns.view
dns_debug
dns_debug_all
dns_mem_info
dst_blacklist.add
dst_blacklist.debug
dst_blacklist.delete_all
dst_blacklist_mem_info
exit
help
license
list
listen
ls
mi
mi_dg
mi_fifo
mi_xmlrpc
ps
quit
server
serverversion
system.listMethods
system.methodHelp
system.methodSignature
tm.cancel
tm.hash_stats
tm.reply
tm.stats
tm.t_uac_start
tm.t_uac_wait
```



# Sercmd

```
Terminal — ssh — 80x25
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

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

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

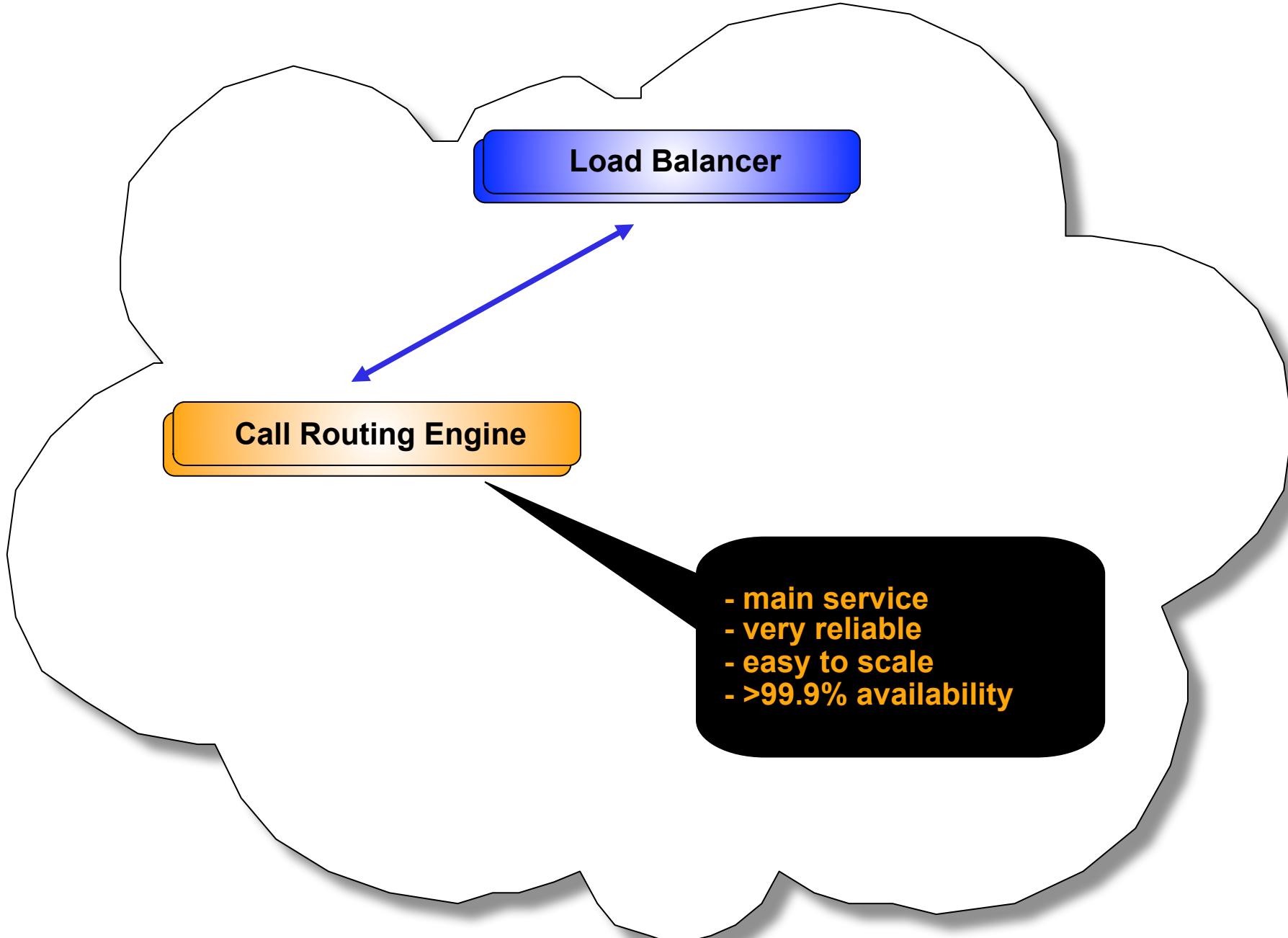
# Load balancing



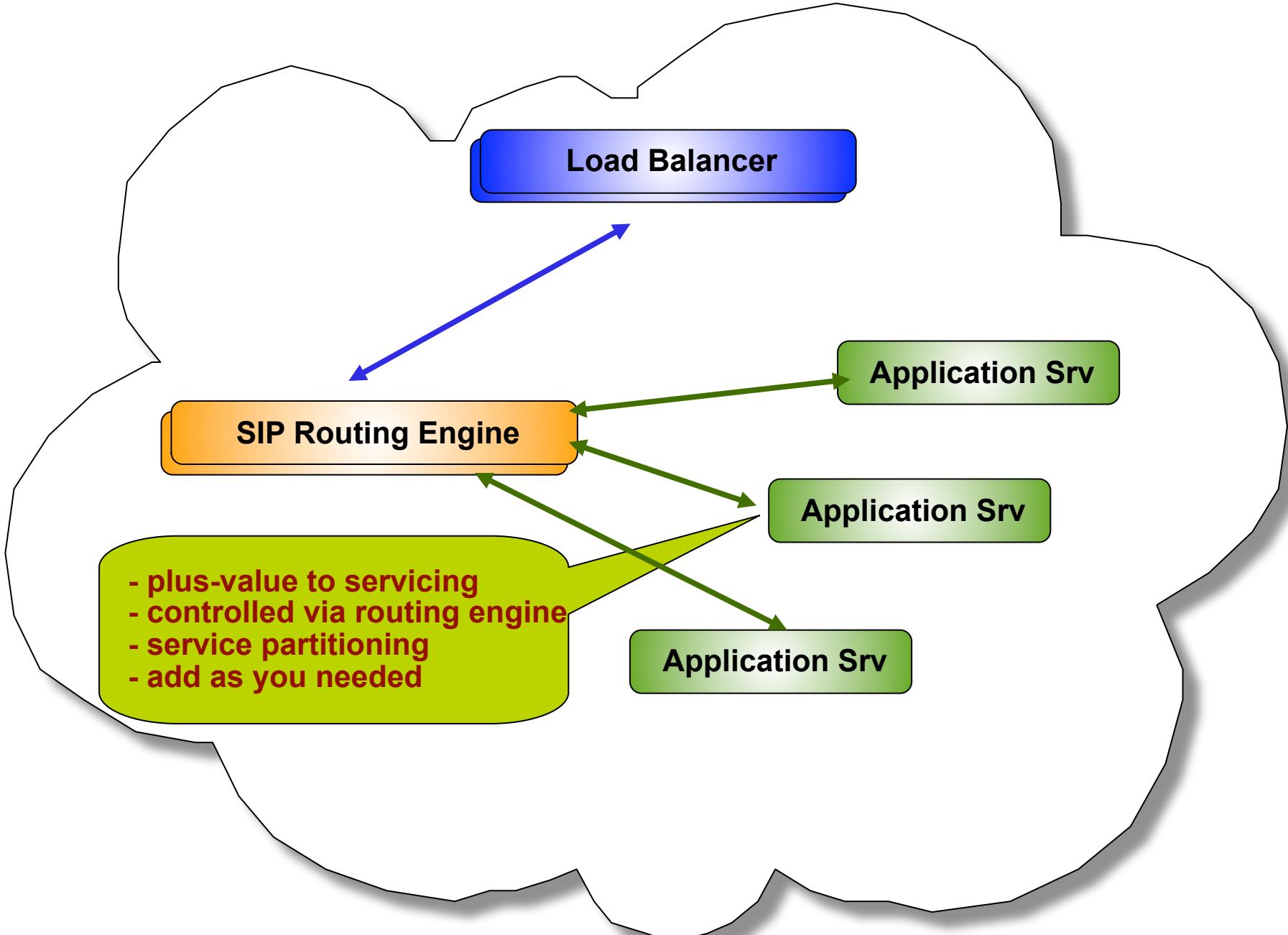
## Load Balancer

- load balancing
- traffic dispatching
- high availability
- security
- as simple as possible

