MANUAL ANSWITCH V6

Manual anSwitch V6 System Description

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Contents

1 Aarenet VoIP Systems

1.1 General

Aarenet delivers turnkey systems, which are tailored to specific customer needs. The systems meet all requirements for professional VoIP products in excellent quality and availability. Each system is assembled and tuned according to the customer's technical, administrative and regulatory requirements. Turnkey systems cannot simply be purchased and deployed "off the shelf": The great technical complexity and the "interpretable" standardization of the interfaces between the individual components require optimal integration of these components into the overall system. Aarenet works directly with the development teams of various suppliers to enable the integration of these products in the required quality.

1.2 Overview

An Aarenet VoIP system consists of the anSwitch V6, a firewall or session border controller and additional equipment such as a provisioning or conference server. The connection to the public switched telephone network (PSTN) is done via SIP trunks or SS7 gateways (dark grey area). Customer equipment (light gray area) is connected via broadband IP (Internet Protocol) connections either directly as in the case of SSIP phones or soft clients or indirectly for existing analog or ISDN equipment via appropriate gateways. Integration into existing or new customer management or billing systems is done via suitable interfaces (e.g. XML).
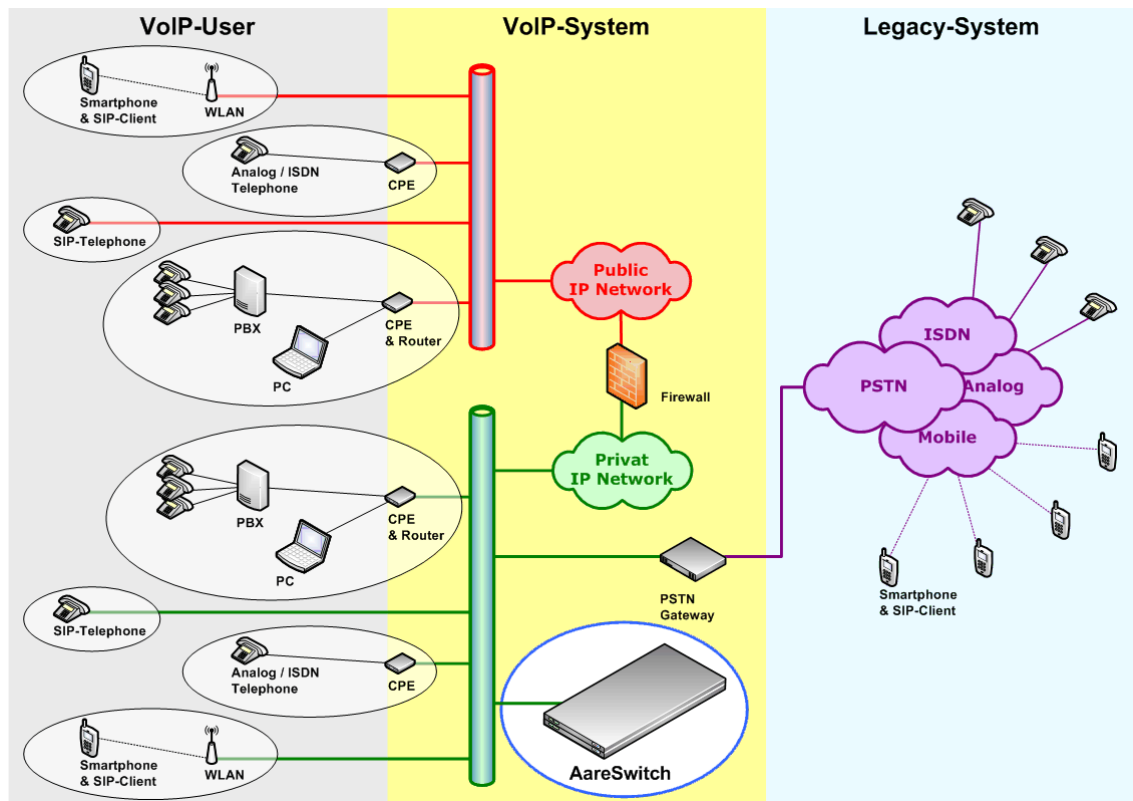
1.3 anSwitch V6

