

foss
asia
2018



BUILD YOUR OWN RTC SERVICE

VOIP - MESSAGING - PRESENCE

DANIEL-CONSTANTIN MIERLA

CO-FOUNDER KAMAILIO PROJECT

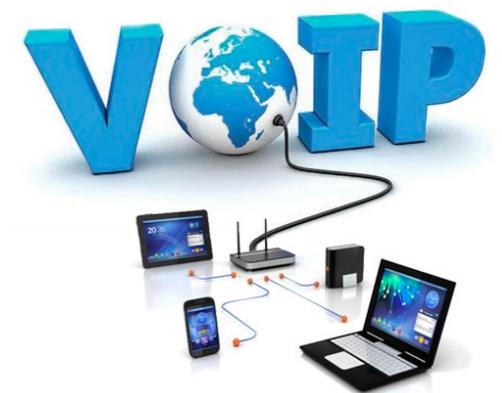
WWW.KAMAILIO.ORG

@MICONDA - DANIEL@ASIPTO.COM

OPEN SOURCE AND KAMAILIO SIP SERVER PROJECT



- ▶ Originally from Romania, living in Berlin, Germany
- ▶ Computer science software engineer
- ▶ Involved in open source real time communications since 2002
- ▶ Shifted from a researcher position to professional consultancy for SIP, Kamailio and all RTC
- ▶ Developing and living only from open source software and services for like 15 years
- ▶ C programmer - mainly VoIP server side infrastructure
- ▶ Co-founder and lead developer of Kamailio
- ▶ Co-organizer of Kamailio World Conference
- ▶ Speaking and promoting OSS RTC at world wide events
- ▶ Enjoying sports and nature, both sea side and mountains
- ▶ Working at Asipto - www.asipto.com

