



Present and Future

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BERLIN, GERMANY

SEP 2, 2011

10 years SER Conference

SIP Express Router & Kamailio

event sponsored by



sipgate

AMOOMA



Fraunhofer
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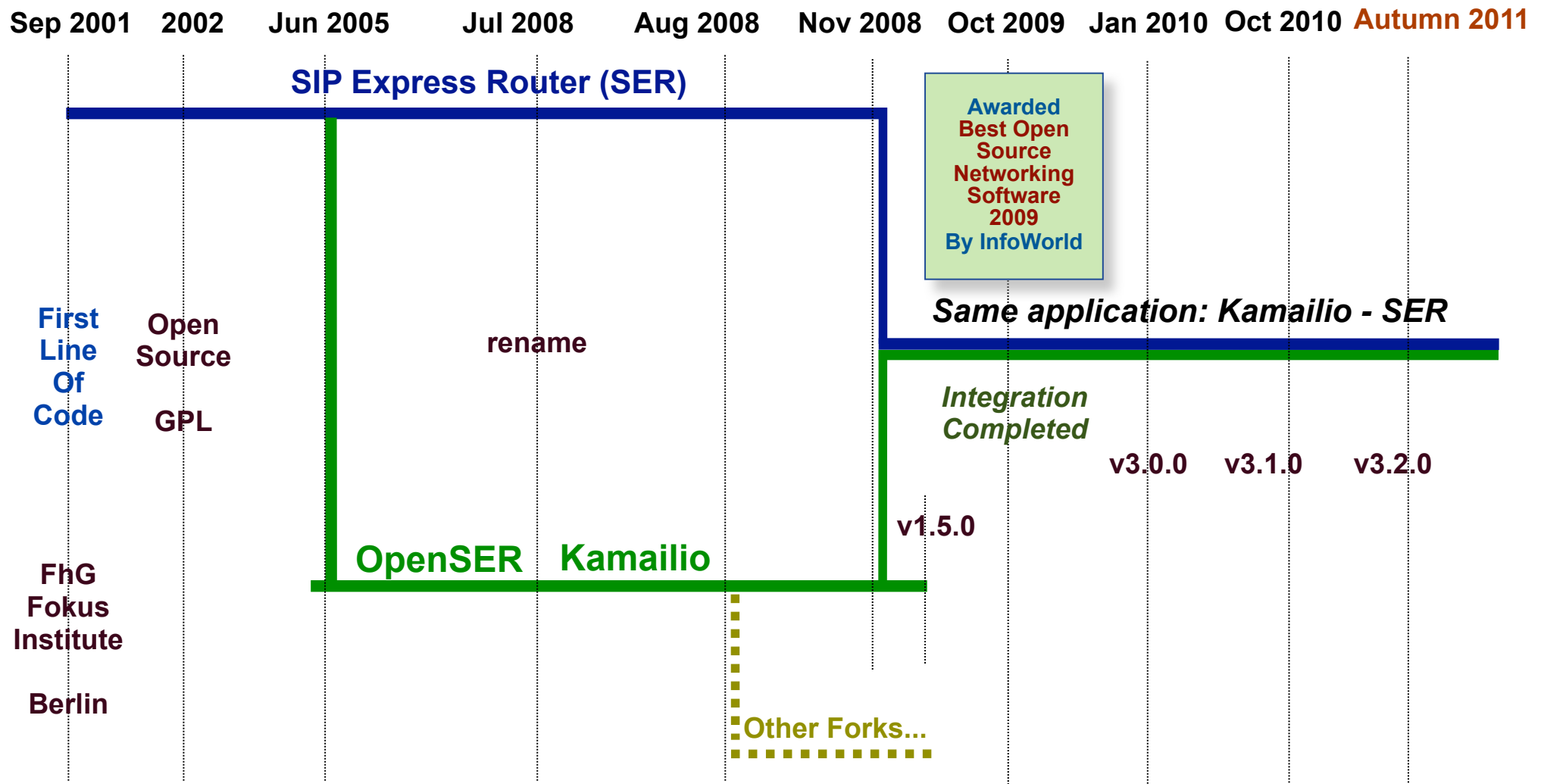


