streamco Smartswitch

Turnkey solution for VoIP business

Streamco Smartswitch

Softswitch	Multi-tenant Virtual PBX	Calling cards access server
Telephony billing system	Traffic analyser/ service controller	Anything you can imagine when it relates to telephony call

Smartswitch is a high-performance and user-friendly software product for telephony calls processing. It's a hybrid Class 4/5 solution that can be used as Softswitch, PBX, Voice Access Server, etc. Smartswitch is a powerful "all-in-one" system for telephony traffic management and accounting in real-time mode.

Reasons to choose Smartswitch

Cost-effective solution

Analysis of available solutions should show you that similar products are significantly more expensive. Other vendors develop and support their telephony core by themselves, which requires significant financial expenses. Smarswitch is based on open-source Asterisk core, which enables us to minimize expenses for development and support, and accordingly, to minimize price for our customer.

Compatibility

Smartswitch is based on open-source Asterisk core, which is tested and used by thousands of users everyday all over the world and in various conditions and combinations with other software and hardware. Compatibility issues are fixed very quickly. This caused Asterisk to be almost 100% compatible on the contrast with similar products.

Full feature set

Open-source multi-functional Asterisk core gives us intellectual feature set and potential which exceed the ones of similar products. The latter often appeared earlier and simply were not expected to be used in modern conditions and demands, therefore many of our core features are implemented as helper modules or not implemented at all.

Performance

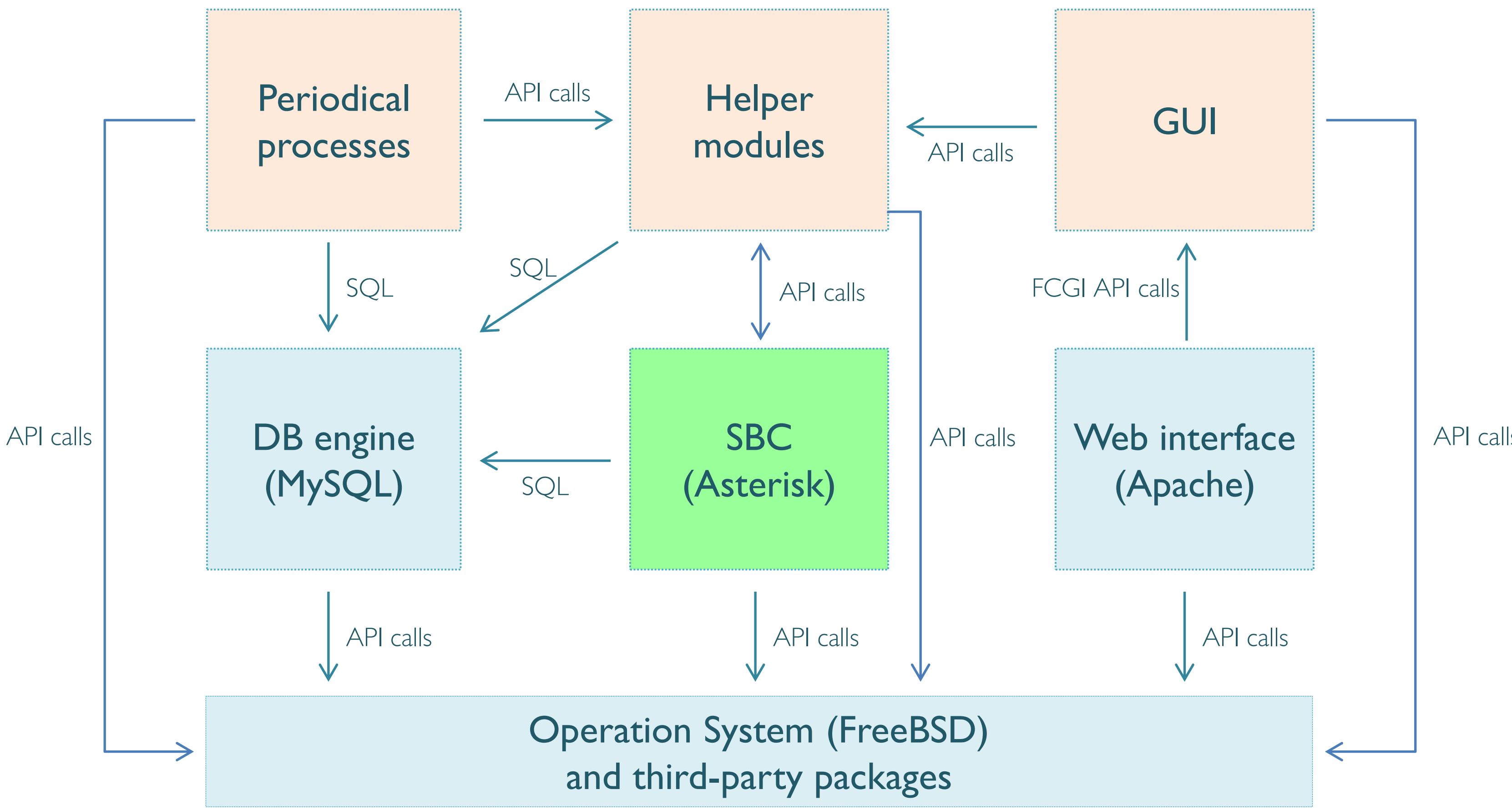
We have Call capturing, which doesn't have analog in the products of our competitors. It allows you to receive .pcap.gz files with full dump of the call from CDR.. Learn more about Call capturing