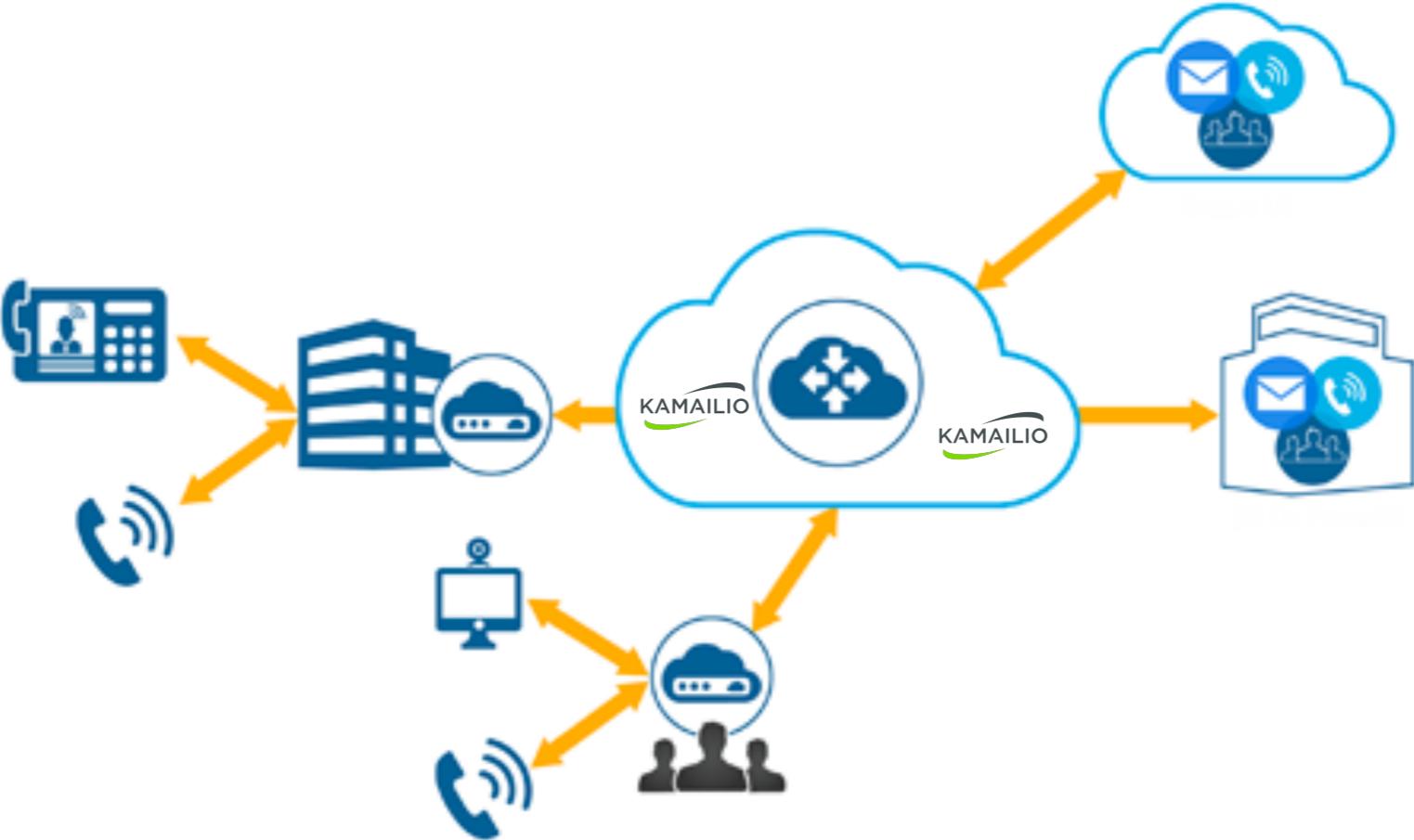


DIRECTION TO BUILD YOUR OWN RTC SERVICE

- ▶ something similar to skype, facetime, whatsapp, wechat, viber, ...
- ▶ using open source
- ▶ using open standards
- ▶ run it for yourself or your community
 - ▶ or start a telephony business
 - ▶ residential or carrier services
 - ▶ straightforward integration with classic telephony/mobile networks
 - ▶ most of the telephony services are using the same protocol (SIP)

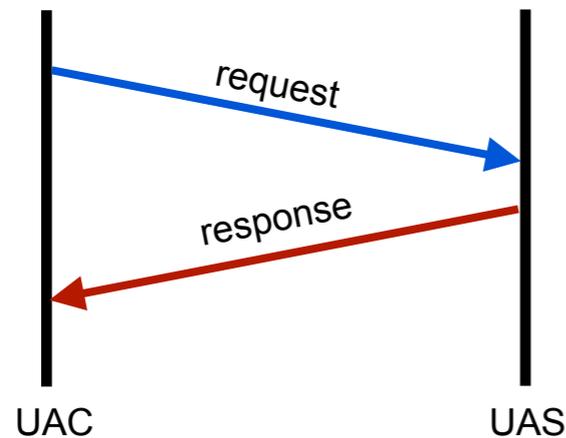


CONNECT IN REALTIME



INTERACTION PROTOCOL

OPEN STANDARD - IETF RFC3261 (+ EXTENSIONS)



request

<i>Start line</i>	INVITE sip:user@sipserver.com SIP/2.0
<i>Message headers</i>	Via: SIP/2.0/UDP 10.10.10.10:5060;branch=z9hG4bKxy From: "Me" <sip:me@sipserver.org>;tag=a012 To: "User" <sip:user@sipserver.org> Call-ID: d@10.10.10.10 CSeq: 1 INVITE Contact: <sip:10.10.10.10:5060> User-Agent: SIPTelephone Content-Type: application/sdp Content-Length: 251
<i>Message body</i>	v=0 o=audio1 0 0 IN IP4 10.10.10.10 s=session c=IN IP4 10.10.10.10 m=audio 54742 RTP/AVP 4 3 a=rtpmap:4 G723/8000 a=rtpmap:3 GSM/8000

response

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.10.10:5060;branch=z9hG4bKxy From: "Me" <sip:me@sipserver.org>;tag=a012 To: "User" <sip:user@sipserver.org>;tag=b034 Call-ID: d@10.10.10.10 CSeq: 1 INVITE Contact: <sip:10.10.10.20:5060> User-Agent: SIPSoftPhone Content-Type: application/sdp Content-Length: 123
v=0 o=audio2 0 0 IN IP4 10.10.10.20 s=session c=IN IP4 10.10.10.20 m=audio 62043 RTP/AVP 0 4

