COMpact 5200/5200R/5500R

Advanced Information

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About this Instruction

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Current Issues

After a firmware update you usually require up-to-date manuals. You find these on the Internet.

Functions, Uses, Concepts

2nd Network Interface
Action URL (HTTP Request)
Address Books
Alarm
Alarm Device
Exchange Line Authorisation, Call Deblocker and Call Restrictor
Exchange Line Request
Exchange Line Transfer
Analogue Connection
Call waiting
PBX Time
Do-not-disturb, Call Deblocker and Call Restrictor
Automatic Reception
Baby Call
Busy-on-Busy
Call Through
Boss/Secretary Function
CLIP Texts
DECT Phones
E-mail
Remote Configuration and Alternative Connection Options
Firmware Update
Fixed Mobile Convergence (FMC)
Follow-me

- activation
- Call Charge Information
- Call Allowance Account
- External Outgoing Call
- Call Data
- Groups
- GSM Gateway
- Hardware
- Powering Down and Switching Off
- Hotel Function
- InterCom Announcement/Hands-Free
- Internal Phone Numbers
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- Configuration Switchover
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- Online Name Search

- Multi-path Call Forwarding
- Pick-up and Call Take-over
- Preliminary Pickup Query and Monitoring
- Pick-up of All Calls to Group Members
- Private (Personalised) Exchange Line Access
- Profiles
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- Provisioning for IP Phones
- Relays (Actuators)
- Call Channel Reservation
- Phone Number Display (CLIP)
- Number presentation
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- Call Distribution
- Switch Input
- Service Data
- Protection from Unauthorised Access
- System messages
- Subscribers
- Announcement before Answering
- Text Messages (Instant Messaging)
- Sound
- Door Functions
- Transmission of the Display Name
- Sub-System Operation
- Transfer and Callback
- Voice mail and fax boxes

VoIP
VoIP/GSM Routing
Preferred Exchange Line
Waiting Field Function
Music on Hold and Announcement
Wake-up Functions
Central Caller List

2nd Network Interface

A second network interface can be configured in the PBX. Accordingly, two IADs (Integrated Access Devices) can be directly connected to the PBX and configured. In this way, it is possible to reach one provider at the LAN 1 network interface and a second provider at the LAN 2 network interface.

The following possibilities for using two network interfaces can be realised:

VoIP provider at the LAN 2 connection

- At the connections LAN 1 and LAN 2 an IAD is connected in each case.
- The external VoIP telephony is realised via LAN 2. There is no Internet connection at LAN 2.
- The internal VoIP telephony and Internet connection are realised via LAN 1.
- The data transmission can be optionally realised via a separate connection.

VoIP internal at the LAN 2 connection

- An IAD is connected to LAN 1
- The data transmission and external VoIP telephony are realised via LAN 1.
- The internal VoIP telephony is realised via LAN 2.

VoIP completely at the LAN 2 connection

- At the connections LAN 1 and LAN 2 an IAD is connected in each case
- The internal and external VoIP telephony are realised via LAN 2.
- The data transmission is realised via LAN 1.

Proceed as follows for configuration:

- Configuring an IAD.
  - Using the Configuration Manager (page **Administration > VoIP Internet gateway**)
- Selecting the used Internet access for the VoIP provider.
  - Using the Configuration Manager (page **Public switched tel. networks > VoIP > Provider > Configure**)
- Selecting the network interface of the VoIP registrar.
  - Using the Configuration Manager (page **Administration > VoIP configuration**)
- Configuring DHCP settings for the 2nd network interface.
  - Using the Configuration Manager (page **Administration > IP configuration**)

Action URL (HTTP Request)

An Action URL is a function that can send requests via a link to a certain device. The device that receives the Action URL subsequently performs this request. The request and the IP address/URL of the Action URL receiving device is entered directly into the Action URL.

Note: Currently, Action URLs in the PBX can only be sent if an alarm was configured and is triggered. However, in this way, all requests can be sent that the Action URL receiving device recognizes.

Note: A maximum of 32 Action URLs can be configured.

The sequence can be as follows:

- Alarm detection via a switch input or a call to the alarm phone number.
- The set alarm delay time is running (0-99 seconds).
- The first alarm call run starts with the call of the first alarm subscriber. At the same time, the Action URL is sent (Alarm type start). If an alarm subscriber acknowledges the alarm, the alarm will be ended. At the same time, an Action URL is sent (Alarm type acknowlegded). If the alarm is not acknowlegded, it ends after all alarm loops. At the same time, an Action URL is send (Alarm type timeout).

Proceed as follows for configuration:

- Configuring Action URLs for a device.
  - Using the Configuration Manager (page **Devices > Action URLs**)
- Linking an Action URL with an alarm.
  - Using the Configuration Manager (page **Devices > Action URLs**)

Address Books

The address books serve to manage contact data (phone numbers, names, contact pictures, addresses and e-mail / internet addresses). The contact data can be used by the connected phones (if supported) to dial and to display on incoming calls.

With connected system phones, existing contacts are automatically offered according to the system phone type and their contact group membership.

Note: If a large number of phones are connected to the PBX, or large volumes of contact details are to be uploaded, it may in certain cases take several hours before the contact data is available to all the phones after the PBX has been commissioned.

Proceed as follows for configuration:

- Create and assign contacts.
  - Using the Configuration Manager (page **User data > Address books > Contacts and assignment**)
- Create contact groups.
  - Using the Configuration Manager (page **User data > Address books > Contact groups**)
- Set contact group authorisations.
  - Using the Configuration Manager (page **User data > Address books > Contact group authorisations**)
- Set Standard contact group for write-access users.

- Using the Configuration Manager (page **User data > Address books > Standard contact groups**)

Alarm

If the PBX function alarm is used in case of an alarm, internal and external phones can be called and sirens can be switched on.

The alarm sequence can be as follows:

- Alarm detection via a switch input (optional, see Technical Data in the Instructions) or via an internal call.
- The set alarm delay time is running (0-99 seconds).
- Non-recurring start of for example a siren via a relay/actuator (optional, see Technical Data in the Instructions).
- At the same time, the first alarm call run starts (alarm calls to the alarm subscribers) by calling the first alarm subscriber for approx. 60 seconds. Internal alarm subscribers are called with a special ringtone (phone-dependent). After picking up the receiver, you repeatedly hear an alarm announcement stored in the PBX. The called party must acknowledge the alarm call within 60 seconds by sending the DTMF number 0.
- If the called party does not acknowledge the alarm call, the next alarm subscriber is called a few seconds after the first call is finished and so on....
- If all the alarm subscribers have been called without receiving an acknowledgement, the alarm loop is repeated after a configurable alarm waiting period (0-99 seconds) has elapsed (up to 9 times).
- The alarm is terminated (and relays assigned to the alarm are switched off) as soon as one of the alarm subscriber acknowledges the alarm. If the alarm input has been configured to **Once**, it is disabled afterwards. If the alarm input has been configured to **Always**, it is enabled again and a new alarm can be triggered.

Configuration procedure:

- Creating an alarm with an internal number.
  - Using the Configuration Manager (**Devices > Alarms** page)

Note: An authorisation for triggering an alarm is required for the corresponding subscriber (page **Subscriber (scr.) > Phone numbers > Configure > Authorisations > Switching of relays and alarms**).

- Creating a switch input and assigning it to the alarm.
  - Using the Configuration Manager (**Devices > Alarms** page)
- Creating a relais and assigning it to the alarm.
  - Using the Configuration Manager (**Devices > Alarms** page)
- Manage and assign audio files for announcements.
  - Using the Configuration Manager (on the pages that follow)
  - **Functions > Music on Hold and Announcements > File management**
  - **Functions > Music on Hold/announcements > File assignment > Alarm**
  - Using a character string on the internal phone (see Alarm page)
- Configuring alarms.
  - Using the Configuration Manager (**Devices > Alarms > Configure** page)
- Activating alarms.
  - Using the Configuration Manager (**Devices > Alarms** page)

- Using a character string on the internal or external phone (see Alarm page)

Alarm Device

The alarm device function permits calls from internal subscribers to be signalled as alarm calls on system phones. Such alarm calls are acknowledged by a confirmation tone and subsequently ended.

