SR-S5000 1.9.2

Functional Specification

SR-S5000

End Office Softswitch & IP PBX

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1 Introduction

1.1 Document Profile

The document describes the architecture of the SR-S5000 software application, provides a list of its main technical and functional specifications, as well as enumerates hardware and software requirements for correct operation of SR-S5000.

1.2 List of Terms and Abbreviations

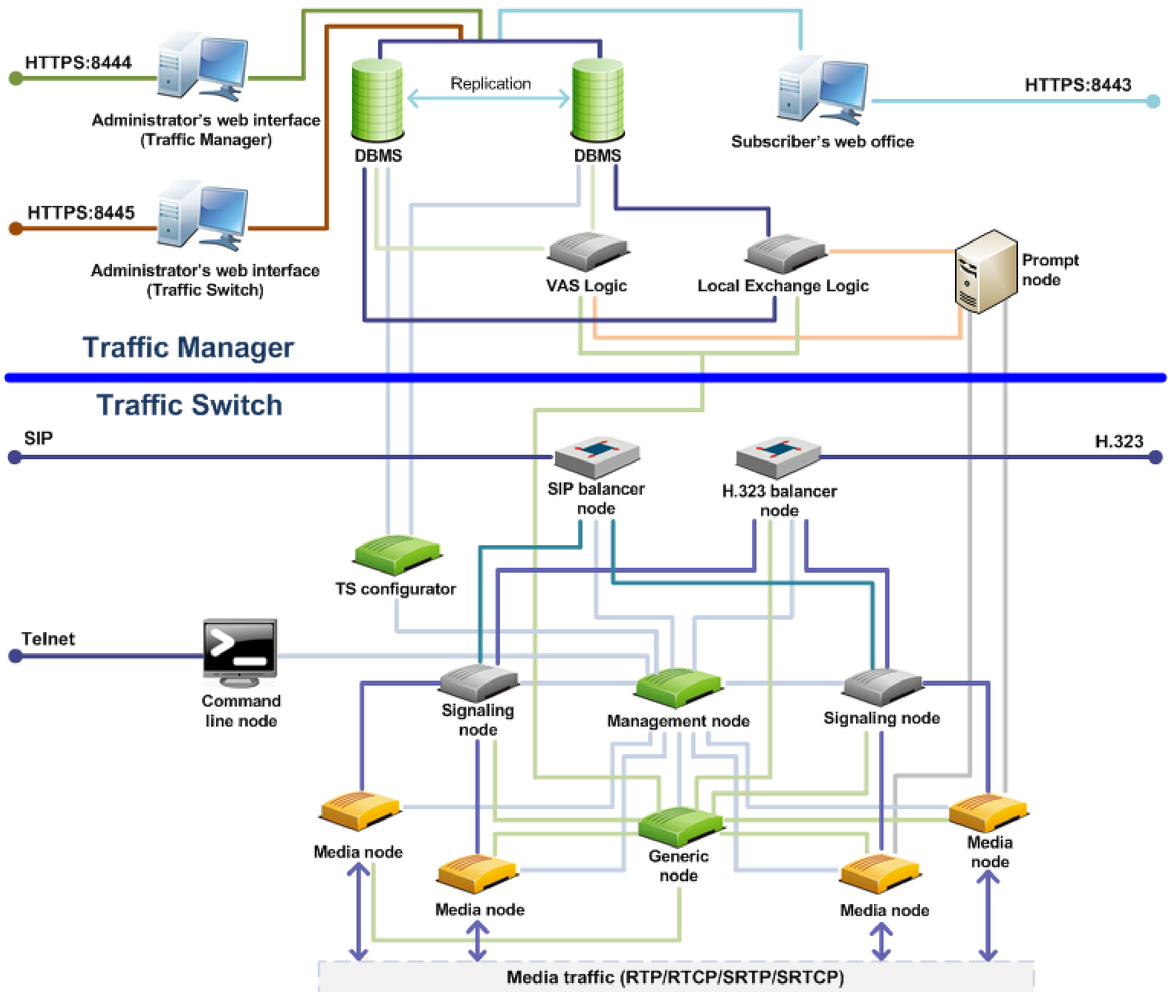
Term / Abbreviation	Explanation
Terms	
Subscribers	End-users presented in the form of subscriber accounts where authentication data for making calls, rules of receiving calls, physical equipment characteristics.
Domains	Logical (virtual) representations of local exchange with individual numbering plan. There are 2 types of domains in SR-S5000: Virtual telecom operator and Hosted PBX. • Domain system – a virtualization mechanism that allows you to create and operate an unlimited number of similar independent logical (Virtual) systems based on a single physical platform. • ROOT domain – parent domain existing in SR-S5000 by default. Inside ROOT domain you can create nested domains in the amount limited by license. • Virtual telecom operator – domain type that features properties of a telecommunications service provider: PSTN number, numbering zone, LI facility connection parameters. Virtual telecom operator can transfer some numbers from its numbering zone to a Hosted PBX domain. • Hosted PBX – domain type that is used for organizing communication within a company. To enable the company employees to make calls to PSTN and also to receive calls from PSTN, several PSTN numbers from the numbering zone of the parent domain are allocated.
Max. number of concurrent connections	Maximum allowed amount of call legs limited by a purchased license. For a successful call, connection should be established on 2 legs.
Numbering zone	A numbering resource, i.e. the totality or part of the numbering options that you can use in the domain.
Abbreviations	
ASR	Answer Seizure Ratio. In SR-S5000, ASR is calculated according to ITU-T Recommendation E.411, paragraph 3.6.3. ASR calculation is used by H.323 balancer and SIP balancer nodes to distribute workload among signaling nodes. ASR calculation is based on data received within the last 5 seconds.
CDR	Call detail record. Set of data fields (call ID, call start and termination time, disconnect reason, etc) used for accounting and billing.
CPC	Calling Party Category.
CPS	Calls per second.
CSV	Comma Separated Values – text format used to represent data in tabular form. Each string in the file is a row of the table. The values of each column is separated by a delimiter, for example, a comma (,), semicolon (;) or a tab symbol. Text values are embraced in double

