PBX Data Sheet

Version 2.0

Product categories

pbxnsip offers products in three categories:

- The embedded versions are usually sold to OEM partners, who include the pbxnsip software with other products like PSTN gateways, routers, firewalls and other hardware products. The product focuses on small memory print, limited file access and performance optimizations. The pbxnsip software will be already preinstalled when the end customer receives the device. Typically, it will also be loaded with a standard configuration, dial plan, extensions. Possible the appliance is sold together with other preinstalled devices, for example phones or terminal adapters.
- The hosted version target service providers. Customers usually receive a phone and a QoS-aware router and subscribe to the service that includes voice and data. The PBX typically runs on blade servers, and it has usually at least one IP address. Those installations are typically loaded to the maximum possible number of extensions and calls per hardware. The PBX typically runs in standard operating systems that can easily be managed by the service provider. The PBX is usually embedded in other applications around it, which take care about provisioning, billing and general customer interaction.
- The CPE versions typically run on a PC located on the customer‘s premises. The focus is here on simple manageability. The CPE versions are also able to deal with several IP addresses, for example private and public IP addresses. Sometimes customers use two PC for redundancy purposes. Most of the installations are performed by system integrators that help the end customer to get the system working and perform special maintenance tasks that can not be done by the customer himself.

Architecture

The pbxnsip software uses a „back to back user agent" (B2BUA) architecture. This architecture is similar to the classical PBX architecture, which makes it simple to implement the features known from most business users.

All SIP request are processed by the PBX software. This has the advantage, that even poor SIP implantations do not affect the interoperability of other connected devices.

The media flows through the PBX. Thanks to the recent advantages in CPU performance, this is no more a limiting factor for real-life deployments. The advantage is that the PBX can detect one-way audio or disconnected devices and automatically disconnect calls. It also opens the possibility to offer features like call barge-in, listening-in and teaching modes.

The PBX supports several domains. This makes it possible to separate departments and customers (hosted environments). Every domain may have one or more domain administrators. The number of calls, the number of extensions and other resources can be restricted on domain-level.

The software is available for Windows and selected Linux operating systems (e.g. RedHat and SuSE).

NAT Support and Multiple Addresses

The PBX is able to run on hosts that have several IP addresses. It is possible to define per port which address the port should be bound to. This makes it possible to run the PBX on both private and public IP addresses, which makes it possible to connect both internal and external workforce to the same PBX without the need for a separate near-end NAT solution. Users can connect to the PBX from home, hotels and other locations with Internet access and fully participate in the PBX traffic.

For far-end NAT support, the PBX automatically detects devices that need periodic NAT binding refreshes. By combining this with the media relay, the PBX is able to connect devices even behind symmetrical NAT.

For trunks to Internet service providers that do not support NAT, the PBX offers the possibility to allocate a public Internet identity using the STUN protocol. This feature can be used with widely available NAT routers that do not block SIP/RTP traffic.

Security Features

The PBX uses standard TLS encryption to protect the traffic between connected devices and the PBX: SRTP is being used en encrypt voice traffic. The key negotiation uses the SDES method. When only one call leg supports SRTP, the PBX performs transcoding between the secure and the insecure channel. By using the „sips" scheme, the user can enforce that the communication is kept private on both call legs.

The PBX also allows secure access to the built-in web server. This way, administrator and users can keep the traffic private and securely exchange passwords.

The PBX offers the possibility to automatically provision passwords using the https transport layer to endpoints. By keeping track on the number of provisions, the PBX can make sure that a password is sent only once to the endpoint, so that later access to the provisioning server will not send the password to other parties.

The PBX respects the routing table of the network configuration and supports the use of multiple IP addresses. By using VPN users can encrypt their voice traffic using a separate network identity to the PBX.

Cell Phone Integration

To deal with today's realities, the PBX has a dedicated support for cell phones. Along with the other settings for an extension, the PBX allows the association of a cell phone number with an extension.