Proceed as follows for configuration:

- Switching alarm device function on.
  - Using the Configuration Manager (on the following pages)
  - For admin: **Subscriber (scr.) > Properties > Settings**)
  - For user: **Subscriber Properties > Settings**)

Note: It makes sense to use the alarm device function in combination with special calling systems such as the GIRA calling system 834 plus, for example. Combined with the transmission of the display name, text messages can be sent and signallised accordingly.

- If necessary, switching transmission of the display name on.
  - Using the Configuration Manager (on the following pages)
  - For admin: **Subscriber (scr.) > Properties > Settings**)
  - For user: **Subscriber Properties > Settings**)

Exchange Line Authorisation, Call Deblocker and Call Restrictor

The exchange line authorisation determines which phone number types can be dialled by a subscriber (e.g., phone numbers in the local phone network).

Call deblockers can be added to the exchange line authorisations to expand dialling options with additions of a certain supply of numbers (phone numbers for specific connections or specific prefixes, released numbers). Since call deblockers have a higher priority than call restrictors, they can be used to release restricted numbers or parts of them.

In order to restrict exchange line authorisation, the phone numbers that should not to be dialled can be put in call restrictors (restricted numbers). There are several ways to do this. In general, phone numbers are restricted that begin with a certain digit sequence and that should not be accessible to every subscriber due to its higher charge tariffs, e.g., 0900 phone numbers. Normally, the first 4 to 5 digits indicate the phone service that cause such high charges.

If an attempt is made to dial a restricted phone number, only a busy signal is heard. If the restricted numbers are not enabled (meaning "not restricted"), restriction via the configured exchange line authorisation is still possible.

Proceed as follows for configuration:

- Creating the call deblocker and restrictor (outgoing).
  - Using the Configuration Manager (page **User data > Special numbers**)
- Configuring the exchange line authorisation and switching the call deblocker and restrictor (outgoing) on.

  - Using the Configuration Manager (on the following pages)
  - For subscriber **Subscriber (scr.) > Properties > Exchange line settings**
  - For groups **Groups > Properties > Exchange line settings**
  - For the exchange line transfer function **Functions > Exchange line transfer**
  - For the Call Through function **Functions > Call Through**
- Configuring the exchange line authorisation, call deblocker and restrictor (outgoing) configuration dependent.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) >Profiles > ... > Exchange line settings**
  - For groups: **Groups > Profiles > ... > Exchange line settings**

Exchange Line Request

The PBX distinguishes three types of exchange line requests:

Internal phone: The user has to dial an external number with preceding **exchange line access number**.

Direct exchange line phone: For an external number no prefix is required. However, the user has to dial an internal number with preceding **.

Automatic exchange line request (factory setting): Neither for the internal nor for the external number a prefix is required (exception: ** for targeted VoIP access). The PBX automatically distinguishes between internal and external numbers with the adjustment of the dialled phone number and the internal numbering plan:

- If the phone number is contained in the phone numbering list, the PBX establishes an internal connection.
- If the phone number is not contained in the phone numbering list, the PBX establishes an external connection.

Note: The short code number is a part of the internal numbering plan and is therefore an internal number. However it leads to an external connection.

Note: In order that the distinction between internal numbers and phone numbers in the local area network functions, the internal numbering plan must not contain phone numbers that also occur within the local area network. To avoid conflicts, phone numbers of the local area network should always be dialled with local area code.

Note: Also when the entered phone number correspond with the internal phone numbering plan, the PBX waits up to 4 seconds for whether

additional digits for an external number are being entered, before it dials the internal phone number. To avoid waiting time, you can subsequently enter #. The PBX therefore recognises the input as finished and instantly starts dialling.

Note: In factory settings, the system-wide automatic exchange line request (for all subscribers) is switched on. Alternatively, the exchange line request type can be set separately for each subscriber. In this case, the automatic exchange line request can only be selected for IP DECT phones, COMfortel 1200 IP and COMfortel C-400. For all other subscribers, only direct exchange line phone and internal phone can be selected.

Proceed as follows for configuration:

- Making settings for exchange line request.
  - Using the Configuration Manager (page **Public switched tel. networks > Exchange line request**)
- Making settings for exchange line request separately for subscribers.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings > Special functions**)

Note: This setting is only visible, if **System-wide automatic exchange line request** is disabled

Exchange Line Transfer

A suitably authorised system phone, for example, the phone exchange, can transfer the exchange line access required for an outgoing call to another internal phone for a short period of time. This is meaningful, for example, when this phone is located in a publicly accessible room and is only occasionally used for external calls by an employee.

The exchange line transfer proceeds as follows:

- The requesting phone must initiate an internal call using the system phone authorised to make the exchange in order to request exchange line access.
- After the function key assigned to exchange line transfer has been pressed on the system phone authorised for making exchanges, a confirmation tone is issued to both internal subscribers and the internal call must be ended.
- The requesting phone now has access to the exchange line settings configured for this purpose. These exchange line settings remain until the first successful, outgoing external call or until the time limitation has elapsed for initiating the external call.

Proceed as follows for configuration:

- Making settings for exchange line transfer.
  - Using the Configuration Manager (page **Functions > Exchange line transfer**)
- Switching the authorisation for transferring external calls externally for the transferred phone on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)

- At minimum, configuring a business exchange line authorisation for incoming calls for the requesting phone.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Exchange line settings**)
- Configuring a function key for the exchange line transfer on the system phone (optional, see the phone manual).

Analogue Connection

The existing analogue connections can be connected to the external analogue ports of the PBX (optional, see Technical Data in the Instructions).

Proceed as follows for configuration:

- Configuring the analogue connection and entering the phone number.
  - Using the Configuration Manager (page **Public switched tel. networks > Analogue**)
- Creating a call distribution for the analogue connection.
  - Using the Configuration Manager (page **Public switched tel. networks > Call Distribution**)
- If necessary, making country-specific modifications.
  - Using the Configuration Manager (page **Public switched tel. networks > Analogue > Configure**)

Call waiting

If you are already in a call, the call waiting function plays the call waiting tone to notify you that another call partner is calling.

Configuration procedure:

Switching on call waiting

- Using the Configuration Manager (**Subscriber (scr.) > Properties > Reachability** page)
- Using the menu/function key on a system phone (optional, see the phone manual)

Note: As of firmware 6.6 (PBX) and 2.4 (VoIP phone) a comparison between phone and PBX will be implemented. This function can then be switched on or off in the phone or the PBX.

PBX Time

The PBX has an internal clock that returns the PBX time (time and date) on the connected end devices. If the end device is so equipped, the PBX time will be displayed on the end device.

Furthermore, the PBX time is required to carry out functions such as switching configurations as well as wake-ups.

During a power failure or PBX reset, the PBX time is lost. There are several ways to update it:

- Manually
- Take from PC
- Via the network
- Using the buffered real-time clock – saves the time for approx. two hours in the case of a power outage.
- Via ISDN/external S_0 port (optional, see Technical Data in Manual)

Note: The switch to/from daylight savings is done automatically for the PBX time.

Proceed as follows for configuration:

- Using the Configuration Manager (page **Administration > Date and time of the PBX**)

Do-not-disturb, Call Deblocker and Call Restrictor

Do-not-disturb offers the option of blocking internal and external calls as well as blocking group calls on a specific phone. A caller hears the busy signal. An internal subscriber can reach a phone with call protection in an emergency using a priority call.

The do-not-disturb function has no influence on outgoing calls, callbacks, wake-up calls or InterCom calls.