Term / Abbreviation	Explanation
	quotes ("); if the text value itself contains double quotes – they are represented by two double quotes following each other.
DB	Database.
DBMS	Database management system.
DTMF	Dual Tone Multi-Frequency.
GW	Gateway.
H.323	An ITU-T recommendation that defines the protocols to provide audio-visual communication sessions on any packet network.
LE	Local Exchange.
NAT	Network Address Translation
OoDRPS	Out-of-dialog requests per second (SIP). Out-of-dialog requests are: • all new requests except INVITEs and REGISTERs; • requests with no tag in the To header field. For example, OPTIONS is generally considered to be an out-of-dialog request. However, received within an already established dialog it does not take part in the OoDRPS limitation.
PoD	Packet of Disconnect (in RADIUS Accounting).
RADIUS	Remote Authentication Dial-In User Service.
RBT	Ring-Back Tone.
RFC	Request For Comments.
RPS	Registrations per second.
RTP/RTCP	Real-Time Protocol / Real-Time Control Protocol.
SBC	Session Border Controller.
SIP	Session Initiation Protocol.
SNMP	Simple Network Management Protocol.
SP	Service Platform.
SRTP/SRTCP	Secure Real-time Transport Protocol / Secure Real-Time Control Protocol.
TLS	Transport Layer Security.
TS	Traffic Switch, an application functioning as a session border controller that handles calls under the control of Traffic Manager.
UDP	User Datagram Protocol. One of the core members of the Internet protocol suite (the set of network protocols used for the Internet).
VAS	Value-added service.
VSA	Vendor-specific attribute.
VoIP	Voice over Internet Protocol.

2 System Overview

SR-S5000 (also known as RTU Class 5) is a software implementation of a telephony system that affords a full range of functionality found in traditional PBXs and provides additional capability enabled by the Internet technology. The SR-S5000 application also lets the owner operate a hosted class 5 switch/hosted IP PBX by lending the system's capabilities to other operators and corporate customers.

From the design viewpoint SR-S5000 comprises two functional components: Traffic Switch (TS) and Traffic Manager. Depending on the scope of extended services, Traffic Manager comprises a single required element, which is Local Exchange Logic, or the required element and an optional module referred to as VAS Logic.



2.1 Traffic Switch

Traffic Switch is the switching layer of the SR-S5000 system. Traffic Switch handles SIP and H.323 protocols, and performs two-way conversion of signaling protocols and voice codecs when necessary. Additionally, Traffic Switch is the primary source of call statistics that is analyzed and visualized by means of Traffic Manager, provides monitoring of the system and is responsible for notifications.

