2018

BUILD YOUR OWN RTC SERVICE

VOIP - MESSAGING - PRESENCE

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CO-FOUNDER KAMAILIO PROJECT

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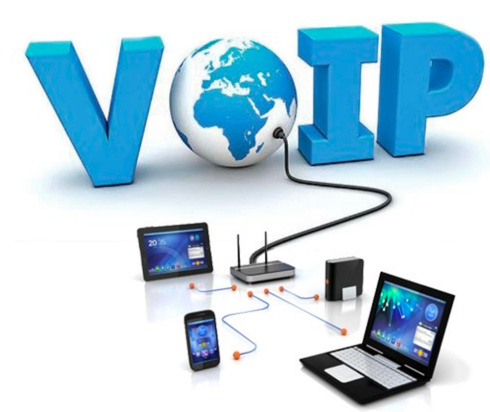
OPEN SOURCE AND KAMAILIO SIP SERVER PROJECT

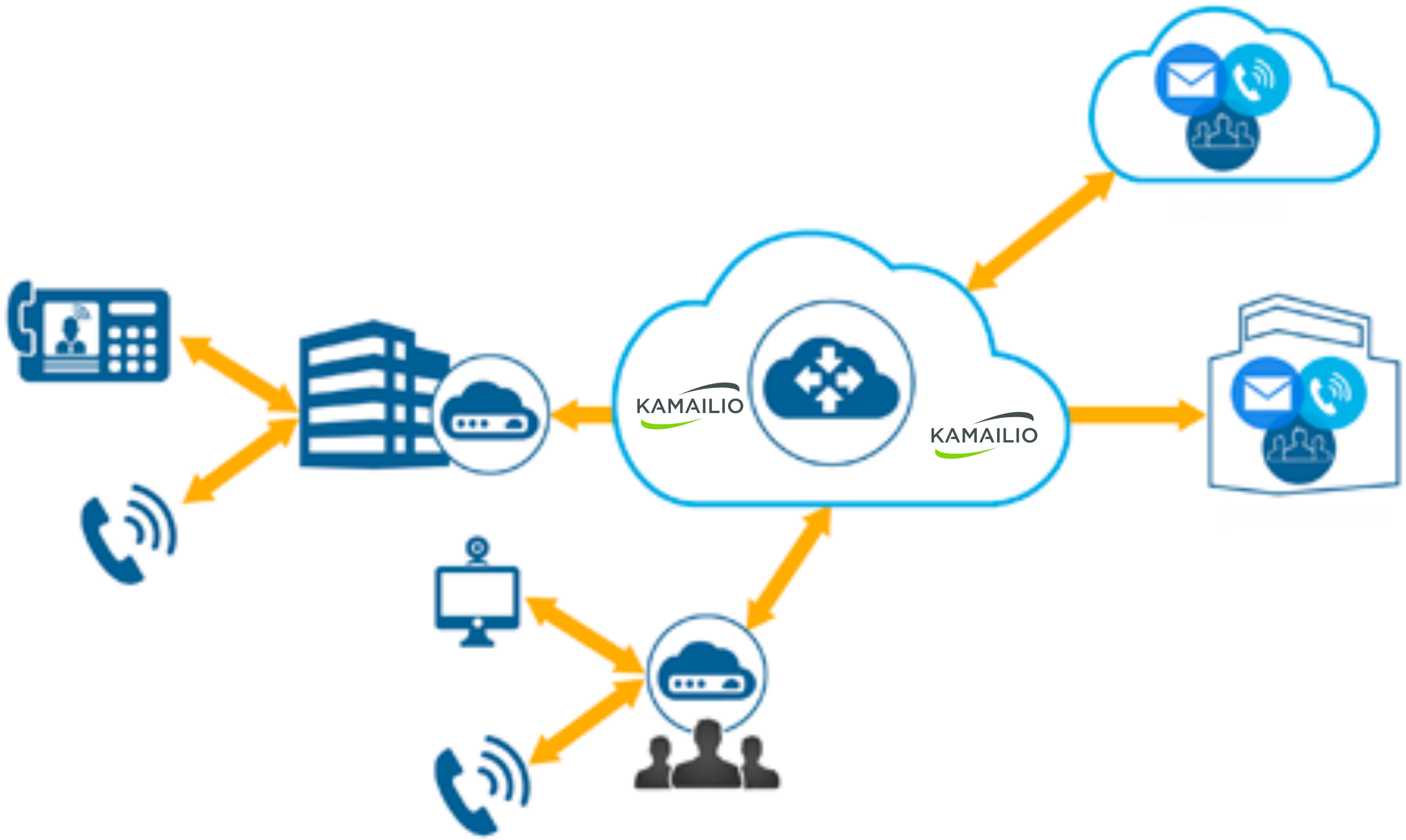
- Originally from Romania, living in Berlin, Germany
- Computer science software engineer
- Involved in open source real time communications since 2002
- Shifted from a researcher position to professional consultancy for SIP, Kamailio and all RTC
- Developing and living only from open source software 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