When a caller reaches an extension, the PBX may include the cell phone

in the group of registered devices for the extension. The delay for including the phone is programmable, so that the cell phone is only included if the extension does not pick up.

The user may also decide to get a call to the cell phone when a new voicemail message arrives.

Calls from the cell phone to the auto attendant lead to a special menu that gives the user the possibility to place outbound calls from his extension. This feature hides the cell phone identity of the caller and it makes it possible to bypass cell phone charges (e.g. for international calls). Alternatively, the caller can directly go his personal voicemail box, where other options are available.

The outbound calling feature is also available from other outbound calls (e.g. 0800 numbers), so that employees can make calls from airport lounges and other places with access to toll-free numbers.

If an extension is busy, the cell phone users can also get an automatic call-back as soon as the extension becomes available again (camp on for external callers).

Mobile Workforce Support

To enhance the support of the mobile workforce, the PBX supports hot desking. Users can log in to a phone using a special code with their PIN code and redirect all telephone traffic to this phone. This feature is independent from specific phone features, so that this feature is available on all SIP-capable devices, including Wifi mobile phones.

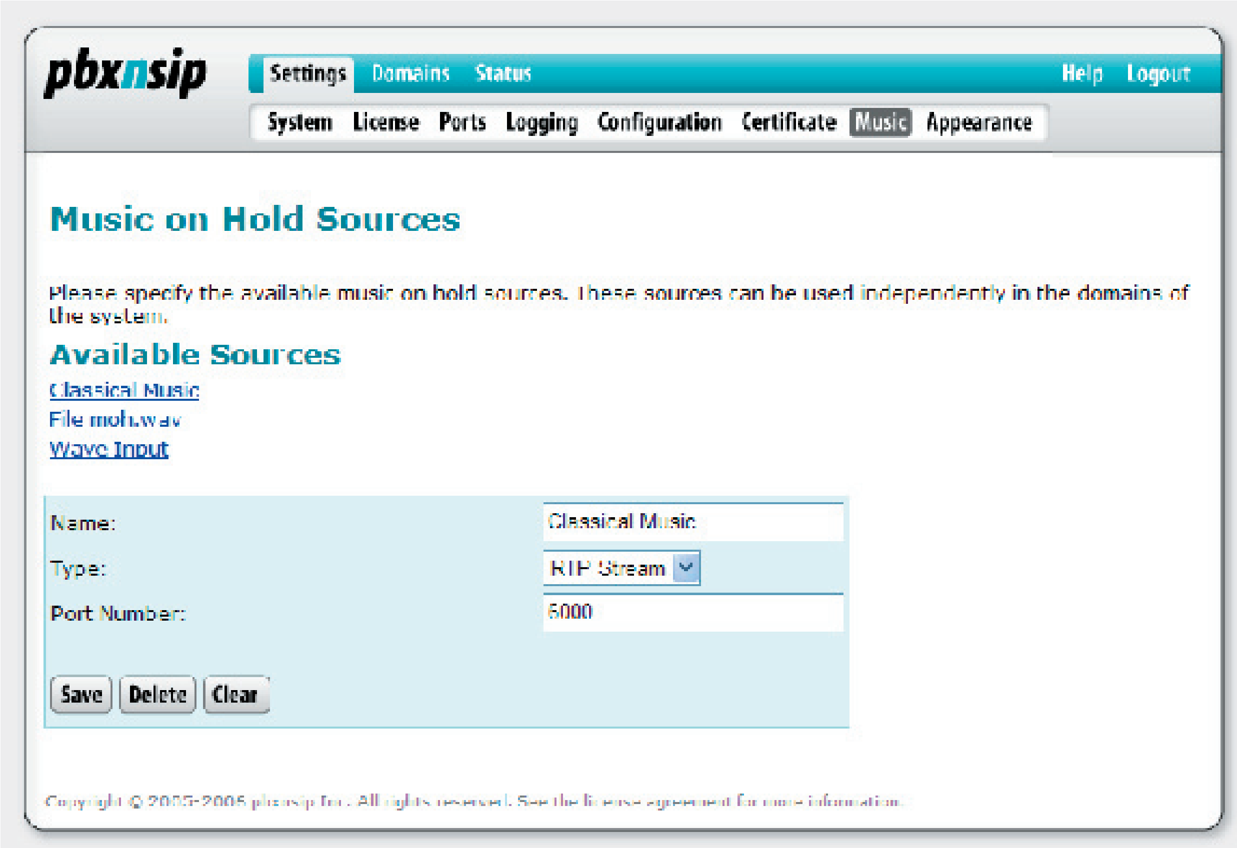
The PBX also allows that every extension has its own time zone and language. This makes it possible to run central corporate PBX for organizations that spread across different time zones and have employees speaking different languages.