If you would still like to allow specific people to call you despite enabling the do-not-disturb function, the external numbers of these people can be assigned a call deblocker for incoming calls in the PBX (VIP numbers). The call deblocker which applies to all subscribers can be enabled on your own phone as needed.

If there are certain persons who you absolutely do not want to call you or if you do not want callers to call an extension, their external numbers and/or all anonymous callers can be assigned to a call restrictor for incoming calls (Robinson numbers). The call restrictor which applies to all subscribers can be enabled on your own phone as needed.

Proceed as follows for configuration:

- Creating the call deblocker and restrictor (incoming).
  - Using the Configuration Manager (page **User data > Special numbers**)
- Switching the do-not-disturb function, the call deblocker and restrictor (incoming) on.
  - Using the Configuration Manager (on the following pages)
  - For subscriber: **Subscriber (scr.) > Properties > Reachability**

- For groups (only call restrictor): **Groups > Properties > Reachability**
- Using a character string on the internal phone (see Do-not-Disturb, Call Deblocker (Incoming)Call Restrictor (Incoming))
- Using the menu/function key on a system phone (optional, see the phone manual)

Note: As of firmware 6.6 (PBX) and 2.4 (VoIP phone) a comparison between phone and PBX will be implemented. This function can then be switched on or off in the phone or the PBX.

- Configuring the call deblocker and restrictor (incoming) configuration dependent.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) >Profiles > ... > Reachability**
  - For groups: **Groups > Profiles > ... > Reachability**

Automatic Reception

This function automatically connects an external caller with the person they want to contact. The caller hears an announcement (e.g. "... If you would like to talk to the Sales team, press 1 ..."). If they then redial, they can call a specific internal subscriber, a group, a voice mailbox, a different automatic reception, or a subscriber whose number has been diverted to an external number. Depending on the configuration, the caller either redials using a DTMF suffix dialling digit of between 0 and 9 or by dialling the entire internal number.

A total of ten receptions can be set up. These can either be switched in parallel or in series. As a result, several automatic receptions can be operated in parallel, for example in multi-company operation or, if required, one main reception with a number of sub-receptions can be set up.

Calls are accepted in the automatic reception in the following way:

- A caller reaches the automatic reception (e.g. via the external call distribution function).
- The caller hears a greeting, which might also be repeated. They then have the option of selecting a call destination by dialling a DTMF suffix dialling digit. They can also redial during the announcement.

Note: The announcement is always played to the caller when their call is received. For this reason, the caller might hear the call tone for a few seconds before their call is accepted, for example, if several callers are listening to announcements at the same time.

Note: Incoming VoIP calls can only be transferred over the automatic reception if the VoIP provider does not block DTMF code presentation.

Note: If an external caller is transferred to the automatic reception by a query call from an internal phone, the person transferring them hears the announcement and can then select the destination themselves. After that, the caller is connected with the selected destination. If the person transferring the calls hangs up before selecting a destination, the caller hears the announcement and can select a destination themselves.

- If the caller enters a valid number, the call destination will be called for a maximum of 120 seconds. The destination subscriber can answer the call during this time.

Note: When a call is transmitted via the automatic reception, it is the caller's phone number that is transferred (not the reception's number).

Note: If the call is not answered within 120 seconds, the announcement if the destination is busy/does not reply is played. In this case, the call is then either ended, the default destination is called, or the caller is reconnected to the automatic reception they previously called (depending on the setting for the system "Reaction if the destination is busy or does not reply").

Note: If the destination number is busy, the call waiting function is enabled if this function is enabled for the destination subscriber. If the destination subscriber rejects call waiting, or if they have not enabled this function, the announcement if the destination is busy/does not reply is played. In this case, the call is then either ended, the default destination is called, or the caller is reconnected to the automatic reception they previously called (depending on the setting for the system "Reaction if the destination is busy or does not reply").

Note: If the caller dials an invalid number, the announcement if the destination is busy/does not reply is played. In this case, the call is then either ended, the default destination is called, or the caller is reconnected to the automatic reception they previously called (depending on the setting for the system "Reaction if the destination is busy or does not reply").

Note: If the caller does not dial a number, they are transferred to the announcements set for the default destination number.

Note: The exchange line authorisations, call restrictors, call deblockers and do-not-disturb settings that apply to in-house call destinations are taken into account when the call is transferred.

Proceed as follows to set up automatic reception:

- Manage and assign audio files for announcements.
  - Using the Configuration Manager (on the pages that follow)
  - **Functions > Music on Hold and Announcements > File management**
  - **Functions > Music on Hold and Announcements > File assignment > Automatic Receptions**
  - Using a character string on the internal phone (see Automatic Reception)

Note: Alternatively, you can use an announcement before answering file (longer).

- Creating and configuring automatic receptions.

  - Using the Configuration Manager (page **Functions > Automatic Receptions**)

- Setting up call distribution for automatic receptions.
  - Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)

Baby Call

The baby call function provides a service in which after a configured delay time after picking up the receiver, a phone number is automatically dialled to call an internal or external subscriber. This means, no key needs to be pressed. However, if a phone is dialled manually within the configured delay time, it has priority. It must be noted that on some phones, the numbers dialled are not sent immediately when the keys are pressed (often there is approx. a half second delay).

Note: The baby call function is not available for VoIP phones.

Proceed as follows for configuration:

- Switching the baby call function on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings**)
  - Using a character string on the internal phone (see Baby Call/ Hotline)
- Configuring the baby call function configuration dependent.
  - Using the Configuration Manager (page **Subscriber (scr.) > Profiles > ... > Settings**)
- If necessary, configuring special dial tone for baby call.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings**)
- Configuring delay time for baby call.
  - Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)

Busy-on-Busy

Busy-on-Busy limits the calls to a group. The caller then hears the busy tone.

A maximum number of connection can be configured from which the group members can no longer be reached. Call forwardings (CFU) and multi-path call forwardings are still possible.

The maximum number of connections can also be configured dynamically. This ensures that the number of maximum connections is automatically adjusted to the number of registered group members so that the call is not put on hold.

Proceed as follows for configuration:

- Assigning a maximum number of connections to a group or switching Busy-on-Busy off.
  - Using the Configuration Manager (page **Groups > Properties > Reachability > Busy-on-Busy**)

Call Through

Call Throughto avoid high telephoning costs allows the use of LCR and VoIP/GSM routing of the PBX even when phoning over e. g. a smartphone. First, call the PBX and get connected to the desired connection. Call Through allows the direct redial of any phone number.

The called person does not receive the mobile number but the company number.

The caller receives the phone number set in the normal exchange line for this transmission. As default setting, the first defined number at this exchange line is used.

Requirements:

- The transmitted subscriber number must be registered as Call Through subscriber
- The transmitted subscriber number must not be in the MSN blocklist
- Configured 6-digit Call Through PIN

Procedure:

- Dialling the predefined external number of the PBX for Call Through.
- The PBX accepts the call and signals this with a special tone.
- Enter the **6-digit PIN** and ***.** The exchange line dial tone is heard.
- Based on the transmitted phone number and PIN, the PBX checks whether the caller has the authorisation for this function.
- Dialling the desired phone number (internal or external with prefix).
- After checking (exception number table, and if necessary LCR table), the connection is established.

Note: If possible, have the mobile phone operator enter the PBK number as special number (higher cost savings).

Proceed as follows for configuration:

- Creating a contact for the Call Through user.
  - Using the Configuration Manager (page **User data > Address books > Contacts and assignment**)
- Making settings for Call Through.
  - Using the Configuration Manager (page **Functions > Call Through**)
- If necessary, configuring LCR.
  - Using the Configuration Manager (page **Routing > Soft-LCR 4.0**)

Boss/Secretary Function

The boss/secretary function protects the boss phone from direct calls by rerouting them to the secretary. The most important difference from call forwarding is that the boss phone can still be called from the secretary phone and the secretary can still transfer important calls to the boss phone.

