OpenSER – the open SIP Server

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About OpenSER

Overview

- OpenSER is an open source, GPLed SIP server with
  - High scalability (up to thousands of calls per second of transactional throughput on a PC)
  - Effective application building (modules and application interface)
  - High flexibility (routing language)
- OpenSER is a multi-functional, multi-purpose SIP server: router, switch, registrar, application server,redirect server, gateway, etc
- OpenSER is only about signaling, but there are media adds-on
- it is not a PBX – that's Asterisk !

Basic Feature

- UNIX-like : IPv4/IPv6 : UDP/TCP/TLS
- NAT traversal
- Extended DB support (Mysql, Postgres, Oracle, etc)
- RADIUS & DIAMETER for AAA
- Security & DOS protection
- Advanced Routing: CPL, OSP, LCR, ENUM
- Gateways: SMS, XMPP, Jabber
- Support for clustering and HA

Who is OpenSER

OpenSER is a public project based on collective effort

- **80% of the project is sustained by Voice System**
- **large number of developers :**
  - 3 core developers
  - 22 main developers
  - ~30 developers
  - ~150 contributers
- **worldwide community of users**
- **OpenSER Summit at VoN Berlin, November 2006**
- **plans for developer's meeting in Paris, June 2007**

Where to find it?

OpenSER can be shaped to run on almost any kind of device:

- embedded devices (routers, firewall, access points)
  - Soma Networks
- medium-size devices
  - Collax
- large architectures (servers, clusters)
  - Cisco – OpenSER is the SIP proxy of Cisco Service Node for Linksys One

Who use it?

- SIP service providers
  - 1und1 , voip-users , babble.com, Arcor
- hosting & white label solutions
  - Voztelecom
- routing & trunking providers
- termination & GW providers
- solution providers
  - Voice System
- integrators
  - Basis AudioNet
- academic institutions
  - MIT,UNC, INRIA, SWITCH

Why use it?

- no vendor trap
- faster development cycle
- split work between parties
- easy synchronization with the main stream by contributions ⇒ unified effort for development
  - Voice System (Presence,XMPP, IM Conference, DNS adds-on)
  - Collax (perl scripting support)
  - Voztelecom (Application Agent)
  - SomaNetworks (Session Timer)
  - Trans Nexus (OSP)
  - Enum.at (Infrastructure Enum, Domain policy)
- performance and flexibility

OpenSER v1.2.0

Presence and Simple

- **modular design for presence**
  - one presence engine
  - several (specialized or not) components to inject presence information
- **this design enables:**
  - presence support for non-SIP entities
    - publish the CPU usage of your desktop
    - publish weather information
    - publish the stock size from your store
  - presence support for old SIP phones
  - easy creation for custom/complex presence extensions
    - BLA/SLA
    - dialog presence
- **instant messaging conferencing (IRC style)**

Presence from Non-SIP device

Management

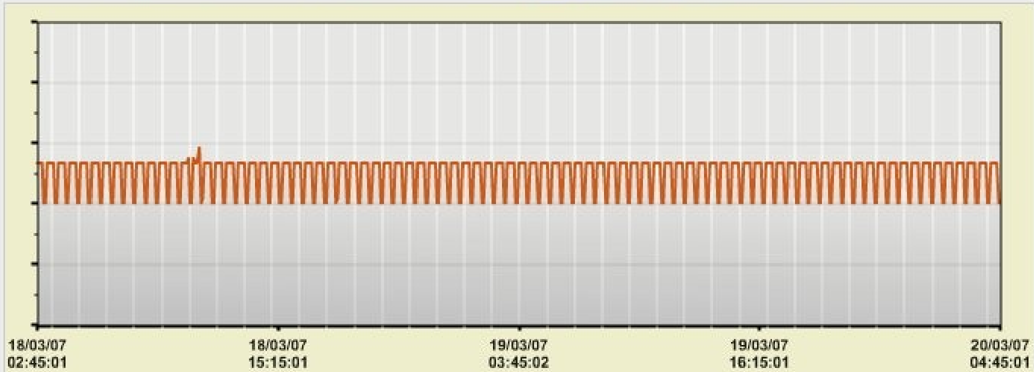
- **Management Interface**
  - pull support for internal statistics
  - push support for external commands
  - different communication layers (local and remote; Ex: xmlrpc)
  - easy to integrate in any external applications (web, shell, perl, etc)
- **SNMP support**
  - built in AgentX subagent for pulling data directly from OpenSER
    - uses standard SIP MIBs
    - OpenSER specific MIBS were added
  - offer ideal hook for traditional SNMP-based platforms
  - this was an external contribution

