AARENET ANSWITCH ENTERPRISE VIRTUAL PBX FEATURE LIST

AARENET VIRTUAL PBX

The Aarenet PBX VoIP System is built up as an independent satellite system. For best performance it is interconnected with an Aarenet Class 5 VoIP Switch. Aarenet Virtual PBX offers outstanding benefits to the voice provider and their customer. Voice providers receive a scalable highly available platform including extensive tools for provisioning, operations and support. To customers the solution of Aarenet features a balanced set of functionalities with a comprehensive offering of devices

PBX

- PBX administrator configuration through Web Portal
- PBX Dashboard with main settings and status
- Defining PBX name and description
- Defining of associated public numbers
- Flexible PBX public prefix number to dial external connections
- PBX limitation of PBX extensions
- PBX limitation of simultaneously busy channels
- Allocation of incoming DDI to private numbers
- Definition of the PBX mode for incoming call handling: Timetable based, night/weekend based, permanent night/weekend based
- Multiple time bands per PBX possible
- Defining PIN in order to call specific destinations
- Define TopStop per PBX or per user
- Set TopStop threshold level(s)
- Alarming e-mail if TopStop threshold is reached
- PBX Timetable for incoming call routing for weekdays and definable holiday calendar
- PBX contacts with short number and destination number
- Call list of all incoming and outgoing calls
- Conference rooms for up to 10 participants, protected through access PIN
- External dial in for conference room

PBX MEMBER

- Web-based Self Care Portal
- configure outgoing CLI number (CLIP)
- Suppress own number (CLIR)
- Set individual language
- Set behavior if call received when busy
- Individual call forwarding CF
- Individual VoiceMail Box VM:
    - List with received messages
    - Connect to VoiceMail Box without PIN
    - Send messages (audio file) by e-mail
    - Allow call back
    - Visual indication of VM message on phones
- Individual call distribution ACD
- Individual interactive voice response IVR
- Phone provisioning for the own private number
- Private contacts with short number and destination number
- Conference rooms for up to 10 participants, protected through access PIN
- External dial in for conference room
- Department handling and management

UNIFIED COMMUNICATIONS ANCONNECT

- Busy status indication of users
- Video conferencing suite
- Chat, group chat and messaging
- Desktop-sharing and file-sending through messaging

CALL FORWARDING AND AVAILABILITY

- Call Forwarding unconditional (user, VM, announcement)
- Call Forwarding busy (user, VM, announcement)
- Call Forwarding no reply (user, VM, announcement)
- Call Forwarding not reached (user, VM, announcement)
- Do not disturb
- Reject anonymous calls

CALL DISTRIBUTION / CONTACT CENTER

- Unlimited call distributions
- Predefined call distributions algorithms (linear, cyclic)
- call distributions with configurable delays for calling the destinations
- Agent login / logout
- Blacklist incoming call filter
- Integration with anContact (*)

IVR FEATURES

- Unlimited IVR message tree
- Menu supported configuration messages and associated actions
- Record IVR messages through connected SIP phone or upload audio files
- Actions per message (direct, wait, not reached, delay
- Actions after message (repeat, forward, start call, end call)
- Actions during destination ringing phase (waiting music, repeat message)
- DTMF interactions (repeat, forward, start call, end call, dial digit)

PHONE FEATURES

- Call hold
- Call query / toggle
- Call brokering and transfer
- Transfer with announcement
- Transfer without announcement
- Speed Dial
- Multiple public lines and identities

SIP PHONE SUPPORT & PROVISIONING

- Centralized provisioning of SIP phones via Web Portal
- Phone configuration via Web Portal
- "an IP-Phone" IOS and Android smartphone client (optional)
- “an IP-Phone” Windows and macOS desktop client (optional)
- Supported SIP phones:
  SNOM: D315, D335, D385, D715, D717, D735, D785
  Yealink: T21P E2, T40G, T53W, T54W, T57W
  Grandstream: GRP 2612, 2613, 2614, 2615. GXP 1615, 1625, 1628, 2130, 2135, 2140, 2160, 2170
- Supported DECT System:
  Yealink: W60B, W80B(*), W70B(*)
- Phone keys configuration of supported SIP phones
- PBX and private phonebook propagated to the phone
- QR code provisioning of smartphone client
- Zero-Touch auto-provisioning via manufacturer's redirection server for SNOM, Yealink, Grandstream and Poly devices.
- Provisioning via the configuration of a config-file
- Integration of SIP devices through the phone provisioning tool.