Requirements:

- At least two System phones, which support function keys for the boss/secretary function (optional, see the phone manual)

Note: The boss/secretary function is configured on the system phones in question. The configuration and operation of the function is described in detail in the manual for the system phone.

Proceed as follows for configuration:

- Configure a function key as a secretary key on the manager's phone (optional, see the phone manual).
- Configure a function key as a boss key on the secretary's phone (optional, see the phone manual).

CLIP Texts

CLIP texts are displayed on the system phones and/or the CLIP-capable analogue phones for specific calls (e. g. for wake-up calls).

Proceed as follows for configuration:

- Enter CLIP texts for specific calls (e .g. wake-up calls) and functions.
  - Using the Configuration Manager (page **Functions > CLIP texts**)

DECT Phones

COMfortel WS-400 IP / WS-650 IP
COMfortel WS-500S / WS-500M

COMfortel WS-400 IP / WS-650 IP

This PBX supports the operation of the IP DECT servers or base stations COMfortel WS-400 IP and WS-650 IP.

The following handsets can be operated in connection with these devices as IP DECT phones on the PBX:

- COMfortel M-100
- COMfortel M-200
- COMfortel M-210
- COMfortel M-300
- COMfortel M-310

Connection and operation of the base station to the PBX is via the Ethernet port and internal VoIP channels (optional, see Technical Data in the Instructions).

The handsets are operated in connectin with COMfortel WS-400 IP or COMfortel WS-650 IP as IP DECT phones on the PBX.

The description for the creation of internal phone numbers for these handsets in the PBX can be found on the page **Settings - Configuration Manager > Subscriber (scr.) > Phone numbers > Management Subscriber**.

The description for registering the handsets at the IP DECT server can be found in the Instructions of th IP DECT server.

How to transfer phonebook netries for the handsets is described here: Information about Transfer via LDAP Server.

COMfortel WS-500S / WS-500M

This PBX supports the operation of the IP DECT base stations COMfortel WS-500S and COMfortel WS-500M.

The following handsets can be operated in connection with these devices as IP DECT phones on the PBX:

- COMfortel M-510
- COMfortel M-520
- COMfortel M-530

Singel-cell system with the base station COMfortel WS-500S

A COMfortel WS-500S can be used to implement a single-cell system with a single base station.

The device has an integrated DECT Manager. it manages the handsets via its own Web Configurator and provides the DECT functions.

It ensures communication between the handset and the PBX.

Up to 20 handsets can be connected.

Multi-cell system with the base station COMfortel WS-500M

One or more COMfortel WS-500Ms can be used to set up a multi-cell system with one or more base stations and a central management station. The COMfortel WS-500M handles various device roles. Different extension levels are possible here.

With the "Integrator" device role, the device manages the handsets via its own Web Configurator.

With the "DECT Manger" device role, it manages the synchronisation of th e base stations and esnures the communication from the PBX to te respective base stations.

Depending on the configuration level, up to 112 handsets (limited by the maximum number of subscribers in the PBX) can be connected.

Further information can be found in the Instructions of the base station.

General information about the configuration

Connection and operation of the IP DECT base stations COMfortel WS-500S and COMfortel WS-500M to the PBX is via the Ethernet port and correspondingly assigned VoIP channels.

Depending on the individual procedure (e. g. use of provisioning or manual configuration), the base stations and handsets are configured and maintained in the Configuration Manager of the PBX and in the Web Configurator of the base station. See the description below.

The description for the creation of internal phone numbers for these handsets in the PBX can be found on the page **Settings - Configuration Manager > Subscriber (scr.) > Phone numbers > Management Subscriber**.

Only the base stations for which the device role "DECT Manager and Intergrator“ is activated are configured at the PBX. Base stations with the role "Base station" are not configured in the PBX but for the mutli-cell operation at a previously determined base station with the role "DECT Manager and Integrator".

The handsets are operated in connection with the base station as IP DECT phones on the PBX.

Registering the handsets at the base stations is described later in this page.

Requirements for first commissioning

For informationen about planning the DECT network, local area (LAN) configuration, and more, refer to the Instructions of the base station.

First commissioning via provisioning

Important: The base station may only be connected to the local area network (LAN) after step 4. Otherwise, the automatic initiation of provisioning is not possible.

1. Configure a base station in the PBX for the COMfortel WS-500S or the COMfortel WS-500M with the future device role "DECT Manager and Integrator" using the MAC address (label on the back of the device) (page **Subscriber (scr.) > COMfortel WS-500x**).
2. Configure a subscriber with the device type **COMfortel M-5x0** for each handset in the PBX and with knowledge of the IPUI (page **Subscriber (scr.) > Phone numbers**). You can obtain the IPUI by reading it from the handset (pressing the control button and entering ***#06#**) or by reading the packaging label.
3. Assign the corresponding base station to each subscriber (page **Subscriber (scr.) > COMfortel WS-500x > COMfortel M-5x0**).
4. Create a provisioning file for the device type **COMfortel WS-500** in the PBX and assign a base station to this file. If the provisioning file has already been created, assign a base station if necessarys (page **Subscriber (scr.) > IP Provisioning > File management**.
5. Connect the base station via LAN cable to the network and wait for the boot process.

6. Only **COMfortel WS-500M**: To set up a multi-cell system with more than one base station, set the "DECT Manager and Integrator" device role for a base station directly on the device.
7. The base station restarts and the provisioning file ia automatically transmitted from the PBX to the base station and the device is configured.
8. After successful provisioning, the base stations remains in login mode for one hour. During this time, register the handset to the base station (on the handset under **Settings > Registration > Register handset**).

Note: Using provisioning, the existing template supplies all handsets with the PBX phone book when LDAP is configured and activated.

For information about connecting the base station, booting, setting the device role, and the LED display on the device, refer to the Instructions of the base station.

Manual first commissioning

1. Configure a subscriber with the device type **COMfortel M-5x0** for each handset in the PBX (page **Subscriber (scr.) > Phone numbers**). Make a note of the internal number, the name and the password for each subscriber.
2. Log into the Web configurator of the base station and carry out the further steps for the configuration of the PBX, the base station and the handsets there.

The description can be found in the Instructions of the base station in **Initial Commissioning > Configure them Manually Using the Web Configurator on the Base Unit**.

Checking for firmware update

After the configuration of the base station, it should be checked via the Web Configurator whether a firmware update is available for the device.

E-mail

The PBX supports the following e-mail functions:

- System e-mails
- Forwarding of voicemail and fax messages

The settings described here must be made centrally in the PBX and can then be freely assigned to the individual e-mail functions of the PBX.

You can configure up to four outgoing mail servers in the PBX. They can be freely assigned to the individual e-mail functions of the PBX (system e-mails, forwarding of voicemail and fax messages).

An outgoing mail server is responsible for the transfer of e-mails. You will receive the access data for the outgoing mail server to be used from your Internet service provider, for example.

The PBX supports a large number of logos (limited by the total memory available). These logos can be freely assigned to the individual e-mail functions of the PBX (system e-mails, forwarding of voicemail and fax messages).

For the e-mails to be sent by the PBX, several language files are included in the scope of supply. The PBX supports additional user-defined which can be generated on the basis of the existing language files.

The language files contain several text modules which are used to generate e-mails. The language files are in *.xml format.

Proceed as follows for configuration:

- Configuring an outgoing mail server.
  - Using the Configuration Manager (page **Administration > E-mail > Outgoing mail server**)
- Assigning an outgoing mail server to a voice mail/fax box.

  - Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Properties > Box settings**)

- Assigning an outgoing mail server to the system e-mails.
  - Using the Configuration Manager (page **Administration > System messages**)
- Managing logos.
  - Using the Configuration Manager (page **Administration > E-mail > Logo management**)
- Assigning a logo to a voice mail/fax box.
  - Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Properties > Box settings**)
- Assigning a logo to the system e-mails.
  - Using the Configuration Manager (page **Administration > System messages**)
- Managing language files.
  - Using the Configuration Manager (page **Administration > E-mail > Internationalisation**)
- Assigning a language file to a voice mail/fax box.
  - Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Properties > Box settings**)
- Assigning a language file to the system e-mails.
  - Using the Configuration Manager (page **Administration > System messages**)

Remote Configuration and Alternative Connection Options

Configuration Options
Configuring the PBX remotely via the Internet with (Auerswald) Remote Access Server
Configuring the PBX Remotely via the Internet (without (Auerswald) Remote Access Server)
Configuring ISDN System Phones via the PBX (Internal)
Configuring ISDN System Phones Remotely via the External S 0 Port on the PBX (Externally)

Aside from configuration over the internal network, there is also the option of remote configuration via the Internet as well as configuring alternative connection options under certain conditions.

The connected system phones can also be configured over the PBX under certain conditions.

You can use the following connection options to configure (either remotely or directly) the PBX and the ISDN system phones connected to it:

Configuring the PBX via the In-house Network

The PBX is usually configured via an internal network connection. The PC you use to do this must be connected to the same network as the PBX.

Configuring the PBX Remotely via the Internet with (Auerswald) Remote Access Server

Remote configuration of the PBX can be accessed via the page (Auerswald) Remote Access on the Internet. This requires a dealer access via the Internet and enabling of the (Auerswald) Remote Access.

Configuring the PBX Remotely via the Internet without (Auerswald) Remote Access Server

You can also configure the PBX remotely via an internet connection (DSL). This requires a fixed external IP address. Alternatively, you can use VPN or DynDNS. The router on the PBX's network must be configured accordingly.

Using the PBX (Internal) to Configure internal ISDN System Phones

You can configure internal ISDN system phones via an indirect connection using the PBX. Before you can do this, you must install the ISDN system phone's COMfortel Set configuration software on the PC you are going to use.

Configuring internal ISDN System Phones Remotely via a PBX's External S_0 Port

You can remotely configure the ISDN system phone indirectly via an external ISDN connection using the PBX and a device which is connected to both the external S_0 port on the PBX and to the PC (for example ISDN PC controller). The PBX must be configured accordingly. Before you can do this, you must install the ISDN system phone's configuration software on the PC you are going to use.

Configuration Options

Aside from configuration via the internal network, there is also the option of remote configuration via the Internet as well as configuring alternative connection options under certain conditions.

The connected system phones can also be configured over the PBX under certain conditions.

You can use the following connection options to configure (either remotely or directly) the PBX and the ISDN system phones connected to it:

Configuring the PBX via the internal network

The PBX is usually configured via an internal network connection. Your PC must be connected to the same network as the PBX

Remote configuration of the PBX via the Internet with (Auerswald) Remote Access Server

Remote configuration of the PBX can be accessed via the page https://remote-access.auerswald.de on the Internet. This requires a dealer access via the Internet and enabling of the (Auerswald) Remote Access

Remote configuration of the PBX via the Internet without (Auerswald) Remote Access Server

You can also configure the PBX remotely via an internet connection (DSL). This requires a fixed external IP address. Alternatively, you can use VPN or DynDNS. The router on the PBX's network must be configured accordingly.

Configuration of internal ISDN system phones via the PBX (internal)

You can configure internal ISDN system phones via an indirect connection using the PBX. Before you can do this, you must install the ISDN system phone's COMfortel Set configuration software on the PC you are going to use.

Remote configuration of internal ISDN system phones via an external S_0 port of the PBX

You can remotely configure the ISDN system phone indirectly via an external ISDN connection using the PBX and a device which is connected to both the external S0 port on the PBX and to the PC (for example ISDN PC controller). The PBX must be configured accordingly. Before you can do this, you must install the ISDN system phone's configuration software on the PC you are going to use.

Configuring the PBX remotely via the Internet with (Auerswald) Remote Access Server

Requirements:

- PBX with existing connection to the Internet (DSL)
- In the PBX configured ARA PIN and remote switching number and also enabled dealer access over the Internet
  - Using the Configuration Manager (**Administration > Remote configuration**)
- Computer with existing connection to the Internet (DSL)
- DTMF-capable phone or DTMF dialler
- Access data for registration at protected Auerswald services
- Knowledge of the serial number of the PBX
- Knowledge of user name and password for the Configuration Manager

1. Start a browser.
2. Open (Auerswald) Remote Access.
3. Enter filter criteria for the search of contacts into the entry fields User name/Customer number and Password/PIN input fields, and click Register.

   The user interface of the (Auerswald) Remote Access Server opens.
4. Register the PBX if this has not been done yet.
   - Click .

- In the entry field **Serial number** entry field, enter the serial number of the PBX.
- Make further entries for the data record and click **Save**.

5. Call the remote switching number.

   The PBX automatically accepts the call. You will hear the external confirmation tone (1 second).

6. Dial with DTMF ##8 ARA PIN *91#.

   You will hear again the external confirmation tone (1 second).

   In the user interface of the (Auerswald) Remote Access Server within a few seconds the connection symbol .

7. Click .

   The login window of the Configuration Manager opens.

8. Log in to the Configuration manager.
9. Make the desired changes.
10. Log out from the configuration manager.
11. On the user interface of the (Auerswald) Remote Access Server click on .

Configuring the PBX Remotely via the Internet (without (Auerswald) Remote Access Server)

Requirements:

- Internet connection between the computer and the PBX
- Permanent external IP address, or the use of VPN or DynDNS
- A router configured for this purpose
- Knowledge of user name and password for the Configuration Manager
- Knowledge of the configured port of the web server

Note: Refer to the operating manual of your router.

Note: The access is possible via **http:** and via **https:** (encrypted access). If the option **Force HTTPS connection** has been enabled, only the https connection can be used. In this case, a port you selected yourself must be rerouted to port 443 of the PBX (access with **https://mein.dyndns.org:[Port]**, for example.

Important: Make sure that no simple passwords have been assigned for all access options, as these pose a security risk.

about configuring your firewall, contact the system administrator responsible for this topic.

1. Start a browser.
2. Enter https:// and the external IP address for the PBX as well as the : character, and then the port, into the address field of the browser.
3. Enter your user name and password.

4. Click OK.
5. Make the desired changes.

Configuring ISDN System Phones via the PBX (Internal)

Requirements:

- PBX with internal ISDN system phones
- In the PBX configured configuration port for system phones (page **Administration > IP configuration**)
- Internet connection
- Installed COMfortel Set configuration software version 3.0, or later
- Knowledge of the user, sub-admin or admin PIN

1. Start COMfortel Set.
2. Click **Options > Interface...**.
3. Select **Interface > IP network**, enter the **IP address** of the PBX and under **IP port (Default: 7002)** the configuration port for system phones on the PBX.
4. Click **OK**.
5. Open the menu **File > Open > Telephone**.
6. Double-click on the PBX symbol displays all connected system phones. Select the phone whose configuration you want to open and click **OK**.
7. Select the desired **Authorisation level**.
8. Enter the associated **PIN** and click **OK**.

9. Make the desired changes.
10. Open the menu **File > Save > File**.
11. Select a memory location in the directory.
12. Enter a name for the *.cf2 file and click **Save**.
13. Open the menu **File > Save > Telephone**.
14. Select the desired phone and click **OK**.
15. Enter the associated **PIN** and click **OK**.

Configuring ISDN System Phones Remotely via the External S $_0$ Port on the PBX (Externally)

Requirements:

- PBX with internal ISDN system phones
- In the PBX configured remote switching number for system phones and dealer access number (page **Administration > Remote configuration**)

Note: If no dealer access number has been configured or the phone number transmitted does not correspond with the configured phone numbers, the PBX must be enabled via an internal phone using a digit sequence for remote configuration.