Statistic charts

Click a statistic to see its chart

dialog:active_dialogs - monitored from 18 Mar 2007, 00:00:01 every 5 minute(s)

shmem:real_used_size - monitored from 18 Mar 2007, 00:00:01 every 5 minute(s)



tm:inuse_transactions - monitored from 18 Mar 2007, 00:00:01 every 5 minute(s)

usrloc:location-contacts - monitored from 18 Mar 2007, 00:00:01 every 5 minute(s)

usrloc:location-users - monitored from 18 Mar 2007, 00:00:01 every 5 minute(s)

Application Server

A more complex configuration:

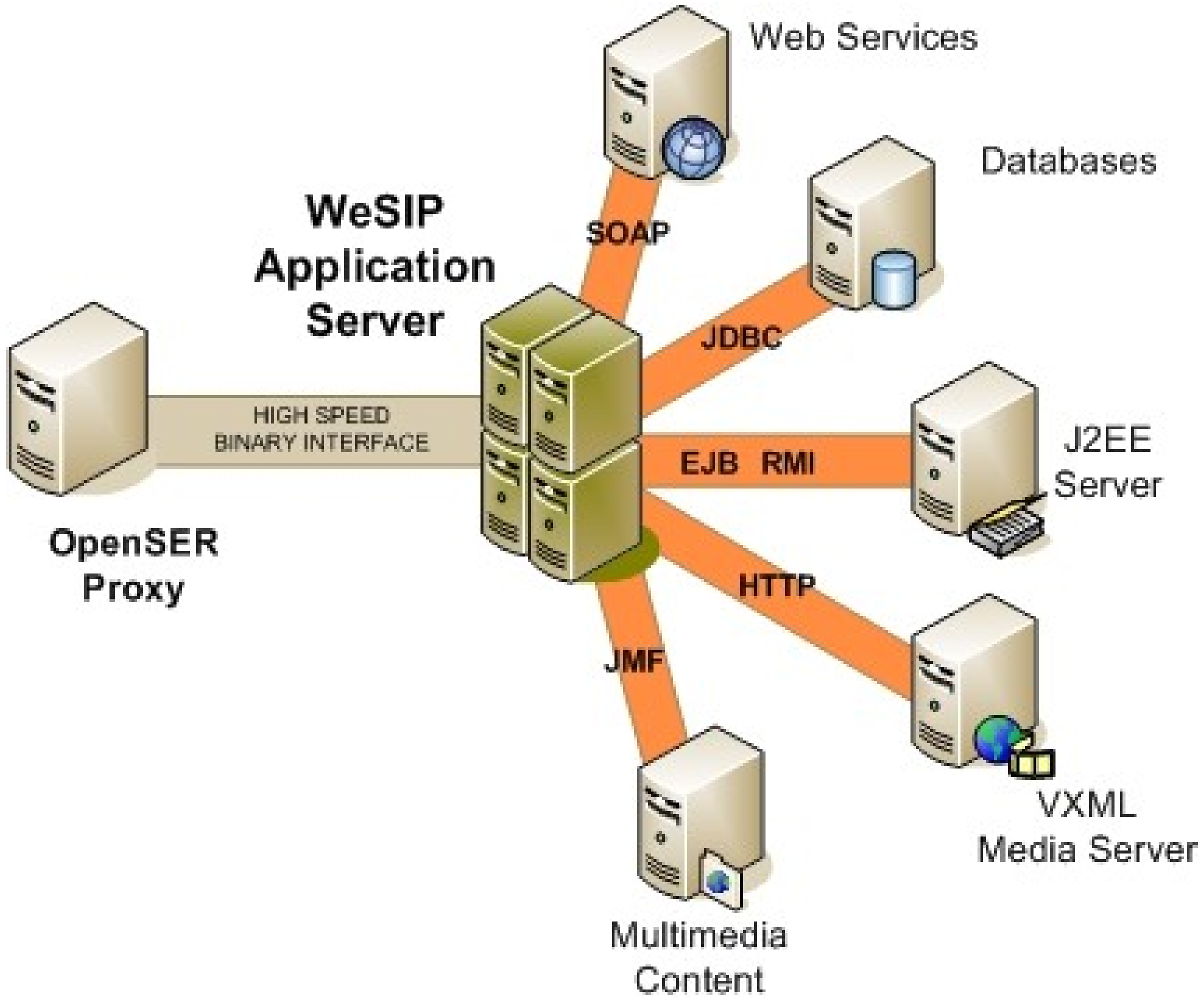
- a separate logical Application Server to be used
- OpenSER reused as SIP stack and transaction engine