24/7 Support & Trainings

Our engineering staff has wide work experience with different solutions and have significant experience in debugging telephony protocol problems. Our developers perform development on operating system kernel level. This means **we can handle absolutely any problem** that you might have.

Architecture

Smartswitch represent itself as a superstructure (additional modules) over Asterisk PBX and uses its capabilities to perform actions on a channel and to bridge signaling sessions and media streams. Items marked orange - developed by Streamco from scratch, green - patched open-source component, blue - original open-source components.



Functionality

Key features

- All-in-one system. No need for separate servers for signaling, media or billing;
- Flexible hunting based on the current configuration and accounting information;
- Configuration changes in real-time mode;
- Custom call handling that is configured via Visio-like Call Handler GUI;
- Integrated billing;
- Fast, user-friendly web interface;
- Great amount of supported hardware;
- Wide support of VoIP protocols and media formats;
- Extendable architecture.

Billing key features

- Works in real-time mode. No separately launched processes/servers needed;
- Integrated in the system's core - all authorized calls will be stored in database, and all of them will be billed according to the actual, at the moment the call was made, configuration;
- Flexible tariff settings allow to set up the billing increment, the connection price and much more;
- Performance. Due to its integration in the core billing is performed very fast;
- Flexibility - every inbound/outbound dial-peer can have its own price-list;
- Reliability - whenever the system crashes, all calls are billed on system respawn and nothing is lost;
- Support for multi-currency in pricing, routing and balance management for companies and end users;
- Possibility to update user balance during the call handling - needed for complex callback and IVR-based services.
- Automatic invoice generation. Learn more about customer invoices.
- Automatic process of adding changes of billing parameters, which come from partners by e-mail. Feature details

Key hunting and routing features

- Any inbound traffic entry point can have its own hunting class (a set of rules that define how hunting table should be built);
- Team-up with the financial information for partner authorization basing on payments processed and current balances;
- Hunting basing on: price (Least Cost Routing), price priority, peer priority, route priority, ASR, ACD, matched code length;
- Time-dependent hunting - implemented using price-list substitution timetable;
- Flexible caller/callee ID substitution using regular expressions;
- You can check in web interface how your routing is performed according to configuration without making real calls.

Web interface key features

- Performance. Interface was developed using the Fast CGI technology and is AJAX-enabled. This makes possible to achieve minimum query reaction time among all other content-generation technologies available;
- Multi-tenant access with flexible roles assignment.
- Advanced reporting features;
- Web portal for administrators, companies and users;
- Internationalization. Full support for english and russian locales.

Key telephony features

- Great amount of supported VoIP protocols (including SIP and H323);
- Direct RTP support;
- T.38 fax support;
- High performance (more than 1000 simultaneous calls per server);
- VoIP protocols are supported in B2BUA mode (on the contrast with proxy mode). This makes possible to execute any codec/protocol transcoding and protocol dialect normalization.
- Compatibility. Many vendors doesn't state that they support each other, but they almost certainly say that they support Asterisk.