# 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

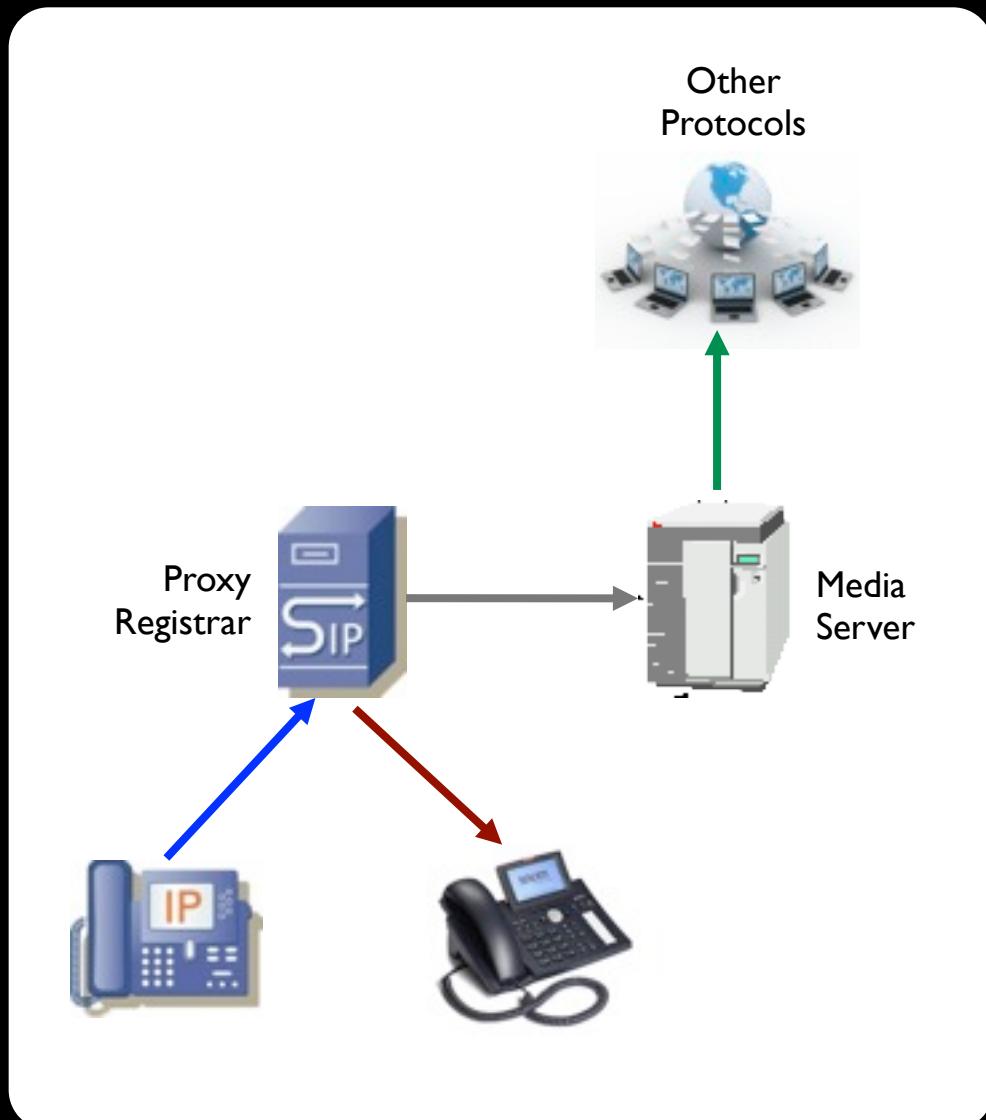


## Adding

- 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 special service or other protocol 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