Next Generation
Voice

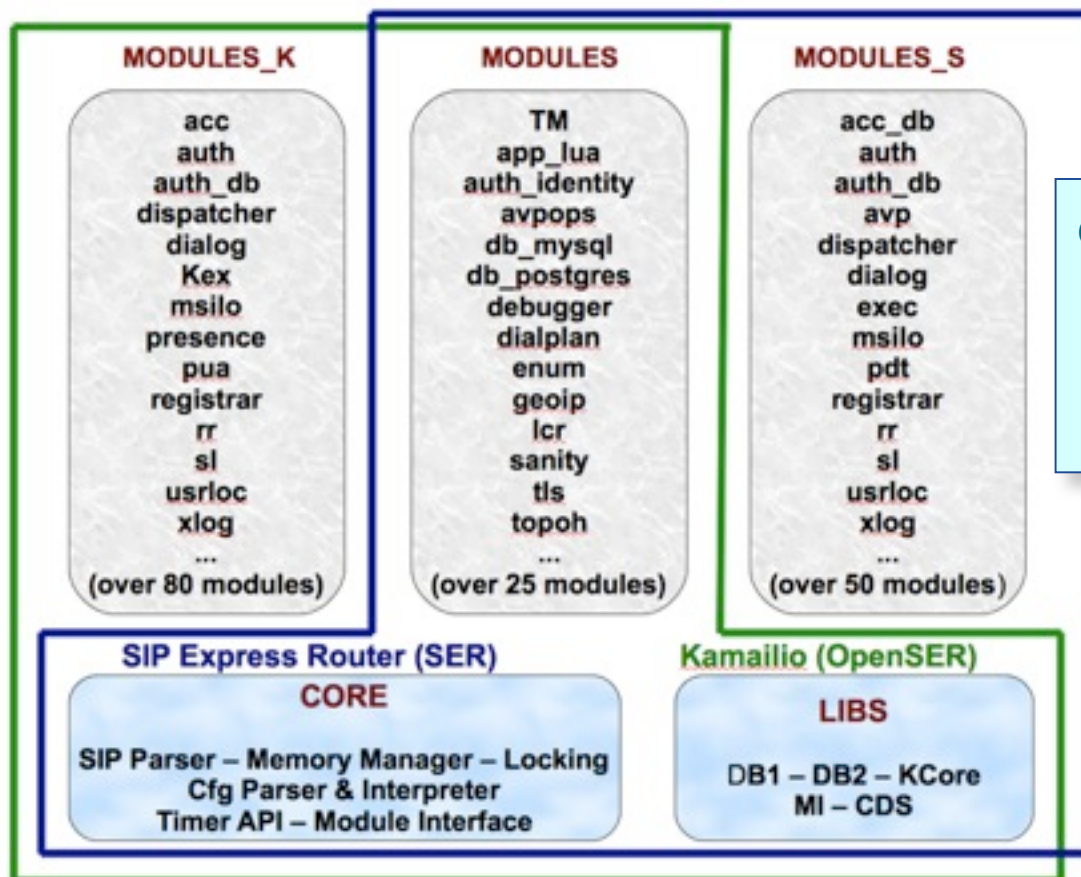
asipto

FRAFOS

A bit of history – 10 years SER



3.x Releases: One application, Two names



Constrained mainly by database schema

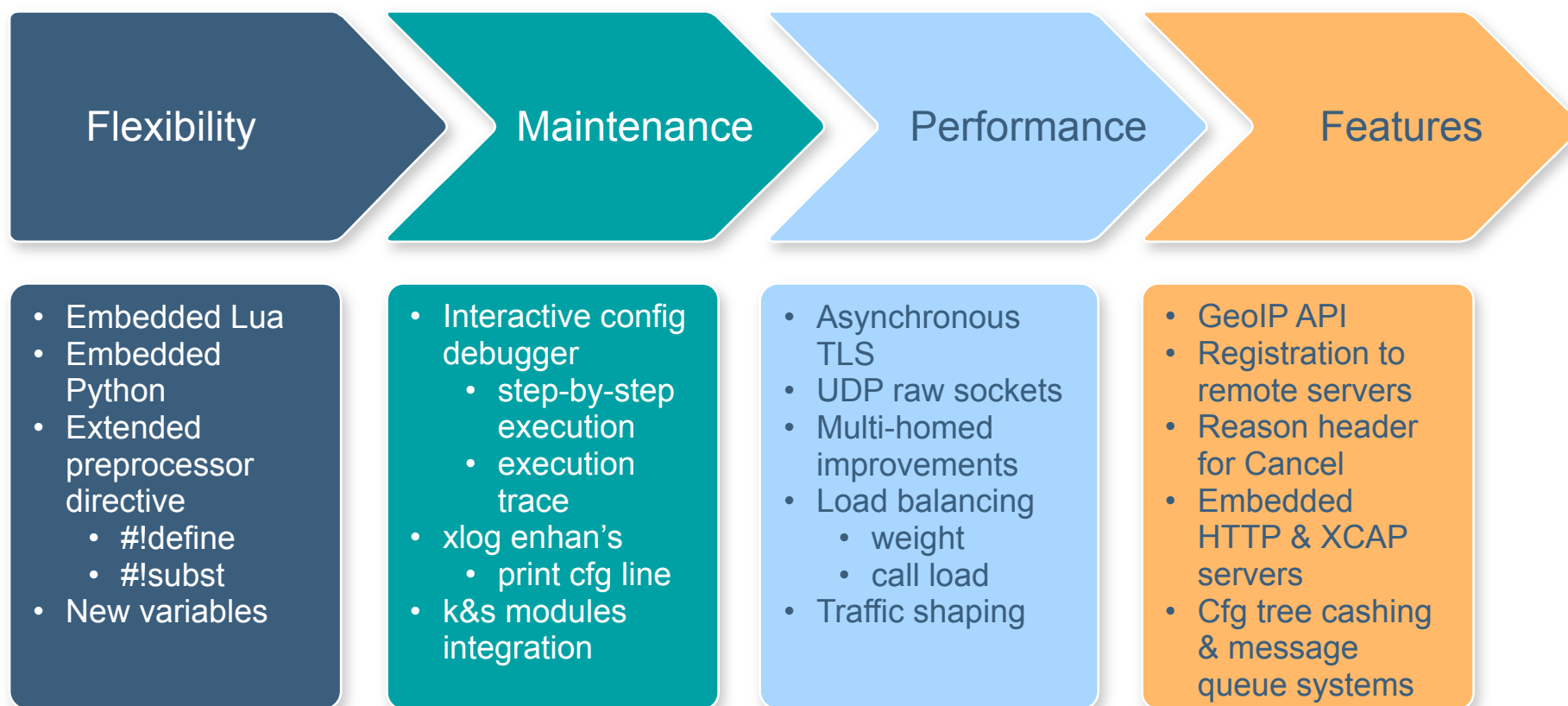
- during 2005 - 2008, SER and Kamailio developed different database structure to store user profiles and routing data
- strong dependency on administration and auto-provisioning systems

Example

- subscriber: username, password, DID, ACL, a.s.o.
- *Kamailio* - table with many columns (one attribute in a dedicated column)
- *SER*: table with many rows of (attribute name, attribute value)

Many duplicated modules were merged meanwhile

New in 3.1.0



<http://www.kamailio.org/w/kamailio-openser-v3.1.0-release-notes/>

State of the project



- Internal architecture refactored for v3.0.0
 - support asynchronous processing
 - TCP and TLS
 - SIP request handling
 - transaction management
 - internal libraries

Right now

- very stable core and main components
 - ➡ topped with our well known scalability and flexibility
- safe framework for future development
 - ➡ your work (extensions and deployments) is safe from now on for many years - there is no need to change the architecture again
- focus is on new features
 - ➡ 3.2.0 (and the next slides) shows that

Scalability (info from public domain)

- services with millions of active subscribers
 - ➡ I&I Germany (> 3M)
- services routing billions of call minutes per month
 - ➡ might be the guy next to you (or pay attention tomorrow)