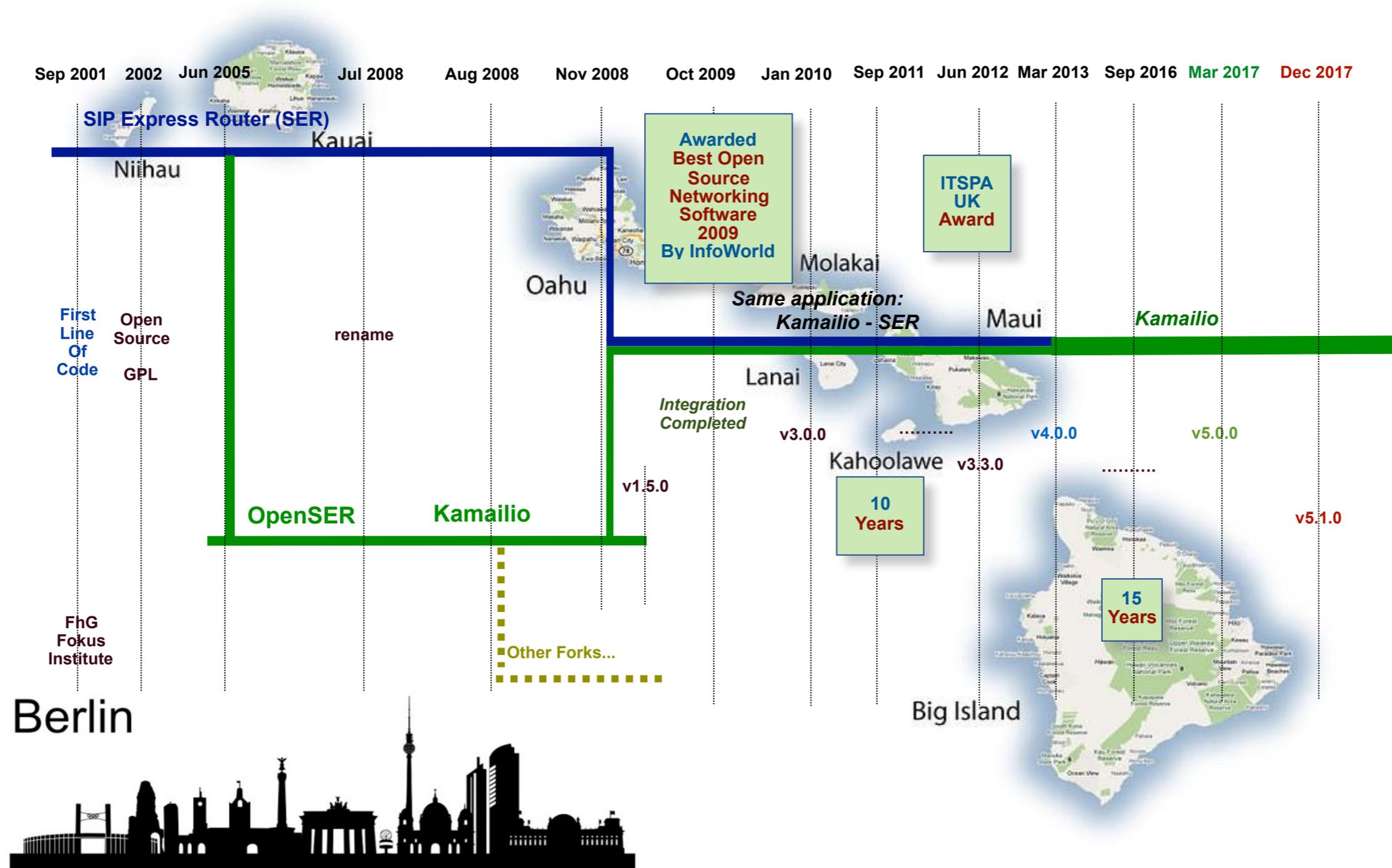
- * telephony over ip
- * VoIP and VoLTE (4G & 5G)
- * voice and video
- * instant messaging, presence