Telephony features

VoIP audio

Supported codecs

- G729 (in proxy mode - G729AB, in transcoding mode - G729A)
- G723
- GSM
- G711 ulaw
- G711 alaw
- G726
- ADPCM
- LPC
- slin
- speex
- iLBC
- G722 (only proxy mode)

Supported protocols

- RTP
- RTCP
- RTP DTMF (RFC2833)

Supported modes

- p2p (peer to peer). Is available for SIP channel;
- proxy. Is avaliable for all codecs. There is application level proxy mode available as well as proxy mode on the kernel level (Media Proxy);
- transcoding. Is vailable between any of the codecs, excluding G722. It's enabled automatically in the cases where in the process of session negotiation peers have chosen a codec scheme that requires transcoding.
- buffering (can be enabled optionally). You can define the duration of voice frame in milliseconds and system will buffer the stream according to the settings.
- reading DTMF from the voice stream (decoding tones on software DSP) - for G.711 ulaw/alaw family of codecs;
- voice recording to file from RTP stream. Supported file formats:
- wav. Voice transcoding mode activates during this mode;
- g729, g723, etc (all codec formats are supported). In case if file format doesn't match current selected channel format - transcoding mode is activated.
- playback from a file to an RTP stream. Supported file formats:
- wav and mp3. Transcoding mode is activated;
- g729, g723, etc (all codec formats are supported). In case if file format doesn't match currently chosen codec- transcoding mode is activated.
- audio generation to RTP stream. Ringback generation.

Telephony features

DTMF

Supported protocols

- RTP DTMF (RFC 2833);
- SIP INFO DTMF (RFC 2976) - for SIP channels;
- Cisco DTMF - for H.323 channels;
- H.245 DTMF - for H.323 channels;
- inband. For G.711 ulaw/alaw.

There is available automatic signalling protocol translation between all supported protocols.

Video codecs

- H261;
- H263;
- H263p;
- H264.

Faxes

Supported protocols of data transmission

- T.38 (UDPTL);
- inband (for G.711 ulaw/alaw).

For T.38 faxes there is supported switch back to voice mode after fax has been transmitted.

T.38 error correction modes:

- redundancy;
- FEC.

Available modes

- proxy. There is available both application level proxy and kernel level proxy (Media Proxy);
- T.38 UDPTL transcoding. It happens in case if peers have incompatible error correction modes and/or buffer size;
- fax-server. Supported modes:
- fax receival in T.38/inband mode with recording to .tiff file.
- transmission of a fax in T.38/inband mode from a .tiff file.

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VoIP signalling protocols

H.323

Supported protocols

- H.323 v4;
- H.225;
- H.245;
- T.38;
- RAS;
- "fast start" procedure support;
- H.245 tunneling. Additional tunneling modes: cisco, QSIG.

Additionally

- support for devices under NAT;
- manipulation with ToS bitfield of IP packets
- RTP transmission modes: proxy, transcoding, passing through the jitter buffer.

There are 2 different stacks you can choose from:

- OpenH323;
- Objective Systems.

Both stacks can be used simultaneously and independently.

SIP

Supported protocols

- SIP v2;
- T.38;
- SIP INFO DTMF (RFC 2976);
- RFC 3261 (ACK, BYE, CANCEL, INVITE, OPTIONS, REGISTER);
- RFC 3265 (SUBSCRIBE, NOTIFY);
- RFC 3311 (UPDATE);
- RFC 3515 (REFER);
- RE-INVITE support.

Additionally

- registration: inbound and outbound. Configurable timeouts and number of registration attempts;
- support for devices under NAT;
- support for system installation under NAT;
- call hangup by RTP timeout;
- RTP transmission modes: direct (p2p) , proxy, transcoding, passing through the jitter buffer;
- video support;
- manipulation of ToS bitfield of IP packets.

Learn more about Smartswitch. Check our DEMO.
Request online demonstration, it's FREE!

Smartswitch Standard edition is FREE. Download here
PRO edition features and price here.

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