New in 3.2.0



**Many native extensions
to Lua**

cfg routing logic all in Lua

**Distributed Message
Queue**

Using SIP and Peer-to-Peer

**SQLite
connector**

use file based
database for
embedded
systems

**Partitioned user location
service**

many nodes sharing location
data

Redis No-SQL

connector from config

New in 3.2.0 – presence server



Reg-Info Implementation

RFC3860
pub-sub service for
location data

Presence Server

data distribution across
many instances through
database

Embedded XCAP server

OMA - specs
If-Match cond

RLS

OMA specs
split NOTIFY bodies
XPath support within doc

Presence User Agent

updates for latest
RL services

New in 3.2.0



async module

run asynchronously parts
of config file
(route blocks)

ipops module

a set of operations for
handling IPv4/IPv6 addresses

sdpops module

**SDP body
management**

json modules

New features in old parts


acc - write full CDR at once
dialog - attach extra attributes
core - more pre-processor directives
pv - new variables and transformations
tmx - export of async TM functions
sqlops - support for xavps
uac - enhancements to remote registration
siptrace - traffic replication enhancements
.....

IMS Extensions

about 10 new modules
(P-CSCF, I-CSCF, S-CSCF...)

SIP:Provider - <http://www.sipwise.com/products/spce/>

- * complete VoIP servicing platform using Kamailio for SIP routing
- * administration interface and user portal
- * ready to roll-out open source Community Edition
- * easy to install with DEB packages - images for VMWare and VirtualBox



The screenshot shows the user portal for 'sip:provider CE Demo Company'. At the top right, there is a 'LOGOUT' link and a status 'LOGGED IN AS CETEST1'. Below the header is a navigation bar with tabs: DESKTOP, CALLS, VOICEMAILS, ADDRESSBOOK, CALL FORWARD, CALL BARRING, REMINDER, and ACCOUNT. A welcome message 'Welcome to your personal desktop.' is displayed. The main content area is divided into three columns. The left column, 'RECENT CALLS', has filters for incoming, outgoing, and forwarded calls. It lists five recent calls with details like number, name, duration, and timestamp. The middle column, 'CLIENTS', shows '1 client devices registered'. The right column, 'CALL FORWARDS', shows 'call forward active'. Below this, 'NEW VOICEMAILS' shows one new voicemail. At the bottom right, a 'REMINDER' section shows 'reminder is active'. The footer contains contact information for 'sip:provider CE Demo Company' and logos for sipwise and sip:provider.

sip:provider CE Demo Company [LOGOUT](#)
LOGGED IN AS CETEST1

DESKTOP CALLS VOICEMAILS ADDRESSBOOK CALL FORWARD CALL BARRING REMINDER ACCOUNT

Welcome to your personal desktop.

RECENT CALLS
incoming outgoing forwarded

+4311002	00:20	30.11.2010 18:37:34
Voicebox	00:20	30.11.2010 18:37:34
agranig@sip.sipwise.com		30.11.2010 18:31:51
+4311001		30.11.2010 18:31:42
Voicebox	00:07	30.11.2010 18:31:25

[all calls](#)

CLIENTS
1 client devices registered

NEW VOICEMAILS
+011002
[all voicemails](#)

CALL FORWARDS
call forward active
[view settings](#)

REMINDER
reminder is active
[view settings](#)

sip:provider CE Demo Company | Some City | Some Street
Tel. +439300815 | Fax +439300815 | E-Mail office@somecompany.at

[sipwise](#) [sip:provider](#)

Information

Logged in as
administrator

» Logout

Statistics

User Administration

Accounts

Subscribers

System Administration

Domains

Administrators

Billing

SIP Peerings

Number Management

NCOS

Dashboard

System Voip

Click&Drag on the graphs to zoom individual ranges.

