



VOLTS

VoIP Open Linear Test Suite

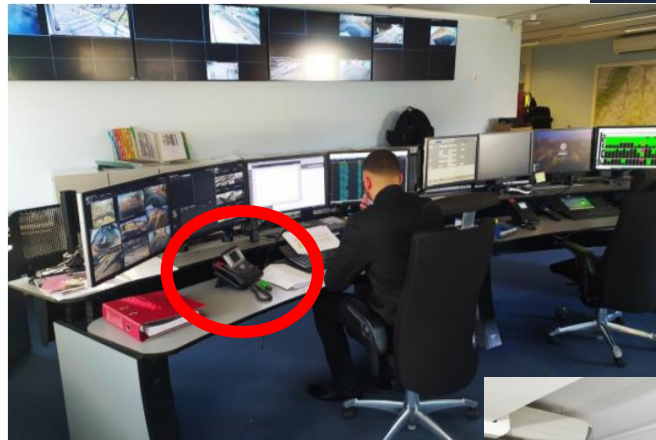
Ihor Olkhovskyi

IT-CS-TR

Telephony in CERN

- Yes, we're still using this "old ways" (cause it works, actually)

- CERNphone
- Call Centres
- Control rooms
- Personal use



Users Portal

List of all the phone numbers you can configure
You can choose to configure or revoke one of the phone numbers listed below

Available phone numbers Go to the CERNphone Resources Portal to create, reassign and delete your numbers

Phone number	Description	Start Date	Responsible	Type	Action
65825		22-10-2020	Ihor OLKHOVSKYI	PERSONAL	Settings Register mobile app
		06-09-2021	Ihor OLKHOVSKYI	SHARED	Settings Register mobile app
		07-09-2021	Ihor OLKHOVSKYI	SHARED	Settings Register mobile app
		23-06-2021	Ihor OLKHOVSKYI	SHARED	Settings Register mobile app
		23-06-2021	Ihor OLKHOVSKYI	SHARED	Settings Register mobile app
		17-09-2021	Ihor OLKHOVSKYI	SHARED	Settings Register mobile app
		07-06-2022	Ihor OLKHOVSKYI	SHARED	Settings Register mobile app
		08-04-2020		SHARED	Settings Register mobile app
		08-01-2021	Ihor OLKHOVSKYI	SHARED	Settings Register mobile app



Based on open source



Edge Proxy

Kamailio



Routing / Application server

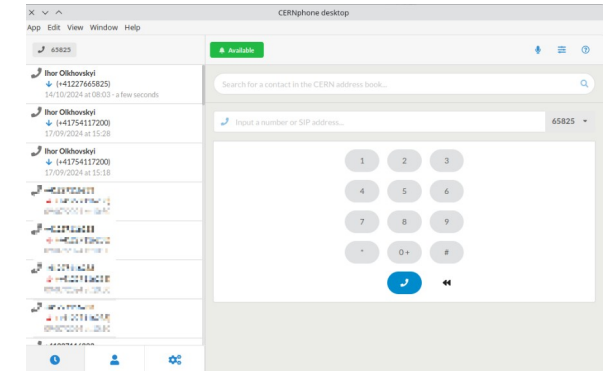
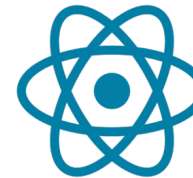
Asterisk

* also to mention MariaDB, Python, GNU/Linux, etc



Mobile client

Linphone



Desktop client

React, Electron, SIP.js...

How telephony system tests looks like?



My desk. Part of it (how it was)



Automated testing?

- No “single approach“ or “industry standard tool” for a testing process
- There are a lot of small CLI tools around to “emulate VoIP device”
- Not what you want?
 - here is SDK. Use C++/Lua/Whatever