Traffic Switch comprises the following functional nodes, each one being an individual process:

- **Management node** ensures distribution of configuration data between other TS nodes, provides centralized control over them, controls licenses and serves as a collection point for SR-S5000 statistics.
- **H.323 Balancer node** serves as the entry point for H.323 traffic. The node handles H.323 registration (RAS) requests and provides load balancing among signaling nodes. When a user (calling device) tries to register with SR-S5000, the H.323 balancer node forwards relevant data to Traffic Manager. Depending on the response received, the registration request is either accepted or rejected. Load balancing is based on the current ASR value of each signaling node.
- **SIP balancer node** balances arriving SIP calls among signaling nodes, and serves as the single exit point for SIP traffic.
- **Signaling node** provides two-way conversion of SIP/H.323 signaling protocols and traffic distribution (load balancing) among media nodes (based on the current CPU load of each media node), as well as between Class 5 control logic modules. Also, the signaling node handles SIP registrations.
- **Media node** handles media flows, functions as an RTP media proxy and performs conversion of voice codecs. The number of media nodes needed in SR-S5000 depends on the anticipated number of concurrent call sessions that involve RTP media proxy operation.
- **Command line node** is a telnet server that allows logging to a switching host using any telnet client.
- **Generic node** ensures interaction between TS and Traffic Manager.
- **TS configurator** transfers configuration data from the TS DB to internal TS tables, as well as monitoring data from internal TS tables to the TS web interface. Monitoring data is viewed and configuration is edited in the **Traffic Switch** category of objects in theTS web interface. The TS configurator interacts only with two TS DBs (primary and failover).

2.2 Traffic Manager

Traffic Manger is designed to make the system function as a class 5 telephony exchange and IP PBX, allowing the provider to render to subscribers both basic and value-added telephony services.

Subscriber calls are handled by the TS part of the application.

Traffic Manger consists of the following parts:

- **Control logics** (Local Exchange Logic and VAS Logic, if available) – software modules that make the whole system function as a telephony exchange and PBX. The module handles incoming and outgoing subscriber calls and renders value-added services (further in this document the server with the VAS Logic component is referred to as the Service Platform (SP)).
- **Database Management System** (**DBMS**) used by both Logics to handle information stored in databases: configuration settings, subscriber accounts, information about provided services, dial plans, as well as gateway, domain and number transformation data. Traffic Manager can contain maximum two DBMSs.
- **Web interface** includes a convenient graphical interface for managing the exchange and PBX operating environment (administrator's web interface) and a subscriber's personal workspace organizer

(subscriber's web office). The web interface for Traffic Switch configuration and monitoring is installed separately.

- **Prompt node** is used to store files (recorded conversations, prompts, etc.) and consists of nginx web server used to read files and Apache web server that allows writing and managing files using the WebDAV protocol.

3 Technical Data and Specification

Carrier-to-Carrier/Carrier-to-Enterprise Connectivity

- Conversion of media codecs:

  – G.729;
  – G.729A;
  – G.729B;
  – G.729AB;
  – G.723.1;
  – G711A-Law;
  – G.711µ-Law;
  – GSM FR;
  – Speex;
  – iLBC;
  – AMR NB;
  – G.726;
  – G.722;
  – G.722.1;
  – G.722.2;
  – Opus.

- Support for and conversion of H.323 and SIP dialects;
- T.38 fax pass-through;
- Transfer to the originator of SIP 181 message sent by the called party;
- Encryption/decryption of SIP calls;
- SIP and H.323 video pass-through using H.261, H.263, H.264 codecs;
- Support of the majority of methods for DTMF transfer, including RFC 2833, SIP INFO, Inband DTMF in G.711 codec (for receiving only), signaling DTMF in H.245, Q.931;
- Support of SIP REFER (RFC 3515), SUBSCRIBE/NOTIFY (BLF) methods.

Supported protocols

The following ITU-T standards are supported:

- H.323 v.2-v.4 “Packet-based multimedia communications systems”;
- H.245 v.7 “Control protocol for multimedia communication”;
- H.225 v.4 “Call signalling protocols and media stream packetization for packet-based multimedia communication systems”.

SIP (the current version of SR-S5000 uses TCP, TLS and UDP as transport protocols for SIP):

Basic signaling protocols:

- RFC 3261 “SIP: Session Initiation Protocol”;
- RFC 3326 “The Reason Header Field for the Session Initiation Protocol (SIP)”;
- RFC 2976 “The SIP INFO Method”.

Privacy:

- RFC 3323 “A Privacy Mechanism for the Session Initiation Protocol (SIP)”;
- RFC 3324 “Short Term Requirements for Network Asserted Identity”;
- RFC 3325 “Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks”;
- SIP Extensions for Caller Identity and Privacy (Cisco proprietary way to handle privacy (Remote-Party-ID).

