



A note from the student behind this lab

This is an **independent, free, learning-only student project**. It respects every company and vendor, and every trademark belongs to its owner. It is **not affiliated with or endorsed by** any of them.

If anyone has a concern, the content will be removed or corrected promptly: please write via <https://cyphermonkeylabs.com/#contact>. Nothing here is sold, and I am happy to volunteer for any organization.

Open-source telephony core

Read the role each project states. Do not treat them as four names for one PBX.

What each page calls itself

The Asterisk history page says the project was published under the GPL and that the idea of an open source PBX caught on. Current SIP configuration for the PJSIP channel driver is documented as `res_pjsip` and `chan_pjsip`, in `pjsip.conf`, with `extensions.conf` as the dialplan.

The FreeSWITCH manual this lab opened teaches SIP profiles, the dialplan, codec selection, media handling, WebRTC, and the Event Socket. It says FreeSWITCH ships as Open Source FreeSWITCH and as FreeSWITCH Enterprise from SignalWire. Kamailio's home page calls it an open source SIP server under GPLv2+, able to handle thousands of call setups per second, and says it can be used to scale SIP-to-PSTN gateways, PBX systems, or media servers such as Asterisk, FreeSWITCH, or SEMS.

rtpengine is the media proxy in this set

The rtpengine README calls it a proxy for RTP and other UDP media, meant to be used with the Kamailio SIP proxy, and a drop-in replacement for other RTP and media proxies. The only supported platform named there is GNU/Linux.

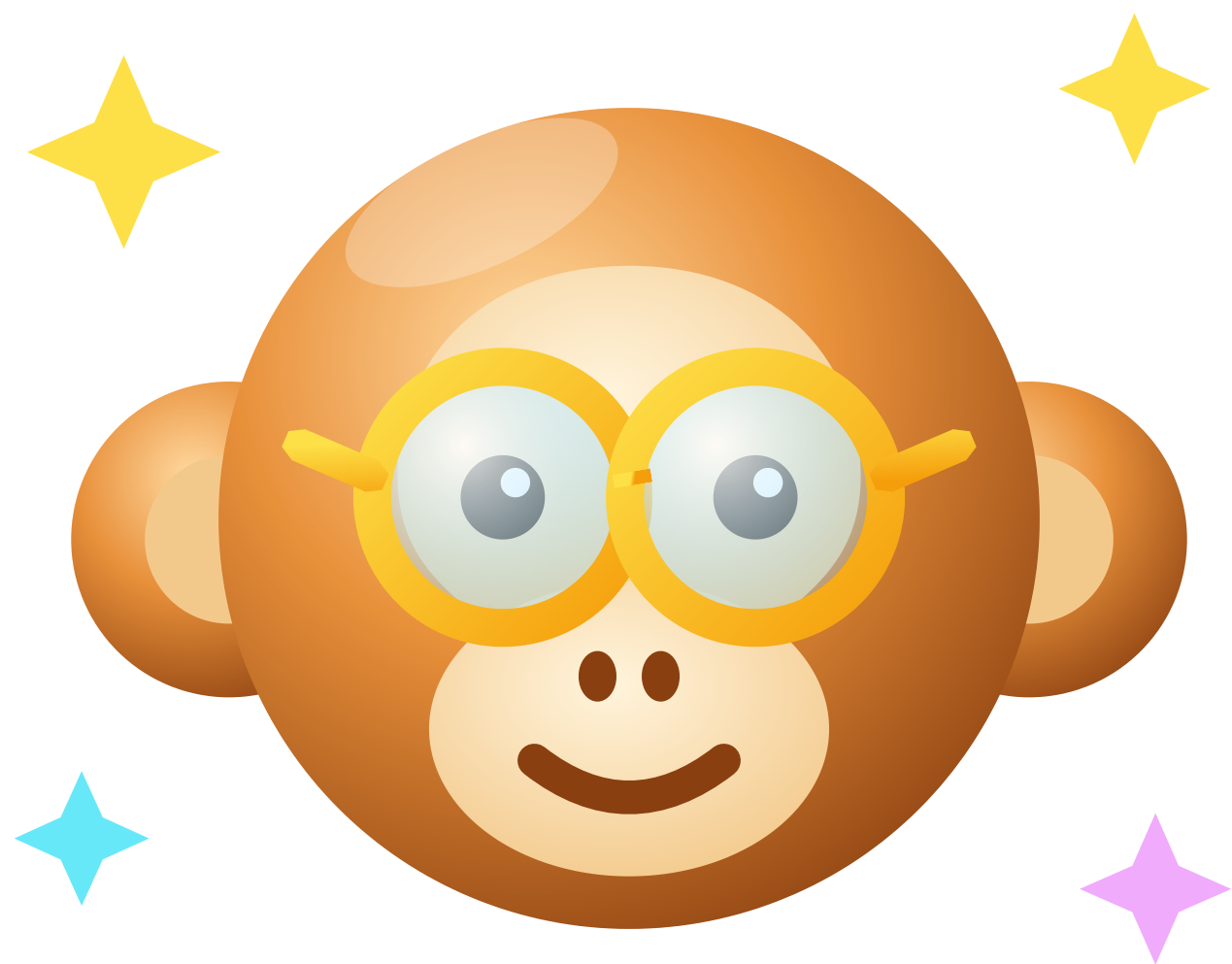
That split is the integration map: Kamailio for SIP signaling roles the project lists, including registrar, edge proxy, and WebSocket support for WebRTC, and rtpengine when the media needs a proxy that can rewrite SDP and bridge RTP with SRTP. Asterisk or FreeSWITCH is where a dialplan lives. Putting all four on one host is a choice these pages do not require.

Room: [Four projects, four roles](#)

Sources and go deeper

- [A brief history of the Asterisk project](#) — last checked 2026-10
- [Asterisk res_pjsip configuration](#) — last checked 2026-10
- [FreeSWITCH users manual](#) — last checked 2026-10
- [Kamailio SIP server](#) — last checked 2026-10
- [rtpengine README](#) — last checked 2026-10

CypherMonkey Labs







© 2026 CypherMonkeyLabs 🐒

[Open app](#) · [View repo](#)