SERVER APPLICATIONS - CLIENT DEVICES & APPLICATIONS



SERVER APPLICATIONS

LET'S SPEAK SIP = E KAMA'ILIO SIP

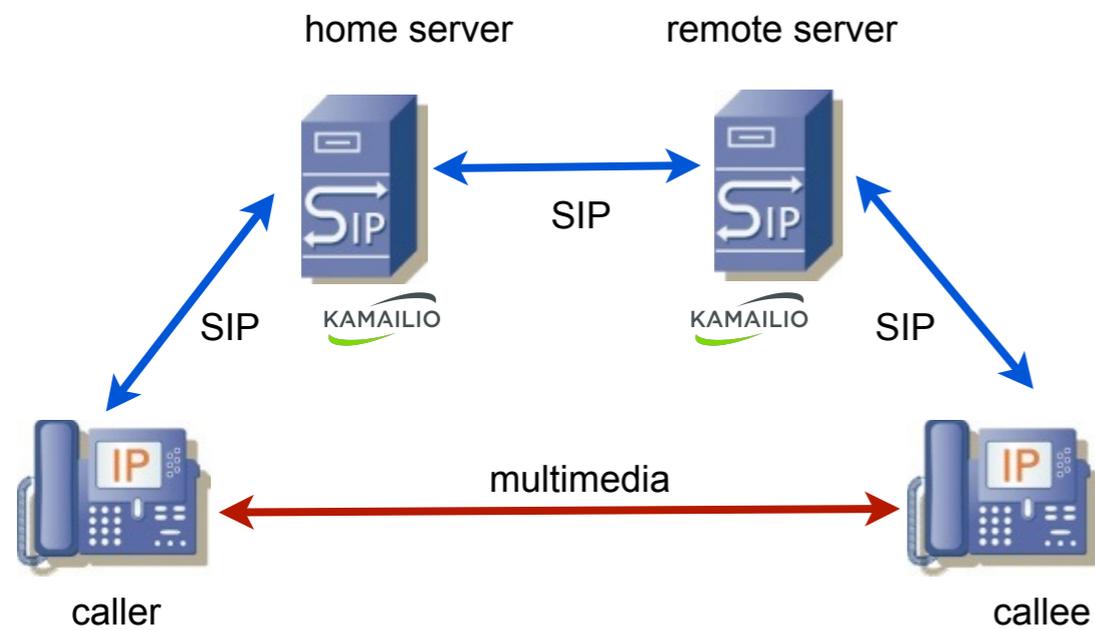


KAMAILIO SIP SERVER IN ONE SLIDE

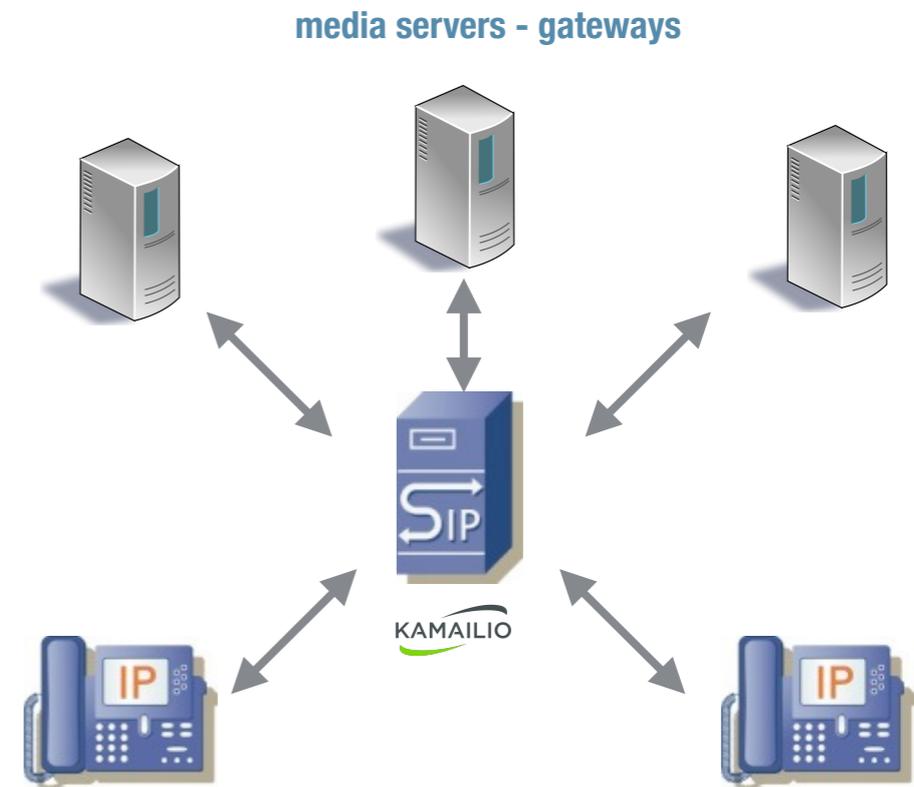


- * Open Source SIP (IETF RFC3261) Signaling Server implementation, developed since 2001
- * Can be used for VoIP (Voice, Video, VoLTE/IMS, SIP-I/SIP-T), Instant Messaging, Presence, WebRTC, IoT, Diameter, SQL and NoSQL backends
- * Designed for modularity, flexibility and scalability
 - * used by large telecoms, mobile operators and OTT services world wide
 - * thousands of call setups per second, hundred thousands of connected phones per instance
- * IPv6/IPv4 - UDP/TCP/TLS/SCTP/WebSocket - asynchronous routing
- * Classic SIP - WebRTC gateway using Kamailio + RTPEngine
- * Embedded interpreters: Lua, Python, JavaScript, Squirrel, Perl, .Net, Java
- * Over 200 modules (extensions) - <https://www.kamailio.org/docs/modules/stable/>
- * Over 50 active developers each year (over 300 contributors over the time)
- * Runs its own conference - Kamailio World
 - * the 6th edition: May 14-16, 2018, in Berlin, Germany: <https://www.kamailioworld.com>

COMMON USE CASES



- * authentication, registration and user location
- * voice, video, instant messaging and presence
- * NAT traversal, RTP relaying, webrtc
- * SIP security firewall - DDoS mitigation, anti-fraud
- * integration with social networking



- * load balancer
- * least cost routing
- * transport layer gateway
- * topology hiding
- * carriers interconnect

ROUTING SIP MESSAGES



- ▶ Deals only with SIP signaling packets
- ▶ Proxy function at the core (not back to back user agent)
- ▶ Initial design for modularity, flexibility and scalability
- ▶ Special focus on innovation, security and privacy
- ▶ Scripting language for configuration of SIP routing
- ▶ Independent applications for routing RTP (media packets) - RTPEngine, RTPProxy
- ▶ No media (audio, video) processing - for announcements, voicebox, conferences, a.s.o., use open source applications such as Asterisk or FreeSwitch