SIP extensions:

- RFC 3262 “Reliability of Provisional Responses in Session Initiation Protocol (SIP)”;
- RFC 3265 “Session Initiation Protocol (SIP) – Specific Event Notification”;
- RFC 3311 “The Session Initiation Protocol (SIP) UPDATE Method”;
- RFC 3515 “The Session Initiation Protocol (SIP) Refer Method”;
- RFC 3581 “An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing”;
- RFC 3891 “The Session Initiation Protocol (SIP) "Replaces" Header”;
- RFC 4028 “Session Timers in the Session Initiation Protocol (SIP)”;
- RFC 5168 “XML Schema for Media Control” (picture_fast_update is supported);
- RFC 5806 “Diversion Indication in SIP”;
- The Calling Party's Category tel URI Parameter.

Access Authentication:

- RFC 2069 “An Extension to HTTP : Digest Access Authentication”;
- RFC 2617 “HTTP Authentication: Basic and Digest Access Authentication”.

SDP:

- RFC 3264 “An Offer/Answer Model with Session Description Protocol (SDP)”;
- RFC 3551 “RTP Profile for Audio and Video Conferences with Minimal Control”;
- RFC 3555 “MIME Type Registration of RTP Payload Formats”;
- RFC 4566 “SDP: Session Description Protocol”.;
- RFC 4568 “Session Description Protocol (SDP) Security Descriptions for Media Streams”.

DTMF:

- RFC 2833 “RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals” (only in SIP);
- SIP INFO Method for DTMF Tone Generation (Cisco specification);
- Inband DTMF in G.711 codec (detection and/or transparent pass-through);
- H.323 signaling messages:
  – by an alphanumeric H.245 User Input message (as digits);
  – by an alphanumeric H.245 User Input message (as string);
  – by H.245 signal;
  – by a Q.931 Facility message with field Keypad.

RTP/RTCP/SRTP/SRTCP:

- RFC 3550 “RTP: A Transport Protocol for Real-Time Applications”;
- RFC 3551 “RTP Profile for Audio and Video Conferences with Minimal Control”;
- RFC 3711 “The Secure Real-time Transport Protocol (SRTP)”.

Monitoring over SNMP:

- SNMP v1 (RFC 1157);
- SNMP v2c, (RFC 1901);
- GET, GETNEXT, GETBULK requests to get counters and dispatch of SNMP traps using Net-SNMP application in Linux OS. For detailed description refer to http://www.net-snmp.org;

RADIUS:

- Draft Sterman 00 “RADIUS Extension for Digest Authentication”;
- Draft Sterman 01 “RADIUS Extension for Digest Authentication”;
- RFC 2865 “Remote Authentication Dial In User Service (RADIUS)”;
- RFC 2866 “RADIUS Accounting”;
- RFC 4590 “RADIUS Extension for Digest Authentication”.

Network security and SBC functions

- NAT traversal:

  The system automatically detects hosts behind NAT devices. For this, it checks a host address written in the last network packet coming from a device. If it differs from the actual address where the packet comes from, the device is considered to be behind a NAT. In this case the system sends all further packets to the address from which this last packet originates.
  Media flow processing is the same. If media source address differs from the one specified in the signaling messages, the system transmits the media stream to the address from which the opposite stream comes.
  Please note that calls to H.323 terminal can't be placed if NAT router ports are defined dynamically.

- Concealment of the owner's network topology.
- Limitation of incoming traffic: the SIP balancer node can be configured to receive traffic only from trusted realms.
- Caller authentication by IP or username based on data stored in the DB.
- Caller authentication by a set of parameters based on:

  — data stored in the DB.
  — data received from RADIUS servers.

- Limitation of incoming calls by their rate (CPS).
- Limitation of incoming SIP and H.323 registrations by their rate (RPS).
- Limitation of the number of concurrent calls.
- Limitation of the number of out-of-dialog requests per second (OoDRPS).

Call routing

- Rerouting on route unavailability.

Native routing capabilities

- Routing based on the calling/called number;
- Day-of-week and time-of-day based routing;
- Route selection based on CPC and other parameters;

External routing (including external Least Cost Routing systems)

- ENUM-aided routing;
- RADIUS-aided routing;
- SIP 302-aided routing.

Statistics and network analysis

- Display of CDRs meeting user-defined parameters;
- Export of CDRs into a text file (including scheduled export).