Free Memory

3:20h - 10s steps

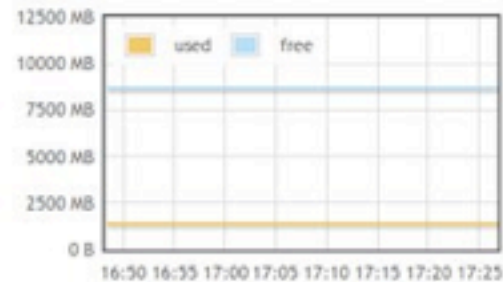
Reset selection



Root Disk

3:20h - 10s steps

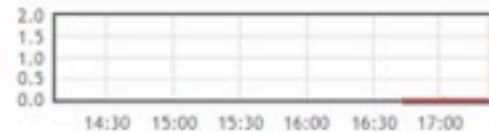
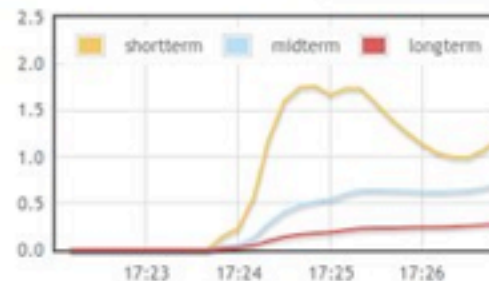
Reset selection



