Building SIP platforms that scales into the future

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Where are we?

A status report.
Open Source VoIP is commonplace
...and nothing new, really.

Many years of guerilla work.

Open Communication with SIP wins.

No one invests in any other VoIP protocol any more.
Everyone talks about Unified Communication

But who knows what it is and where the wind is blowing...

Current servers have enough CPU power.

A standard HP DL380 server can handle 10,000 concurrent calls. That's more than enough.
The network is the computer. Finally.

Thin clients and network computing is renamed to "CLOUD COMPUTING"

Telcos are moving down the stack.

The old dinosaurs are fighting to stay in service control.
Case studies

Open Unified Communication is reality. Today.

OUC for callcenters

- Customer with existing call center software adopting it to Asterisk
- From 50 to 500 agents
- From 50 to 1,500 concurrent calls
- Immediate failover if server or application crashes
- Delivered as a service (cloud) or as an application
...for the public sector

- Universities in Portugal and Norway
- One seat 15.000 phones
- Portugal installation - 500 servers, 100.000 phone lines.
- Scalable SIP networks with Kamailio and Asterisk
- Interfacing to legacy PBXs over ISDN for migration
- Many additions contributed to FreePBX and Asterisk
- Portuguese project with partners IT Center & WaveCom

...for a regional body

- Replacing 15 pbxs with 5000 phones
- Schools, hospitals, dentists, daycare, offices
- Adding distributed presence to Asterisk (assisted by Kamailio)
- Project with Edvina partner TeleKompetanse in Oslo
...and for Service providers

- Kamailio in the core
- Asterisk for feature services and PSTN handling
- Scalable platforms for 1,000-1,000,000 phones
- Many users in this sector
- Many additions contributed to the Asterisk SIP channels over the years

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What’s pushing us forward?
#1: The Internet.

- The Internet is truly always available
- The networks are getting better
- IP Telephony over 3G works today
- 4G will get there in 10 years time. HD video anyone?

#2: Open Standards making it possible

- The TCP/IP protocol suite
- SIP
- XMPP
- Unicode
- Freely available codecs
#3: Open Source

- FreeSwitch, Asterisk, Kamailio, SIP-router, ejabberd, Linux, FreeBSD, Adhearsion
- The application stack is available without license fees (I won’t say for free)
- Quick adaption to your business possible, you have the source and an open market available
- **Innovation** is pushing these platforms forward

#4: Voice interfaces

- Speech recognition and synthesis
- Natural interfaces for services
- Price/channel has gone down
#5: Commodity hardware

- Standard PC servers can run 10,000 audio channels - over 5,000 calls. Today.
- Investment/call is going down

#6: API's

- Programmer interfaces to telephony are freely available
- Lowers cost of implementation
- Asterisk AGI/AMI, Adhearsion, FreeSwitch libraries, Kamailio MI, LUA and Python
- Service providers should run these
  - You don’t need your own PBX to add voice to your apps!
#7: Unified, federated identity and authorization

LDAP
OpenID
SAML 2.0
Oauth

#8: New codecs, built for IP networks

• OPUS!
9. Security requirements

- Sadly, too few customers require security
- We might have to follow Gemeinschaft 4 philosophy:
  *We know better even if the customers don’t*

So where are we going?

...today and tomorrow...
A new generation of customers

- Telephony for the old generation is replaced by IM & Presence on the cell phone for the new generation
- Powered by social networks, mobile apps and IM systems
- Multimedia on top of this is emerging

Presence and location will drive everything.

Cellphone location
Set-top-box
Bot providers
The Car Bot
Safety alarms
Answer-my-mother-service

SIP and XMPP are important parts of this puzzle
Without the Internet you are on an isolated island.

So how do we get away from it?

Why are we building with PSTN focus?

Isolated islands of SIP!
Put the Internet where it belongs

One Unified Communications Network, one cloud service

IPv6 is the new glue

No NAT. Enough network addresses to build anything you need. Go and have fun. Discover the new opportunities with IPv6!
The road ahead:
10 bullet points to remember!

1. Use Open Network Protocols

TCP/IP * SIP * XMPP
2. Open Source

COOPERATIVE SOFTWARE

3. Unification

One address that rule them all!
4. Integrity and Security

BUILDING TRUST FOR NEW SOLUTIONS

5. Climate Friendliness

VIDEO * SYSTEM MANAGEMENT
6. Social Responsibility
Building and connecting to a network for everyone.

OPEN SOURCE = AVAILABILITY FOR THE 3RD WORLD

7. Everything is and should be building blocks
EVERYTHING SHOULD HAVE AN OPEN API
8. Keep an open network

9. Underestimate yourself

You just don’t know about the future
10. Get a new mindset

The new telephony platform is not about telephony.

SIP evolution

Number of endpoints or pages of RFCs?
Where did we loose the grip?

- SIP is now equal to PSTN over IP.
- That wasn’t the idea, at least not for me.
- It’s rather boring.
- We need to move beyond PSTN over IP.

Remember?

- freeworlddialup.com (dead)
- iptel.org free service (still running)
- IAXtel (dead)
Do we have enough power to change?

- Can we break the one site-syndrom?
- Can we build an open and secure federation?
- Let's discuss this in the breaks and tonight.

Where is the IETF

- Seems like the inventors of SIP are now working with RTCweb, trying to fix their errors
- SIP seems to be dominated by PSTN people
- We need to do something about this, the gap between implementations and IETFs work is far too wide.
Kamailio’s role in my world

- Kamailio - sip-router - ser - operates the base for the platform I am describing here
- Me and my partners have over 600 proxys running in enterprises and public sector
- For me, it’s been rock solid with no security issues. (Quite boring from a hacking point of view :-(
- It fits right in - security, IPv6, presence, multimedia calls and more
- Thank you, Team SIP-router!

...And please remember at least two things from this presentation:
IPv6 needs to be on your agenda.

Cloud services not running on IPv6 will not be fully reachable from anywhere and anytime if it's not connected to the IPv6 Internet.

OUC != pstn-over-IP
Open Unified Communication
Connecting Internet Users in real time.