The core element of an Aarenet VoIP system is the anSwitch V6. It ensures the cooperation of the different components and provides the central functionality of the system. The anSwitch V6 is a Class-5 soft-switch, it provides the end user (subscriber) with the same quality, variety and availability of services as conventional Time Division Multiplex (TDM) systems. The subscriber does not notice any difference to the conventional systems, as the anSwitch V6 supports all functions such as emergency call routing, abbreviated dialing, */# stimulus procedures, VAS numbers, carrier selection, number portability, fax services, data service, digit blocking and many

other services. Furthermore, the system offers functionalities that cannot be offered with conventional systems, such as web access for the end user or virtual PBX functionality.

1.4 Quality of Service

The call quality with Voice over IP VoIP depends on many different parameters. Appropriate measures are necessary to ensure high quality. The last mile to the customer is particularly critical, because in many cases only a limited bandwidth is available there. The main factors influencing call quality are transmission delays, packet losses and call compression.

In order to have sufficient bandwidth available for telephony, it must be ensured that there is always sufficient bandwidth available for telephony on lines that are also used for data traffic or other services such as IP TV. This can be ensured, for example, with VPN connections or other procedures that allow bandwidth reservation. Transmission delays are usually not a problem in modern networks. Packet losses, on the other hand, can severely disrupt voice transmission. They occur, for example, in the case of overloaded connections or transmission paths that are not properly coordinated (duplex, half-duplex, etc.). The call compressions used depend on the available bandwidths. Call quality deteriorates the higher the compression. Whenever possible, no compression should be used; this not only provides the best call quality, but also ensures reliable fax transmission.

1.5 Terminals

Gateways, SSIP phones and SIP soft clients can be used as end devices. VoIP gateways enable the use of existing equipment such as analog devices, ISDN phones or PBXs with multiple base or primary lines. SSIP phones replace traditional analog or ISDN phones and SIP soft clients are used on PCs or mobile devices (e.g. iPhone).

1.6 Interconnection

A selection of special carrier gateways ensures the connection with other networks. They enable any connections between SIP, ISDN/DSS1 and SS7. These gateways are available in various configuration levels and ensure the connection to the PSTN and other networks. Today, interconnections are more and more realized via SIP trunks - i.e. without transition to TDM - directly IP-based. It should be noted, however, that SIP trunks are often not functionally equivalent to SS7 transitions.

1.7 Security and availability

Security is of course of the utmost importance. For example, the authentication of the end devices is fully encrypted. Voice transmission can also be encrypted, provided that this does not conflict with legal regulations. The VoIP system itself is protected by firewalls or session border controllers. By using redundant, high-quality components, the required operational reliability is achieved with the 99.999% availability usual for carrier systems. In order to meet even higher requirements, the system can also be set up site-redundantly without any problems.

1.8 Provisioning, Operation and Maintenance

In Aarenet VoIP systems, configurations for various customer devices can be generated directly and loaded fully automatically into the customer devices. The loading of the configurations into the customer devices can be done before their delivery or via download after their installation and commissioning.

Aarenet VoIP systems offer customer service staff of the provider the possibility to access the customer equipment directly from the system administration and to query operationally relevant information. This allows end-to-end monitoring of the system and thus also to locate possible errors at the customer site.

Furthermore, special system tools support the analysis of problems and thus facilitate troubleshooting.

1.9 Additional Services

Value added services (VAS) such as conference services, call centers or unified messaging can be easily coupled with the Aarenet systems. Via special interfaces, data can be automatically exchanged and synchronized between the systems as required.

1.10 Integration with Existing Customer Management Systems CMS

Thanks to their flexibility, Aarenet systems can usually be integrated into existing customer management or billing systems without major effort. Data is exchanged automatically in both directions via suitable interfaces.

1.11 Customized Systems

Each VoIP system is tailored and adjusted to the customer's needs and the existing network infrastructure. Due to the modular design and flexibility, new, additional features can be added if necessary. The number of end customers or parallel calls can be varied in large ranges.

2 System Deployment Scenarios

Carrier systems are essentially divided into three deployment scenarios: Class 5, Class 4 and Virtual PBX or Centrex.