Billing

- Single CDR collection point;
- Great number of fields in CDRs for detailed analysis and debugging;
- Generation of interim CDRs to store accounting data on active calls;
- Integration with external billing systems using RADIUS protocol;
- Cisco VSA;
- Authorization of users in the external billing system based on the data provided by SR-S5000.

Number and URI translation

- Flexible number translation options based on regular expressions;
- Separate number translations for routing and SORM LI (on gateways, when calls enter/leave the system, for pre-routing and post-routing);
- Translation of CPC and other call parameters.

Configuration management

- Managing configuration via web interface supporting a flexible system of roles;
- Secure authentication and authorization of the system users', including configurable web password policies;
- Console interface via telnet;
- Provisioning the DB via web API over XML/HTTP(S);
- Exporting data from the DB into CSV files.

Logging and debugging

- System trace logs with selectable information detail level;
- Call simulation;
- Logs of users' actions in the web interface.

Fault tolerance and availability

The fault tolerance of SR-S5000 is achieved due to its modular architecture. It is possible to run a whole set of nodes of the same type that and backup each other.

The fault tolerance of the DBMS is ensured by installing an additional DBMS and configuring replication between DBMSs.

Scalability

The scalability of SR-S5000 is achieved due to the possibility to run several sets of nodes of the same type (on the same server or on different servers) that increase the overall system performance.

It is possible to use several active control logics that can be installed on one or on different servers and to allocate different logics to different domains. The number of subscribers in this case is not limited by the performance of one server or one Local Exchange Logic.

Besides installing nodes on different servers, there is also a possibility to use a separate server for call recording.

Geographically distributed configuration

- Modular design;
- Locations intended to unite geographically close nodes that should interoperate with each other only;
- Dynamic distribution of licenses among locations.

Value-added services (services)

The availability of services in SR-S5000 is assured by the software manufacturer only if their provisioning and delivery to subscribers is carried out in accordance with the tested delivery scenarios.

Services allowed by the LE Logic:

- **Call Hold** – capability to put the caller on hold, for example, in order to make a call transfer or another call;

- **Call Transfer** – redirection of the answered call to another telephone or extension by means of DTMF input or the SIP REFER method;
- **3-party Conference Call** – communication session involving three communicating parties simultaneously;
- **Multiparty Conference Call** – communication session in which multiple subscribers participate;
- **Call Waiting** – notification of a subscriber engaged in a telephone conversation about a newly arrived call waiting to be answered;
- **Call Forward** – redirection of the arrived call to some other number configured by the subscriber:

  — unconditional;
  — on busy;
  — on no answer;
  — on unavailability.

- **Do not Disturb** – call processing feature that prevents calls from reaching the called subscriber (when the LE Logic is used without the Service Platform the DND management is possible by means of the web-based interface only);
- **Black/White Lists** – rejection of calls originating from the numbers that the subscriber included in the black list and letting through calls from numbers on the white list;
- **CLIP/CLIR** – Calling Line Identification Presentation is a service that allows the telephone customer to see who is calling before answering the call; Calling Line Identification Restriction is a service that gives calling subscribers the ability to block the delivery of their telephone numbers from being displayed on a caller identification device;
- **BLF** – Busy Lamp Field. Allows you to see the current status (busy, free or alerting) of the phone number you intend to call;
- **Speed Dial** – ability to use abbreviated numbers (typically consisting of one or two digits) instead of lengthy ones. Abbreviated number identifiers are individually unique for all subscribers (when the LE Logic is used without the Service Platform the speed dialing management is possible by means of the web-based interface only);
- **Call Intrusion** – allows a subscriber to reach another subscriber, even if the latter is busy;
- **Conversation Recording** – allows recording conversations pursuant to recording rules. The rules are based on prefixes of calling and called numbers. The administrator of the system can create, edit and delete recording rules as well as to manage recorded conversations.

