
SIPTEL
FIRENZE

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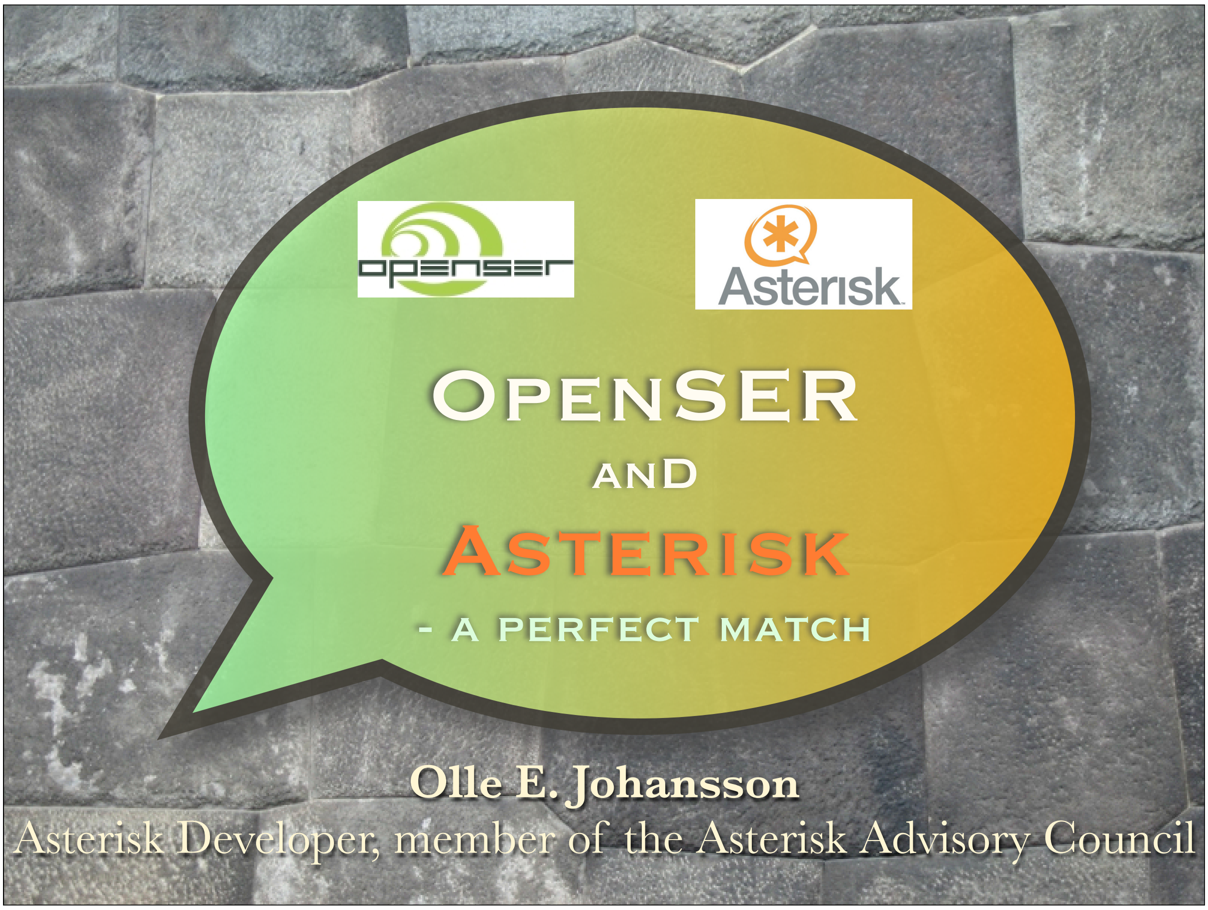
opensER
Asterisk
OpenSER
and
Asterisk
- a perfect match
Olle E. Johansson
Asterisk Developer, member of the Asterisk Advisory Council

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What is Asterisk?
~ An Open Source Modular Multiprotocol PBX
Asterisk

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Not Asterisk
~ A scalable IP communications platform
~ A user-focused platform
~ A secure platform (not yet :-))
Asterisk

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I love Asterisk,

but it’s telephony!!!

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The future

Audio

Text

Realtime

IP

Communication

White-
board

Video

3D
Hologr

????

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...so in my daily work

- Asterisk is not the focus, the main server
- SER/OpenSER is the main platform
- Asterisk delivers services on the SIP network
  - *Voicemail*
  - *PSTN gateway*
  - *Conference*

Which is why I ended up improving the Asterisk SIP stack!

Too separate islands on the net...

Asterisk.org	Postfix.org	Ejabberd
Ekiga	OpenSER	MythTV
VideoLAN	Speex.org	SEMS
ReSIProcate	KDE	????

What happens if we cooperate more?