2.1 Class 5 Systems

Class 5 systems are by definition systems to which end customers are connected, also known as local exchanges. The anSwitch V6 is a Class 5 carrier system with the corresponding end user functionality. It can be directly connected to different interconnection carriers and supports individual call routing, if necessary for each destination and source number.

2.2 Class 4 Systems

Class 4 systems are by definition systems to which no end customers are connected, also known as transit exchanges.
The anSwitch V6 is not a typical Class 4 system. However, it supports Class 4 functionality to a certain extent. Calls do not necessarily have to be terminated in the system, but traffic can also be routed from one carrier to another carrier according to various criteria.

3 anSwitch V6 VoIP Switch

The anSwitch V6 serves as a central switching, control and management element for modern NGN telephone networks. It ensures the cooperation of the different components in the network and is responsible for the central functionality of the system. It switches the calls in private or public IP networks and connects to the traditional public switched networks (PSTN) via SS7 gateways. This allows operators to make full use of their legacy infrastructures - be they private or provider-owned facilities - while reducing operating costs. The anSwitch V6 provides everything needed for an end-to-end "carrier-grade" solution.

3.1 Architecture

The anSwitch V6 has a modular design, which allows optimal adaptation to the respective customer requirements. In a small, non-redundant system, all system components are installed on one server. Fully redundant carrier grade systems consist of at least two ServiceCenter, one ConfigurationCenter and two LoadBalancers. For larger systems, additional ServiceCenters are then added. With this architecture, site-redundant systems can also be implemented without any problems.
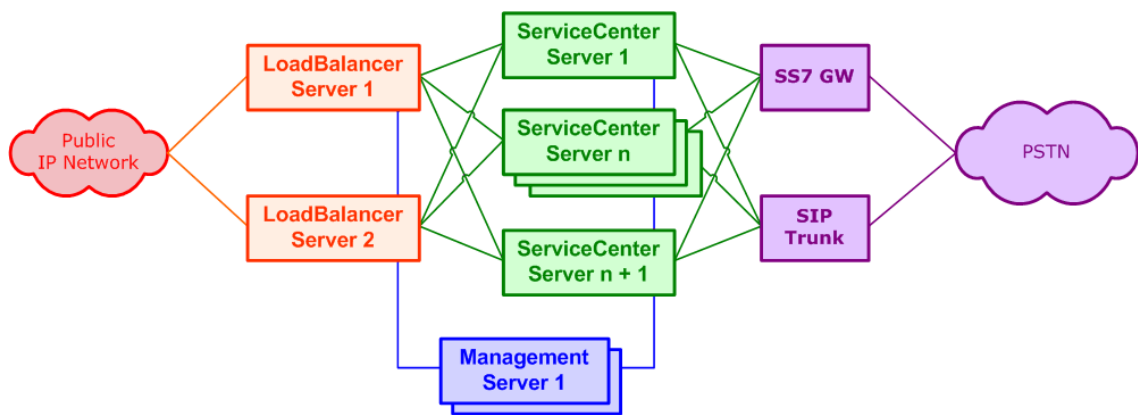
3.2 SW Components