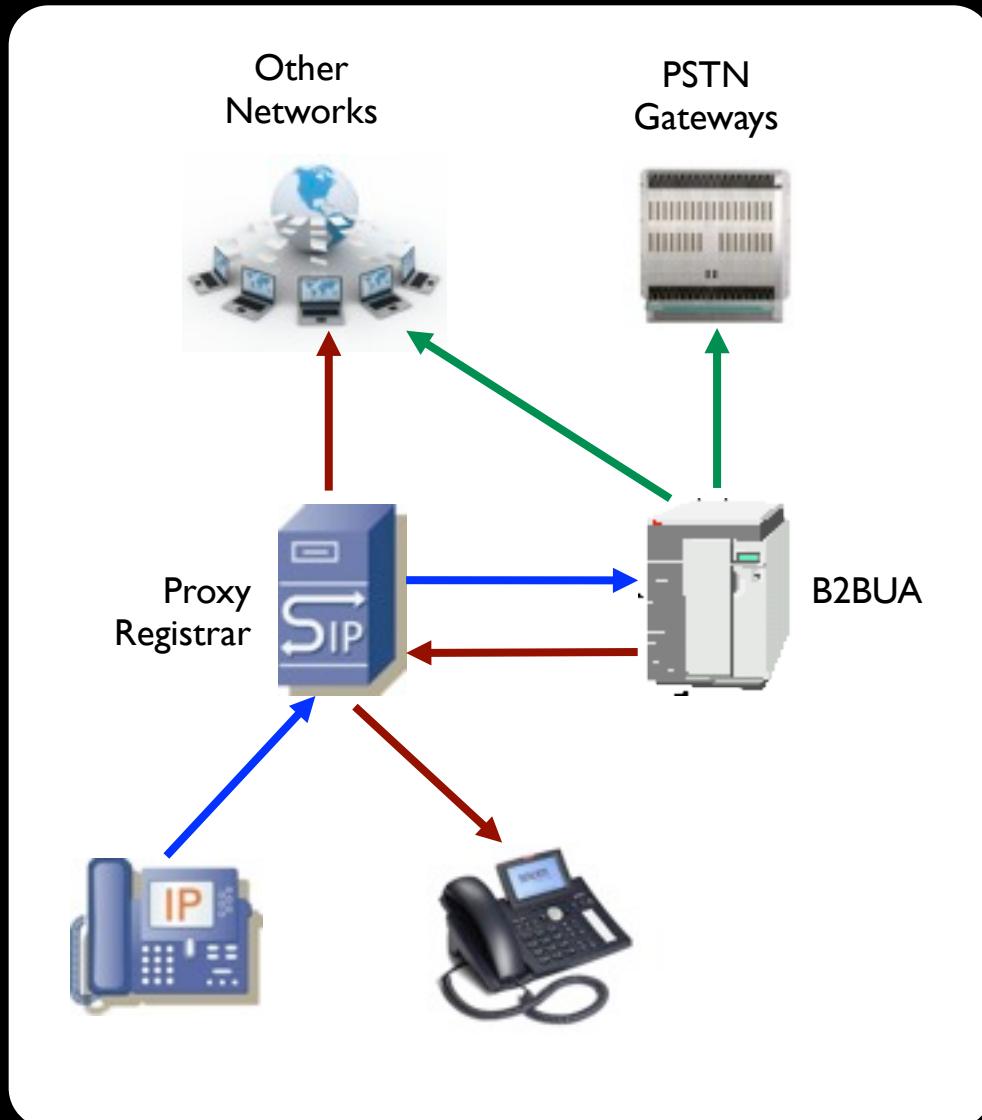


## Adding

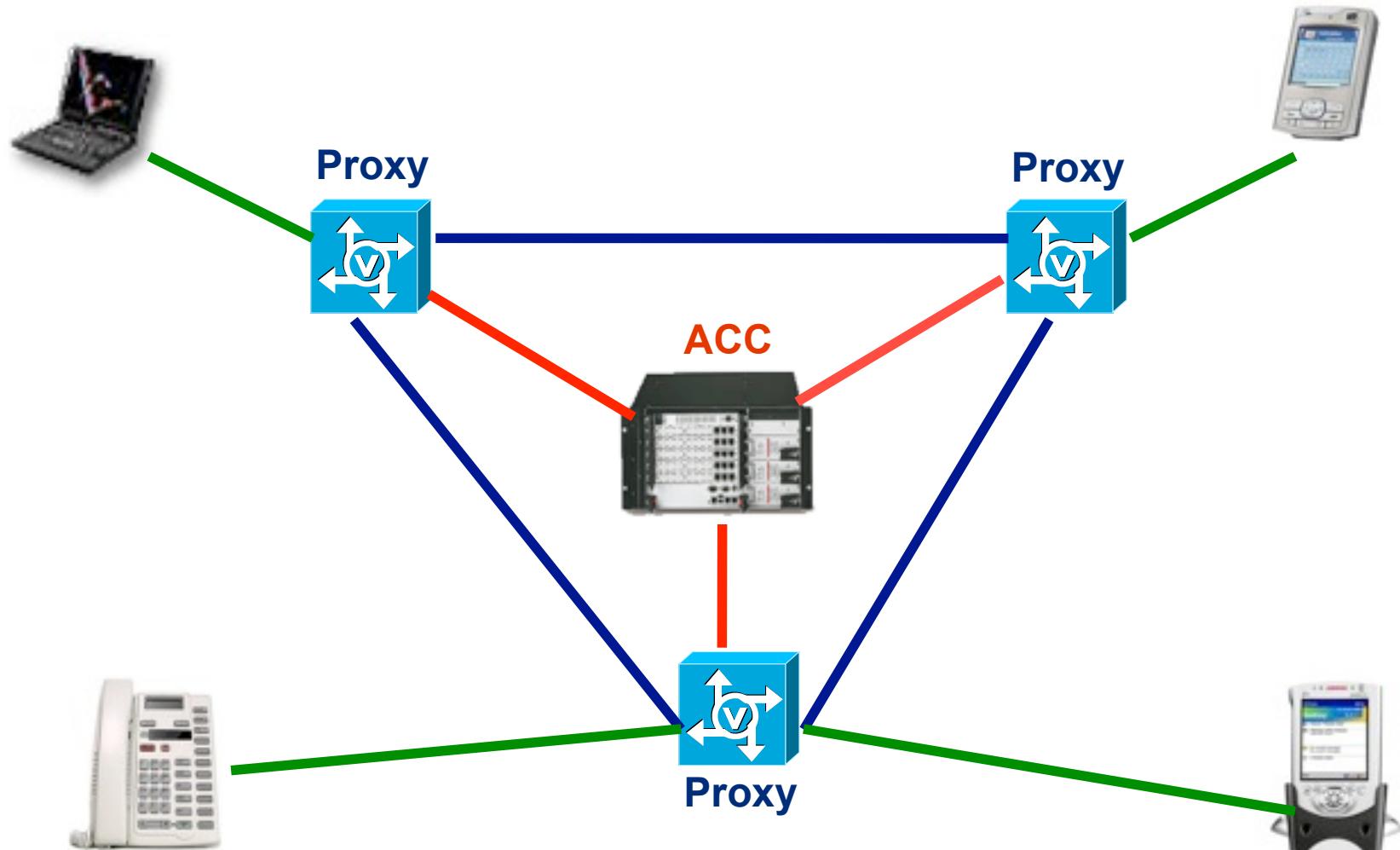
- session control (e.g., rtp timeout)
- topology hiding
- one-to-one SIP hop
- prepaid billing
- calling cards, call queues
- call recording