- Let's build an Open Realtime Platform!
- We need to teach other non-telephony Open Source projects!

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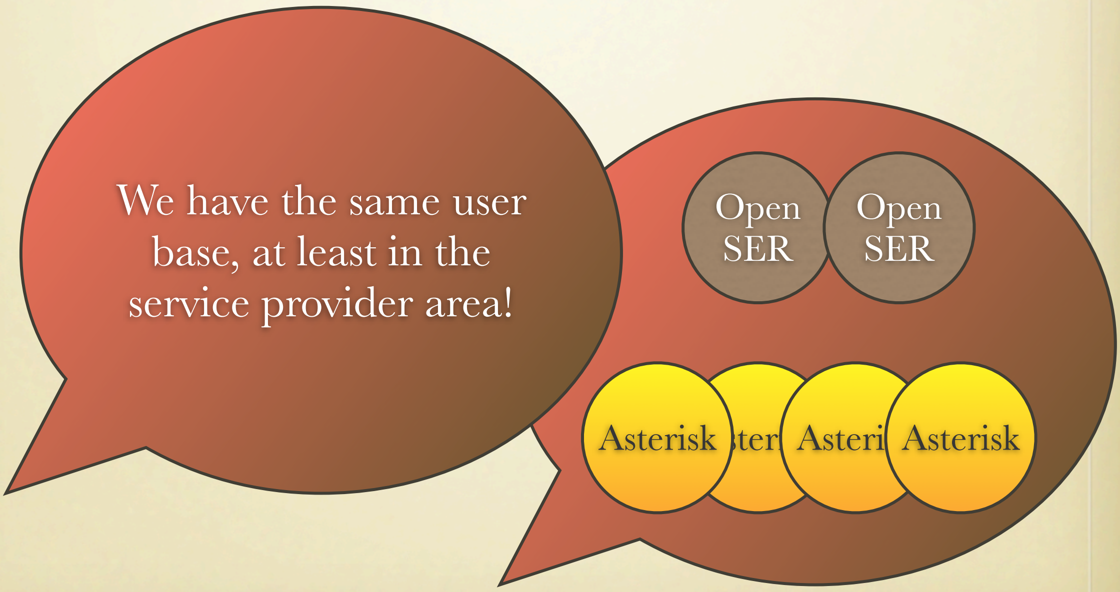
Service providers build like this



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Things I miss in Asterisk/OpenSER integration

- MWI notification to unregistered users by sip URI
- Presence integration
    - *Asterisk handles call states*
    - *Simple/jabber is user states*
- Dialog between OpenSER and Asterisk developers/ users

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Big changes

- Asterisk 1.0 was managed by Mark and two additional committers with small dedicated areas of source to manage
- Asterisk 1.2 was managed by Mark and Kevin
- Asterisk 1.4 has been managed by a larger team - over 10 committers working on all or parts of the code under Kevin's supervision
- An development advisory council is formed to manage the process

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Generic Jitterbuffer

- A jitterbuffer for all channels
- IAX2, SIP, Skinny, zap, jingle
- Developed by Securax in Belgium

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No re-invites needed

- If we know at call setup that we can release media, we will do that directly
- This replaces the re-invites Asterisk used in earlier versions

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SIP transfers

- Enhanced support for REFER
- Support for INVITE/Replaces
- Ability to control REFER support
  - `allowtransfer = yes | no`

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Blinking lamps

- Continued improvements
- You can now disable SUBSCRIBE
  - `allowsubscribe=yes | no`
- Support for parking lots and conferences (meetme)

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Video support improved

- You can now enable video support per peer in sip.conf
- You can also set maximum bitrate allowed
- Asterisk will not include video stream in outbound call when there's no video in the inbound call
- Passthrough support for H.264

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Other 1.4 News in short

- Tons of bug fixes
- Timed RTP transmission
- T.38 fax passthrough support (UDPTL)
- Configurable RTP packetization
- Separate ToS settings for SIP, Audio and Video

Asterisk 1.4

THE SUMMER OF '06 RELEASE

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Going forward

- Sign the Edvina.net NDA and I'll tell you...

Codename

PineApple

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Psst... Codename Pineapple!

- Forking from the standard chan_sip
  - *Not a single 17.000 line source code file*
- Configuration per SIP domain
- Adding transaction states
- Support for forking SIP proxies (branch/tag etc)
- No pedantic mode!
- No more peer/user type's
  - *Trunk, Service, Phone*
- New realtime model
- Preparing for new things
  - *SIP outbound*
  - *GRUU*
  - *Remote RTP handling*
  - *TCP/TLS*

...when?

- Depends on funding...
- Current sponsor: Voop, Norway


A final word:
Security in VoIP...

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Asterisk
Thank You!
oej@edvina.net
Codename
PineApple

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