VoipSwitch is a software platform allowing for rapid VoIP services roll-out. It contains all necessary elements required in successful implementation of various VoIP services.

Our customers can make money on whole range of services starting with wholesale voice and SMS termination, through Calling cards (phone to phone) and all types of Callback, to the services offering full replacement of traditional phones, namely broadband phone calling, which enables users to benefit from class 5 services.

Internet Telephony Service Providers (ITSP) can enhance their offer by adding softphone/communicator with SIP, Instant Messenger and other cutting edge functionalities.
Another popular group of phone services are callshops which are also fully supported by our software.
All services can be offered through resellers who can manage their end-users accounts, rates sheets, see reports, active calls and more via web interface.

The system is multilayered, with the core which is the session border manager (softswitch) integrated with own, built-in, billing system. The core is responsible for switching the traffic, accepting incoming connections in different protocols, authorization, billing, reconciliation of different protocols or dialects and proxying the packets.

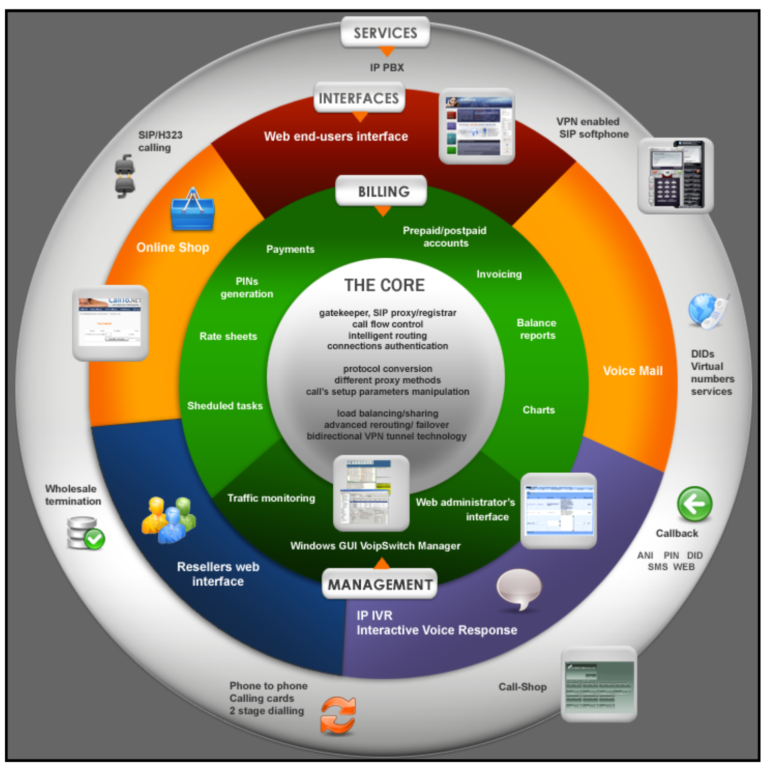
The higher layers consists of the applications extending the core of additional functionality like Callback, Interactive Voice Response system, Callshop engine, Resellers module and others.
In addition the platform offers variety of tools for management and reporting such as VSM, Web Configuration, Event Manager and others.

Another group of software are applications designated for end-users to which belong VSportal– endusers web interface, PC softphones – Vippie, SIPLink, Mobile softphones, Callshop web interface and Online shop.

Main characteristics

System

- Single platform solution
- Rapid Service roll out
- Pre-configured templates
- Scalability – Phased investment
- Quick, flexible reporting
- Open database architecture allowing for extended reporting functions and use of external scripts.



Switching

- Carrier-grade softswitch, gatekeeper, registrar and proxy functionality, SIP and H323 protocol support
- Protocols transparent conversion
- Network protection capabilities
- Advanced routing, single entry point into VoIP infrastructure
- Flexible proxy methods
- Video support

Class 5 features

- Hold, music on hold
- Call transfer: blind and attended
- Conditional find me/follow me, different rules depending on caller ID
- Voicemail with personalized voice messages for different callers
- Ring/Hunt groups

Billing

- Various authentication methods Prepaid/Postpaid accounts
- Robust engine, full integration with the softswitch. Cooperation with MySQL
- Advanced billing features, support for monthly calling plans, prepaid and postpaid accounts

Management interface

- Real-time traffic and systems
- Performance monitor, Windows graphical interface, web-based interface

Web interface for end-users

- Customizable customer self care web interface
- Admin section
- Ready to use design templates
- CDRs, payments, invoices, address book, users profile and others
- Sending SMS, web Callback
- Flash Web phone
- Instant messenger with video conferencing
- Contacts
- IP IVR (Phone to Phone calling cards module)

- Customizable IVR scenarios (XML), ready to use templates, Caller ID recognition
- One or two stage calling procedures, support for DIDs as access numbers

Softphones – Vippie and SIPLink

- Based on SIP protocol, proprietary VoIP tunnel technology – Making/Receiving calls even behind VoIP blockades
- Full integration with the web interface, quick access to Voicemail
- Sending SMS, Instant Messaging – Chat, presence, file transfer
- Address book shared with the Web Portal interface
- G729, G722, G711 and other codecs

Callback

- Authentication by caller ID or PIN, various methods of realizing Callback
- Support for DIDs as Callback service numbers, SMS and Web triggered Callback
- Embedded IVR system

Resellers system (VSR)

- Multilevel structure, Web-based comprehensive interface
- Support for multi-currency
- Customizable web interface for end-users, integration with the E-Shop

Callshop

- Real-time monitoring and billing, booths visualization
- Standalone and web based versions invoicing

E-Shop

- Various payments processors Credit cards, PayPal, Moneybookers
- Automatic sign-up procedure, recharge by vouchers/PIN
- Basket for selling products online.