Load

3:20h - 10s steps

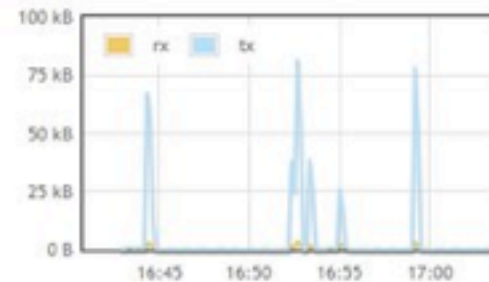
Reset selection



Network Traffic

3:20h - 10s steps

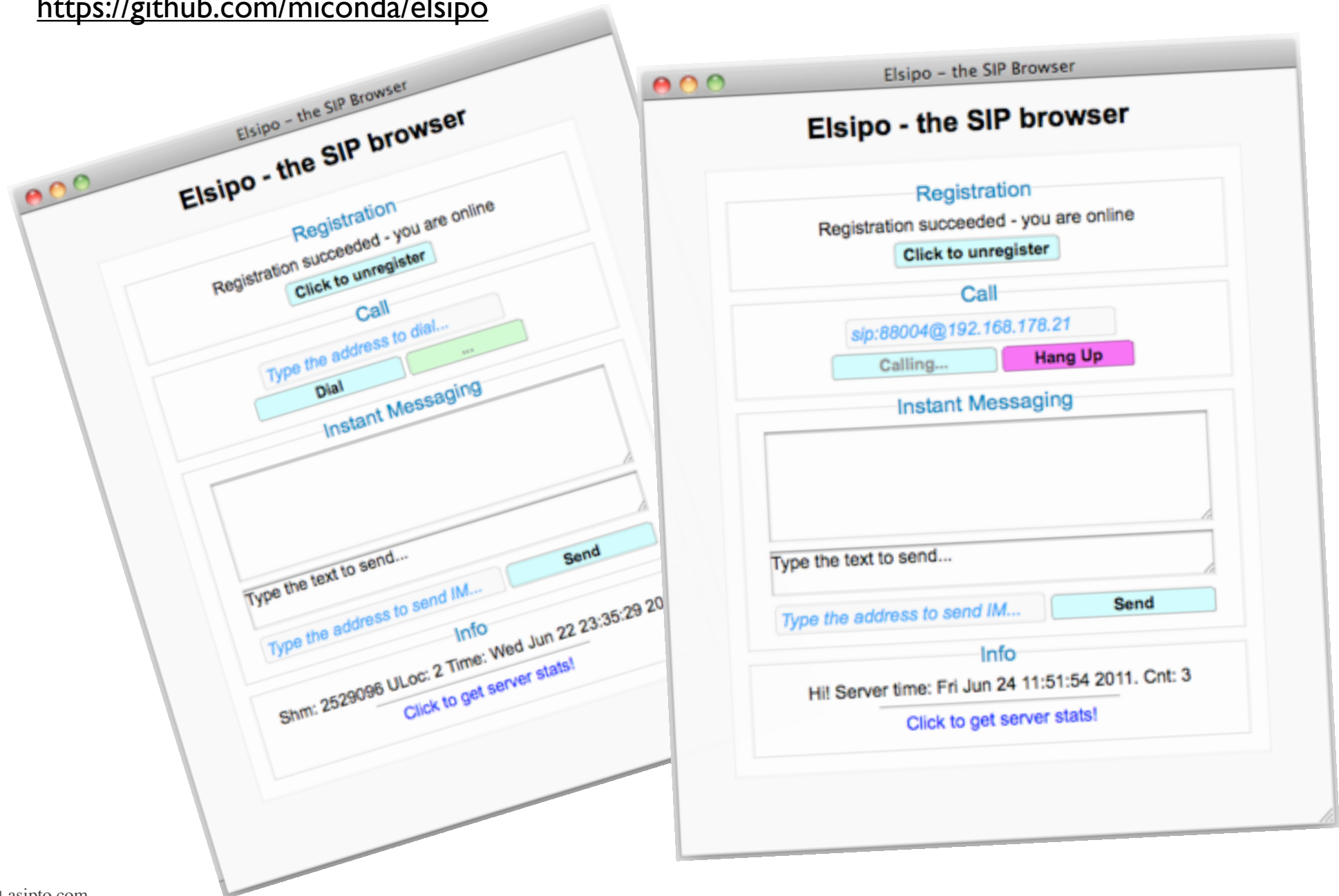
Reset selection



Elsipo – this SIP browser



<https://github.com/miconda/elsipo>



Homer Project



HOMER 2.0
because sip debugging make sense

Search | Advanced Search | Help | Account

Time / Date Parameters
Location ☒ Dusseldorf ☐ Acme
Date Today [21-06-2011]
 Today [21-06-2011]
 Yesterday [20-06-2011]
 30-06-2011

Home • Advanced Search

User Details

PUF User (B-Number)
To User (B-Number) 02115424991
From User (A-Number) 02115424991
PID User (A-Number)
Contact User
Auth User
Logic OR in search ☒

Call Details

Call ID
B2B Call ID ☐
From Tag
To Tag
Via Branch
Method / Reply
Reply reason

Header Details

PUF
Via 1
Division
Cseq
Reason
Content-Type

trace.pcap

212.202.83 sip 213.148.136 5063 213.148.136 sip 213.148.136 391

1 INVITE sip:02115424991@212.202.83 5063 user-phone w/SOP
2 100 Trying - your call is important to us
3 INVITE sip:02115424991@213.148.136 39138 user-phone w/SOP
4 100 Trying
5 180 Ringing
6 180 Ringing
7 200 OK w/SOP
8 200 OK w/SOP
9 ACK sip:02115424991@213.148.136 39138 user-phone
10 ACK sip:02115424991@213.148.136 39138 user-phone
11 BYE sip:02115424991@213.148.136 5063
12 BYE sip:02115424991@213.148.136 5063
13 200 OK
14 200 OK