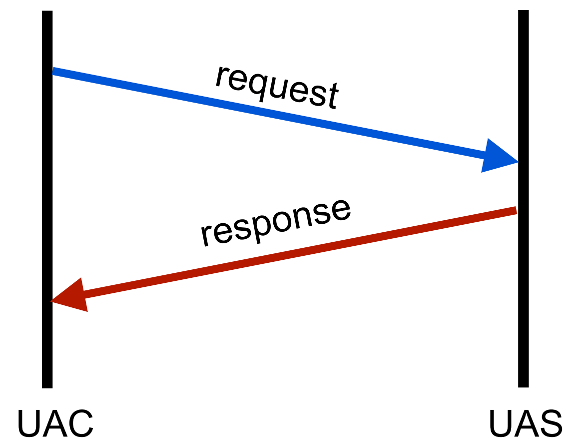
- something similar to skype, facetime, whatsapp, wechat, viber, …
- using open source
- using open standards
- run it for yourself or your community
  - or start a telephony business
    - residential or carrier services
  - straightforward integration with classic telephony/mobile networks
  - most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

CONNECT IN REALTIME



INTERACTION PROTOCOL

OPEN STANDARD – IETF RFC3261 (+ EXTENSIONS)



request

Start line

```
INVITE sip:user@sipserver.com SIP/2.0
```

Message headers

```
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
```

Message body

```
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

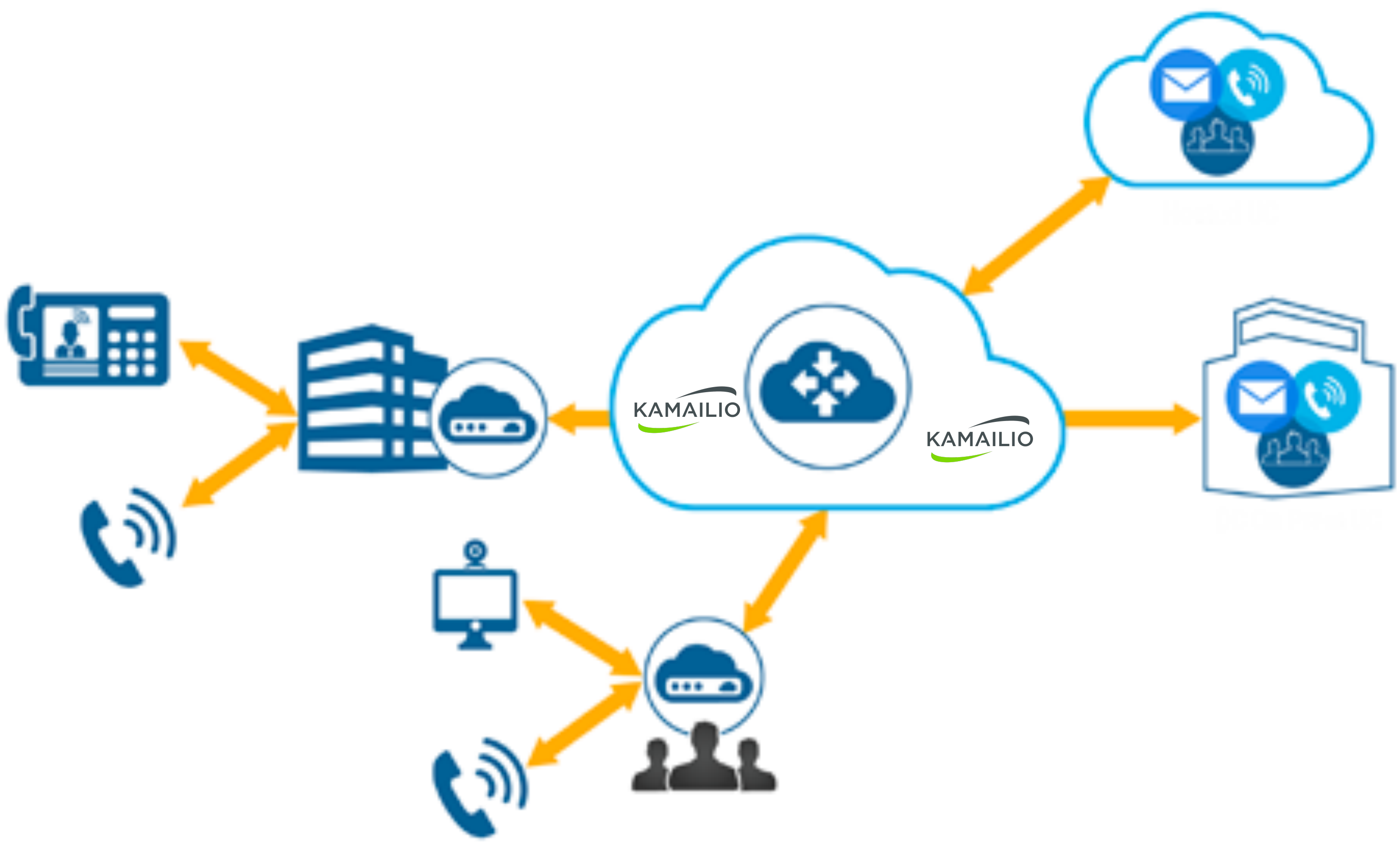
```
SIP/2.0 200 OK
```