- **Standalone Application Server (weSIP-www.wesip.eu)**
  - implements SIP servlets (in Java)
  - uses a connector to talk to OpenSER
  - offers an ideal base for building SIP application with high complexity without dealing with the low details of the SIP part
- **Perl programming interface**
  - triggering PERL scripts from the openser config file
  - similar to Asterisk AGI

Benefits?

- use a robust and fast SIP implementation
- easy and fast creation of high level SIP applications (like PBX)

weSIP architecture



XMPP

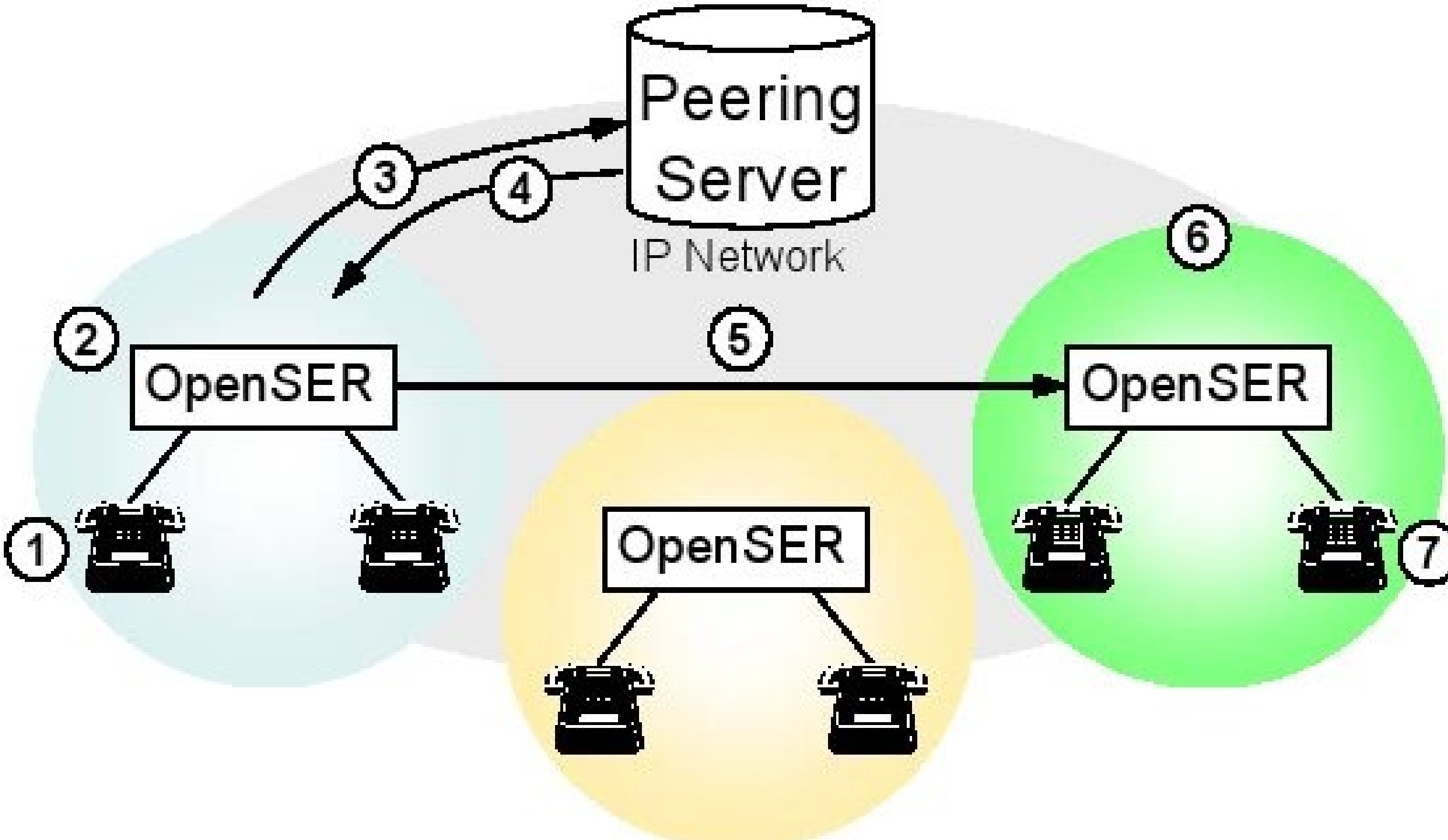
- built-in XMPP gateway for instant messaging
  - transparent translation
  - chat with GoogleTalk® or Jabber buddies
  - flexible routing based on DNS (protocol discovery based on NAPTR)
  - join IM conferencing on XMPP servers
- presence (upcoming)
  - SIMPLE XMPP rich presence translation
  - sub-status mapping
- voice (future)
  - only at signaling level ?!
  - avoid keeping translation states