The software is written in Java and technologies like Ajax and Soap are used. The software is completely modular. The most important components are briefly described below.
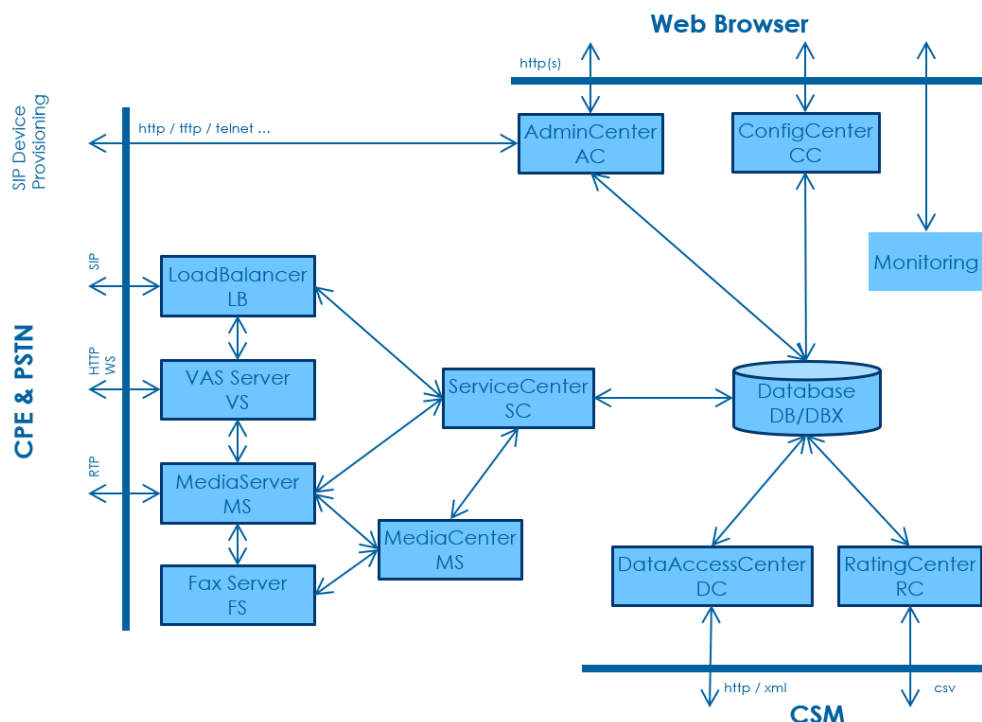
3.2.1 LoadBalancer Component

The LoadBalancer receives and sends SIP messages from the terminals to the service centers. It ensures that messages are always routed to the service center that controls the call. For new calls, it selects a ServiceCenter and takes the configured load balancing into account.

In redundant systems, there are at least two LoadBalancer: One LoadBalancer is active and distributes the incoming calls. The other is hot-standby, always ready to step in as soon as the active LoadBalancer fails.

3.2.2 ServiceCenter Component

This component controls and manages the calls. It authenticates the callers, determines the routing of the calls and creates call data records (CDR) so that the calls can also be billed accordingly. The ServiceCenter components are designed redundantly so that availability can be increased on the one hand and the overall system is scalable on the other: If the number of subscribers grows, new instances can simply be added.

3.2.3 MediaServer Component

The MediaServer fulfills two tasks: On the one hand, it is used as a "media proxy" to mediate the media streams (the "voice channel") between end devices if they cannot exchange media data directly (e.g. if they are behind a NAT).

On the other hand, the MediaServer also serves as a media endpoint in that it can send voice messages or signaling tones to end devices as a media stream.

3.2.4 MediaCenter Component

While the MediaServer only processes the media streams, the MediaCenter handles the signaling messages and thus represents a virtual terminal. It receives calls, processes them and instructs the MediaServer to play or record the desired media files. The MediaCenter contains, for example, the Voice Mail with functions such as IVR Interactive Voice Response, multilingualism, callback, sending messages by mail, voice storage, etc.).

3.2.5 ConfigurationCenter Component

The configuration of the system is done via the web browser. The various data can be displayed and modified in a window system. An authorization concept allows different access levels to be defined so that, for example, a first-level support team has read access to configuration data or can modify it to a limited extent.

All system-relevant configurations can be made via the ConfigurationCenter, these are essentially:

- User accounts, clients and basic settings
- Emergency call routing
- Routing to and from PSTN, rulesets, routing tables, numbering plans
- Gateways
- Customer Accounts, Addresses
- Features on Level Account
- Rating
- Lawful Interception
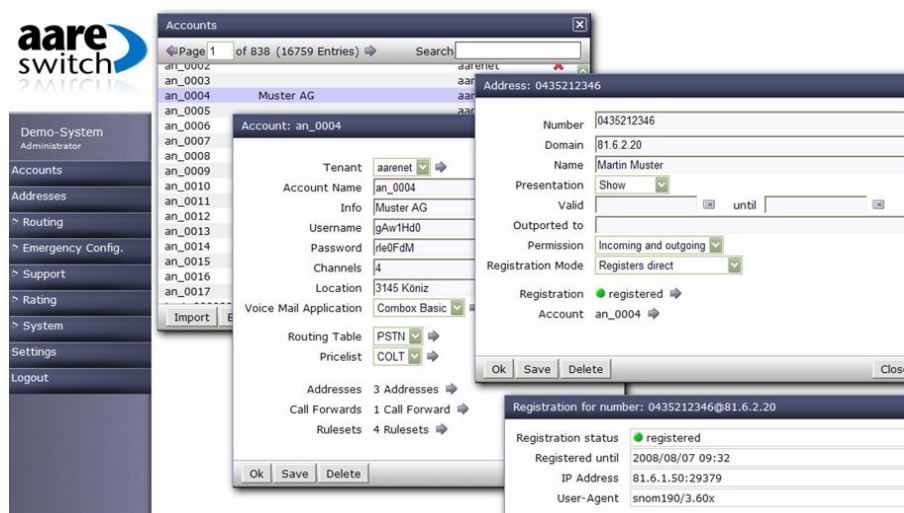
The ConfigurationCenter also provides import and export functions so that even large amounts of subscriber data can be quickly and easily created or modified in an Excel document.