Supervision Features

Traditional PBX are offering features for call supervision. The pbxnsip PBX offers the following features:

- Call Barge In: The supervisor can jump into a call and establish a three-way party call. The interception is announced with a tone. This feature is typically used by secretaries that try to „save" colleagues from talkative callers.
- Teach Mode: In this mode, the supervision also establishes an instant conference, but only the internal extension can hear what the supervisor is saying. This mode is useful for training purposes when the outside caller should not be aware that someone is helping an agent.
- Listen Mode: This mode is similar to the teach mode, but neither side can hear what the supervisor is saying, and there is not audible indication. This can be either used for call supervision without the knowledge of either person or it can be used in parallel with instant messaging, where the agent receives instant messages with hints.

These features can be combined with the recording feature of the PBX. Jumping into a call does not stop the recording of the call. Depending on the mode and the recording point, the PBX will record also the speech of the supervisor.



Plug and Play Support

To simplify routine work during the setup of endpoints, the PBX generates configuration files on the fly. The PBX supports TFTP and HTTP/HTTPS provisioning. This feature is available for selected standard business phones.

The plug and play support includes provisioning of time zones (including daylight savings) and dial plans for the end devices. For devices that automatically assign lines to keys, the PBX can provide the number of available lines to the endpoint. The PBX is also able to deal with the different flavours of indicating intercom traffic and alerting tones. Address book information can also be provisioned from the user's personal address book to the endpoint.

Additional devices can be added by using XML-based configuration files which contain configuration templates.

The PBX itself can also be configured automatically. This makes it possible for large network providers to maintain a central provisioning system for CPE devices.

Trunking Interface

The PBX uses trunks to connect to the public switched telephone network (PSTN) and to other SIP-capable devices. The SIP traffic on those trunks is kept separate from the SIP traffic with the extensions, so that features like transfer or call pickup are transparent on the operators side. This dramatically reduces the complexity in connecting to service providers and to standard PSTN interfaces.

The PBX allows a separate codec preference on the trunk side. If necessary, the PBX performs media transcoding to translate between the different call legs.

The PBX supports routing of calls to ENUM locations. If performs the necessary steps to resolve the destination of the call and is able to send the call directly to the other party. Inbound calls on ENUM addresses can be terminated using unbound trunks.

When trunk calls fail, the PBX is able to perform failover to other trunks that are listed in the used dial plan.

Trunks support the emulation or CO-lines known from traditional PBX. This way, the traffic on trunks can be limited to a certain number of calls. It is also possible to control the number of inbound and outbound calls on a trunk. Extensions can subscribe to the state of trunks and display the connected parties (for example, on LED).

DID routing and Dial Plans

Every account on the system can be associated with one or more DID numbers, so that calls may bypass the auto attendant and directly go to the extension. Trunks can use the Request-URI or the To-header to determine the destination of the call. A powerful pattern matching algorithm is available, e.g. for using parts of the SIP-header for routing purposes.