Security with OpenSER

Secure Peering

- **TLS**
  - encryption at transport level
  - authentication and peering policies via certificates
  - flexible, but still low level feature
- **OSP**
  - no encryption
  - authentication and validation
  - a higher level of standardization for usage
  - eliminates burdensome bilateral peering agreements
- **Domain Policy**
  - no encryption, no authentication, no validation
  - flexible engine for defining federation memberships and inter-domain routing policies
  - based on DNS

OSP-based peering



Server security

- **Authentication & authorization**
  - DIGEST authentication
  - IP authentication (not really secure for UDP)
  - ACL support
- **DOS detection**
  - dynamic monitoring of SIP traffic to detect DOS attacks (mainly based on flood)
  - self protection mechanism
- **IP blacklists**
  - static or dynamic lists containing forbidden destinations
  - lists can be activated based on the type of destination (Gws, subscribers, Media servers, etc)

Convergence & Distribution

Convergence versus Distribution

The explosive development and deployment of VoIP force different - even opposite, in many cases - approaches at each conceptual level.

- at service level, the key word is convergence
  - fix mobile convergence
  - peering heterogeneous IP networks
  - unified messaging
- at architectural level, the key word is distribution.
  - geographic
  - failover
  - scalability

To get them all in a single solution, you need to rely on a cooperative underlying software, like OpenSER

Convergence

- peering heterogeneous networks
  - focus on XMPP (Jabber/GoogleTalk)
  - IM already there
  - coming presence and voice
- facilitate fixed mobile convergence
  - mobile VoIP
  - based on new generation of “smart” phones
  - handover between SIP and GSM – we still need a bit of cooperation from the phones
  - detection can be at signalling or media level
- unified messaging
  - web/email integration
  - SMS gateway
  - SMPP gateway/integration

Distribution

- types of distributions
  - scaling reasons
    - load-balancing
    - traffic dispatching
    - distributed user location
  - high-availability reasons
    - dns-based failover
    - fault detection based on traffic
    - synchronization support
  - geographical distribution reason
    - complex routing for managing remote components
    - centralized decisional logic, but local/distributed processing

Thanks for your attention
You can find more at www.openser.org

Questions are welcome