```
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
```

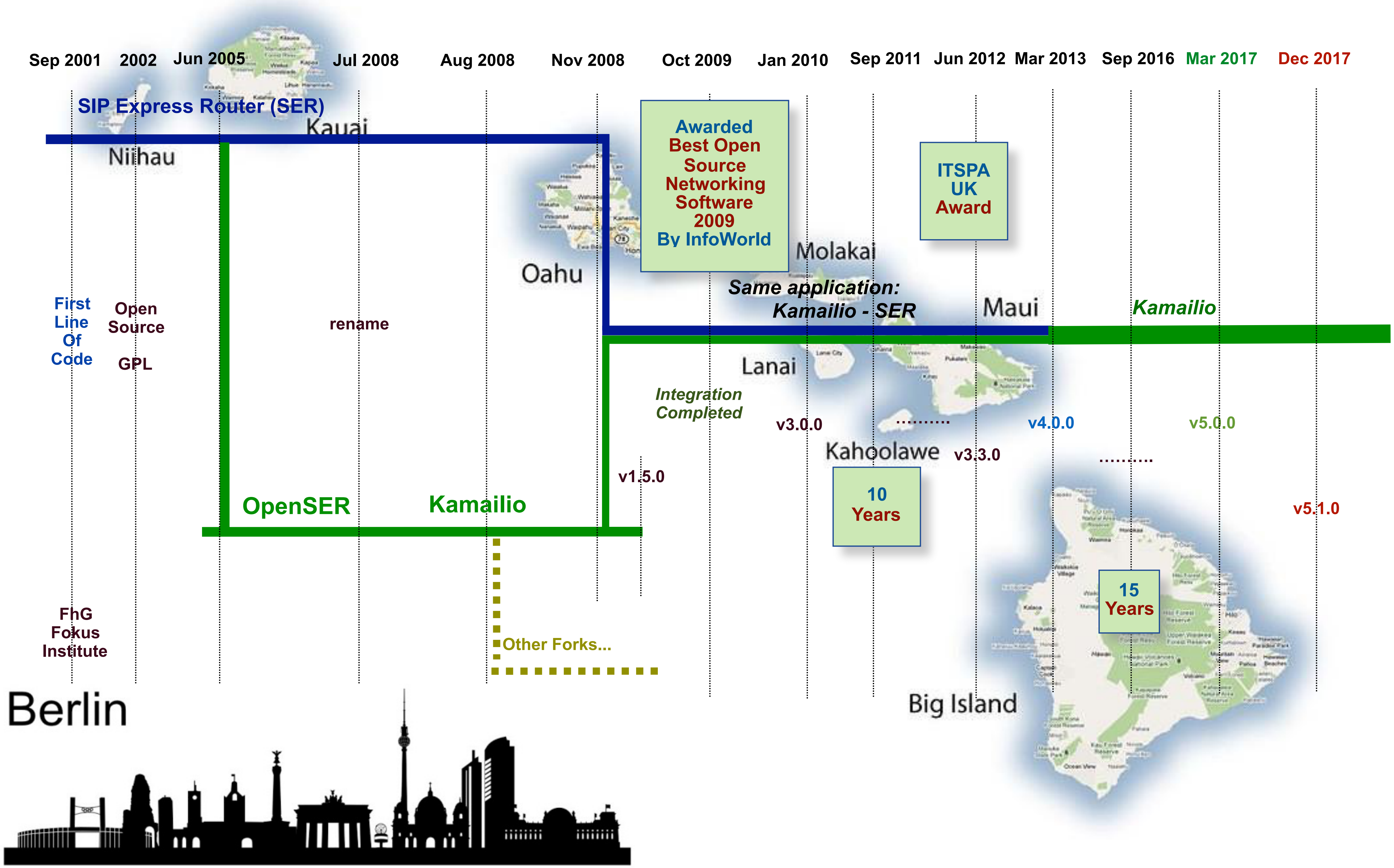
- telephony over ip
- VoIP and VoLTE (4G & 5G)
- voice and video
- instant messaging, presence

SERVER APPLICATIONS - CLIENT DEVICES & APPLICATIONS



SERVER APPLICATIONS

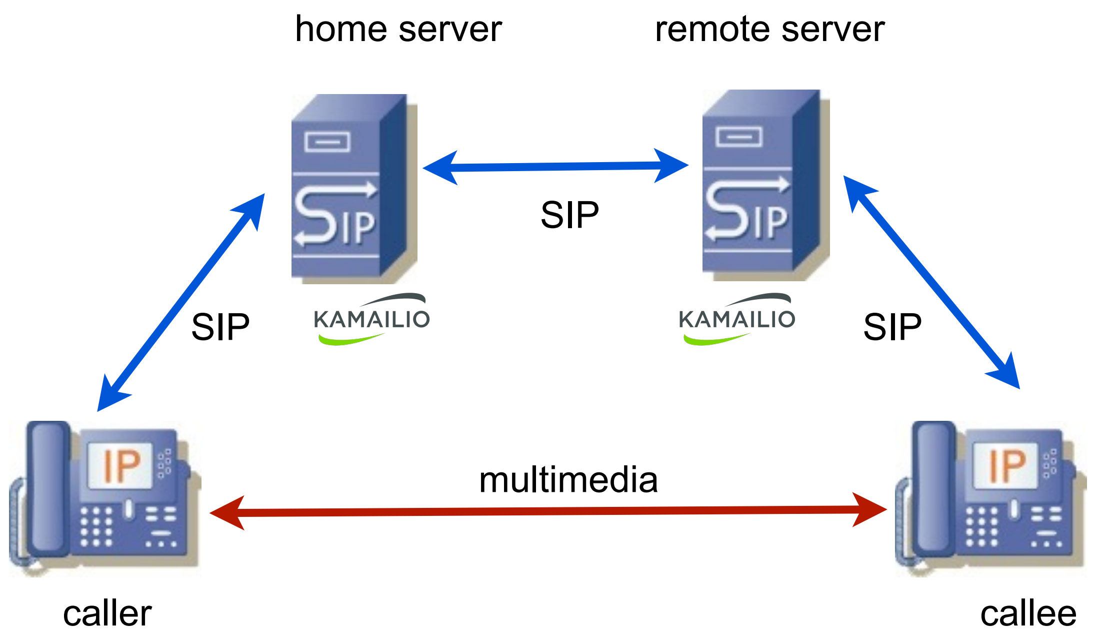
LET'S SPEAK SIP = E KAMA'ILIO SIP