- An existing external ISDN connection between the PC and the PBX including the drivers
- Installed COMfortel Set configuration software version 3.0, or later
- Knowledge of the user, sub-admin or admin PIN
- Activation of the PBX immediately before remote configuration (see section Short Reference) if no dealer access number has been configured or the transmission of the configured numbers is not possible on the computer being used

1. Start COMfortel Set.
2. Open **Options > Interface...**.
3. Under **Interface**, select **ISDN card (CAPI 2.0)**.

4. Click **OK**.
5. Open **File > Open > Telephone**.
6. Enter the remote switching number for system phones (including local city code and exchange line access number), and click **OK**.
7. Double-click the PBX symbol to display all connected system phones. Select the desired phone and click **OK**.
8. Select the desired **Authorisation level**.
9. Enter the associated **PIN** and click **OK**.
10. Make your changes.
11. Open **File > Save > File**.
12. Select a memory location in the directory.
13. Enter a name for the *.cf2 file and click **Save**.
14. Open **File > Save > Telephone**.
15. Select desired phone and click **OK**.
16. Enter the associated **PIN** and click **OK**.

Firmware Update

Information and Options
Distributing Firmware

Information and Options

When firmware is updated, errors are corrected or new features are installed. To do this, the device establishes a connection to a server in order to download the most current firmware (the device settings remain intact). The update is free of charge.

Note: If you have just performed a firmware update and an error message appears when you log into the Configuration Manager, please empty your web browser's cache or wait for about five minutes before logging in again.

Note: Some of the functional upgrades provided as part of a firmware update require additional configuration parameters. Sometimes not all these parameters can be determined automatically from the previous firmware version.This is why you need to edit some parts of the configuration manually in this update type.

Note: If the regular firmware update option has been enabled, the status messages tell you whether you need to edit the current update manually. Click **Update** to download an update for the device firmware from the update server.

Note: In order to use new features after a firmware update, you normally need a new version of the Manual. This is available on the Internet (see Auerswald: Auerswald Service Portal, FONtevo: FONtevo Support).

Firmware update options:

- Manual firmware update of the PBX.

  - Using the Configuration Manager (page **Administration > Firmware update and restart**)

- Firmware update of the PBX via update server.
  - Using the Configuration Manager (page **Administration > Firmware update and restart**)
- Firmware update for system phones.
  - Using the Configuration Manager (page **Administration > Firmware update and restart**)
- Restoring the firmware of the PBX.
  - Using the Configuration Manager (page **Administration > Firmware update and restart**)
  - By pressing a button on the PBX

Distributing Firmware

A new version of the firmware can be distributed from one system phone to other system phones connected to the PBX. To configure an IP system phone as firmware distributing phone, see Advanced Information of the phone. Under **Administration > Firmware update and restart**, you can see under **Firmware update for Voip system phones** the current status.

Note: The PBX must have a firmware version as of 6.6.

Note: The phone that receives the firmware must have a firmware version as of 2.4.

Note: Distributing the firmware is possible from one COMfortel 3600 IP to other COMfortel 3600 IP and from COMfortel 1400 IP/2600 IP to other COMfortel 1400 IP/2600 IP.

Proceed as follows:

- Using the phone and automatically via the PBX

Fixed Mobile Convergence (FMC)

FMC supports the merging of mobile and landline. The user enables the FMC behaviour of the PBX for a subscriber via a smartphone app and is supported by it when initiating calls.

For incoming calls, ther user at the smartphone gets the correct phone number signalled. Outgoing calls via the smartphone app to an external number are analogous to Call Through calls. But here the caller is notified of the MSN of the PBX under which the caller can be called back and not the Call Through MSN or the phone number of the smartphone. When using CLIP no screening, a phone number you define can be transmitted.

Note: An outgoing dialling is only possible when FMC behaviour is enabled. In addition, there is the reachability on the smartphone via the call to the extension.

Note: The phone number of a smartphone can be entered and enabled for only one extension.

An FMC call is shown as such under **call data list** > **Call type**.

Requirements:

- On the smartphone installed app **COMfortel Mobile Business**
- Provider supporting CLIP no screening (possibly with costs)
- CLIP no screening configured at the provider
- Enabled FMC behaviour (page **Subscriber (scr.)** > **Features** > **availability** page)

Procedure for outgoing calls:

- Dialling the Call Through MSN of the PBX.

- Dialling the user PIN **6-digit PIN** + * + destination number (internal or external without exchange line access number). This is unambiguously and makes it clear that this is an FMC call.
- The caller is identified by the number for the Multi-Path call forwarding.
- The required data (PIN, phone number of the caller) is checked.
- When all data match, the caller can initiate a call (e. g. via the smartphone app).

Configuration procedure

- Enabling FMC behaviour and entering destination number
  - Using the Configuration Manager (**Subscriber (scr.)** > **Features** > **availability** > **Multi-path call forwarding / FMC** page)

Note: When the option **Call signalling ... (multi-path call forwarding)** (setting in the app) is selected, this setting at the PBX is automatically enabled and disabled and compared with the phone number of the smartphone stored in the app.

- Enabling CLIP no screening for the account/connection
  - Using the Configuration Manager (**Public switched tel. networks** > **Expert configuration of the corresponding Account/ connection** > **Configure** > **Call forwarding** > **Use Clip no screening for call forwarding**).
  - Using the Configuration Manager (**Public switched tel. networks** > **Expert configuration of the corresponding Account/ connection** > **Configure** > **Number presentation** > **Clip no screening**).
- Making settings for FMC

- Using the Configuration Manager (**Functions** > **Call Trough** page)

Note: For FMC, the limits (call duration, number of calls) that are set for Call Through apply.

Follow-me

Follow-me is like a call forwarding unconditional for subscribers or groups that can be configured on any phone (instead of on the forwarding phone). This gives you the option of moving from room to room and taking the calls with you.

Follow me can be configured from different phones:

Configure Follow-me to an internal destination phone: With the help of this function you can go from room to room and take your calls with you. You can configure the call forwarding of your calls from the phone (destination phone) where you currently are. But you can also take the calls of several subscribers/groups at your phone.

Configure Follow-me from any available internal phone: If you have already left your desk, you can forward your calls to another phone. This way it is possible to forward the calls to your mobile phone or forward it to an internal representative.

Configure Follow-me from external: If you have already left the desk, you can forward your calls to another phone. This way it is possible to forward the calls to your mobile phone or to forward it to an internal representative.

Note: In the case of an external call for switching, the PIN of the diverting subscriber is required.

Proceed as follows for configuration:

- Switching authorisations for the involved internal subscribers on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)

Note: The forwarding subscriber needs an authorisation e.g. for external destinations. For configuring using a character string on an internal phone the subscriber needs an authorisation for controlling of the PBX.

- Configuring sufficient exchange line authorisations for the involved internal subscribers.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Exchange line settings**)

Note: The destination subscriber needs a sufficient exchange line authorisation for taking external calls.

- Switching Follow-me on.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) >Properties > Reachability**
  - For groups: **Groups > Properties > Reachability**
  - Using a character string on the internal or external phone (see Follow-me)

activation

Basic equipment

The following unlockable functions/resources are already available in the PBX:

- 4 (COMpact 5200/R) or 8 (COMpact 5500R) VoIP channels for internal IP and Internet telephony (SIP-compliant in accordance with RFC 3261)
- Support of VoIP Codecs G.711, G.729, G.722, G.726, iLBC
- 1 voice mail channel for voice mail functions with one voice mailbox
- Wait 10 Configurations
- 18000 Call data records
- Project-related calls (project numbers)
- Call Through calls (Call Through for all lines)
- LAN-TAPI for 4 subscribers
- LCR for all subscribers

Important: Operating the PBX without system activation is not possible.

Important: Using the voice mail channel results in the loss of one VoIP channel.

Important: It is not possible to use the voice mail channel without inserted USB memory stick (not included in the scope of delivery).