SCRIPTING SIP ROUTING

- Two main roles
 - Kamailio application initialization
 - Done once at startup (passive scope)
 - Global parameters, loading modules and modules' parameters
 - Many values can be changed at runtime via RPC (no restart)
 - Rules for handling SIP traffic
 - Done during runtime to decide the routing of SIP messages
 - No reload without restart for native kamailio.cfg scripting language
 - KEMI routing scripts can be reloaded without restart (v5.0+)
- Scripting languages
 - Native scripting language
 - Initially designed in 2001-2002, built from scratch
 - Kamailio Embedded Interface (KEMI) languages
 - Introduced in v5.0
 - Reuse existing scripting languages
 - Support for Lua, Python, JavaScript, Squirrel language
 - Allow reloading of scripts without restart
 - Inline execution of scripting languages or REST-API based routing
 - Can be executed inside native scripting language
 - Support for Lua, JavaScript, Python, Perl, .Net (C#, ...), Squirrel, Java

```
# global settings
#!define FLT_ACC 1
debug=9
fork=no
listen=192.168.1.34:5060
...
pstn.gw = 1.2.3.4" desc "pstn gateway ip"
...

# module settings
mpath="/usr/local/lib/kamailio/modules/"
loadmodule="tm.so"
...
modparam("tm", "fr_inv_timer", 30000)
.....

# routing blocks
request_route {
    xlog("request received from $si\n");
    if($si=="10.1.2.10") {
        route(REDIRECT);
    } else {
        $rd = "10.1.2.5";
    }
    t_on_reply("LOGRPL");
    t_relay();
}
route[REDIRECT] {
    $rd = "10.1.2.3";
    send_reply("302", "Redirected");
    exit;
}
onreply_route[LOGRPL] {
    xlog("response received from $si\n");
}
...
```

MIX TO BUILD THE TELEPHONY ROUTING SYSTEM

- ▶ Authentication
- ▶ Authorization
- ▶ Accounting
- ▶ Registration
- ▶ Location
- ▶ Least cost routing
- ▶ Load balancing
- ▶ Encryption
- ▶ ...



LOAD BALANCING VOIP CALLS

Dispatcher Module - Destinations

- text file with destinations
 - each record per line
 - comments start with #

```
# setid    destination                flags    priority    attributes
# (int)    (sip uri)                (int,opt) (int,opt)    (str,opt)

# proxies
2        sip:127.0.0.1:5080;transport=tcp    0        10        class=4;prefix=448;strip=2;
2        sip:127.0.0.1:5082;px=vx      0        5         socket=udp:192.168.0.125:5060

# gateways
4        sip:127.0.0.1:7070            0        0         duid=xyz;maxload=20;
4        sip:127.0.0.1:7072            0        5
4        sip:127.0.0.1:7074
```

- database support
 - **dispatcher** table
 - each field is a column in database table

LOAD BALANCING VOIP CALLS

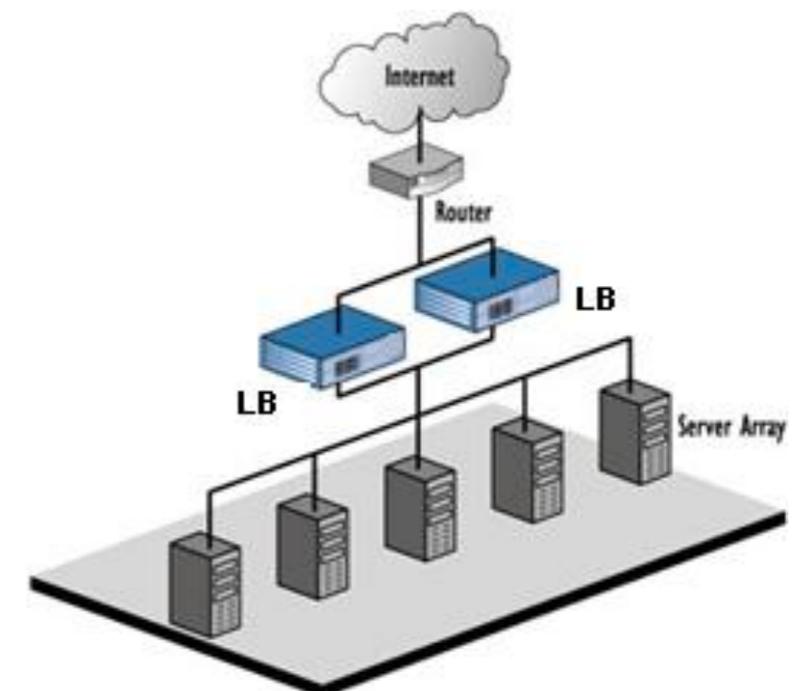
```
loadmodule "dispatcher.so"

# ----- dispatcher params -----
modparam("dispatcher", "db_url", DBURL)
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", "$avp(AVP_CNT)")
modparam("dispatcher", "sock_avp", "$avp(AVP SOCK)")
modparam("dispatcher", "attrs_avp", "$avp(AVP_ATTRS)")

# Dispatch requests
route[DISPATCH] {
    # round robin dispatching on gateways group '1'
    if(!ds_select_dst("1", "4")) {
        send_reply("404", "No destination");
        exit;
    }

    xlog("L_DBG", "--- SCRIPT: going to <$ru> via <$du>\n");
    t_on_failure("RTF_DISPATCH");
    route(RELAY);
    exit;
}
```

```
# Try next destinations in failure route
failure_route[RTF_DISPATCH] {
    if (t_is_canceled()) {
        exit;
    }
    # next DST - only for 500 or local timeout
    if (t_check_status("408|5[0-9][0-9]")) {
        if(ds_next_dst()) {
            t_on_failure("RTF_DISPATCH");
            route(RELAY);
            exit;
        }
    }
}
```



