Kamailio with FreeSWITCH B2BUA

ClueCon 2023

Fred Posner • qxork.com

Agenda

Lets talk about the talk…

- Who am I?
- What is Kamailio?
- (You should know what FreeSWITCH is, this is ClueCon)
- Kamailio TOPOH/TOPOS and Scaling
- Kamailio with FreeSWITCH B2BUA
- Questions

Hi. I’m Fred.

Fred Posner

qxork.com

- VoIP Consultant
- Member of Kamailio Project
- Created APIBAN
- Proud Papa
- Based in Florida

What is Kamailio?

Kama what?

Kamailio is a hawaiian word. Kama’ilio means talk, to converse.

Kamailio

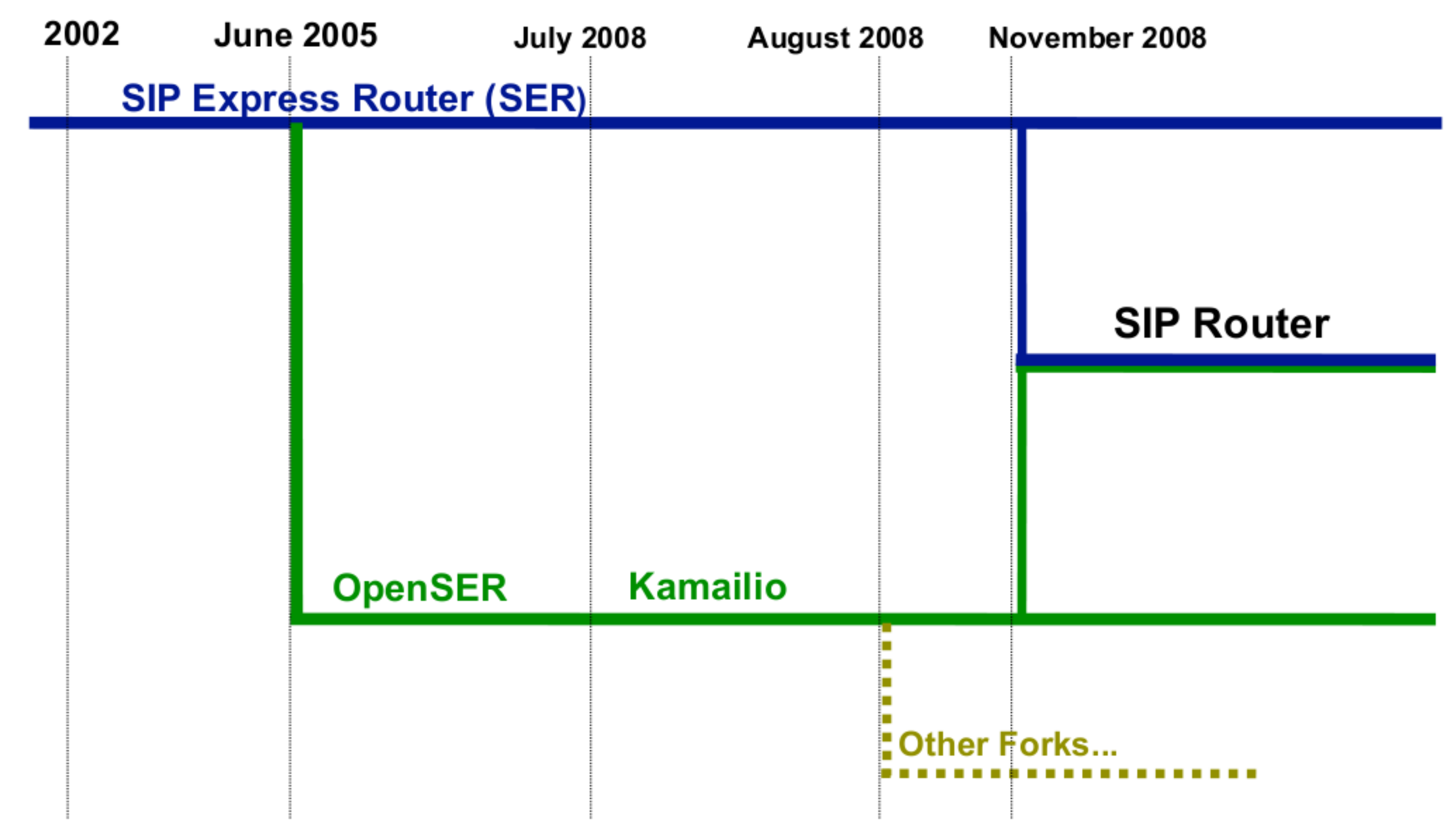
kamailio.org

- Open Source SIP Server
- Module based
- SIP Proxy
- SIP Registrar
- SIP Application Server
- SIP Dispatcher/Load Balancer
- SIP Websocket Server

Brief History