SMARTPHONE CLIENT

- IOS and Android supported
- QR code secure auto provisioning
- Selectable skins
- Own app on Apple App store and Google Play store to be realised as a separate project
- Full-Service support
- Push message support
- Seamless in-call WIFI2NET and NET2WIFI handover
- Local recording
- In app call conference
- Video calls (H.264 codec support)
- Calls routed through mobile network
- Message waiting display
- Busy state indication for favorites
- Rich messaging incl. picture and document sending
- Outgoing call number rewriting
- Codec settings and provisioning
- OPUS, iLBC, G.711, G.722, GSM-FR codec support
- Selectable P-time
- Business features incl. video calls
- SIP SIMPLE messaging
- Transport protocol UDP, TCP supported
- Support Callkit on IOS
- Busy Lamp Field for favorite contacts
- Web Portal self-care access
- Native contacts and PBX telephone book access
- Configurable tabs / icons
- Auto push of new features to registered endpoints
- Multi language support
- Selectable ring tones
- Call history
- Selectable behavior for incoming GSM calls
- SIP log
- 3G/WiFi behavior selection
- Echo suppression
- Noise suppression
- Bluetooth support

ROUTING

- Definable dialing rules
- Definable blocking rules
- SIP message modification rules
- Definable number normalization rules
- Rule creation supported with regular expressions-based Regex rewriting
- SIP Zones for serving dedicated IP subnets
- Gateways for the interconnection with PSTN peering devices
- Rule Based Routing
- Domain Based Routing
- Gateway Based Routing
- Destination Based Routing
- Source Based Routing
- Websocket CSTA interface (ECMA-323) Call Control API

RATING

- Multiple PriceList support
- Assigning different PriceList to Gateways, Tenants and PBXs
- Real-Time rating engine
- Billing Exports in .CSV

WEB BASED PORTAL

- HTTP/HTTPS for configuration, provisioning, support and maintenance
- User access with different roles to assigned resources
- multiple languages support
- LDAP authentication service

SUPPORT AND MAINTENANCE

- Centralized support via Web Portal
- SNMP support, SNMP Traps, AareSwitch MIB
- Built in Diagnostic Tools for SIP and Media
- Support of Local Number Portability
- Configurable Event Alarming via Email
- Call list of all incoming and outgoing calls with access to SIP trace and media analysis
- Access to all log data

SIP RELATED

- RFC3261: SIP: Session Initiation Protocol
- Registrar Server
- MD5 authentication
- NAT Traversal Support
- Connectivity through SIP Trunk

MEDIA RELATED

- Integrated MediaControl
- Media proxy
- Audio Codec transcoding
- In band announcements
- DTMF automatic mode for RFC2833 and SIP-INFO

SYSTEM DEPLOYMENT

- Legacy redundant server deployment
- Deployment on customer server farms virtual environment
- Deployment on Cloud Platforms

SYSTEM ARCHITECTURE

- Multi-Tenant Support and Multi-Tier Structure
- DataBase organized in Organizational Units OrgUnit
- OrgUnits provide parameter inheritance from parent to subsequent child resources

TECHNICAL DATA

- Open architecture
- IPv6 enabled
- High availability, N+1 sparing, routing redundancy
- Load sharing
- Standard SIP control of media gateways
- SIP interconnect to remote soft switch
- DTMF transport type negotiation
- T.38 fax relay, fax bypass, modem bypass
- Open REST API for configuration, provisioning and billing

SOFTWARE

- Linux OS (CentOS)
- Virtualisation by Docker Containers
- SW languages: C++, Java
- SQL database technology

SUPPORT AND MANAGEMENT

- Support and Service Levels
  - Standard: 8/5 plus updates
  - Premium: 24/7 plus updates plus support
  - Custom: flexible
- Enhanced System monitor and External alarming to centralized NOC

LICENCING FLEXIBILITY

- Account licenses
- Number of concurrent calls license