OpenSIPS

Building multi-tenant VoIP platform

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Stas Kobzar, Modulis.ca Inc., Developer

Adrien Laurent, Modulis.ca Inc., Founder and CEO

Very beginning

ÉTS
Le génie pour l'industrie

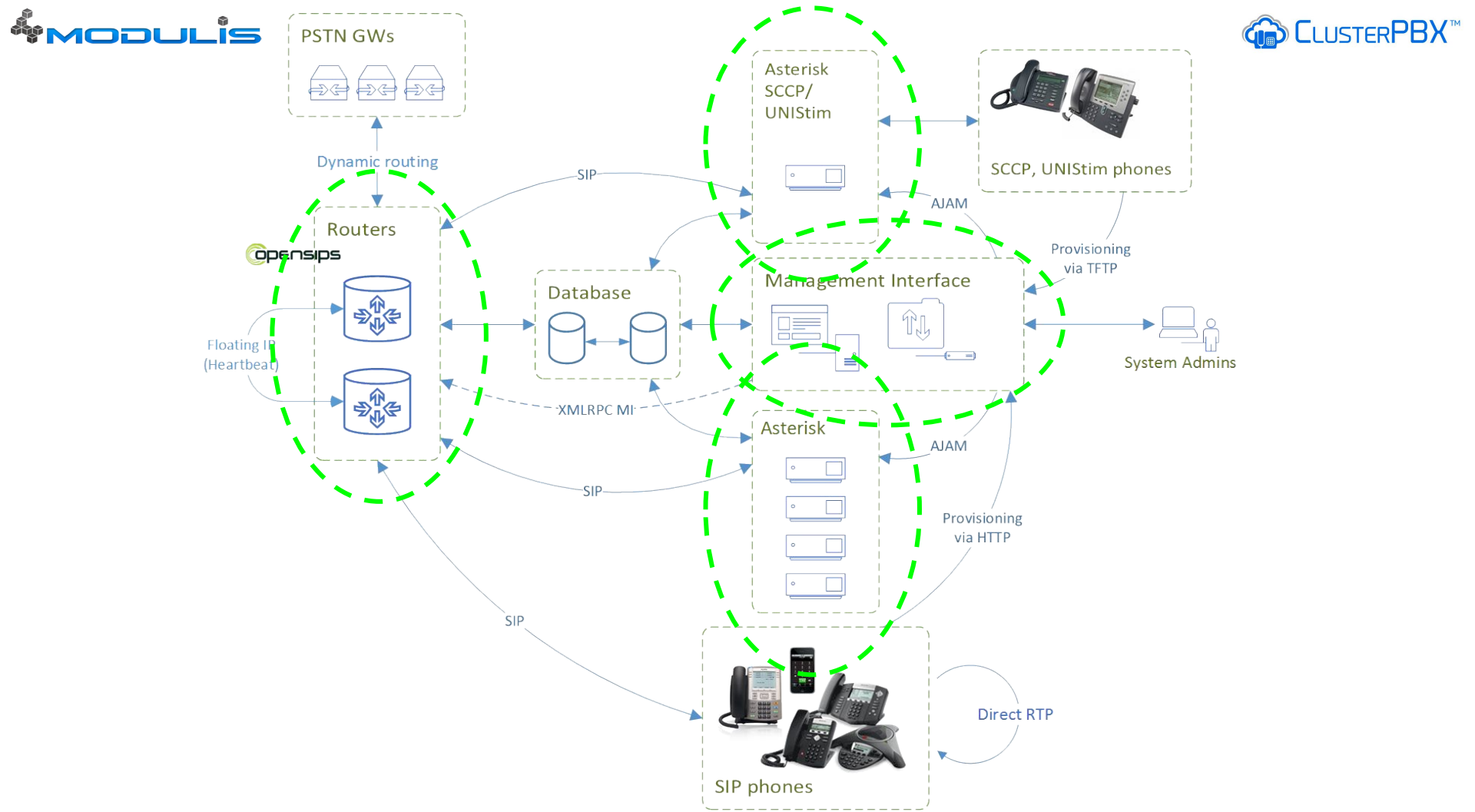
- Started at 2013 with ÉTS
- Replace old Nortel VoIP PBX CS1000
- More than 1500 Avaya phones
- Requirements:
  - Open Source
  - Scalability
  - Virtualization
  - Multi-level management (admins, users)
  - Gradual migration process

Challenges

- Three protocols: SIP, UNIStim, SCCP
- Unified management
- PBX features
- No single points of failure