Continuous Development since 2001

- 2001: FhG FOKUS Project
- 2002: SER project released GPL
- 2005: OpenSER branched
- 2008: OpenSER -> Kamailio
- 2008: Other Forks
- 2008: SER + Kamailio = SIP Router
- 2010: Kamailio 3.0 (with full SER)
- Now: Kamailio 5.7.x



5000+

Calls/Sec as Load Balancer in stateless mode

300,000+

Registrations with just 4GB RAM

150+

Modules

What is Kamailio?
Typical Use Cases

- SIP Edge Router / “SBC”
- Load Balancer
- LCR
- WebRTC / TLS / TCP Bridge

What isn't Kamailio?

- Kamailio does not handle Media
  - There is RTPENGINE…
    and Media Servers (like FreeSWITCH)
  - Also a couple modules
- Kamailio <> B2BUA
  - TOPOS module
  - TOPOH module

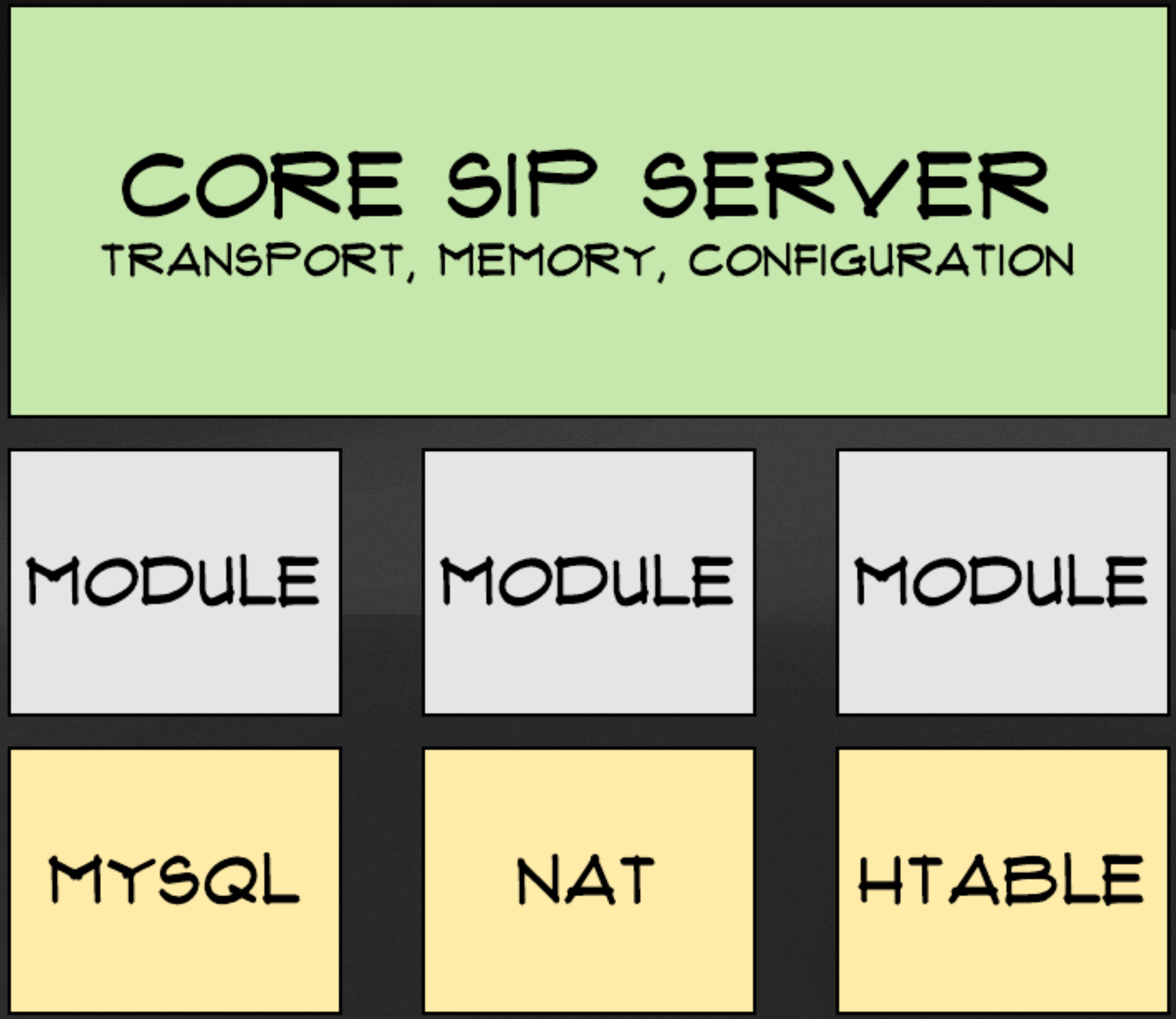
Kamailio TOPOH/TOPOS (and Scaling)