Furthermore, a range of support tools is also available, which allow rapid and targeted problem analysis in the event of a malfunction.

3.2.6 AdminCenter Component

Via the AdminCenter, the end user can view and modify his account-specific data via PC or PDA. There is an authorization level on account level (e.g. for the person responsible for telephony in a company) and one on number level. The following functions can be changed or viewed:

- ‣ Call lists
- ‣ Call forwarding permanent/when busy
- ‣ Call forwarding
- ‣ Parallel call
- ‣ Substitute target (fallback)
- ‣ Quiet before the phone
- ‣ CLIR
- ‣ Enable and disable voice prompt for calls with suppressed number
- ‣ Activating and deactivating lock sets
- ‣ Viewing fee information

3.2.7 RatingCenter Component

The RatingCenter is a real time rating engine. It calculates call costs for all calls made based on price lists stored in the system. Call prices can be calculated in anSwitch V6 for different rating entities like account, client and system. There can be any number of price lists in the system. Thus, even a single customer (account) or closed user group (client) can be assigned its own price list.

Gateways can also be assigned to price lists. This allows costs to be shown and charged for calls made via a specific gateway or to a specific carrier.

A	B	C	D	Peak		Off-Peak		Night/Weekend	
Destination	Type	Timeband	Postrating	Minute	Setup	Minute	Setup	Minute	Setup
Switzerland Default	PSTN & ISDN	CH		0.0100	0.0110	0.0050	0.0550	0.0000	0.0000
Switzerland-Aarau-Olten	PSTN & ISDN	CH		0.0100	0.0110	0.0050	0.0550	0.0000	0.0000
Switzerland-Basel	PSTN & ISDN	CH		0.0100	0.0110	0.0050	0.0550	0.0000	0.0000
Switzerland-Bern	PSTN & ISDN	CH		0.0100	0.0110	0.0050	0.0550	0.0000	0.0000
Switzerland-Biel-Neuenburg	PSTN & ISDN	CH		0.0100	0.0110	0.0050	0.0550	0.0000	0.0000

Different time bands are also supported for the rating (peak, off-peak, night/weekend). By means of import, charges not shown in the price list, such as INA Offline and connection costs of ported mobile subscribers, can also be taken into account. The formats of the reports CDR and XDR can be configured very flexibly, customer-specific.

3.2.8   DataAccessCenter Component

External systems like a customer management system get access to configuration and running data like active calls, charges, redirections and registrations etc. via this component. The data is transferred in XML format via the HTTP protocol and can be individually adapted to the needs of the peripheral systems used with the help of XSLT scripts.

In addition, several different access accounts can be configured so that certain information is only accessible to selected systems and the confidentiality of the data is maintained (such as charges).

3.3   HW Components

Standard hardware is used without proprietary components. Aarenet usually uses Dell Power Edge servers with dual power supply and mirrored hot plug disks (RAID 1). Linux (Rocky 9) is used as operating system.

3.4   Maintenance, Monitoring

All important system components are continuously and automatically monitored. If a malfunction is detected, the system can trigger automatic alarms via SMS or email. The status of the individual system components can be seen at a glance and detailed data of a system component can be queried or statistics can be generated in the simplest way.

The system also provides tools that allow preventive monitoring and verification of the system status.