For outbound calls, the caller-ID presentation can be controlled on per-trunk bases using settings of the calling extension. Extensions can be programmed to send anonymous calls out. The PBX supports several methods for caller-ID presentation and privacy indication.

Each domain can have several dial plans. Each dial plan may have several match patterns and replacement rules. Simplified dial plan rules can be used to set up telephone-number based dial plans. For each extension the domain administrator can define which dial plan should be used. This makes it possible to assign e.g. international dialling permissions to specific extensions or avoid sending FAX over Internet-based routes.

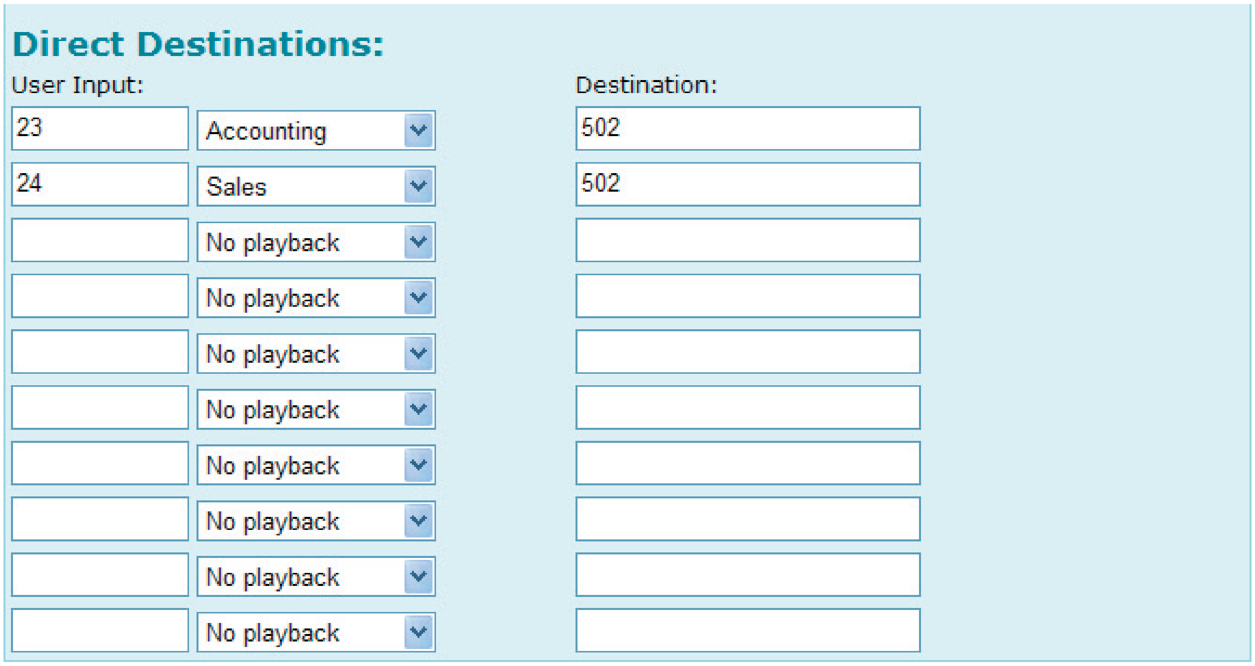
Music on Hold

The PBX automatically inserts music on hold when an extension puts a call on hold. The music can come from a standard WAV audio file or from a RTP stream. RTP streaming makes it possible to use external tools to generate life audio streams. In Microsoft Windows environments, it is also possible to use the audio input jack to provide music on hold (e.g. for connecting standard CD/MP3 players and radios).

Paging and Intercom

The PBX supports both unicast and multicast paging. While unicast paging is suitable for small paging groups or specialized equipment (like SIP-enabled overhead paging equipment), multicast mode is suitable for large groups.

Intercom initiates a two-way audio call sp that the other party does not have to pick up the handset. If the number of lines for the extension is limited, the PBX checks if the destination is busy and if that should be the case, offers camp on.