Services allowed by the VAS Logic (Service Platform):

- **Fax sending form personal web office (Web to Fax)*** – facsimile transmissions from a personal web office. Supported fax quality grades include: Standard, Fine, SuperFine;
- **Voice Mail** – allows you to receive, leave and keep voice messages in your personal voice mail box as well as to deliver a recorded voice message to the desired email address;
- **Follow Me**** – automatic diversion of the arrived call to the number or several numbers at which the subscriber can answer it at the moment;
- **System/Personal** IVR** – flexible call handling mechanism that allows creation of customized call handling algorithms by means of an illustrative graphical tool;
- **Identity-based Access** – allows calling from any phone with the privileges and rights afforded to the caller by its or somebody else's account as per the provided access credentials;
- **Last Number Redial**** – automatic dialing of the number that was dialed last or the number of the call that came last;
- **Auto Redial**** – unattended repeated redialing of the number that is currently busy or unreachable;
- **Call Back** – service that involves two successive calls – first made to a specialized phone number and then from that number to the destination number with subsequent connection of the two communication links. The Call Back service allows the reduction of long-distance and international long-distance calling costs;
- **Auto Redial with Dial Back**** – unattended repeated redialing of the number that is currently busy or unreachable and calling back the caller when the call is finally answered;

- **Call Pick-up**** – ability to answer the call meant for somebody else and ringing at somebody else's phone;
- **Call Park**** – addition of a newly arrived call to the pool of calls on hold (parked calls) in order to allow call making. Call parking implies the capability to revert to conversation with the caller of the parked call at any time;
- **Save Prompt**** – allows subscribers to make their own recordings of greetings, prompts and other voice messages in form of .wav sound files;
- **Alarm** – pre-configured reminder or alert sent to the subscriber in form of a telephone call made at a preset time;
- **Alarm Settings Query** – current alarm settings delivered in response to a query;
- **CF Settings Query** – current call forwarding settings;
- **Current Time Query** – current system time delivered in response to a query;
- **Service List Query**** – listing of services available to the subscriber;
- **Speed Dial Settings Query** – current speed dial settings delivered in response to a query;
- **Direct Inward system Access (DISA)** – out-of-office subscriber with access to the inner telephone network;
- **Group Call** – single call fork to all phones of the designated group. When the call is answered at one of the group phones all other phones stop ringing;
- **Chat Room** – communication session with limitless number of participants. A multi-point communication session is initiated by inviting new participants to join by means of the web-based interface or by a call to a pre-designated number;
- **Calling Card Platform** – pre-paid card-based telephony service allowing customers who are not system subscribers to make calls with prior PIN-based or number-based RADIUS-aided authorization by the calling number;
- **Multiple Endpoints**** – ability to use simultaneously several terminals associated with the subscriber's account. All operated terminals have identical settings and differ by their unique registration names;
- **Hunt Group** – allows the deployment of a call center. To establish a common call center (CC), a telephone number is defined and several telephone numbers of the CC operators are configured. If all the CC operators prove to be busy when a call arrives, the newly arrived call is put on the call queue until anyone of the operators is free to answer it;
- **Ad-sponsored Call** – allows the service provider to include advertising in subscriber call sessions;
- **Televoting** – allows opinion polls, surveys and vote gathering through counting the number of ingress calls arriving at designated phone numbers. Vote gathering results are presented as bar charts viewable both during and after the voting session.
- **Missed Call Notification** – allows notifying subscribers about missed calls.
- **System Voice Mail** – allows saving voice messages for departments, subscriber groups, etc.
- **Mass calling** – allows organizing automatic dialing of a defined list of numbers in order to play voice messages to the called party or to connect the called party with an agent.
- Some of the services can be additionally managed by means of the voice menu over the telephone set:

  — request/perform call forwarding settings (Query/Set Forward);
  — request/perform speed dialing settings (Query/Set Speed Dial);
  — request/perform alarm settings (Query/Set Alarm);

* – fax receiving and sending with the help of the Web to Fax service and Fax to Email service is guaranteed under the T.38 protocol only.

** – these services require Local Exchange logic for proper operation, for other services third party vendor platforms may be used.

Traffic Manager distinct features

- Connectivity to SIP networks that require registration;
- Domain partitioning including control over the number of both intra-domain and outward calls; control over the number of intra-domain restrictions, support of the Virtual PBX functionality;

- Use of aliases (any subscriber can have a limitless number of additional short phone numbers also known as aliases);
- Virtual numbers for subscribers;
- Control over the personal communication environment afforded by a subscriber's web office with the entry point independent of that for the admin's web interface;
- Centralized system for managing phone terminal settings (auto provisioning), supporting automatic loading of the configuration over TFTP and HTTP protocols.

4 Software Requirements

The system is supplied as a bundle of software applications running on Debian GNU/Linux 7.0 (amd64 Wheezy) with 64-bit kernel and 64-bit operating environment.

The SR-S5000 database server uses the MySQL DBMS.

Supported web browsers:

- Mozilla Firefox starting from 23.x

Other versions of web browsers may not be fully compatible with the web interface.

