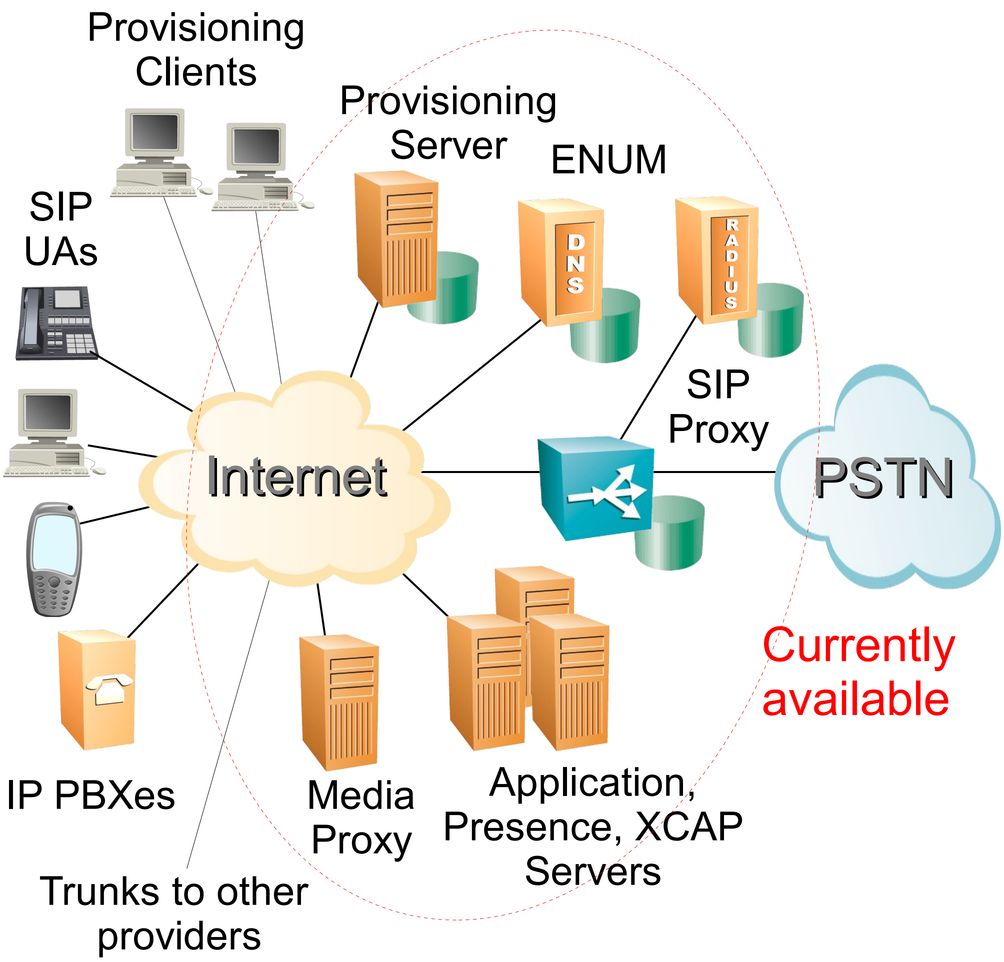
OpenSIPg - OpenXg SIP Platform

Product Overview

OpenXg SIP Platform (OpenSIPg) is a SIP based environment for providing telephony, instant message, and presence services to consumer, enterprise, and operator interconnect markets. Operators and ISPs can use OpenSIPg "as is" to offer telecommunication services to consumers. OpenSIPg (together with modern SIP phones) can also be used to provide centrex style PBX services to enterprises. In addition, OpenSIPg 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 are SIP Proxy based on SIP Router, Presence and Resource List Server based on OpenSIPS, XCAP Server based on OpenXCAP, as well as web based provisioning servers for service provides, domain owners, and end users. Other open source building blocks of OpenSIPg include Radius server based on FreeRADIUS, NAT traversal solution based on Mediaproxy, Application Server based on SEMS, and ENUM server based on PowerDNS. Backend storage engine for all these components is MySQL. Resiliency of OpenSIPg environment is achieved by industry standard high-availability components DRBD, Heartbeat, and Pacemaker.

 info@openxg.com

Feature Summary

- High-performance SIP proxy/registrar, messaging, and presence/RLS/XCAP server
- 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 APIs for integration to customer's IT environment
- User controllable, time/day/month based call diversion either unconditionally, or when the user is unavailable, busy or does not answer
- User definable speed dial by dialing <digit>*
- User controllable display of calling party number (always or per-call)
- User access to registration and call history information for placed, received, diverted, failed and missed calls together with click-to-dial capability
- User controllable barring of calls to given prefixes, such as 700 numbers
- Parallel and sequential forking of SIP requests including “follow-me” capability
- Allocation of phone numbers blocks to service providers, resellers, and enterprises
- Domain and peer specific least cost routing of PSTN bound calls
- Database based configuration of national 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 ENUM updates (user, infrastructure, and private ENUM trees)
- Easy provisioning of trusted peers, such as IP PBXes and other operators
- Domain specific dial-plans and dial prefix replacements
- Automatic handling of SIP UAs behind NATs
- Voicemail application with choice of email, web, and/or call access to messages including support for Message Waiting Indication
- Call Center application for implementing user specific contact centers
- Conferencing application for provision of user specific voice conferences with both dial-in and dial-out capability
- Domain and user specific directories with click-to-dial capability
- Authentication of OpenSIPg users to WiFi APs using EAP-PEAP MS-CHAPv2
- User, IP network, or peer specific forced use of Media Proxy
- Redundant, fault tolerant configuration and monitoring of all OpenSIPg components

System Requirements

- OpenSIPg is distributed as a set of Debian 5.0 (Lenny) Linux packages for PC (i386) architecture.
- Minimum of one PC with 1 GB of main memory and 40 GB of disk space