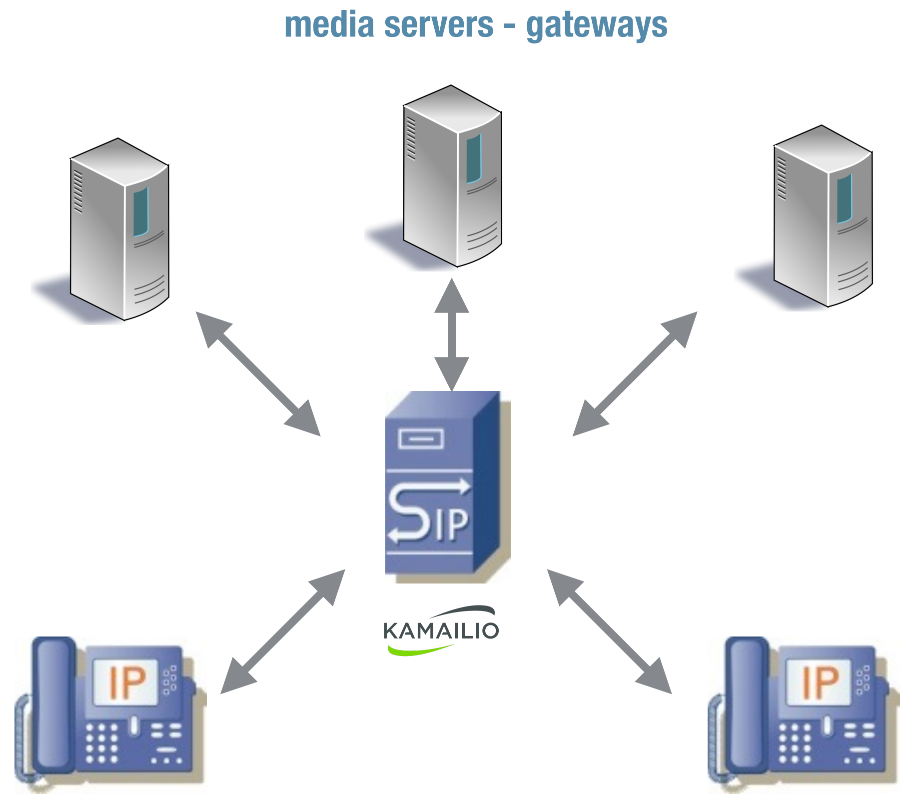


KAMAILIO SIP SERVER IN ONE SLIDE

- 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

KAMAILIO

- Deals only with SIP singaling 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

Asterisk 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

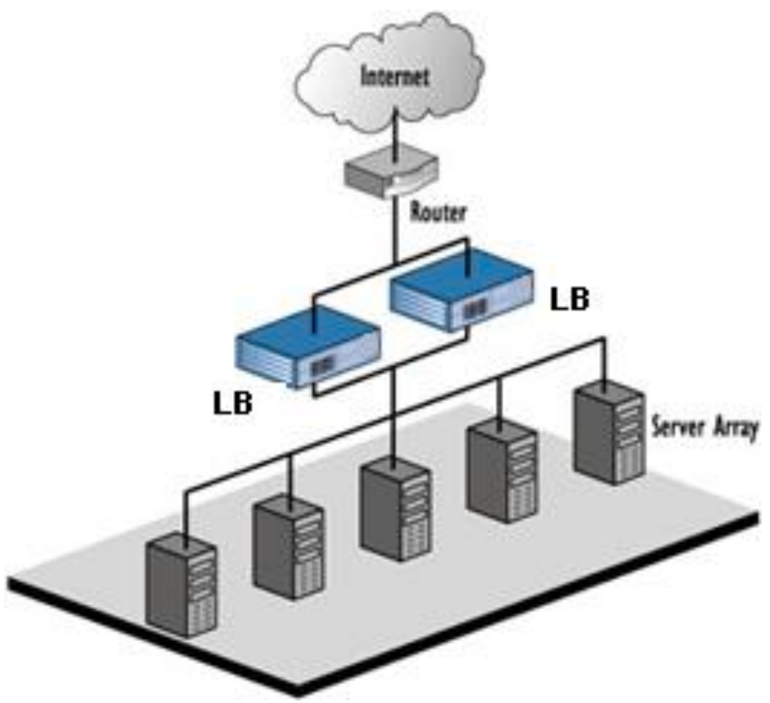
```
loadmodule “dispatcher.so"