LESS THAN 5 MIN ON DEBIAN

Run:

```
apt install kamailio kamailio-mysql-modules mysql-server
```

Edit /etc/kamailio/kamctl and set DBENGINE=MYSQL

Edit /etc/kamailio/kamctl and add next snippet after the first line

```
#!/define WITH_MYSQL
#!/define WITH_AUTH
#!/define WITH_USRLOCDB
```

Run:

```
kamdbctl create
```

Add users:

```
kamctl add alice@sipdomain.com secret
```

Start kamailio:

```
systemctl start kamailio
```

Configure sip phones/apps and start talking.

PLATFORM ADMINISTRATION

SIREMIS

<https://www.siremis.org> - <https://github.com/asipto/siremis>

SIREMIS 3.2 Administration | My Account | Help | Logout

Administration | SER Menu

Main System > SER Menu > Chart Services > Acc Summary

admin <admin@yourcompany.com>

December 14, 2011, 10:10 am
Timezone: America/Los_Angeles

Period	INVITE - ALL	INVITE - 200	INVITE - 404	INVITE - 487	INVITE - XYZ	BYE - ALL	ALL RECORDS
Last 72 Hours	143	143	0	0	0	162	305
Last 48 Hours	105	105	0	0	0	121	226
Last 24 Hours	68	68	0	0	0	80	148
5 To 4 Hours Ago	9	9	0	0	0	8	17
4 To 3 Hours Ago	1	1	0	0	0	2	3
3 To 2 Hours Ago	8	8	0	0	0	8	16
2 To 1 Hours Ago	4	4	0	0	0	4	8
Last Hour	0	0	0	0	0	0	0