## 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();
}
```



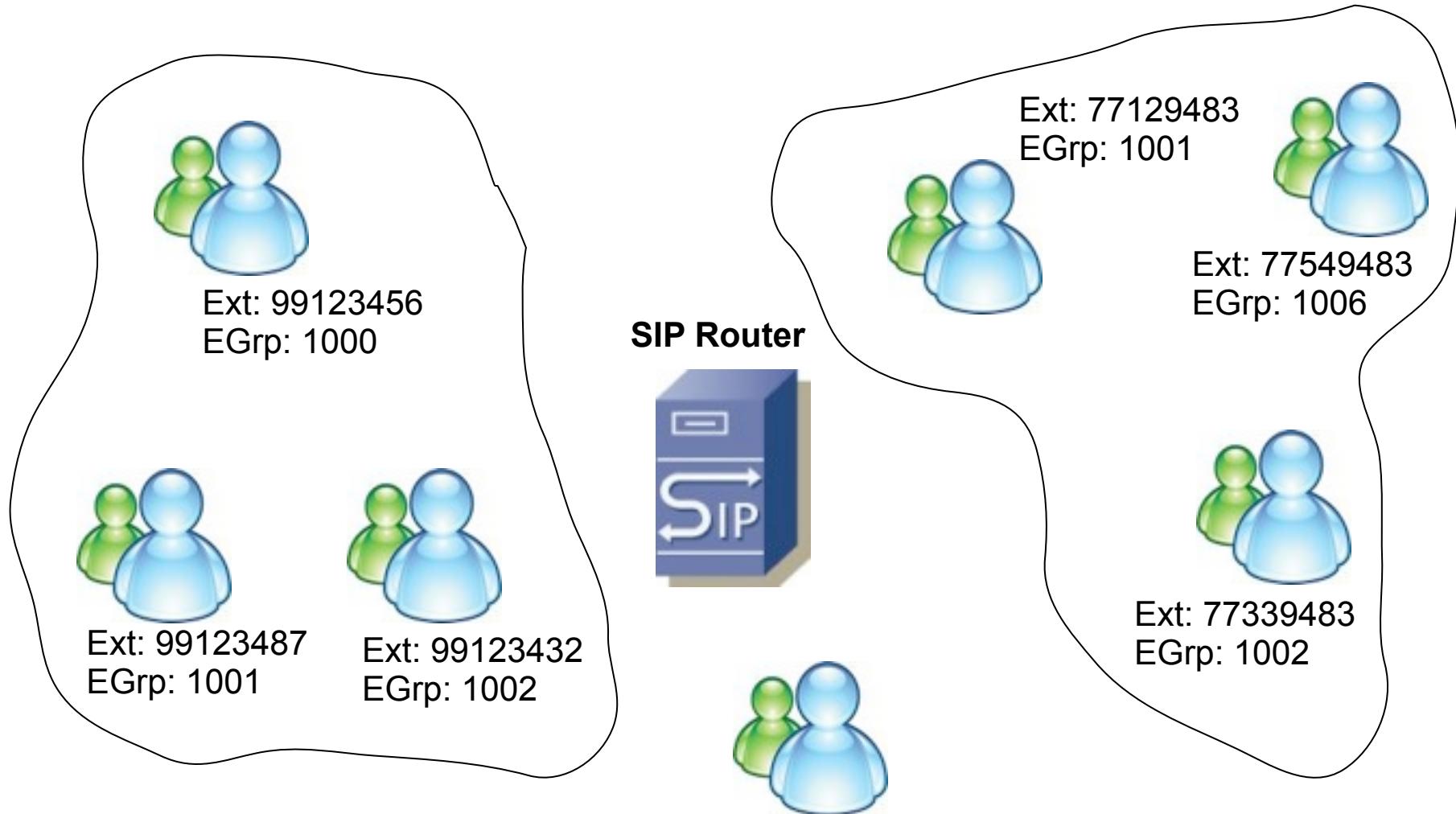
```
route {
    if(method=="ACCOUNTING" && $rU=="store")
    {
        sql_query("ca",
            "insert into accounting
                (timeval,callid,ftag,ttag,src,dst)
            values ('$(rb{s.select,0,,})',
            '$(rb{s.select,1,,})', '$(rb{s.select,2,,})',
            '$(rb{s.select,3,,})', '$(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(\$db(\$ra=>rows)>0) \$rU = \$db(\$ra=>[0,0]);  
....



**Thank you!  
Questions?**

**Daniel-Constantin Mierla**  
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<http://www.asipto.com>



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