# ----- 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:

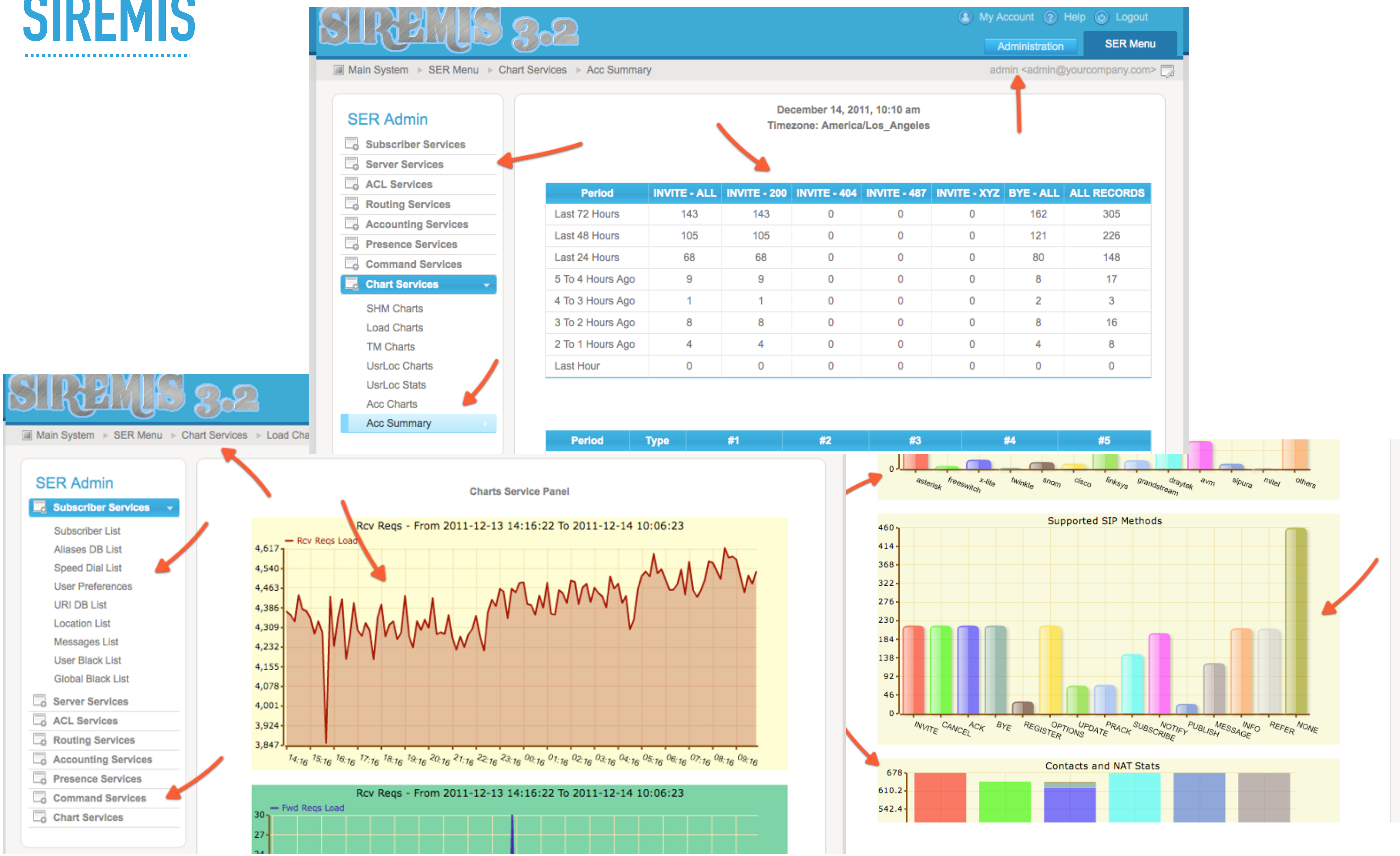
```
systemctl start kamailio
```

Configure sip phones/apps and start talking.