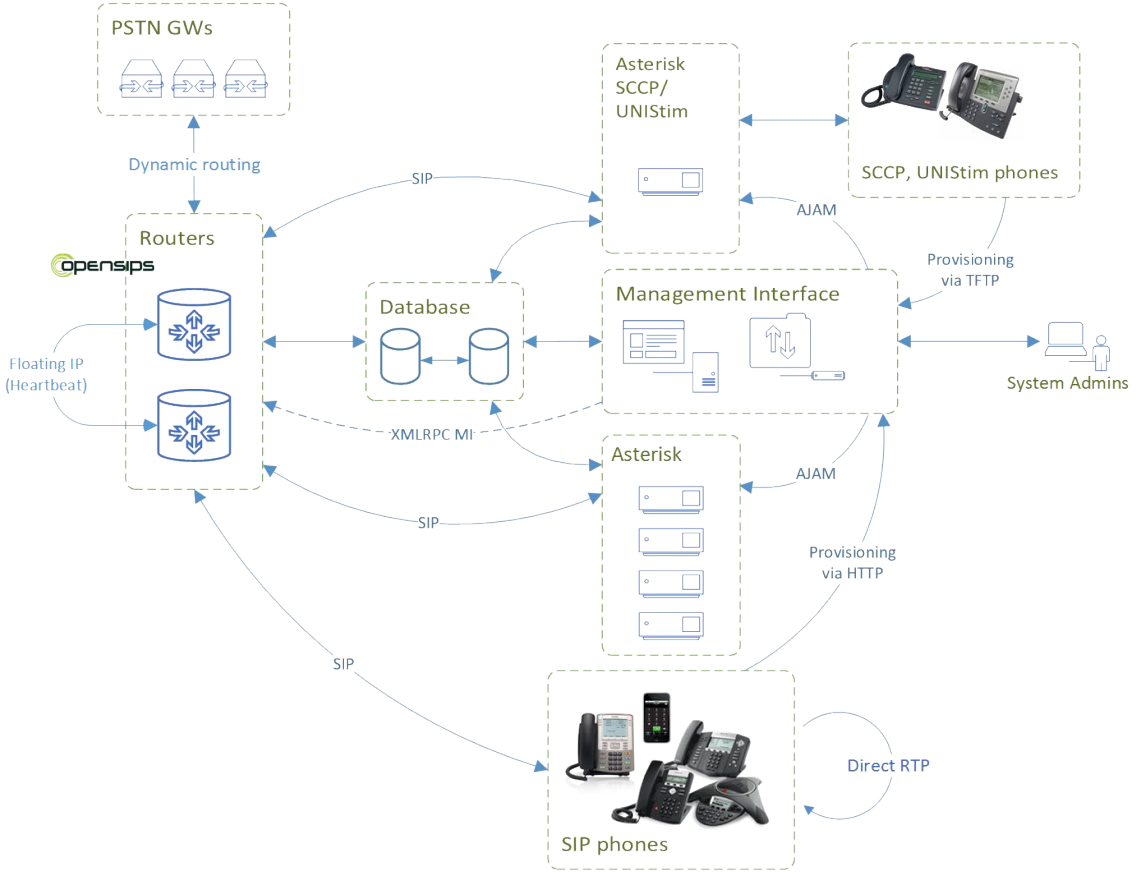
Platform design

- OpenSIPS
- Asterisk PBX features
- Asterisk SCCP/UNIStim gateway
- Admin web-interface (Rails + ExtJS)
- REST API
- Provisioning
- Monitoring and alerts (monit)


MODULIS
CLUSTERPBX™
PSTN GWs
Dynamic routing
Routers
opensips
Floating IP
(Heartbeat)
SIP
Asterisk
SCCP/
UNIStim
SCCP, UNIStim phones
AJAM
Provisioning
via TFTP
Database
Management Interface
System Admins
XMLRPC MI
Asterisk
AJAM
SIP
Provisioning
via HTTP
SIP
Direct RTP
SIP phones

