Specifications

Possible Uses

- Rating & Routing Server
- Transcoding B2BUA
- IVR & Announcement Server
- Conference Server
- Voicemail Server
- SBC (Session Border Controller)
- Basic Topology Hiding Session Border Controller
- DAHDI, Khomp, PIKA, Rhino, Sangoma and Xorcom Hardware Support
- Fax server
  - T.38 gateway, termination, and origination mode
  - T.30 to T.38 and T.38 to T.30 gateway
  - See also: mod_spandsp
- And, of course, a PBX

Features

- WebRTC support
- Centralized User/Domain Directory (directory.xml)
- Nano Second CDR granularity
- Call recording (in stereo caller/callee left/right)
- High Performance Multi-Threaded Core engine
- Configuration via cURL to your HTTP server (mod_xml_curl).
- XML Config files for easy parsing.
- Protocol Agnostic
- ZRTP support for transparent RTP based key exchange and encryption
- Configurable RFC 2833 Payload type
- RFC 2833 is obsoleted by RFC 4733.
- Inband DTMF generation and detection.
- Software based Conference (no hardware requirement)
- Wideband Conferencing
- Media / No Media modes
- Proper ENUM/ISN dialing built in
- Detailed CDR in XML
- Radius CDR
- Subscription server
  - Shared Line Appearances
  - Bridged Line Appearances
- Enterprise/Carrier grade Eventing Engine. (XML Events, Name Value Events, Multicast Events)
- Loadable File formats and streaming
- Stream to and play from Shoutcast and Icecast
- Multi-lingual Speech Phrase Interface
- ASR/TTS support (native and via MRCP)
- Basic IP/PBX features
- Automated Attendant
- Custom Ring Back Tones (Early Media)
- XML-RPC support
- Multiple format CDRs supported
- SQL Engine provides session persistence
- Thread Isolation
- Parallel Hunting
- Serial Hunting
- Mozilla Public License
- Paid support available
- Free support via IRC & E-mail
- Many supported codecs
  - CELT (32 kHz ahd 48 kHz)
  - G.722.1 (wideband)
  - G.722.1C (wideband 32 kHz)
  - G.722 (wideband)
  - G.711
  - G.726 (16k, 24k, 32k, 48k) AAL2 and RFC 3551
  - G.723.1 (passthrough)
  - G.729AB (Requires a license unless using passthrough)
  - AMR (passthrough)
  - ILBC
  - Speex (narrow and wideband) with RFC 5574 fmp4 support
  - LPC-10
  - DVI4 (ADPCM) 8 kHz and 16 kHz
  - SILK
  - OPUS - RFC 6716
• Video Codecs (passthrough):
  • Theora
  • H.261
  • H.263
  • H.264
  • MP4
• See also: codecs
  • Live Migration of calls from one FreeSWITCH box to another. See Freeswitch_HA

Applications

• Voicemail
  • Multitenancy - Enterprise/Carrier configuration
  • Time of Day Greetings
  • Urgent Message Tagging
  • E-mail Delivery
  • Playback and Rerecord messages before delivery.
  • Keys are templates so you can rearrange to fit your needs.
  • Callback support from inside voicemail.
  • Podcast of Voicemail (RSS)
  • Message Waiting Indicator (MWI)
• Support for Queues (via mod_fifo or mod_calcenter)
• Parking (via mod_fifo)
• Conference
  • Software based Conferencing without any hardware requirements.
  • Wideband conferences.
  • Multiple on-demand or scheduled conferences with entry/exit announcements
  • Play files into the conference or a single member.
  • Relationships
  • TTS integration
  • Transfers
  • Outbound Calling
  • Configurable Key Lay
  • Volume, Gain and Energy level per call.
  • Bridge to Conference transition
  • Multi Party outbound dialing.
  • RFC 4579 SIP CC Conferencing for UAs
  • Automatic or on-demand recording
• RSS Reader
• Fax endpoint, gateway and passthrough mode.
  • T.30 (G.711) Audio Fax (via mod_spandsp) formerly known as mod_fax.
  • T.38 faxing (gateway, endpoint and passthrough)

Protocols

• SIP with mod_sofia
  • UDP, TCP, SCTP and TLS transports for full SIP compliance.
  • SIP v.2.0 (RFC 3261)
  • IPv6 Support
  • SIP Session timers
  • RTP Timers
  • RFC 3263 (SRV and NAPTR)
  • RFC 3325
  • RFC 4694
  • SRTP via SDES (Works with Polycom, Snom, Linksys and Grandstream)
  • Blind SIP Registration
  • STUN Support
  • Jitter buffer
  • NAT Support
  • Distributed SIP registrations
  • Late Codec Negotiation
  • Multiple SIP registrations per user account.
  • Multitenancy - Multiple SIP UAs
  • SIP Reinvites.
  • Can act as an SBC (Session Border Controller)
  • Manage Presence
  • SIP/SIMPLE (can gateway to other chat protocols)
  • SIP Multicast Paging support for Linksys and Snom
  • Intercom/AutoAnswer support.
  • Call features like Call Hold (Re-INVITE), Blind Transfer (REFER), Call Forward (302), etc.
• Jingle with mod_dingaling
• Interop with Google Talk and Telepathy
  • H.323 with mod_opal (opalvoip.org)
  • H.323 with mod_h323 (www.h323plus.org)
  • IAX2 with mod_opal (opalvoip.org)
  • mod_skinny - Skinny Call Control Protocol (SCCP)
Languages

- JavaScript (Using the Google V8 JavaScript engine.)
- ODBC Support from inside your JavaScript
- Extendable modules for JavaScript
- Tone Generation
- Ruby
- Python
- Perl
- Lua

Cross Platform

- Builds native on Windows in MSVC
- Builds on Mac OS X, Linux, Solaris and *BSD.

Minimum/Recommended System Requirements

- 32-bit OS (64-bit recommended)
- 512MB RAM (1GB recommended)
- 50MB of Disk Space

System requirements depend on your deployment needs. We recommend you plan for 50% duty cycle.

Performance

- Tested under load for over 100 hours
- 10,000,000+ calls
- At rates exceeding 50 CPS

Performance will vary depending on application. You will need to test for your particular situation.