http://kamailio.org/docs/tutorials/devel/kamailio-install-guide-deb/

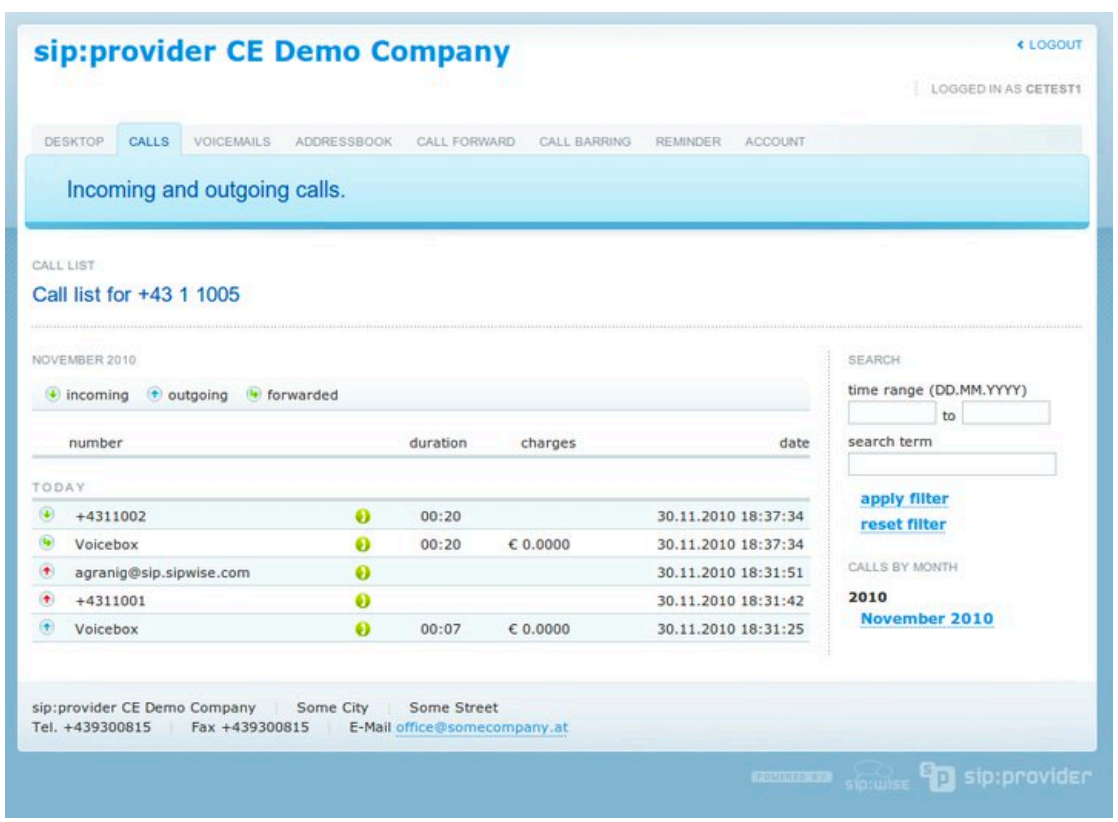
SIREMIS

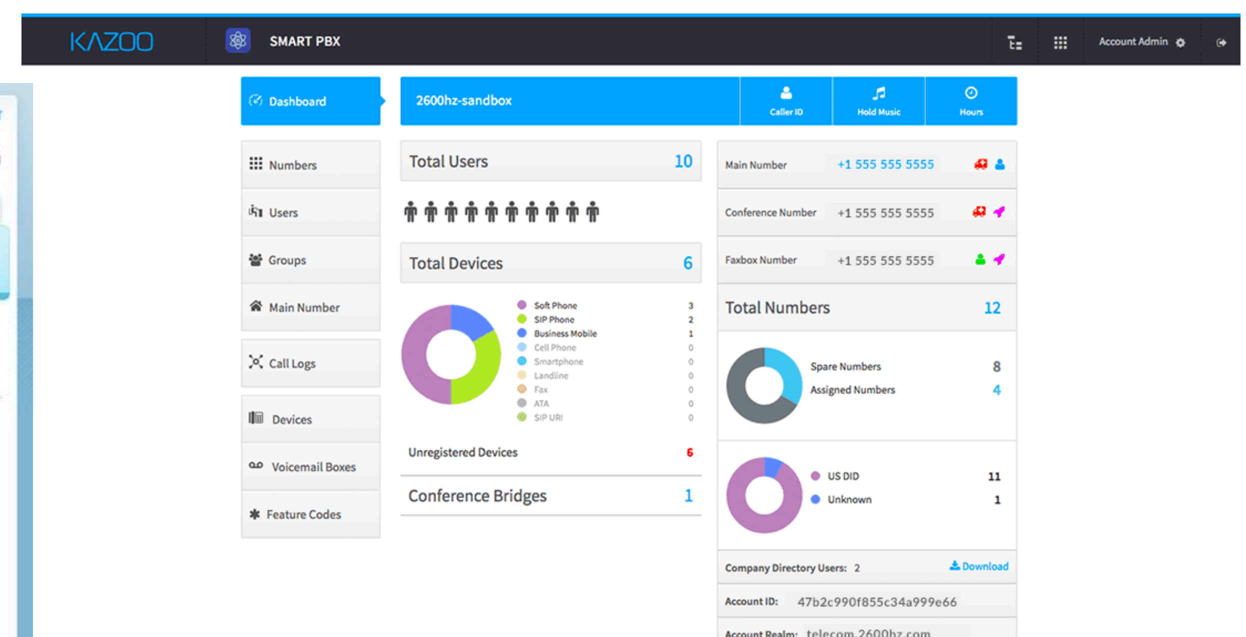
https://www.siremis.org - https://github.com/asipto/siremis