5 Hardware Requirements

The minimum hardware requirements for SR-S5000 are as follows:

- 8 core CPU.
- 16 GB of RAM.

The recommended platform is **HP Proliant DL360**.

Deployment of the system in virtualization environments is only allowed at the testing stage. For commercial usage it is only permitted to run the system on physical servers. Otherwise, the manufacturer does not guarantee failure-free operation of the system.

If system redundancy is employed, each server used in the redundancy scheme should have at least two network interfaces.

6 Version History

Below are major changes and improvements implemented in previous versions of SR-S5000. For the full list of changes and improvements, refer to Release Notes available on our Help Desk.

6.1 1.9.1 > 1.9.2

- Notifications on problems in SR-S5000 operation are now configured in the Traffic Switch web interface. It also became possible to deliver notifications in form of SMS messages.

Traffic Switch

- Now it is possible to configure encryption/decryption of SIP calls.

Traffic Manager

- The Traffic Manager scalability was improved: you can now use several active control logics (Local Exchange Logic & VAS Logic), and allocate a separate Local Exchange Logic to a domain, which allows you to have a much greater number of subscribers.
- Now it is possible to use a separate server for call recording.
- The logical nodes are now managed by the Phoenix process. This allows starting, stopping, and restarting all logical nodes using one start, stop, or restart command.
- Graceful shutdown mechanism was implemented: you can now configure unloading of Class 5 logical nodes upon completion of calls.
- Now it is possible to make calls through a registered gateway without replacing Destination number with the gateway registration name.
- The format of recorded conversation files can be configured now in the web interface (**Call recording** > **Common settings** page).
- To provide users of Mayak Mobile application with more detailed information about their contacts, we implemented the possibility to use vCards. To upload subscriber vCards, open the **Value added services** tab in subscriber account settings. To upload external contacts' vCards, open the **Subscribers** > Contact **groups for Mayak Mobile** page.
- To provide ROOT domain subscribers who use Mayak Mobile application with the possibility to get a contact list of all subscribers of this domain, we made it possible to create a Virtual group in the ROOT domain (**Subscribers** > **Contact groups for Mayak Mobile** page).
- Now it is possible to view detailed information about the amount of allocated and used objects and services of a domain you are logged in and its child domains (**Basic configuration** > **Licenses** > **Objects usage information** page).
- Now it is possible to make calls over the internal protocol (NULL) without proxying media traffic.
- A new service, **Mass calling**, was created. The service allows organizing automatic dialing of a defined list of numbers in order to play voice messages to the called party or to connect the called party with an agent.
- The **schedule** functionality was extended: a special schedule can now be configured for specified dates, the same schedule can be applied to several rules (routing, call forwarding, agent's schedule, etc.), the schedules are created based on **templates**, which makes it faster to create and edit schedules.

6.2 1.9.0 > 1.9.1

- All SR-S5000 components now run on a 64-bit platform. System operation on a 32-bit platform is no longer supported.

Traffic Switch

- SIP messages can be transferred over TCP.

- Traffic Switch includes two new nodes: SIP balancer node and TS configurator.
- Traffic Switch has its web interface that is used to configure SIP balancer node, H.323 Balancer node, to perform monitoring of Traffic Switch and configure SR-S5000 disconnect codes.
- Mechanism for setting limits for SIP and H.323 traffic is more flexible.
- The System is capable of transferring SIP 181 responses "Call is being forwarded" from called device to calling device working over SIP protocol.
- RFC 4028 standard is supported to keep the SIP session alive by sending periodic re-INVITE or UPDATE requests.

Traffic Manager

- The System sends push notifications to the Mayak Mobile application by means of the APN (for iOS devices) and GCM (for Android devices) services. The push node is no longer used.
- A **ToS** (Type of Service) header can be specified in the endpoint profiles settings.
- When searching for originating gateway, the System can use not only an IP address but a port as well.
- A special timer was added to disable logging at all levels except for Fatal, Error, Warning and Info.
- CDRs contain fields with numbers resulted from billing translations.
- Two new pages **List of subscribers endpoints** and **List of endpoints registrations** with filtering option were added to the web interface. Using filtering it is possible to display endpoints behind a particular gateway. Information about registered equipment (data from the **User-Agent** headers of registration requests) is also displayed next to the endpoint registration IP address.
- It is possible to pick up calls of other subscribers using their aliases.
- The **System IVR** and **DISA** services contain a new parameter that defines how long the System will wait for a subscriber to dial a last digit before it decides that the input is completed.
- Calls limitation through the **DISA** service becomes more flexible. You can now restrict calls through DISA to system services only.
- **System IVR**:
  - contains a new block that provides a caller with two options: either to select a voice menu item or to dial an extension.
  - the **Match Input String** block was improved: it is possible to set prompts and input timeout.
  - the **Authenticate** block was improved: it is possible to configure greeting and error prompts, input timeout and maximum number of input attempts.
  - Blocks **Disable service package** and **Enable service package** are supported in logic.
- It is possible to limit the amount of phone numbers that can be called within one scenario step of the **Follow Me** service.
- If there is no available agent in the main queue of the **Hunt Group** service, the call does not wait a specified time before going to the additional queue.
- It is possible to view calls statistics within Hunt Group instances of the domain and configure automatic export of statistics.
- The Opus codec is supported.