SER Admin

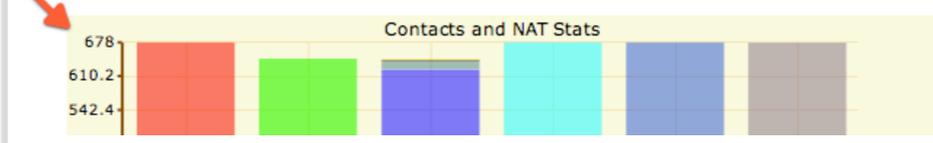
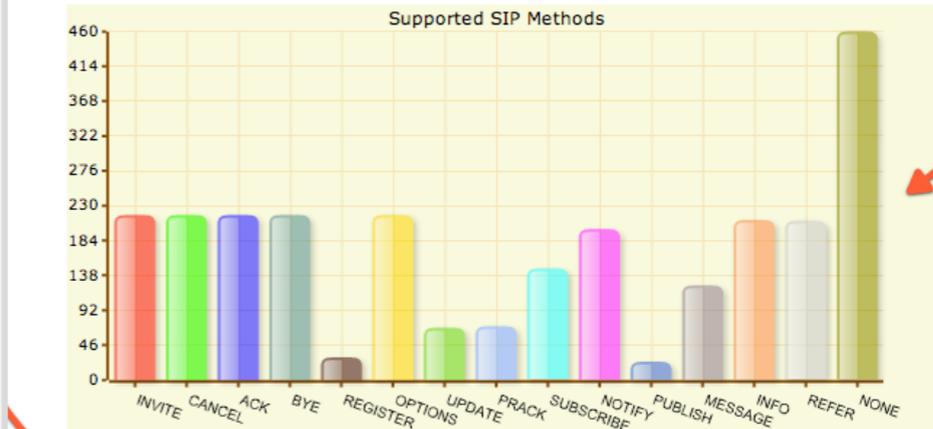
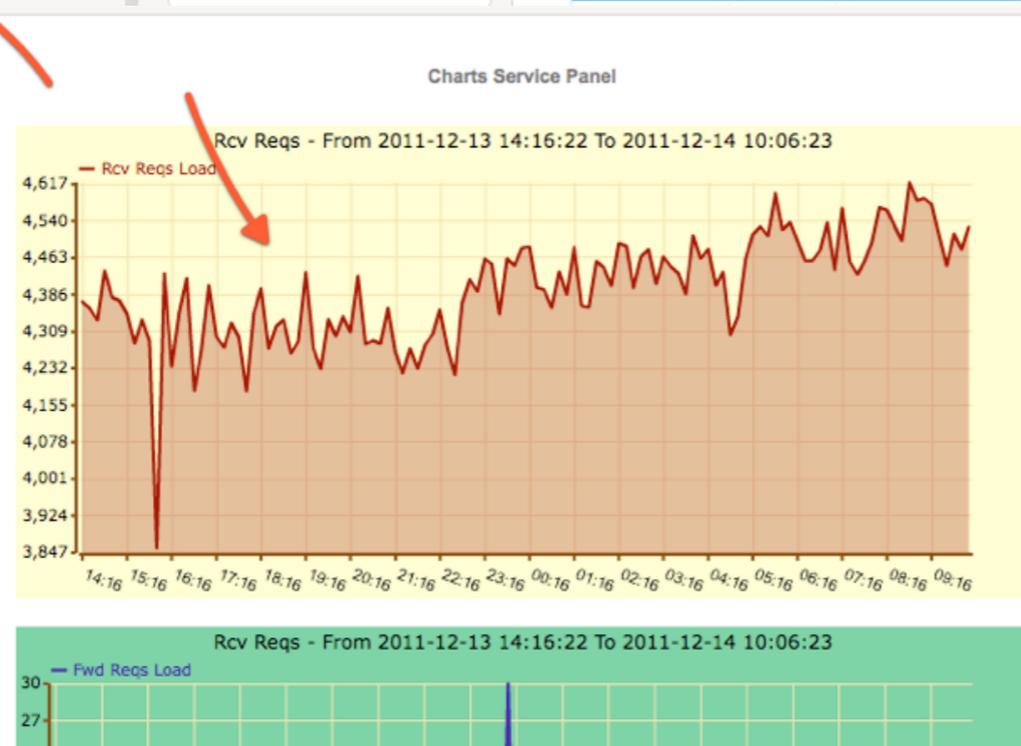
- Subscriber Services
- Server Services
- ACL Services
- Routing Services
- Accounting Services
- Presence Services
- Command Services
- Chart Services**
 - SHM Charts
 - Load Charts
 - TM Charts
 - UsrLoc Charts
 - UsrLoc Stats
 - Acc Charts
 - Acc Summary**

SIREMIS 3.2

Main System > SER Menu > Chart Services > Load Cha

SER Admin

- Subscriber Services**
 - Subscriber List
 - Aliases DB List
 - Speed Dial List
 - User Preferences
 - URI DB List
 - Location List
 - Messages List
 - User Black List
 - Global Black List
- Server Services
- ACL Services
- Routing Services
- Accounting Services
- Presence Services
- Command Services
- Chart Services



FREE OPEN SOURCE WITH KAMAILIO

- ▶ SIP:Provider CE - <https://www.sipwise.org/products/spce/>
- ▶ Kazoo - <https://github.com/2600hz>
- ▶ iVOZProvider - <https://github.com/irontec/ivozprovider>
- ▶ dSIPRouter - <https://github.com/dOpensource/dsiprouter>

The screenshot shows the user interface for 'sip:provider CE Demo Company'. At the top, there's a navigation menu with options like DESKTOP, CALLS, VOICEMAILS, ADDRESSBOOK, CALL FORWARD, CALL BARRING, REMINDER, and ACCOUNT. Below this, a section titled 'Incoming and outgoing calls.' is visible. The main content area displays a 'CALL LIST' for the number '+43 1 1005'. It includes a search bar, filters for 'incoming', 'outgoing', and 'forwarded', and a table of call records. The table has columns for 'number', 'duration', 'charges', and 'date'. Below the table, there are links for 'apply filter' and 'reset filter', and a 'CALLS BY MONTH' section showing 'November 2010'. At the bottom, contact information for 'sip:provider CE Demo Company' is provided, including telephone, fax, and email addresses.

The screenshot displays the 'KAZOO SMART PBX' dashboard for a user named '2600hz-sandbox'. The dashboard is divided into several sections. On the left, there's a sidebar menu with options like Numbers, Users, Groups, Main Number, Call Logs, Devices, Voicemail Boxes, and Feature Codes. The main area shows several key metrics: 'Total Users' (10), 'Total Devices' (6), 'Unregistered Devices' (6), and 'Conference Bridges' (1). A donut chart shows the distribution of device types: Soft Phone (3), SIP Phone (2), Business Mobile (1), Cell Phone (0), Smartphone (0), Landline (0), Fax (0), ATA (0), and SIP URI (0). On the right, there's a 'Main Number' section with details for '+1 555 555 5555', including 'Conference Number' and 'Faxbox Number'. Below this, a 'Total Numbers' section shows 'Spare Numbers' (8) and 'Assigned Numbers' (4). Another donut chart shows the distribution of numbers: 'US DID' (11) and 'Unknown' (1). At the bottom right, there's a 'Company Directory Users' section with 2 users and a 'Download' link. The footer contains account information: 'Account ID: 47b2c990f855c34a999e66' and 'Account Realm: telecom.2600hz.com'.