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





KAMAILIO DEPLOYMENTS

1&1

2nd telecom in Germany
over 10M phone numbers

sipgate

cloud telephony service

2600Hz

KAZOO cloud pbx

Libon

ott service orange™

sip:wise

ip telephony operators

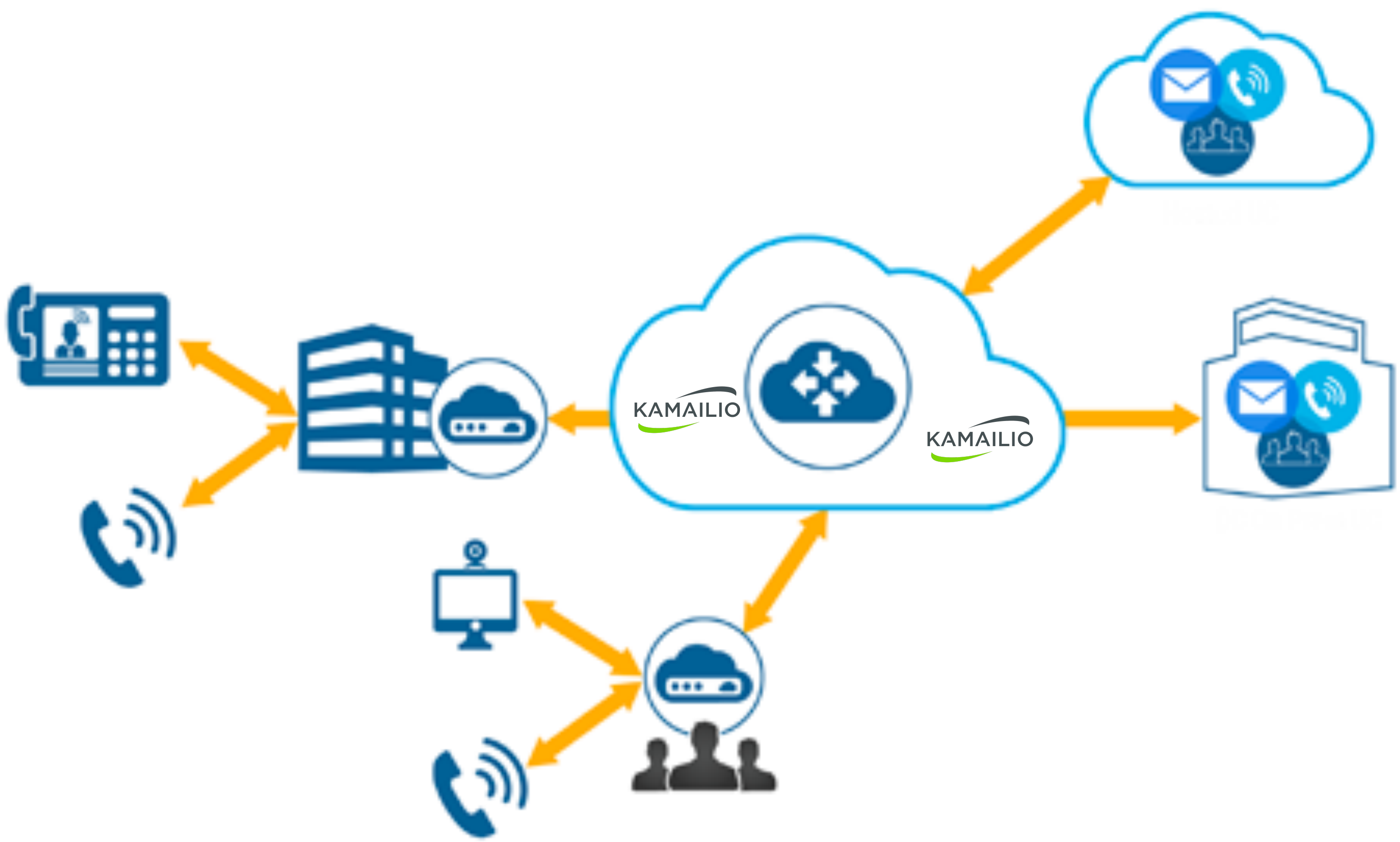
plivo

telephony
api platform

ngvoice

ims - volte

SERVER APPLICATIONS - CLIENT DEVICES & APPLICATIONS



CLIENT DEVICES & APPLICATIONS