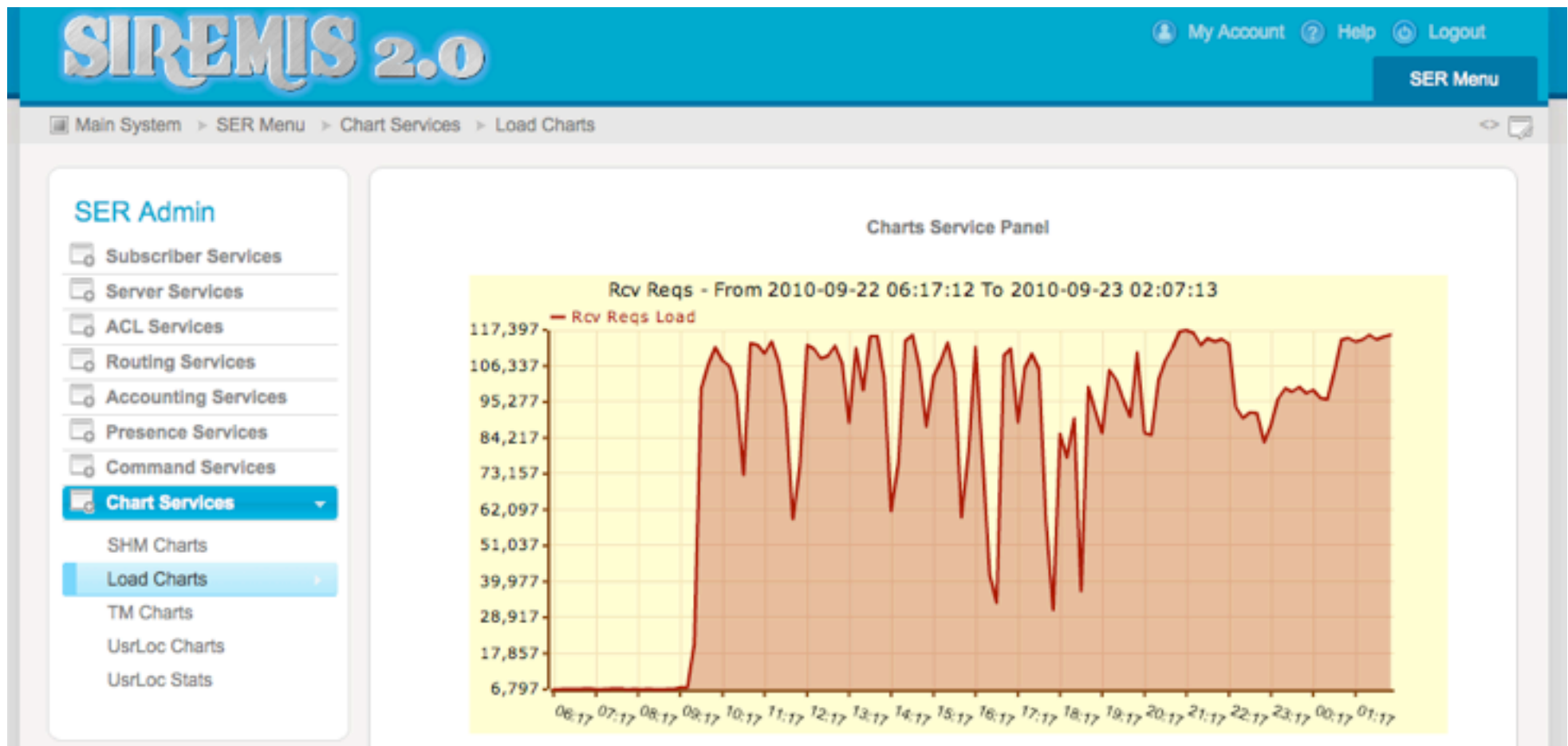
Static | Dynamic | Frames
Trace source

Frame 11 (489 bytes on wire, 489 bytes captured)
Arrival Time: Jun 21, 2011 13:40:23.111627000
Internet Protocol, Src: 213.148.136, (213.148.136)
User Datagram Protocol, Src Port: 39138 (39138),
Session Initiation Protocol
Request-Line: BYE sip:02115424991@213.148.136
Method: BYE
Request-URI: sip:02115424991@213.148.136
Request-URI User Part: 02115424991
Request-URI Host Part: 213.148.136
Request-URI Host Port: 5060
[Resent Packet: False]
Message Header
Via: SIP/2.0/UDP 213.148.136:39138;branch
Transport: UDP
Sent-by Address: 213.148.136
Sent-by port: 39138
Branch: z9hG4bKE3ed86w6d51e7173ca4ed
Route: <sip:212.202.83>;lr/etag=18h6ug
Call-ID: 7y1ih1lmus55y5g7ymgumco8y16yme
From: <sip:02115424991@213.148.136>;user=p
SIP from address: sip:02115424991@213.148.136
SIP from address User Part: 02115424991
SIP from address Host Part: 213.148.136
SIP tag: 211818439
To: <sip:02115424991@213.148.136>;user=p
SIP to address: sip:02115424991@213.148.136
SIP to address User Part: 02115424991
SIP to address Host Part: 213.148.136
SIP tag: 18h6ug11-CC-30
CSeq: 1 BYE
Sequence Number: 1
Method: BYE
Max-Forwards: 70
User-Agent: 5450 IP/0211700000000
Content-Length: 0

Siremis 2.0



<http://siremis.asipto.com/>



- a research project about Green VoIP
 - by Columbia University, NY
 - using complete config file, with user authentication and NAT traversal
 - injected traffic captured from an European ITSP



<http://www.kamailio.org/w/2011/05/green-voip-energy-efficiency-and-performances-of-v3-0/>