FreeSWITCH
FreeSWITCH
WEST
Kamailio
SQL
Public
Redis
EAST
Kamailio
FreeSWITCH
FreeSWITCH

Scaling

Kamailio Built to Scale

- High CPS, Low CPU
- HTABLE
- DMQ
- Load Balancer
- KEMI
- APIs, N+1



TOPOH

Topology Hiding

- Hides SIP Routing Headers
- Stateless or Stateful
- No external dependencies

TOPOS

Topology Stripping

- Strips Topology Headers
- Needs Database
- Very clean

RTPENGINE
SIPWISE

- Richard Fuchs is here at ClueCon!
- Powerful RTP proxy, runs in kernel
- Can use multiple nodes
- Failover
- 4000+/node
- DTLS/SRTP Bridge / Transcoding

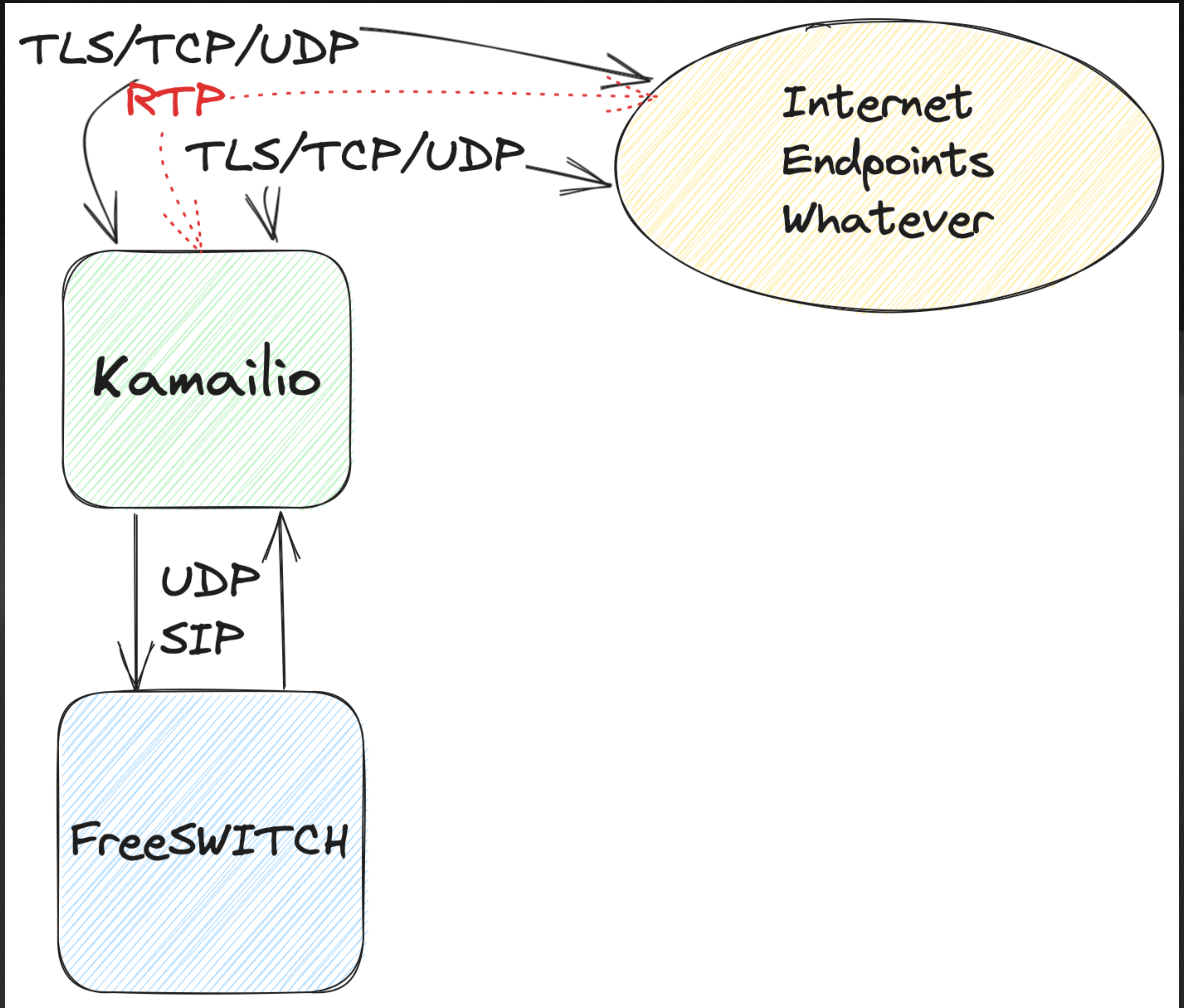
Kamailio + FreeSWITCH B2BUA

Basic Concept

Using FreeSWITCH as B2BUA

- No media
- New Call-ID
- Stripped down modules
- Configs: GitHub.com/fredposner

  https://github.com/fredposner/cluecon2023



```
<!-- b2bua -->
<extension name="b2bua">
  <condition field="destination_number" expression="^b2bua-(.+)$">
    <action application="log" data="INFO entered b2bua extension"/>
    <action application="set" data="sendto=${sip_via_host}:5080"/>
    <action application="set" data="bypass_media=true"/>
    <action application="set" data="sip_copy_custom_headers=false"/>
    <action application="bridge" data="[sip_h_X-PGPX=${uuid}]sofia/
internal/$1@${sendto}"/>
    <action application="hangup"/>
  </condition>
</extension>
```

```
2023/08/14 12:35:02.696520 192.168.8.3:5060 -> 192.168.25.86:5060
INVITE sip:8597211046@192.168.25.86:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.8.3:5060
From: "15555551212" <sip:15555551212@192.168.8.3:5060>;tag=1
To: "8597211046" <sip:8597211046@192.168.25.86:5060>
Call-ID: 1-318583@192.168.8.3
Cseq: 1 INVITE
Contact: sip:sipp@192.168.8.3:5060
Content-Type: application/sdp
Content-Length:   245

v=0
o=user1 53655765 2353687637 IN IP4 192.168.8.3
s=PGPX.IO Media Test
t=0 0
c=IN IP4 192.168.8.3
m=audio 6000 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=maxptime:20
a=sendrecv
```