The image shows two overlapping terminal windows. The top window is a terminal for 'baresip-0.4.3' showing its help menu and execution output. The bottom window is 'ocadmin@marigot:~/SIPP/Sipp-0.3-src.new' displaying a detailed call scenario report.

```
marius@marius-desktop: ~/Desktop/baresip-0.4.3
-d Daemon
-e <commands> Exec commands
-f <path> Config path
-h -? Help

marius@marius-desktop:~/Desktop/baresip-0.4.3$ baresip softpedia.com
baresip v0.4.3 Copyright (C) 2010 - 2013 Alfred E. Heggstad <aeh@db.org>
Local network address: IPv4=eth1:10.10.0.236
audec: PCMA 8000Hz 1ch
audec: PCMU 8000Hz 1ch
```

```
ocadmin@marigot:~/SIPP/Sipp-0.3-src.new
File Edit Settings Help

----- Scenario Screen ----- [1-4]: Change Screen --
Call-rate(length) Port Total-time Total-calls Remote-host
500 cps(0 ms) 5060 20.02 s 10014 192.168.200.175:5060(UDP)

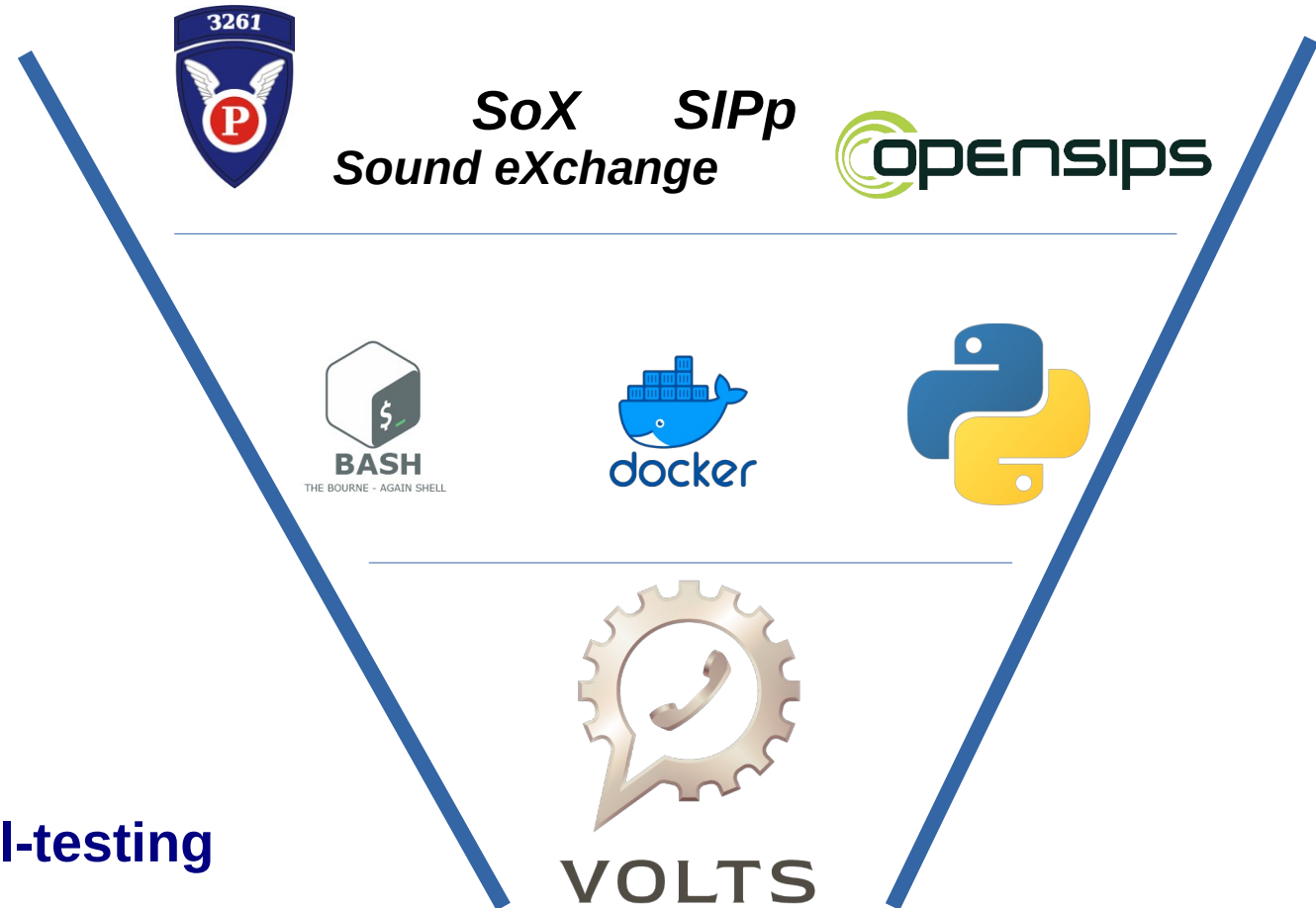
2505 new calls during 5.010 s period 1 ms scheduler resolution
12 concurrent calls (limit 1500) Peak was 15 calls, after 0 s
0 out-of-call msg (discarded)
1 open sockets

Messages Retrans Timeout Unexpected-Msg
INVITE -----> 10014 1 0 0
100 <----- 0 0 0 0
180 <----- 10014 0 0 0
200 <----- StopRTD 10014 0 0 0
ACK -----> 10014 0 0 0
[ 0 ms]
BYE -----> 10014 0 0 0
200 <----- 10002 0 0 0

----- [ + | - | * | / ] : Adjust rate ---- [ q ] : Soft exit ---- [ p ] : Pause traffic -----
```

VOLTS – VoIP Open Linear Test Suite

- Just a BASH/Python glue around other open source components
- Run multiple scenarios at once
- Templates - based
- Test-driven-development
- GPLv2



<https://gitlab.cern.ch/cernphone/functional-testing>
<https://github.com/igorolhovskiy/volts>

Q&A?



home.cern