3.5   Expansion levels and Licensing

3.5.1   System Sizes and Scalability

Thanks to the modular design of the anSwitch V6, there are theoretically no limits to the size of a system. The maximum number of subscribers that can be operated on a VoIP switch depends mainly on the performance of the hardware used. With the standard server types used by AareNet, 5000 subscribers can be supported per VoIP switch. A system with 10000 subscribers therefore needs two servers, if it is redundant three.

A significant advantage of this modular concept is also the fact that only one additional server with 5000 subscribers is ever required for a redundant system, rather than two fully expanded systems.

3.5.2   Expansion Stages
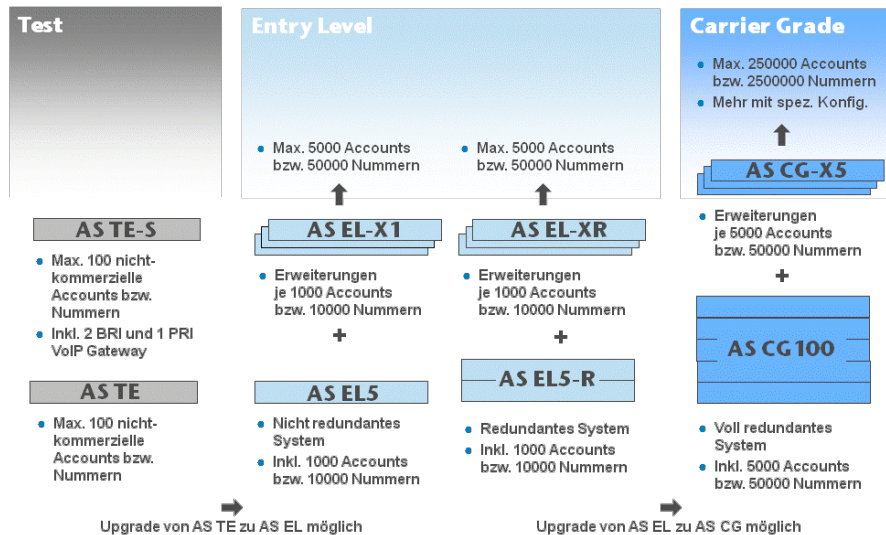
3.5.2.1   Test System

Test systems are used by customers for evaluation purposes and as test or pre-production systems. Test systems support the full system functionality. All required SW components are installed on a server. These systems are equipped with non-productive licenses and allow the operation of a maximum of 100 participants.

3.5.2.2   Entry Level Systems

Entry level systems are available with an expansion of 1000 to 5000 accounts. They are available redundant with two servers as well as non-redundant with one server. These systems cannot be expanded further, but if more than 5000 accounts are required, it is necessary to change to a carrier grade system.

3.5.2.3   Carrier Grade Systems

Carrier grade systems have a minimum expansion of 5000 accounts. Such systems can be expanded to a system size of several hundred thousand accounts without interruption. Such systems are always built redundantly and can also be built redundantly. They are usually designed and built according to the specific requirements (e.g. additional servers for voice mail, etc.).

3.5.3   Licensing Model

Accounts and numbers are decisive for licensing. Furthermore, a few optional functions such as VoiceMail and CPECenter are subject to licensing.

3.5.3.1   Account

An account is a registered SIP client. This can contain one number as in the case of an SIP phone or several numbers as in the case of a PBX.

3.5.3.2   Number

A number is an active number configured in the system, which is individually configured and individually routed.

3.5.3.3   System Size

The size of an anSwitch V6 is always defined by a combination of accounts and numbers. For example, a system is licensed for a maximum of 20000 accounts or a maximum of 200000 numbers, whichever number is reached first.

3.5.3.4　　Base Number

Basic numbers are required if DDI number ranges are allocated when using PBXs, of which only a part is used. This is often the case in countries with open numbering plans or e.g. with customers like hotels, which map the floors in their numbering plan and thus may use only a small part of the number ranges. Such unused base numbers can be configured in anSwitch V6 and will be routed to a predefined number (e.g. main number). No licenses are required for such base numbers.