Extension by activation

By chargeable activation in the Auerswald/FONtevo Online Shop, the PBX can be extended by the following functions/resources:

- System activation
- Automatic receptions
- Auerswald hotel function for all subscribers
- LAN-TAPI for all subscribers
- COMpact 5200/R: 4 additional VoIP, voicemail and fax channels (maximum of 16)
- COMpact 5500/R: 8 additional VoIP, voicemail and fax channels (maximum of 32)
- COMpact 5200/R: 6 additional VoIP exchange line channels (maximum of 20)
- COMpact 5200/R: 12 additional VoIP exchange line channels (maximum of 32)

- 20 additional voice mailboxes and fax boxes for subscribers and groups (maximum of 40)
- **SIP convenience package** / **SIP convenience package brand plus** (e. g. BLF, MWI) for VoIP end devices
- Support PBX Call Assist

Important: The system activation is mandatory for the operation of the PBX.

Note: The PBX Dongle required for the extension is part of the basic device.

Configuration procedure:

- Activate further functions.
  - Using the Configuration Manager (**Administration > Activation** page)

Call Charge Information

To monitor the charges accrued on the connection and also on the internal phones, the charges are recorded by the PBX for each subscriber separately. From time to time, the total charge may be reviewed on the corresponding phone and deleted, if necessary. The deletion has no influence on the call charges recorded in the call data memory of the PBX.

To check the charges during a call or directly afterwards, the charges are transmitted to ISDN phones unchanged at the moment they are accrued even if the receiver is on-hook.

Proceed as follows for configuration:

- Switching charge information at the external connection on.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > ISDN > PTMP connection > Configure**
  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles > Configure**

Call Allowance Account

A call allowance account is a restricted amount of currency units (credit) that is configured for each phone. As soon as the configured credit is used up, it is no longer possible to make external calls on the corresponding phone until the credit has been increased or reset.

If the remaining amount is used up during a call, the subscriber hears a short attention tone repeatedly. Afterwards, the call is disconnected.

Note: For providers who do not provide billing information, a configured billing account will not work.

Note: If the LCR procedure Soft-LCR 4.0 is used, the charges are calculated based on the length of the call and the tariff table set up separately. For this purpose, you need to specify a price per minute as well as the length of the charge pulse per switching time and provider.

Proceed as follows for configuration:

- Configuring LCR.
  - Using the Configuration Manager (page **Routing > Soft-LCR 4.0**)
- Making settings for call allowance accounts.
  - Using the Configuration Manager (page **Functions > Call allowance accounts**)
- Setting a credit.
  - Using the Configuration Manager (page **Functions > Call allowance accounts**)

- Using a character string on the internal phone (see Call Allowance Account)

External Outgoing Call

An outgoing external call is made as follows:

- A user dials an external number.
- The PBX checks whether the phone number or the phone number range is specified in the exception numbers (VoIP/GSM routing) and switched on for the current configuration.
- If the phone number is found, the PBX automatically forwards the dial attempt to the connections set up there, while taking into account the dialling operations.
- If the phone number is not in the list of exception numbers (VoIP/GSM routing), or it is not enabled for the currently valid configuration, the system checks whether a preferred exchange line has been configured.
- If no preferred exchange line has been configured, or the preferred exchange line is a land line connection, the system checks the phone number, on the basis of the LCR networks (if LCR is configured and enabled), and then dials it on a land line connection.

Note: The VoIP/GSM routing is performed depending on the destination number, if the exchange line settings (subscriber, group) do not prevent this (e.g. exchange line authorisation only for selected connections, preferred exchange line, call restrictor).

Note: If a call is initiated over a targeted VoIP access point, VoIP/GSM routing is not carried out.

Note: The special numbers provided by the VoIP provider, for checking one's account status or dialling, for example, usually do not include a

local code. It is not possible to route numbers via the exception table. In order to correctly dial these phone numbers, use targeted exchange line assignment with the account number (targeted VoIP access point).

Note: When you dial one of the local numbers (phone number without a prefix) listed in the exception number table over a VoIP access point, the PBX automatically converts the phone number as specified under **Number presentation (outgoing)** for the VoIP provider.

Note: For all DSL connections that are not made over a dedicated line, some VoIP provider reset the DSL line at least once within 24 hours. This terminates calls currently in progress.

Call Data

Overview of Setting Options
Contents of the Call Data Record
Opening the Call Data List

Overview of Setting Options

The call data memory of the PBX, can store up to 18,000 call data records. The call data memory is secured against power failures.

Proceed as follows for configuration and using:

- Set call data management and SFTP transfer.
  - Using the Configuration Manager (page **User data > Call data > Settings**)
- Currency name and set the cost per call unit.
  - Using the Configuration Manager (page **Functions > Call allowance accounts**)
- Set billing factor.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings**)
- Configure the columns of the call data list.
  - Using the Configuration Manager (page **User data > Call data > Print options**)
- Configure filter.
  - Using the Configuration Manager (page **User data > Call data > Filter**)
- Management call data.
  - Using the Configuration Manager (page **User data > Call data > Call data list**)
- Delete call data records.

- Using a character string on the internal phone (see Call Data (Single Call Record))

Contents of the Call Data Record

A call data record contains the following information:

- Date at the beginning of the call
- Time at the beginning of the call
- Duration of the call
- Prefix number of the LCR provider – if used
- Number of the external call partner – if known
- Name of the external call partner – if known
- Number of the internal phone (billing + real)
- Name of the internal phone (billing + real)
- Your own external number
- Call charges
- Billing factor
- Call direction
- Billing type
- Call type
- Project number
- Billing number for room phones

Opening the Call Data List

The call data list shows the call data records recorded in the call data memory.

Proceed as follows:

- Using the Configuration Manager
  - As admin: **User data > Call data > Call data list**
  - As user: **Call data list**.

Note: Sorting criteria (max. 3) and sequence can be changed via the column headlines. Main sorting criteria is the last one selected.

Groups

Membership in a group does not necessarily mean that a subscriber receives calls sent to the group. A subscriber that is logged out takes on no group functions. This can be especially important for employees of a support department, who should not be available to customers around the clock, but rather rotate in and out of the hotline.

Note: If a group (subgroup) has been configured in the PBX as member of a group (main group) and a subscriber is at the same time member of these two groups, he will only be considered as a member of the main group when a call comes in.

Note: Subgroups (i.e. groups that have been configured as members of other groups) are **permanently logged in as incoming** and **permanently logged out as outgoing**.

Note: Phones **permanently logged in** cannot **log out** of a group.

Note: Group calls can also be placed in the waiting loop, always provided that at least one subscriber is **logged in as incoming**. The group will be called as soon as a subcriber who is logged in as incoming is free.

There are three different ways to log in:

Only incoming

When logging into a group as **incoming**, the subscriber then finds himself in the call distribution of the group in question for internal, exchange line and door calls.

Only outgoing

When logging into a group as **outgoing**, the subscriber acquires a number of characteristics/access rights from the group. These replace his own characteristics/access rights as an individual subscriber for outgoing work-related calls. A subscriber can only be logged on in a single group as **outgoing** even if he is a member of more than one group.

Note: If a subscriber logs into different groups as **outgoing** one after the other, he will only remain logged in as **outgoing** in the group that he logged into last.

Incoming and outgoing

When logging into a group as **incoming and outgoing**, the subscriber then finds himself in the call distribution of the group in question for internal, exchange line and door calls and also acquires a number of additional characteristics/access rights from the group. These replace his own characteristics/permissions as an individual subscriber for outgoing work-related calls.

Note: If a subscriber logs into more than one group as **incoming and outgoing**, he will only remain logged in as **incoming and outgoing** in the group that he logged into last. And in all other groups, he is then only **incoming and outgoing**.

Proceed as follows for configuration:

- Creating groups.
  - Using the Configuration Manager (on the following pages)
  - **Groups > Phone numbers**

  - **Groups > Properties**

- Logging Subscribers in/out.
  - Using the Configuration Manager (page **Groups > Properties > Basic settings**)
  - Using a character string on the internal phone (see Groups)
  - Using the menu/function key on a system phone (optional, see the phone manual)

GSM Gateway

A GSM gateway can be connected to an external port on the PBX. This allows direct access to be configured on the world-wide GSM network (Global System for Mobile Communications).

