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
- 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
- Support
    - 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 fmtp 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_callcenter)
- 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.