DESK PHONE FEELING

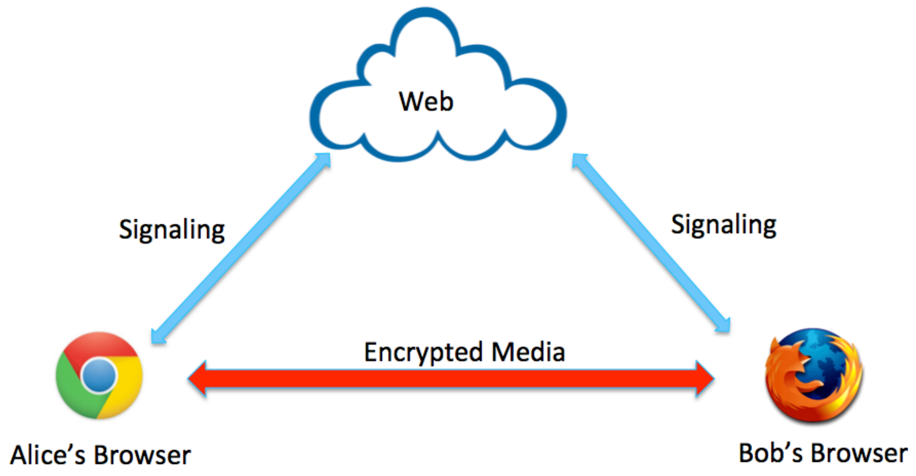
- Cisco
- Polycom
- Yealink
- Grandstream
- Snom
- Mitel
- Panasonic

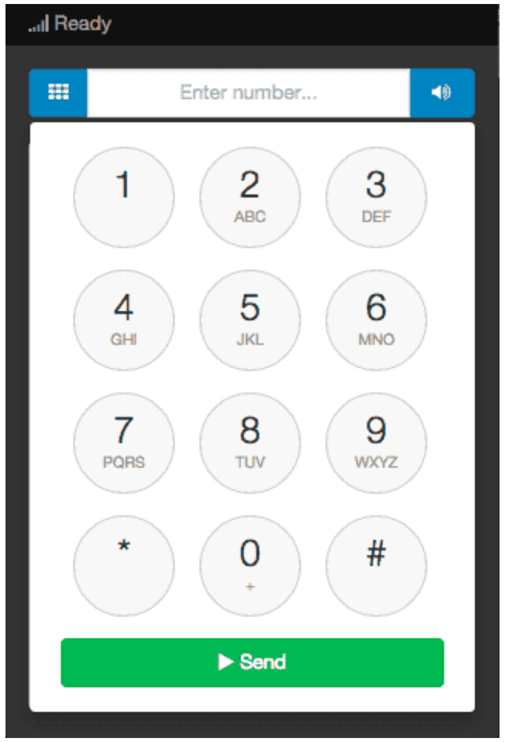
MOBILE, DESKTOP AND WEB APPS

- 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 - collecttix.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

KAMAILIO WORLD
CONFERENCE & EXHIBITION
BERLIN GERMANY MAY 14-16, 2018

foss asia