PBX features

Auto Attendant

The auto attendant supports one or two languages. Callers can decide on the language by pressing a DTMF tone when reaching the auto attendant. The language will be kept for the further ongoing conversation.

Direct destinations are available to emulate extensions. There are several standard texts available to announce the destinations (e.g. „For accounting, press 123‟).

The auto attendant supports day/night mode. The day/night mode may have several duty periods per day and it is possible to define holidays. The day/night mode may also be set manually. The programming can also be used for other accounts like hunt or agent groups.

The dial-by-name feature supports parallel searching for first and last name. If sufficient numbers have been collected, the auto attendant generates a menu where the caller can select the extension number.

It is possible to define extensions that have the permission to override the DND mode of an extension (e.g. secretary). It is also possible to define which number can not be dialled from a auto attendant. A domain may have several auto attendants.

When a call from an anonymous call hits the auto attendant, the PBX may intercept that call and ask the caller for the name. Then the extension will receive a call with the recorded name and the user can decide if the call should be put through. This mechanism can also be enabled for all calls coming from an outside like to fight the increasing threat of SPIT and unsolicited sales calls. For each extension, the PBX maintains a list of „black list‟ and „white list‟ address book entries. When callers call from a white list, the PBX never intercepts the call. Callers in the black list are always intercepted.

Conference Mixer

The conference mixer is a simple, dial able account that mixes the audio streams of the conference participants. It performs a speaker detection to minimize the noise on the conference floor. The entering and leaving of a conference participant is announced by a tone which is mixed into the conference.

The entry to the conference can be protected by a PIN code. Each domain can have several conference rooms.

Mailbox

Each extension may have a mailbox. A PIN code may be used to protect the access to the mailbox. A special prefix can be defined to allow direct access to the mailbox without having to wait for a timeout on the extension.

When listening to mailbox messages, the mailbox user may stop, restart, rewind and fast forward the audio stream using the DTMF keys of the connected phone. Envelope information can be read out on demand. If the caller left a caller-ID, the PBX is able to call that person back from the mailbox menu.

Mailbox messages are stored in a compressed format. Mailbox messages can be forwarded to other extensions after listening to a message. They can also be sent out as Email-attachment.

The PBX sends message waiting indications to registered SIP endpoints.

It is possible to use an external mailbox system. For this purpose, the PBX uses a trunk to an external SIP-enabled mailbox system.

Waiting Queues

The PBX supports agent groups which are used for serial processing of incoming callers.

The PBX uses the music on hold input when callers are in the waiting queue. Per queue, up to ten voice recordings can recorded, which are mixed into the music on hold stream in a round-robin fashion. The first recording is used as introduction prompt and is being played back even if there is no one else in the queue.

Extensions may log in and log out for the participation in the agent groups. The agent queue limits the number of agents which are called at a time and uses a pseudo-random algorithm to pick the agents that are receiving the calls. Other agents are gradually included into the ringing process until the call connects.

If a timeout is being defined, the call can be escalated to an internal or external destination. Callers can use DTMF to jump to direct destinations.

The PBX may indicate a distinctive ring tone for calls coming from an agent group.

Agent groups may use the day/night feature like the auto attendant.

Agents may be giving a recovery time after a call to prepare for the next call and to give them the opportunity to log out. In this silence period, they are not considered for calls from an agent group. The PBX generally does not consider agents that are busy with a call that is active on the PBX.

The web interface can be used to show the status of the queue. A SOAP interface is available that posts the queue status to external sources.