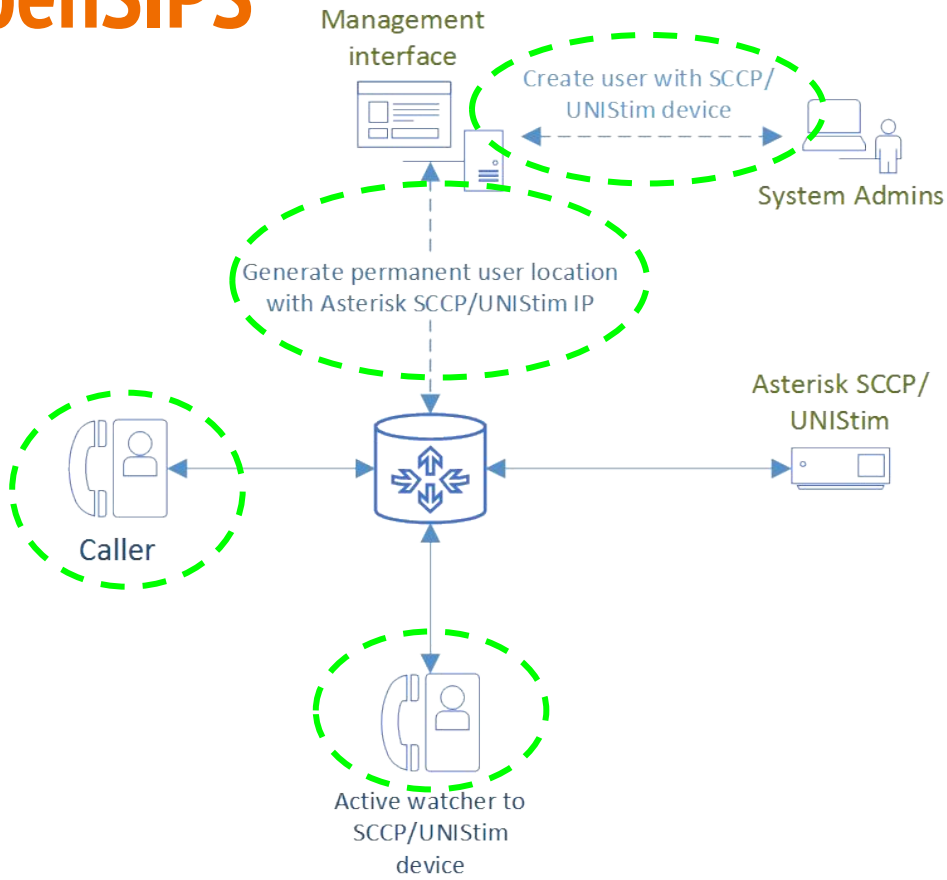
OpenSIPS roles

- Registrar
- Proxy
- Load balance
- Presence server
- PSTN SIP trunking
- Calls Profiling
- Exceptions handling
- Security