3.6　　Performance Features

Call related functionality:

- Rule Based Routing
- Subscriber Based Routing
- Domain Based Routing
- Gateway Based Routing
- Destination Based Routing
- Least Cost Routing (static)
- Source Based Server Group Routing
- Multiple Route Choice
- Basic Announcements
- Dial Rules
- Numbering Plan based Routing
- Regular expressions based Number Rewriting
- Emergency Routing
- Number Blocking
- Calling Line Presentation (CLIP)
- Calling Line Presentation Restriction (CLIR) Default and Call by Call
- Call Forwarding Unconditional (CFU)
- Call Forwarding Busy (CFB)
- Call Forwarding Fallback (CFF)
- Flexible routing on multiple gateways (balanced, fallback etc.)
- Abbreviate dialing
- 8xx Number Translation
- Ingress and Egress ANI Phone Number Normalization
- Call Blocking/Barring
- Free Calls
- Subscriber Authentication
- Redirection Execution
- Support of Local Number Portability
- Support of Lawful Interception
- Do Not Disturb
- Call Waiting, Call Hold
- Call Transfer (ECT, BCT)
- Call forking
- Credit Limit per Account (Top Stop)
- Web Access for End Users
- Extension dialing plan
- Hunt groups (sequential, random)
- Simultaneous ring

- Call Inquiry
- Call forward (All, Busy, No answer)
- Speed dialing
- Call restriction / permission per subscriber
- Do not disturb / scheduled forward to voicemail
- Music on hold
- Caller ID
- Call details (completed / missed / unanswered)
- Multi IVR capability via optional module
- Voice Mail
- Private Numbering Plans
- Call Distribution/User Groups (global, linear, etc.)
- Presence (Busy, Meeting, Break, Holidays, etc.)

However, the functionality (call forwarding, toggling, conference, etc.) of a SIP system also depends on the functionality of the SIP clients/CPEs used.

SIP related functionality
- RFC3261: SIP: Session Initiation Protocol
- Registrar Server
- MD5 authentication
- Several Numbers per Account
- Integrated Media Proxy and MediaServer (Announcements)
- NAT traversal support
- SIP Trunk
- Support info (SIP traces, problem log)

Operation and maintenance
- Web HTTP / HTTPS Interface for Configuration
- Multi Tenant Support and Multi Tier Structure
- Centralized Configuration
- Centralized Provisioning
- Centralized CDR's
- Real-Time Rating Engine
- Prepaid Application
- Provisioning Interface
- Configurable event alerting via email
- SS7 support via gateways
- High availability via N + 1 redundancy
- SOAP interface
- SNMP Access
- Built in diagnostic tools

4　3rd Party VoIP System Components

4.1　Firewall and SBC

Unless it is operated in a private network that is not accessible from the outside, a VoIP system must usually be protected with a firewall. Depending on the size of the system and its use (public or private), this must be more or less powerful. If different IP networks are to be served, SBC (Session Border Controller) functionality is also required. The anSwitch V6 already supports several typical SBC functions like NAT support or protocol conversion.

4.1.1　Firewall

Aarenet offers firewalls as an option for its VoIP systems. These are also available as redundant clusters and meet the highest requirements.

4.1.2　Session Border Controller

A session border controller transfers VoIP calls from one IP network to another IP network. This includes signaling and voice (RTP). Various additional functions can also be performed:

‣ Complete conversion of calls from one network to another network

‣ Firewall, only exactly defined VoIP calls are allowed.

‣ Protocol conversion, VoIP calls are transferred from one protocol to another.

‣ NAT functions, this means the transition between private and public IP networks.

‣ Lawful interception, if all calls go through the same SBC, lawful interception can also be realized there.

4.1.3　SBC Functionality Integrated in the anSwitch V6

An anSwitch V6 system provides the following functions:

‣ Protocol converter, there is the possibility to intercept the peculiarities of different SIP terminals with the anSwitch V6 so that they understand each other.

‣ NAT function, the anSwitch V6 concept of the media server allows SIP clients to operate behind NAT in private networks.

‣ Lawful interception, the basic functionality for lawful interception is included in the system.

4.2　SS7 Gateways or SIP-Trunks

The transition from a VoIP network to the PSTN or other VoIP networks is made either by SIP/SS7 gateways or SIP trunks.