SMS module

- Support for HTTP, SIP and SMPP protocols
- Extended routing plan with support for multiple SMS providers integrated with the softphone and the end-users web interface

Software Features

The softswitch part is the main element of the platform, which merges the functionality of the following VoIP architecture elements:

- SIP registrar
- SIP proxy
- H323 switch
- H323 gatekeeper
- SMS gateway

Supported protocols:

- H323 v.2 (H245 v7, H225 v4) with/without "fast start"
- SIP (RFC 3261)
- SMS through SIP, HTTP and SMPP

The main characteristics of the softswitch include:

- Simultaneous and transparent support of SIP and H323 protocols (SIP<->H323 translator).
- Various types of proxy methods e.g. full proxy (with RTP-proxy), signaling proxy and other options, possibility of selecting a proxy method per destination, route or per client.
- Full interoperability with industry standards compatible VoIP equipment (gateways, switches, ATAs, terminals).
- Bidirectional NAT supports both for SIP and H323 equipment.
- Advanced routing system (support for internal virtual prefixes that allows the creation of separate dialing plans for different groups of customer accounts).
- Routing based on prefixes, priorities per routes, depending on allowed voice Codecs per destination.
- Video calling
- Support for failover (rerouting), configurable end reasons initiating fail over, support for priorities.
- Load sharing support – Advanced algorithm taking care of traffic being evenly distributed according to defined percentages for multiple routes.
- Least calls routing.
- Quality routing.
- Internal numbering plans support.

Various authentication methods:

- By IP address
- By caller ID (ANI)
- By H323 ID
- By SIP credentials
- PIN

Flexible methods for call setup data modifications (for clients and/or for destination in the dialing plan):

- Modifying dialed number, adding prefixes/suffixes, wild cards, max/min number length.
- Modifying caller ID/SIP display – Adding prefixes/suffixes, wild cards, max/min number length.
- Defining allowed and/or primary Codecs for clients and for terminators.
- Codecs auto negotiation.
- Import/Export accounts and dialing plan from/to excel or TXT file.
- Settings stored in the MySQL database.
- Scalability is supported due to a cluster configuration, where multiple VoipSwitch servers run connected to each others in what is known as “cluster”, sharing the same SQL database server, thus increasing performance by dividing the traffic among the multiple servers while retaining a central point of management with one main IP address for clients (load balancing).
- Redundancy support for seamless traffic handover in case of the main server failure, the service allows for controlling availability of particular ports (for example SIP, H323 listeners) real-time SQL data backup procedure.

Telephony features:

- Hold
- Music on hold
- Do not disturb
- Call transfer: blind and attended
- Follow me/Find me (based on caller ID of incoming call), sequential or ring to all
- Voicemail boxes with personalized voice greetings for different caller IDs
- Hunt/Ring groups

Unified messaging:

- Voicemail to email with attachment (Mp3)
- Voicemail notification to SMS or email
- Voicemail transcription to SMS
- SMS forwarding (e.g. internal SIP SMS forwarding to external GSM numbers)

System Requirements

Capacity of the platform depends on various factors. The major of them are internet bandwidth and hardware.

With regards to the required internet link please use bandwidth calculator where you can specify the number of **anticipated channels/ports** and the calculator will show the needed bandwidth.

For the system to work in most cases enough is **one public IP address**. **The IP address should be also static**, using private or dynamic IP addresses for the server is not recommended (notice: this requirement is only for the server side, the clients can use private and/or dynamic IP addresses without any limitations).

The VoipSwitch software can run on up to **5 IP addresses per one server**. To each IP can be added all types of listeners i.e. SIP, H323, Callback, Callshop, Voipbox listeners.

Hardware – Recommended are servers based on Intel Xeon processors, example configuration below:

- Dual core or Quad core, in configuration of single or dual CPU
- RAM minimum 2 GB
- HDD 100 GB or more
- A server with a Dual or Quad core CPU and 4GB RAM should be able to handle on average 1000 concurrent calls (proxy mode) in wholesale and retail SIP/H323 calling, similar results have been achieved in real production environment also for Calling cards and Callback.

With higher hardware specification it is possible to increase the load on a single server.
Other method for dealing with higher traffic, recommended also for **failover and backup**, **is to build a cluster** consisting of two or three servers running on one IP address and dividing the traffic among them.

Operating system:

- **VoipSwitch runs on Windows Operating System**. The system will work on any edition of Windows starting from Windows 2000 through Windows XP, Windows 2003, Vista and to Windows 2008, however **it is recommended to use server** edition for example **Windows Server 2003 or Windows Server 2008.**

For the server can be used Web Edition with minimal users license i.e. no additional user licenses are needed.