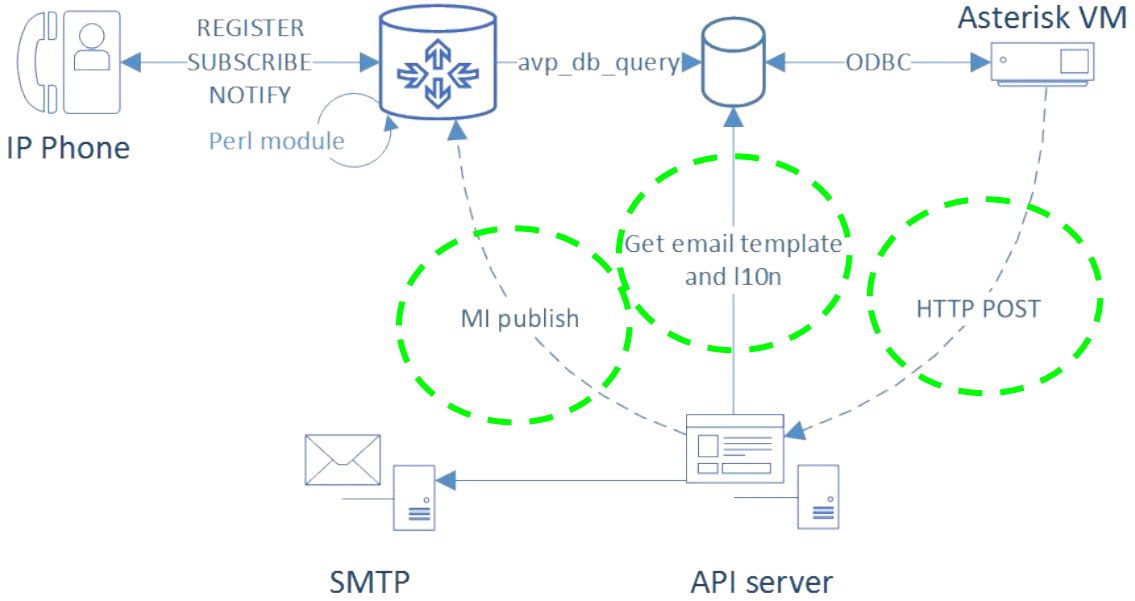
None-SIP devices via OpenSIPS

- Management
- OpenSIPS proxy
- Sharing presence state

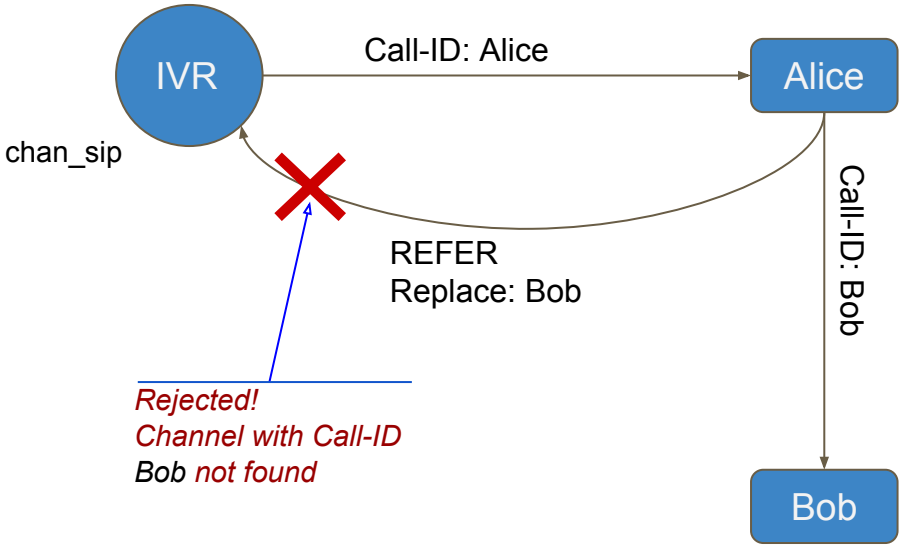


Presence

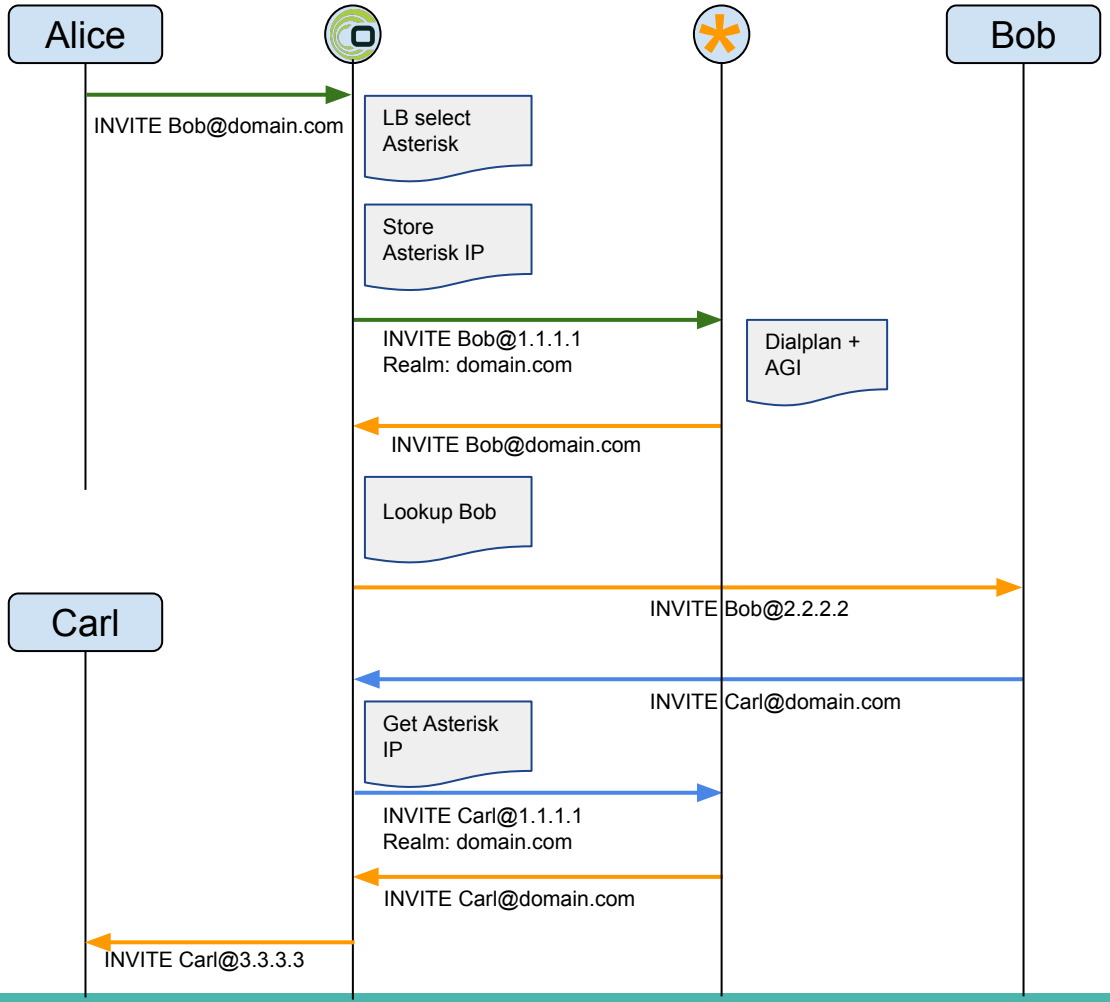
- RLS and xcap for Avaya phones
- Call pickup
- MWI (Message Waiting Indicator)



Remote attended x-fer and Asterisk challenge



Call flow


Alice
Bob
Carl
INVITE Bob@domain.com
LB select Asterisk
Store Asterisk IP
INVITE Bob@1.1.1.1
Realm: domain.com
Dialplan + AGI
INVITE Bob@domain.com
Lookup Bob
INVITE Bob@2.2.2.2
INVITE Carl@domain.com
Get Asterisk IP
INVITE Carl@1.1.1.1
Realm: domain.com
INVITE Carl@domain.com
INVITE Carl@3.3.3.3

Challenge : MySQL

- MySQL cluster
- MySQL Master/Master simultaneous write.
- MySQL Master/Master active/passive

Testing with SIPP

- Call flows (register, call, presence)
- Stress tests.
- OpenSIPS benchmark

```
---------------------------- Scenario Screen -------- [1-9]: Change Screen --
  Call-rate(length)   Port   Total-time  Total-calls  Remote-host
2.0(10000 ms)/1.000s   5060     572.27 s         1144  10.130.8.20:5060(UDP)

  2 new calls during 1.004 s period      1 ms scheduler resolution
  24 calls (limit 60)                    Peak was 28 calls, after 20 s
  0 Running, 90 Paused, 9 Woken up
  11 dead call msg (discarded)           353 out-of-call msg (discarded)
  3 open sockets

                             Messages  Retrans   Timeout   Unexpected-Msg
     INVITE ---------->         1144      0         0
        100 <----------         1122      0         0         0
        407 <----------         1142      0         0         0
     INVITE ---------->         1142      0         0
        100 <----------         1164      0         0         0
        180 <----------         924       0         0         0
        183 <----------         0         0         0         0
        200 <----------  E-RTD1 1141      0         0         0
        ACK ---------->         1141      0
     INVITE <----------         1125      0         0         15

        200 ---------->         1125      0         0
        ACK <----------  E-RTD1 1125      0         0         0

      Pause [    10.0s]         1125                           1
        BYE ---------->         1104      0         0
        200 <----------         1104      0         0         0

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