Hunt Groups

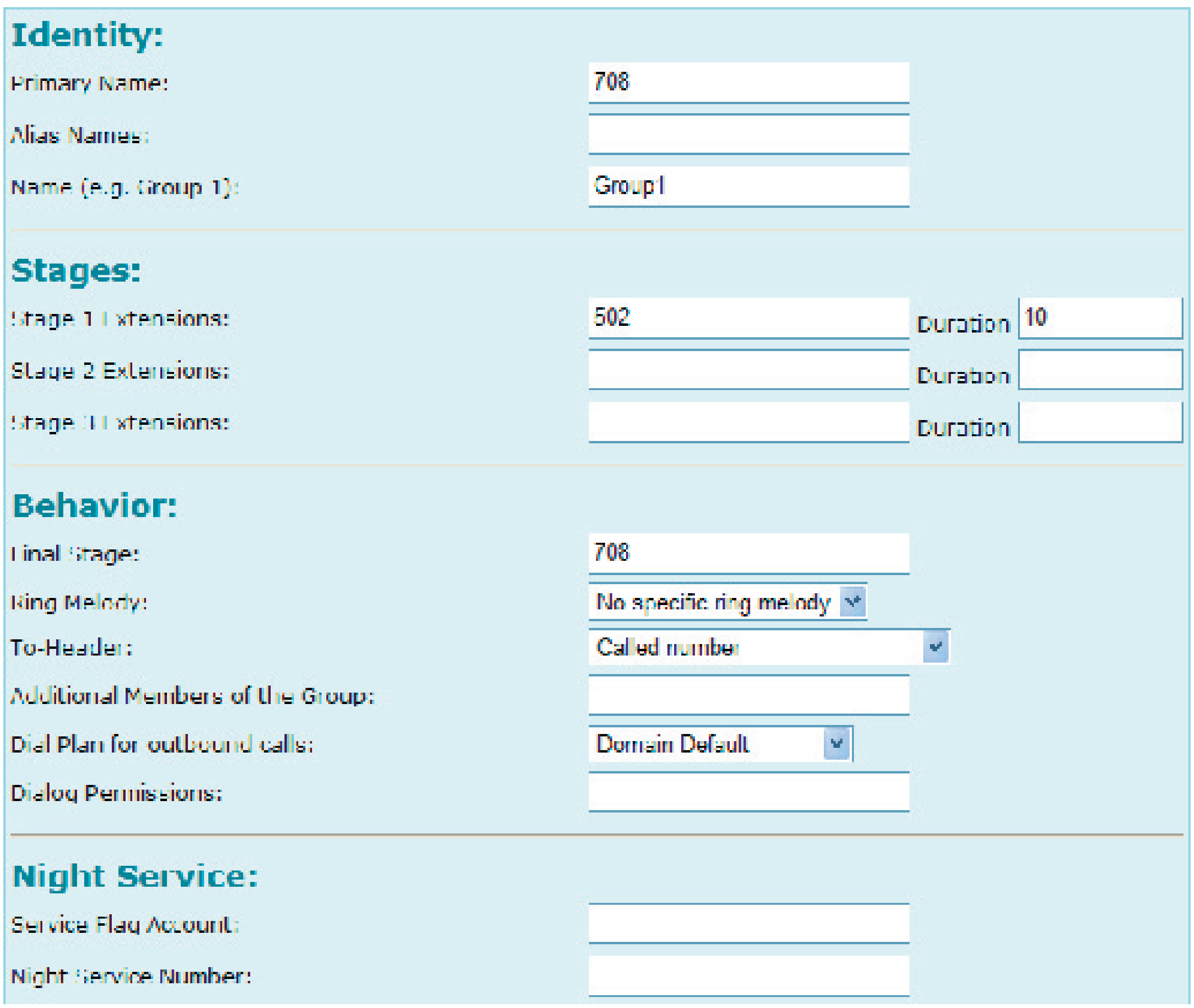
While agent groups line calls up in a serial fashion, hunt groups perform a parallel processing of calls. The PBX supports up to three hunt group stages.

Every hunt group stage ring several extensions. The duration is programmable per hunt group stage. If all hunt group stages fail, the call can be escalated to an internal or external destination.

As with the agent group, distinctive ringing and day/night mode can be programmed.

