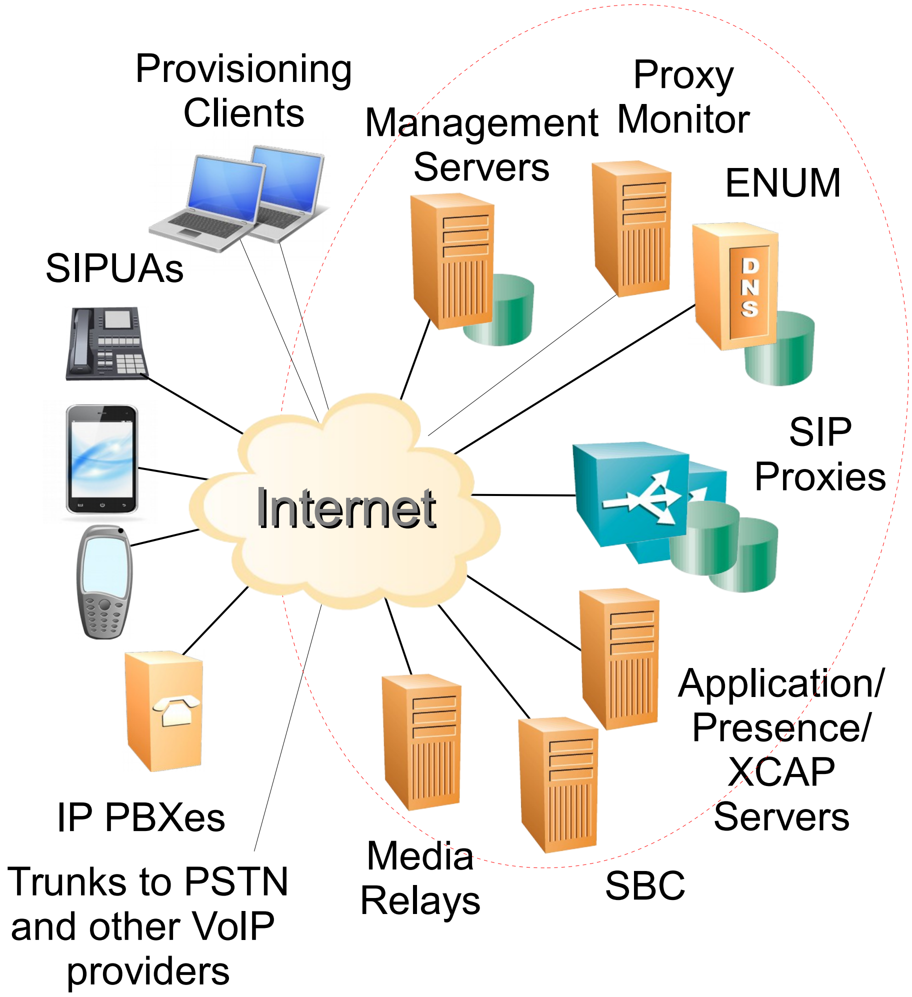
2016-12-26

OpenSIPg SIP Platform

Product Overview

OpenSIPg SIP Platform is a SIP based environment for providing telephony, instant message, and presence services to consumer, enterprise, and telecom operator markets. Operators and ISPs can use the platform “as is” to offer telecommunication services to consumers. Together with modern SIP phones, OpenSIPg can also be used to provide centrex style virtual PBX services to enterprises. In addition, OpenSIPg SIP Platform can serve as a backend for IP PBXes from other vendors and as an interconnection point for telecom service providers.



The key components of OpenSIPg SIP Platform are SIP Proxy, Presence, RLS and XCAP Servers based on Kamailio as well as Application Servers and Session Border Controller based on SEMS. In addition, web based management systems are provided for operators, domain owners, and end users. Backend storage engine for all these components is MySQL. Resiliency of OpenSIPg SIP Platform is achieved via single server active/hot standby environment controlled via industry standard high-availability tools Corosync and Pacemaker or via distributing requests to more than one active “outbound” servers that run independently of each other.

Feature Summary

- High-performance SIP proxy/registrar, messaging, and presence/RLS/XCAP servers
- Support for UDP, TCP, TLS and WebSocket transport protocols over IPv4 and IPv6 together with bridging between incompatible transports and RTP media profiles
- Support for multiple independent domains for easy telecom service hosting
- Hierarchical web based configuration and provisioning system: platform owner, service provider, enterprise, reseller, end user
- PHP and XML-RPC management APIs for integration to customer's IT environment
- Comprehensive fraud protection including fail2ban blocking of attacker IP addresses
- User controllable, time/day/month/caller based call diversion either unconditionally, or when the user is unavailable, busy or does not answer
- Parallel and sequential forking of SIP requests including “follow-me” capability
- User access to registration and call history information for placed, received, diverted, failed, and missed calls together with click-to-dial capability
- Service provider and user controllable barring of calls to given prefixes, such as 700 numbers, as well as GeoIP based limiting of allowed calling countries
- Database based configuration of multinational and peer specific dial plans including local calling without area code
- Routing of emergency calls to nearest emergency center
- Support for number portability and automatic user and infrastructure ENUM updates
- Domain and peer specific least cost routing of PSTN and peer bound calls
- Easy provisioning of trunks to PSTN gateways, IP PBXes, and other operators
- Domain specific dial-plans, user definable speed dialing, as well as user specific carrier pre-selection options
- Automatic handling of SIP UAs behind NATs in addition to user, IP network, and peer specific forced proxying of media to geographically distributed media proxies
- Voicemail application with choice of email, web, and/or call access to messages including support for Message Waiting Indication
- User specific Call Center and Voice Conference applications
- Session Border Controller for implementing caller anonymity as well as URI specific limiting of parallel calls
- Easy integration of user developed application services that can be written either as SEMS DSM diagrams or generated automatically from textual descriptions
- Local and remote monitoring of all components

System Requirements

- OpenSIPg SIP Platform is distributed as a set of Debian 8 (Jessie) Linux packages for amd64 architecture.
- Bare metal, cloud, or container based installation (also behind NAT).