KAMAILIO DEPLOYMENTS



2nd telecom in Germany
over 10M phone numbers



ip telephony operators



SERVER APPLICATIONS - CLIENT DEVICES & APPLICATIONS



CLIENT DEVICES & APPLICATIONS

DESK PHONE FEELING

- ▶ Cisco
- ▶ Polycom
- ▶ Yealink
- ▶ Grandstream
- ▶ Snom
- ▶ Mitel
- ▶ Panasonic



MOBILE, DESKTOP AND WEB APPS

▶ Open source



▶ Linphone (all platforms) - <http://linphone.org>

▶ Jitsi (desktop) - <http://jitsi.org>



▶ CSipSimple (android)

▶ Ekiga (desktop) - <http://ekiga.org>

▶ SIPDroid (android) - <http://sipdroid.org>

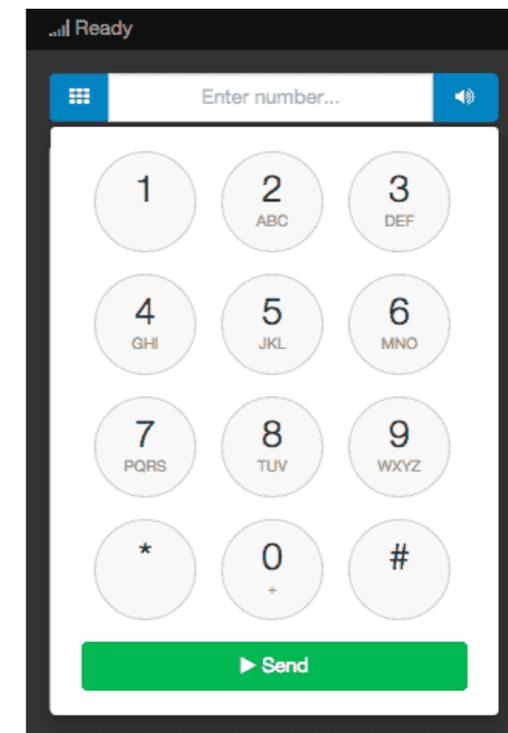
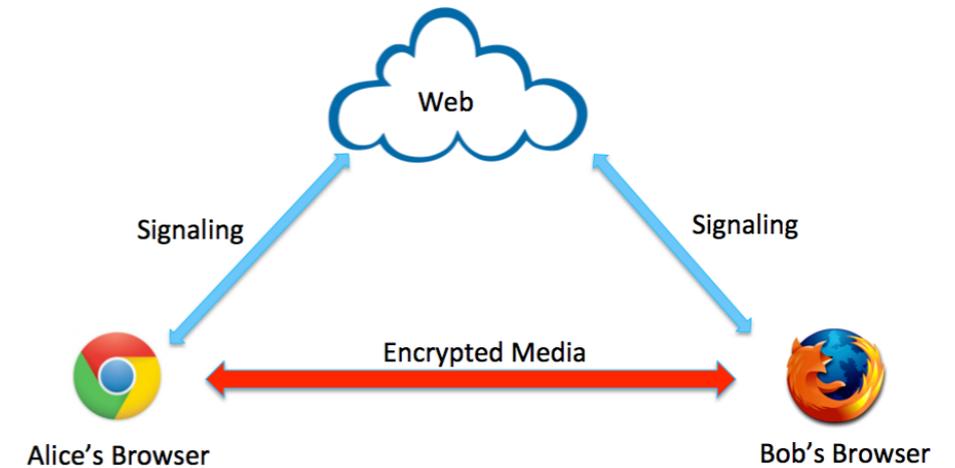
▶ Free to use

▶ Zoiper (all platforms) - <http://zoiper.org>



MOBILE, DESKTOP AND WEB APPS

- ▶ Browser - WebRTC
 - ▶ ctxPhone - collectix.github.io/ctxSip
 - ▶ JsSIP - jssip.net
 - ▶ SIPML5
 - ▶ SIP.js - sipjs.com
- ▶ Open source libraries
 - ▶ pjsip - pjsip.org
 - ▶ baresip - github.com/alfredh/baresip
 - ▶ libosip - antisip.com/doc/osip2



Kamailio SIP Server Project
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THANK YOU!

Hope to see some of you at Kamailio World 2018!
www.kamailioworld.com

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