SIP to Kamailio

```
2023/08/14 12:35:02.700891 192.168.25.86:5080 -> 192.168.25.87:5060
INVITE sip:b2bua-8597211046@192.168.25.87 SIP/2.0
Max-Forwards: 10
Record-Route: <sip:192.168.25.86:5080;r2=on;lr=on;ftag=1>
Record-Route: <sip:192.168.25.86;r2=on;lr=on;ftag=1>
Via: SIP/2.0/UDP 192.168.25.86:5080;branch=z9hG4bK4701.a79db2cb537b7805013899b335e40cef.0
Via: SIP/2.0/UDP 192.168.8.3:5060;rport=5060
From: "15555551212" <sip:15555551212@192.168.8.3:5060>;tag=1
To: "8597211046" <sip:8597211046@192.168.25.86:5060>
Call-ID: 1-318583@192.168.8.3
Cseq: 1 INVITE
Contact: <sip:sipp@192.168.8.3:5060;alias=192.168.8.3~5060~1>
Content-Type: application/sdp
Content-Length:   245

v=0
o=user1 53655765 2353687637 IN IP4 192.168.8.3
s=PGPX.IO Media Test
t=0 0
c=IN IP4 192.168.8.3
m=audio 6000 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=maxptime:20
a=sendrecv
```

SIP to FS

```
2023/08/14 12:35:02.771134 192.168.25.87:5060 -> 192.168.25.86:5080
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.25.86:5080;branch=z9hG4bK4701.a79db2cb537b7805013899b335e40cef.0
Via: SIP/2.0/UDP 192.168.8.3:5060;rport=5060
Record-Route: <sip:192.168.25.86:5080;r2=on;lr=on;ftag=1>
Record-Route: <sip:192.168.25.86;r2=on;lr=on;ftag=1>
From: "15555551212" <sip:15555551212@192.168.8.3:5060>;tag=1
To: "8597211046" <sip:8597211046@192.168.25.86:5060>;tag=NN08X74gD4t1Q
Call-ID: 1-318583@192.168.8.3
CSeq: 1 INVITE
Contact: <sip:8597211046@192.168.25.87:5060;transport=udp>
User-Agent: FreeSWITCH-B2BUA
Accept: application/sdp
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY
Supported: timer, path, replaces
Allow-Events: talk, hold, conference, refer
Content-Type: application/sdp
Content-Disposition: session
Content-Length: 257
Remote-Party-ID: "+18597211046" <sip:+18597211046@192.168.25.86>;party=calling;privacy=off;screen=no

v=0
o=FreeSWITCH 1692011120 1692011121 IN IP4 192.168.25.61
s=FreeSWITCH
c=IN IP4 192.168.25.61
t=0 0
m=audio 28480 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtcp:28481 IN IP4 192.168.25.61
a=ptime:20
```

Answer from FS

```
2023/08/14 12:35:02.710570 192.168.25.87:5060 -> 192.168.25.86:5080
INVITE sip:8597211046@192.168.25.86:5080 SIP/2.0
Via: SIP/2.0/UDP 192.168.25.87;rport;branch=z9hG4bKNFea9yH0cFeBH
Max-Forwards: 9
From: "15555551212" <sip:15555551212@192.168.25.87>;tag=pyS1Z2NmaDHmK
To: <sip:8597211046@192.168.25.86:5080>
Call-ID: 813970a0-b57c-123c-958b-000c293a91c9
CSeq: 71467059 INVITE
Contact: <sip:mod_sofia@192.168.25.87:5060>
User-Agent: FreeSWITCH-B2BUA
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY
Supported: timer, path, replaces
Allow-Events: talk, hold, conference, refer
Content-Type: application/sdp
Content-Disposition: session
Content-Length: 233
X-PGPX: b41b9c3d-cc44-4f16-80d1-f386d965fab4
X-FS-Support: update_display,send_info
Remote-Party-ID: "15555551212" <sip:15555551212@192.168.25.87>;party=calling;screen=yes;privacy=off

v=0
o=user1 53655765 2353687637 IN IP4 192.168.8.3
s=PGPX.IO Media Test
c=IN IP4 192.168.8.3
t=0 0
m=audio 6000 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=maxptime:20
```

FS B-Leg INVITE

Why?

Performance and More

- uuid_ options
  kill, media, record, etc
- Change of Call-ID*
- Highly Performative
- 1000s calls per box (no media)
- High CPS

Questions?

Questions?

qxork.com

- Kamailio World 2024!
- kamailio.org
  - Matrix
  - Mailing List
- Daniel’s talk
- Kamailio Lunch and Learn