```
---------------------------- Scenario Screen -------- [1-9]: Change Screen --
 Port   Total-time  Total-calls  Transport
 5060     608.01 s         1159  UDP

 2 new calls during 1.001 s period      1 ms scheduler resolution
 28 calls                               Peak was 34 calls, after 44 s
 0 Running, 92 Paused, 7 Woken up
 8 dead call msg (discarded)
 4 open sockets

                                Messages  Retrans   Timeout   Unexpected-Msg
 ----------> INVITE             1159      0         0         0

 <---------- 180                1159      0
 <---------- 200                1159      0         0
 ----------> ACK         E-RTD1 1158      0         0         1

 ----------> INVITE             1158      0         0         0

 <---------- 200                1158      0         0
 ----------> ACK         E-RTD1 1158      0         0         0

 ----------> INVITE             1138      0         0         0

 <---------- 200                1138      0         0
 ----------> ACK         E-RTD1 1138      0         0         0

 ----------> BYE                1138      0         0         0
 <---------- 200                1138      0
 [   4000ms] Pause              1138                           0
---------------------------- Sipp Server Mode -----------------------------
```

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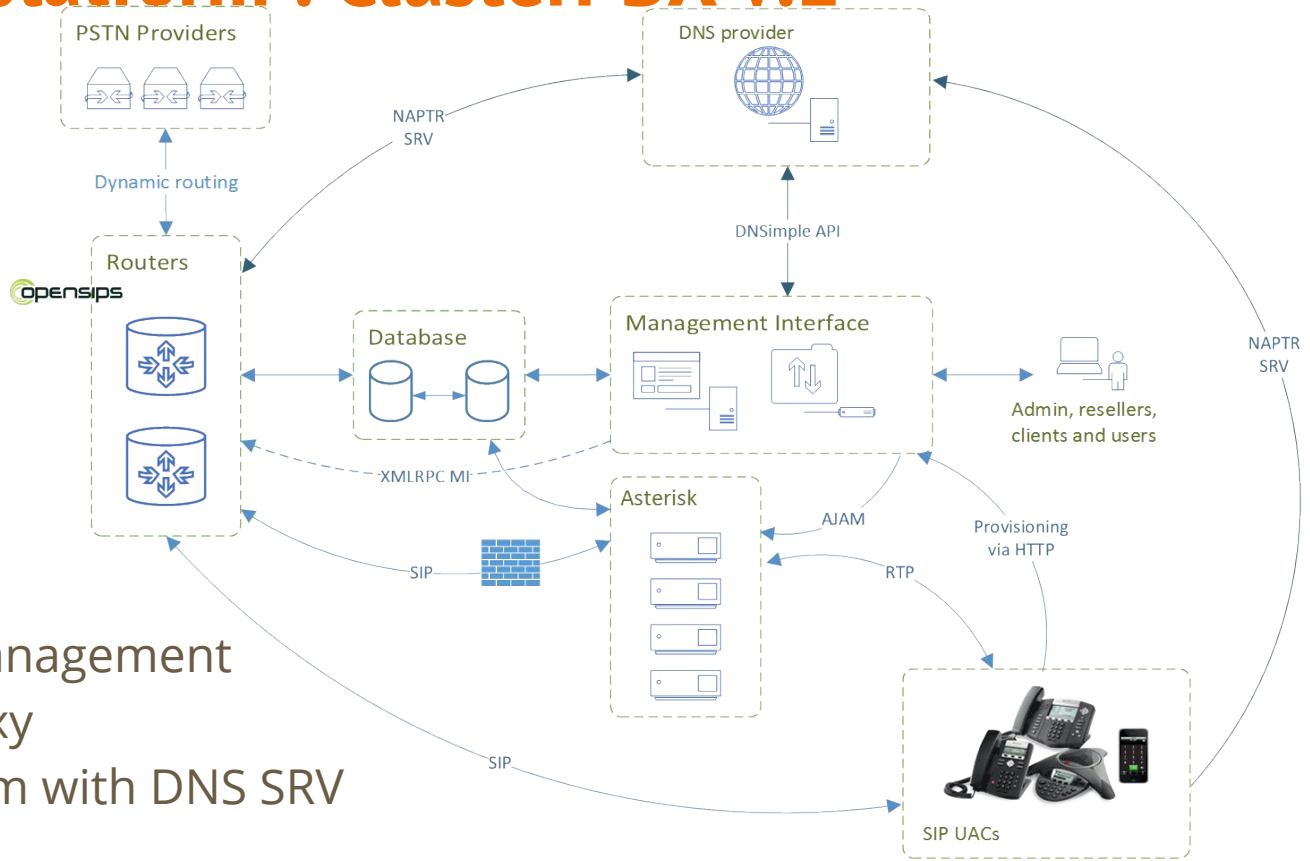