Custom IVR Dialogs

If necessary, customers can create their own IVR dialogs. The PBX supports flexible pattern-matching schemes. It is also possible to use an external application server for process user input or the caller-ID to determine the destination of the call.



Start	From	To	Duration
2007/01/24 16:00:09	Jim Smith (501)	Jim Smith (501)	00:22 M
2007/01/24 18:54:43	Freddy Quinn (507)	Carl Calera (506)	
2007/01/24 19:12:54	Freddy Quinn (507)	Carl Calera (506)	
2007/01/24 21:47:57	Klara Überhaß (503)	Carl Calera (506)	00:05
2007/01/24 21:49:00	Klara Überhaß (503)	Carl Calera (506)	
2007/01/24 21:49:58	Klara Überhaß (503)	Freddy Quinn (507)	00:24
2007/01/24 21:50:59	Carl Calera (506)	Klara Überhaß (503)	00:16 M
2007/01/24 22:03:45	Klara Überhaß (503)	Carl Calera (506)	
2007/01/24 22:10:39	Carl Calera (506)	Klara Überhaß (503)	
2007/01/24 22:10:48	Carl Calera (506)	Klara Überhaß (503)	00:04
2007/01/25 21:55:46	Karl Klammer (502)	Freddy Quinn (507)	00:03

Call Recording

Selected calls may be recorded, even if SRTP is being used.

The call recording is selected on per-extension basis. Within the extension, the administrator can select if calls from agent groups, hunt groups or direct extension calls shall be recorded. For outbound calls, the administrator can decide if internal and/or external shall be recorded.

The recording can be written as WAV files to the file system (offline recoding). Alternatively, calls can be sent to an external recording entity by using standard SIP traffic. This makes it possible to use standard SIP phones for real-time quality monitoring. It also makes it possible to employ specialized recording equipment.

Address Book

Each domain and each extension may have an address book. Address book can be entered through the web interface. Alternatively, the user may add address book entries marked as „white list‟ or „black list‟ entry through a special code. Address book entries may have a speed dial index.

Incoming calls are matched against the address book. If there is a match, the PBX inserts the name into the other call leg, so that the extension will see the display name of the caller.

Codecs

The PBX supports the standard codecs G.711, G.726 and GSM. The PBX supports transcoding between these codecs.

Email Integration

The PBX may use standard SMTP to send emails.

Emails may be sent when a voicemail arrives, an extension misses a call or when a new voicemail is available.

The PBX is also able to send daily reports about the performance of the system.

Presence and Instant Messaging

The PBX passes instant messages transparently from extension to extension. This makes it possible to use Instant Messaging in the company without having to use public services. Employees can focus their attention on business-related messages without interference from private messages.

The PBX acts as a presence agent. If a user agent provides presence information, the PBX stores and forwards that information to subscribed users.

System Management

Most of the system features can be managed by a web browser using the http and https protocol. A certificate can be loaded into the PBX, so that web browsers do not warn about the automatically generated server certificate.

In order to ensure that the CPU is not running out of resources, the PBX monitors the performance of the media thread. The administrator can define the threshold where the PBX rejects additional calls. By using this dynamic way of controlling the performance, the system can be safely run close to the system capacity.

It is possible to change the appearance of the PBX. By loading new images and changing the texts of the PBX, installers can easily give the PBX their corporate appearance. By loading customized html text into the system, the PBX can be completely redesigned.

The PBX includes a SNMP agent. This agent can provide fundamental information about the state of the system (e.g. number of ongoing calls, registrations). The agent uses version 1 of the protocol.

The logging subsystem displays a specified number of last messages conveniently in the web interface. It is also possible to write the log data into a file and include the date in the file name. Several flags control which log messages find their way into the system log. For example, logging of SIP messages can be controlled by the SIP message class.

Call logs are available on system and domain level. By using SOAP requests, call data records (CDR) can be exported to external databases. Active calls can be seen on the web interface.