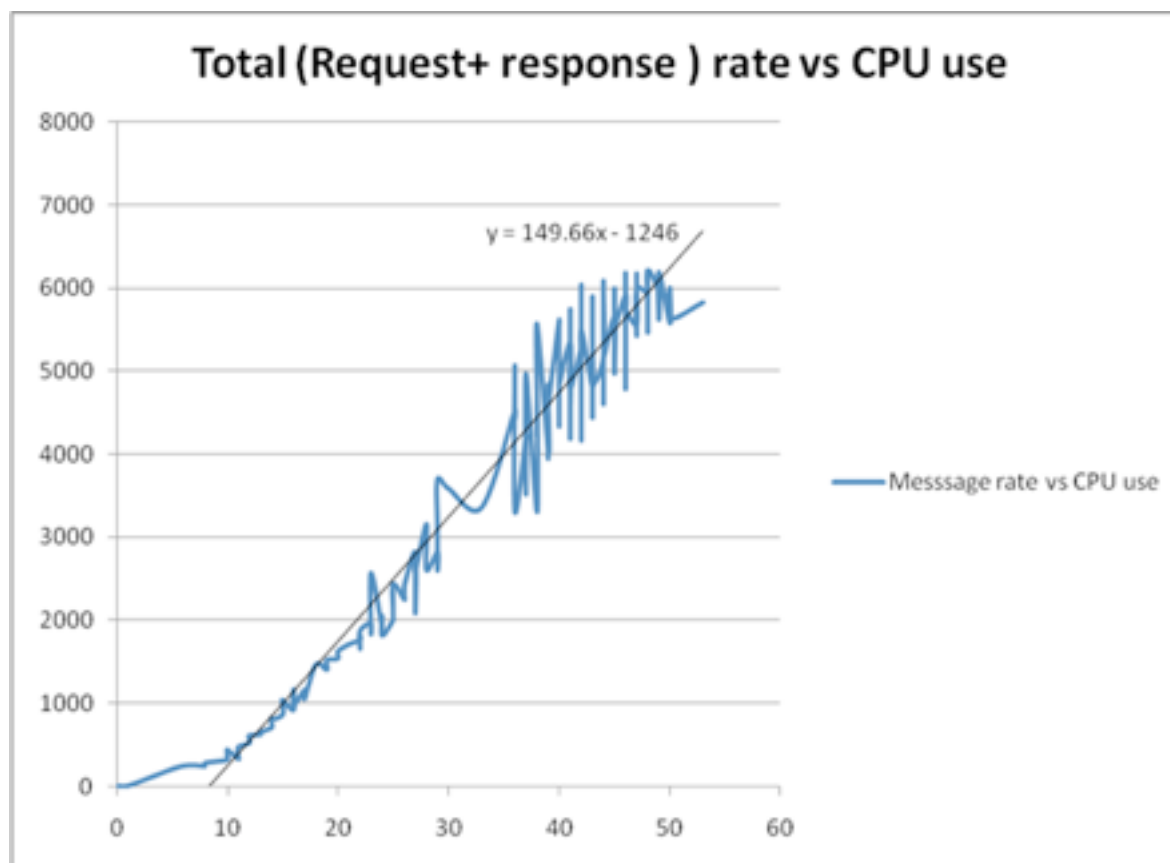
Some interesting results:

- one instance of SIP server with *500 000 online users* (mixed users – behind and not NAT routers) – consumed energy *210W*
- one instance of SIP server with *1 000 000 online users* (no NAT involved) – consumed energy *190W*
- on a 32-bit machine with 4GB of memory and with 2.5GB reserved for SIP server, the server could support *43 000 simultaneous TLS connections* – consumed energy *209W*
- one SIP server instance with *80 000 permanent TCP connections*, the SIP server could still handle at least *1000 requests per second* and a connection arrival rate of *1000 new connections per second*, done for 20 000 new connections. CPU load generated by the SIP server was from *6% to 8%*.

TLS – Stress tests



- private company lab environment
 - Kamailio 3.1.x with 8 children and 2 GB memory
- traffic stress
 - 6000 SIP messages/second for 2 weeks
- socket stress
 - created over 4000 connections
 - released the connections immediately
 - at the same time created more connections



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Voice

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Thank you!
Questions?

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