CLUSTERPBX™

ClusterPBX v2

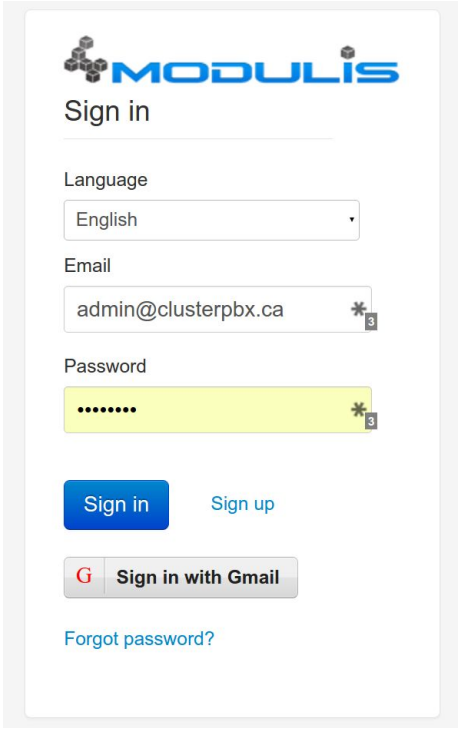
Multi-tenant platform : ClusterPBX v.2



- DNS NAPTR/SRV management
- Asterisk as RTP proxy
- Asterisk has problem with DNS SRV

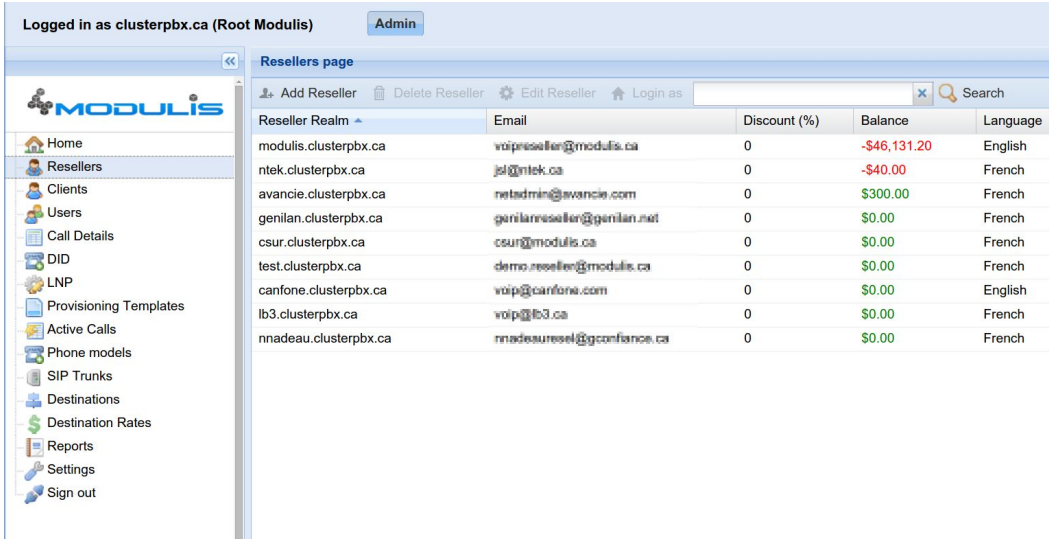
Management

- Multilingual
- OpenSIPS MI and Asterisk AJAM
- Multi-tenant
- Admin -> resellers -> clients -> users
- Domains client.reseller.tld



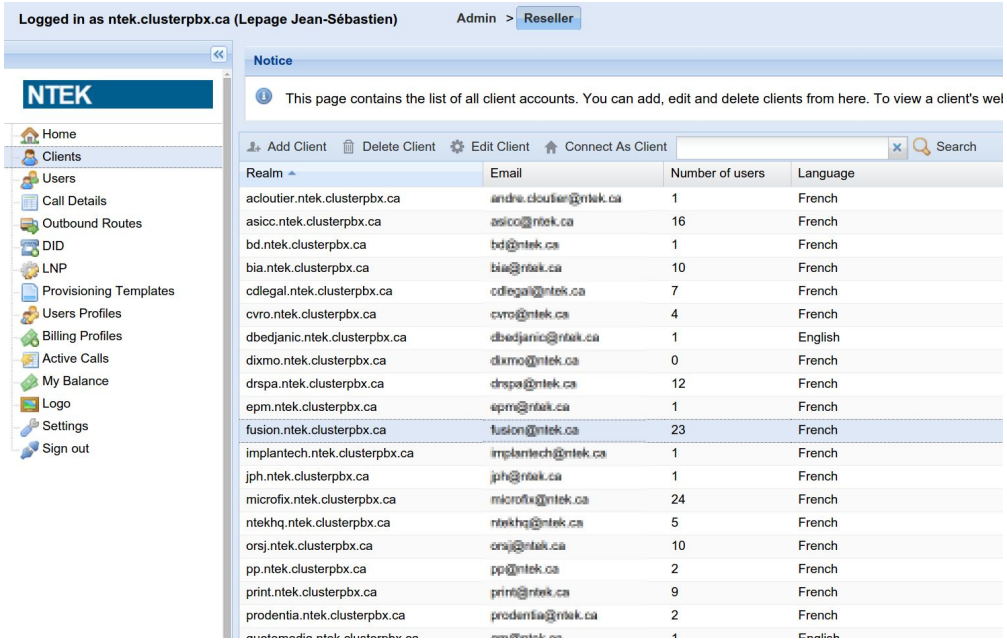
Top level : admin

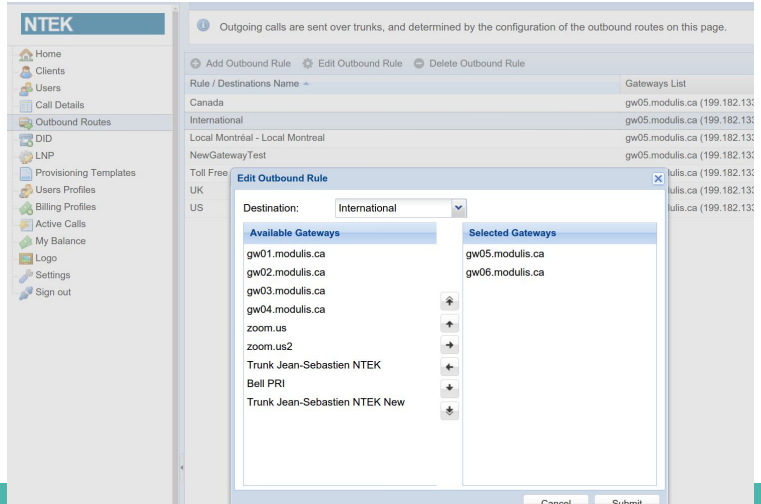
- Access to any sub level
- DID management
- Provisioning templates
- SIP trunks
- Routing rules



Resellers level

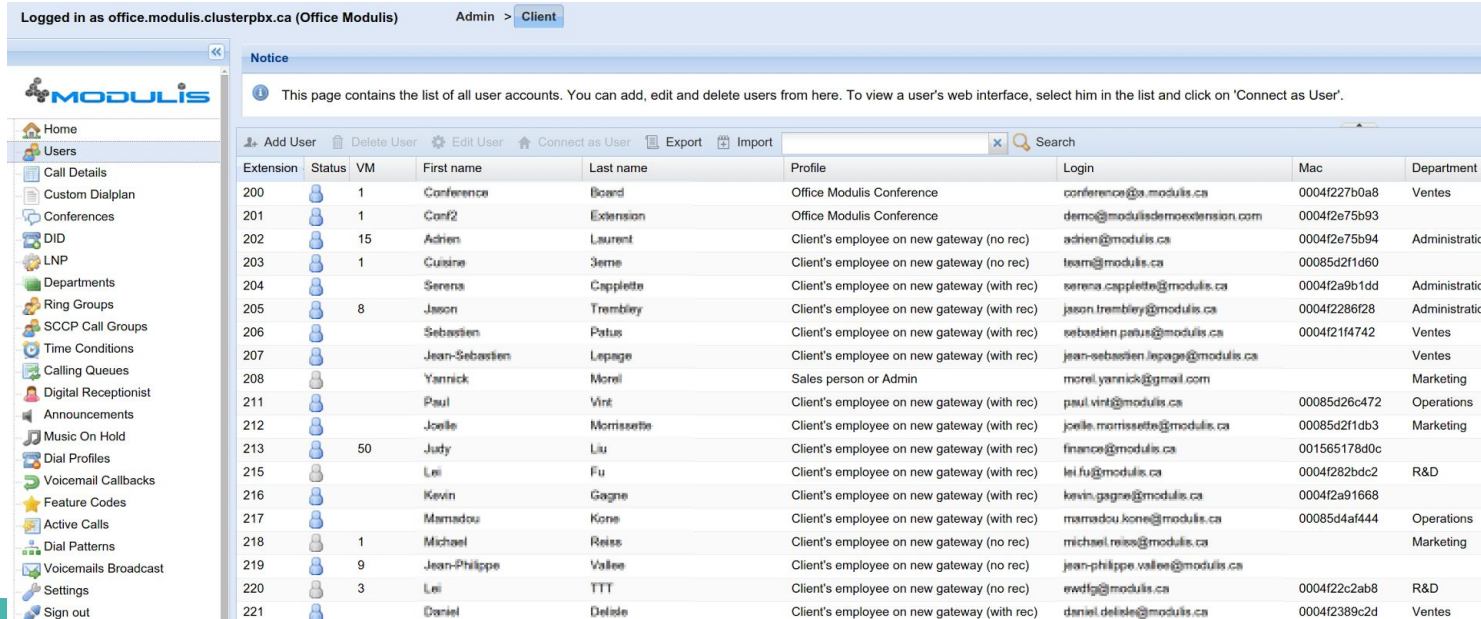
