

Open Source Telephony Applications & Opportunities

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Introduction

- What is Open Source?
- Open source doesn't just mean access to the source code.
- The distribution terms of open-source software must comply with the following criteria:
 - **Free Redistribution**
 - **Source Code**
 - **Derived Works**
 - **Integrity of The Author's Source Code**

Introduction (continued)

- **No Discrimination Against Persons or Groups**
- **No Discrimination Against Fields of Endeavor**
- **Distribution of License**
- **License Must Not Be Specific to a Product**
- **License Must Not Restrict Other Software**
- **License Must Be Technology-Neutral**

Introduction (continued)

- The first true open source telephony application was Vocal (Vovida Open Communications Application Library) followed closely by the OpenH323 project in the late 90's.

VOCAL

- Vovida Open Communications Application Library.
- Set of libraries and applications used as VoIP Building Blocks.
- Acquired by Cisco in 2000, most projects have not been updated since 2003.
- Includes a number of applications including STUN Server, Load Balancer, SIP to H.323, SIP to MGCP, B2BUA, and a provisioning server.

Asterisk



- Perhaps the most well know Open Source application.
- Highly scalable, extensible beyond its original purpose.
- Highly complex to configure. Thus the need for 3rd party front-ends.
- Compete distributions on disc including OS, Asterisk, Front-end, back-end and other apps. Most popular being TrixBox & Elastix.

Asterisk (continued)



- Runs on legacy machines quite well. Also versions for embedded processors such as the AMD Blackfin.
- Good opportunities for aggressive, forward looking carriers to bundle DSL, SIP Trunks and IDD to SOHO, SME clients.

FreeSwitch



- FreeSwitch originally started life as a fork of asterisk.
- Now, it's been total re-written and the authors claim its much faster, more extensible and solves many problems associated with running asterisk in a Service Provider environment.
- Can provide services such as: a Rating & Routing Server, a transcoding B2BUA, IVR & Announcement Server, Conference Server, Voicemail Server, a basic SBC as well as a basic topology hiding SBC

FreeSwitch (continued)



- is capable of doing about 6,000 channels per chassis with slight transcoding on a modern machine (Dual Quad Core 3.0 GHz processors).
- FreeSwitch is the engine behind a large number of VoIP services, most notably being used along with Telco Bridges hardware to power the VoIP services of TruPhone.

SIP Express Router



- SER, the Sip Express Router, has been around since 2002 and is a high performance, highly configurable, free SIP server.
- SER can be configured to perform specialty services such as load-balancing. In conjunction with other open source software it can also function as a media relay or a SBC.
- In June of 2005, some of the original team that developed SER forked SER into a new project OpenSER.

SER (continued)



- quickly outpaced its parent to become the defacto SIP server among service providers.
- In July 2008, OpenSER was fractured into two different groups with the original OpenSER project being renamed to Kamailio and the splinter group being called OpenSIPS.
- OpenSIPS and Kamailio can be used as a SIP Presence Agent, a SIP IM Server (chat and end-2-end IM), SIP to SMS gateway (bidirectional), SIP to XMPP gateway for presence and IM (bidirectional) as well as a SIP NAT Transversal unit.

SER (continued)



- As of the 4th of November, it seems the Kamailio people have joined with the original SER group to form the SIP Router Project.
- Is this just like a soap opera or what!

OpenSBC



- The objective of the OpenSBC project is to create a Session Border Controller that is fully compliant with the mandates of RFC 3261.
- can be used as a SIP router, media anchor for far-end NAT traversal, SIP egress and ingress trunking among others.
- OpenSBC has been under development and in use for over 7 years in high volume applications.

- sipXecs is the other free IP PBX.
- Often lost in the Asterisks shadows, its proponents claim it is a far superior solution to asterisk both in manageability as well as scalability.
- SipXecs is easy to use and made for non-technical people.
- sipXecs powers the Nortel SCS500 Internet Phone System.

SIPp



- SIPp is a free Open Source test tool / traffic generator from HP.
- features the dynamic display of statistics about running tests .
- SIPp can also send media (RTP) traffic through RTP echo and RTP / pcap replay.
- It is also very useful to emulate thousands of user agents calling your SIP system.

Vovida STUN Server

- This project implements a simple STUN server and client on Windows, Linux, and Solaris. The STUN protocol (Simple Traversal of UDP through NATs) is described in the IETF RFC 3489, available at <http://www.ietf.org/rfc/rfc3489.txt>

Open SS7

- OpenSS7 is a project to provide a robust and GPL'ed SS7, SIGTRAN, ISDN and VoIP stack for Linux and other UN*X operating systems.
- The overall project is made up of smaller projects that include:

• ISUP	• MTP Level 3
• BSSAP	• SS7 SM
• TCAP	
• SCCP	

- VICIDIAL is a set of programs that are designed to interact with Asterisk to act as a complete inbound/outbound call centre suite.
- VICIDIAL can function as an ACD for inbound calls.
- VICIDIAL allows for seamless integration of traditional TDM and VOIP.

Open Source is not just Software!

- The Free Telephony Project is a complete embedded 4 port PBX.
- Powered by Asterisk and the Asterisk NOW! GUI.
- Hardware schematics and PCB designs are available free.

Free Telephony Project



Free Telephony Project (continued)

- IP04 as manufactured by ATCOM in Shenzhen.



Conclusion

- Searching the term “telephony” returns 114 projects on sourceforge (a central repository of open source projects).
- Searching “VoIP” returns 235 projects.
- Most companies already running some open source software in the form of Linux.
- If you use a project commercial, please give back to the community in the form of \$\$\$ or by contributing code for the benefit of all.

URLs

- Vovida: www.vovida.org
- Asterisk: www.asterisk.org
- TrixBox: www.trixbox.org
- Elastix: www.elastix.org
- FreeSwitch: www.freeswitch.org
- SER: www.iptel.org/ser/
- OpenSIPS: www.opensips.org
- Kamailio: www.kamailio.org

URLs (continued)

- SIP-Router Project: www.sip-router.org
- OpenSBC: www.opensourcesip.org
- sipXecs: www.sipfoundry.org
- SIPp: sipp.sourceforge.net
- OpenSS7: www.openss7.org
- ViciDial: www.vicidial.com
- Free Telephony Project: www.rowetel.com/ucasterisk/
- Sourceforge www.sourceforge.net



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