6.3 1.8.1 > 1.9.0

Traffic Manager

- Users can now create domain profiles. Profiles allow for flexible and convenient management of the intra-domain restrictions.
- The domain creation procedure was simplified. Now with the help of domain profiles users can automatically create intra-domain services.
- Users can now create access groups for subscriber services and include these access groups into packets of services. This change significantly simplified the process of administering access to subscriber services.
- A new VAS "Missed Call Notification" was added. Thanks to this service subscribers can now receive notifications about missed calls via email and the Mayak Mobile application.

- A new VAS "System Voice Mail" was added. It processes incoming calls received in the organization and stores voice messages.
- Auto Attendant services (all-system and personal) were deleted. During the update to 1.9.0 all previously created accounts of these services will be automatically converted into VAS "IVR" and "Personal IVR".
- Users can now configure web interface access rights.
- The System now saves CDRs on the disk and only then moves them to the DB. Users can now configure CDR rotation.
- Users can now manage the following VAS:

  — DISA;
  — Chat room;
  — Virtual Fax;
  — System Voice Mail;
  — Ad-sponsored Call;
  — Call Back;
  — Group Call;
  — Identity-based Access;
  — Televoting;
  — System IVR.

- API commands for working with subscriber services were added. Users can now configure numbers and other parameters of subscriber services.
- API commands for working with audio files and voice messages were added.
- An additional interface was implemented to create and delete Virtual PBXs, add/remove additional domain profiles and external numbers ("Protocol for VAS VPBX Accounting Systems").

6.4 1.7.4 > 1.8.1

Traffic Manager

- For convenient work with equipment configuration, profiles were added. A profile is designed to group the equipment settings and can be used in multiple gateway/subscriber accounts.
- Now it is possible to configure individual equipment settings for each subscriber's endpoint.
- Now it is possible to disable some pages in the subscriber's web office.
- The conversation recording feature is enhanced. Now work of the feature is defined by conversation recording rules.
- The service Hunt Group is added.
- It became possible to configure every domain to interact with its own RADIUS server.

6.5 1.7.3 > 1.7.4

Traffic Switch

- The original duration of DTMF signals can be saved and transmitted now when they pass through the system.
- Now it is possible to transmit key frame within video stream as per RFC 5168, to enhance video quality.

Traffic Manager

- Domains are now divided into two types: Virtual telecom operators and Hosted PBX.
- The administrator can delegate some user account control functions to numbering zone administrators and operators.
- The Call Hold service now works independently of Call Transfer.

- All services being configured via the web interface, are now created using the service wizard.

6.6 1.7.2 > 1.7.3

Traffic Manager

- To ensure safety of Class 5 Component operation, newly created passwords are checked for complexity. Now it is also possible to generate complex passwords automatically.
- A prompt server is implemented that stores audio files with prompts.
- Automatic call authentication without registration.
- Total amount of audio files created in a domain is limited (available only to the administrator of ROOT domain). Audio files include added or customized welcome prompts, voice mail messages and recorded conversations.

6.7 1.6.0 > 1.7.2

Traffic Switch

- Support of codecs G.722, G.722.1 (including Annex C), AMR-WB/G.722.2.
- Ability to dispatch SNMP traps.

Traffic Manager

- The major portion of VAS is enabled by a standalone software module VAS Logic (Service Platform). A Class 5 deployment may comprise several Service Platforms which ensures correct forwarding of the call through various Class 5 services.
- When Class 5 Component includes the Service Platform, there appears the possibility to use multiple subscriber terminals associated with the same subscriber account (for details, see Technical Data and Specification).
- A single CDR is formed upon each call instead of several EDRs as it was before.