- Sub-level management
- Profiling
- Routing
- Branding



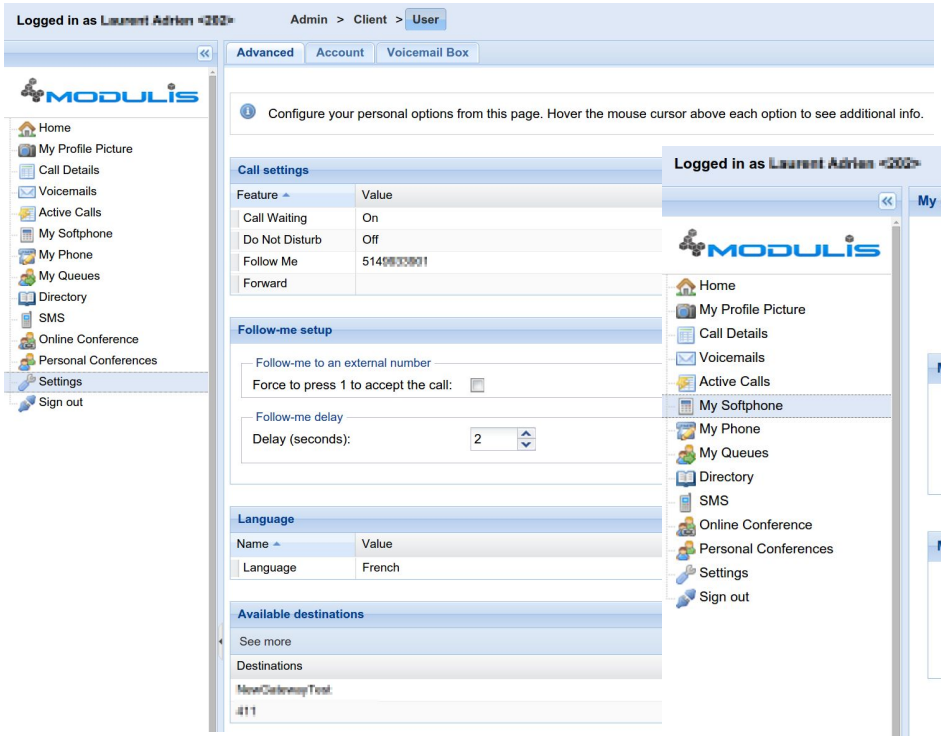


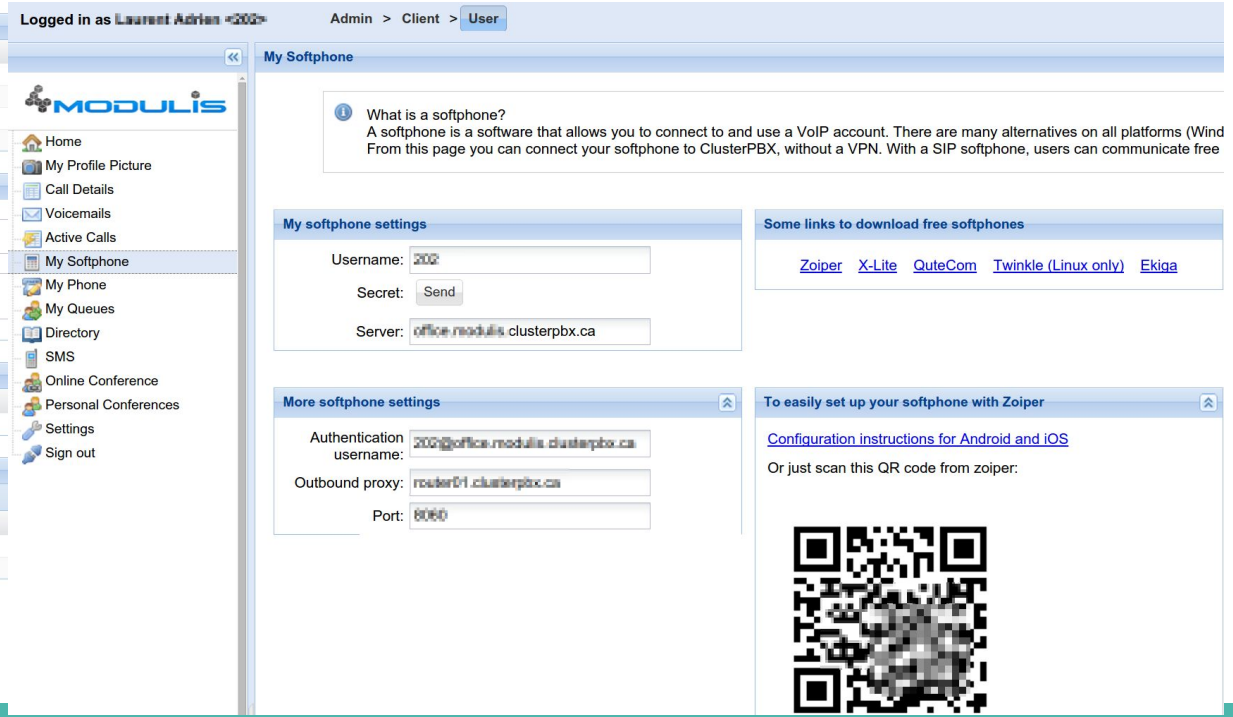
Client level

- Users management
- PBX features (conferences, ring groups, MoH etc.)
- DIDs assign
- Reporting
- IVRs
- Dialplan



Multi-tenant : User





Ruby gems

- opensips-mi (https://github.com/modulis/opensips-mi)
- asterisk-ajam (https://github.com/modulis/asterisk-ajam)

Plans

- Migration to OpenSIPS v2
- Locations and dialogs binary replications
- RabbitMQ multiple OpenSIPS servers
- Investigating Percona

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Thank you!

Questions?

Stas Kobzar
Adrien Laurent
info@modulis.ca