Intercarrier connection via SS7 is one of the most important functions of a VoIP system. It is also possible to operate multiple transitions at different locations (e.g. in different countries) and to different networks or different carriers. The system is capable of switching calls along the optimal or most cost-effective path. This becomes necessary, for example, if certain destinations cannot be reached via a carrier or can only be reached with insufficient quality or at non-competitive prices.

4.2.1 SIP/SS7 Gateways

For the SIP/SS7 variant, Aarenet uses the IMG products from Dialogic, which have been optimally integrated into the Aarenet systems. These are available in versions with 1 to 4 E1 and 3 to 24 E1 and can be operated in redundant clusters.

4.2.2 SIP-Trunks

In the case of the SIP-trunk variant, it must be carefully checked in each case what functionalities are supported by the corresponding SIP trunk provider. SIP trunks are usually used for interconnecting different VoIP systems.

4.3 Client Gateways

In principle, there are no restrictions regarding the client gateways to be used. However, it is strongly recommended to use only tested devices. Aarenet provides a current list of tested client gateways on request. Aarenet offers specially selected, tested and partly with anSwitch V6 specific additional functionality (e.g. Advice of Charge, Overlap Dialing) provided Client Gateways for the residential and business area.

4.3.1 Business Gateways

Aarenet uses Patton-Inalp equipment as standard.

These devices are available with analog and ISDN interfaces in the following configurations and variants:
- Analog versions from 2 to 32 ports
- ISDN versions from 1 to 5 BRA or 2 to 8 voice channels and 1 to 4 PRI or 15 to 120 voice channels
- High Precision Clock version for PBXs with DECT systems
- QoS on the last mile
- Integrated IP Router
- Integrated ADSL and SHDSL modems

These devices are supported by the CPECenter for configuration, auto-provisioning and monitoring.

4.4 SIP Phones

In principle, there are no restrictions on the devices that can be used with SIP phones. However, it is strongly recommended to use only tested devices. Aarenet provides a current list of tested SSIP phones on request. Aarenet offers selected and tested devices. These are available in the following variants and designs:
- Corded SIP phones in 5 comfort levels

▸ IP DECT and WLAN mobile devices in various comfort levels

▸ Depending on comfort level Entry Level to full featured Business Phone functionality

▸ Business devices expandable with additional keypads

▸ Line and freely programmable function keys

▸ POE, integrated switch, etc.

5 System Specifications

5.1 Technical Data

System:
- Open architecture
- Standard SIP control of media gateways
- SIP interconnect to remote soft-switch
- High availability, N+1 sparing, routing redundancy
- DTMF transport type negotiation
- T.38 fax relay, fax bypass, modem bypass
- RFC4040 (clear mode)

Software:
- Linux OS (Rocky 9)
- Java application
- SQL, Regex
- Ajax, Soap

Hardware:
- DELL PowerEdge
- 1U Rack-Mountable Chassis

Environmental conditions:
- Operating Temperature: 10º C to 35º C
- Operating Relative Humidity (non-condensing twmax=29C): 20% to 80% non-condensing

Power supply:
- Two hot-plug high-efficient 502W Energy Smart PSU

5.2 Expansion Stages

5.2.1 10-002A Entry Level anSwitch redundant supporting SIP

- 2 server softswitch
- Maximum capacity 100'000 accounts/numbers
- 1'000 concurrent calls
- 72'000 BHCA Busy Hour Call Attempts

5.2.2 10-003A Carrier Grade Base anSwitch redundant supporting SIP

- 6 server softswitch
- Maximum capacity 500'000 accounts/numbers
- 3'000 concurrent calls
- 216'000 BHCA Busy Hour Call Attempts

5.2.3 10-004A Carrier Grade anSwitch redundant supporting SIP

- 8 server softswitch
- Maximum capacity 1'000'000 accounts/numbers
- 5'000 concurrent calls
- 360'000 BHCA Busy Hour Call Attempts

5.2.4 10-005A Carrier Grade anSwitch redundant supporting SIP

- 12 server softswitch
- Maximum capacity 1'000'000 accounts/numbers
- 10'000 concurrent calls
- 720'000 BHCA Busy Hour Call Attempts

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