Note: GSM gateways take external calls automatically. When this is done, costs may be charged to the caller, even if nobody takes the call on the internal destination side.

Note: LCR ist not available for an external port connected to a GSM gateway.

Note: If the PBX has no connection to the fixed network, outgoing external calls can only be initiated via a targeted exchange line access using the phone numbers (and the corresponding configured access) entered on the page **Routing > VoIP/GSM routing > Exception numbers** or via a configured preferred exchange line (**Exchange line settings** for subscribers or groups).

Proceed as follows for configuration:

- Switching GSM at the external connection on and entering the freely selectable phone number.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > ISDN > PTMP connection**
  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles > Configure**

- Entering the connection type on the ISDN connection and the phone number.
  - In the GSM gateway configuration (see manual of the GSM gateway)

Hardware

After the first initialisation or the insertion of new modules, these modules have to be configured using the Configuration Manager. Assign the existing modules to the used expansion slots and perform the port configuration.

Port configuration is the basic set-up for the inserted modules. The connection assignments for the ports must be set depending on the proper use and must be adapted to the actual connection options on site.

Note: Please find the basic information for the use of the modules in the Installation and Commissioning Instructions.

Proceed as follows for configuration:

- Selecting modules.
  - Using the Configuration Manager (page **Hardware > Selection of modules**)
- Setting port configurations.
  - Using the Configuration Manager (page **Hardware > Port configuration**)

Powering Down and Switching Off

If it is necessary to separate the PBX from the electric current (pull the power plug of the PBX out of the socket and also of the accessory equipment where required), the PBX will first have to be powered down properly.

Proceed as follows for powering down and switching off the PBX:

- Immediate powering down.
  - Using the Configuration Manager (**Administration > Firmwareupdate/restart** page)

Important: When shutting down the PBX, all of the connections to the web server, all calls and all data services are terminated. Unsaved data is lost.

Hotel Function

The hotel function offers room and call data management with the following functions:

Hotel reception phones

The hotel reception phone supports various control and query functions via the display and the LEDs of the function keys. For each room, a function key is set up on the reception desk, via which various functions can be performed.

Before check-in, the room status (cleaned, unclean, blocked) can be checked and changed.

As soon as check-in is complete, the following configuration and query functions can be used.

- Call charge queries including check-in date
- Interim call charge calculation printout
- Wake-up time configuration
- External access block
- Check-out

Hotel room phones

When used as a hotel room phone, the programming and transferring functions as well as spezific call initiation types are blocked on the devices. The guest can initiate and accept calls and configure a wake-up time. The cleaning staff can also indicate the room status.

In addition, the administrator can prevent the initiation of calls between room phones (Room2Room).

The function of the hotel room phone are operated the same way as described for normal phones. Exception: A function key assigned with **Hotel room** allows the configuration of a wake-up time as well as the inquiry of the call costs and the check-in time.

The following functions are still possible:

- Making and excepting internal and external calls
- Deleting data and call charges (this does not affect the call charge counter on the PBX)
- The use of the caller/call/redial list as well as of the phone book (parts of the phone book can be blocked in the phone)
- Scheduled call/reminders and power dialling
- Call charge query and wake-up time configuration using a function key

Print function

Via the hotel reception phone the incurred call costs of a guest can be printed out as an invoice or interim bill with itemized bill of identification.

Requirements:

- Hotel function enabled (optional, see activation)
- ISDN System phone, that supports function keys for reception phones (optional, see the phone manual)

Proceed as follows for configuration:

- Selecting reception and room phones.
  - Using the Configuration Manager (on the following pages)
  - **Functions > Hotel function > Hotel reception subscribers**

- **Functions > Hotel function > Room subscribers**

Note: The system phone cannot be used at the same time as waiting field reception phone and as hotel reception phone.

Note: To ensure that the costs are fully charged to the hotel guest in the case he wants his call to be transferred, the function **Call charge recording changes with scr** (page Functions > Call allowance accounts) for the hotel reception phone function must be enabled.

Note: To ensure that the hotel always remains available by phone, the hotel reception phones can be grouped together and one or more call channels can be reserved for them. Alternatively, exchange line authorisations for only a part of the available external ports can be assigned to the hotel room phones.

- Configuring the printing function.
  - Using the Configuration Manager (page **Functions > Hotel function > Create print form**)
- Configuring a hotel room function key for each hotel room phone (optional, see the phone manual).

Note: During the check-in or check-out process, the hotel reception phone is not available for incoming calls. Fir such cases, **Call forwarding on busy** should be configured.

Note: If unsuccessful calls are to be documented in the hotel reception phone, the recording of the internal calls in the caller list must be enabled.

- Assigning a room key on each hotel room phone for call charge queries and for setting up wake-up times (optional, see the phone‘s manual).

Note: The value configured in the system phone for cost per call unit (optional, see the phone‘s manual) should match that in the Configuration Manager.

- Configuring the SFTP transfer and switch on the SFTP server if an evaluation programme shall have access to the provided data (e. g. call data list).
  - Using the Configuration Manager (page **Administration > IP configuration > SFTP server**)

InterCom Announcement/Hands-Free

The InterCom function enables an announcement to system phones (one phone or group) from any internal phone without someone having to actively receive a call (for example, in a doctor's office).

Furthermore, the caller's system phone can be programmed to switch on the microphone in addition to the loudspeaker (hands-free) so that a person nearby can speak with the caller using this intercom system.

InterCom announcement and InterCom hands-free are carried out as follows:

- A single system phone or a group is called using the InterCom function.

Note: If a group is called using the InterCom function, all system phones of the group that have given the InterCom permission and are ready to take calls will be called. The group status (logged in, logged out) will not be taken into account. Hands-free calling is not possible in the case of an InterCom call to a group.

- After a few rings, the system phone/system phones of the group members automatically pick up the call.
- Announcement or hands-free calling is executed.

Note: Announcement or hands-free calling is limited to 120 seconds due to security reasons in order to prevent possible misuse of this function (for example, eavesdropping from remote rooms).

- After 120 seconds has elapsed, the connection is automatically interrupted unless the system phone receiver is picked up during this time.

Requirements:

- System phone, that supports InterCom (optional, see the phone manual)

Proceed as follows for configuration:

- Switching the InterCom permission for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings**)
- If necessary, assigning a function key for InterCom on the system phone (optional, see the phone manual).

Internal Phone Numbers

Internal numbers must be configured for the internal ports so that the connected end devices are accessible. Each connected end device receives a subscriber number.

If multiple internal end devices need to be accessible via a phone number, a common group for the internal subscribers must first be configured.

Internal phone are also needed to make e. g. connections to devices, voip accounts or functions.

Note: It is a good idea to create a phone numbering plan for the internal numbers before allocation and then use the following steps to transmit this to the PBX.

Proceed as follows for configuration:

- Create internal phone numbers for subscribers.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers**)
- Create internal phone numbers for groups.
  - Using the Configuration Manager (page **Groups > Phone numbers**)
- Create internal phone numbers for Voice mail and fax boxes.
  - Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Phone numbers**)
- Create internal numbers for door stations and announcement outputs.
  - Using the Configuration Manager (page **Devices > Device wizard**)
- Create internal numbers for alarms.

  - Using the Configuration Manager (page **Devices > Alarms**)
- Create internal phone numbers for automatic receptions.
  - Using the Configuration Manager (page **Functions > Automatic Receptions**)
- Create internal numbers for short-code dialling numbers as part of contacts.
  - Using the Configuration Manager (page **User data > Address books > Contacts and assignment >**)

ISDN Connection

The existing ISDN can be connected to the external S_0 ports of the PBX (optional, see Technical Data in the Instructions).

Proceed as follows for configuration:

- Configuring the connection type for the ISDN connection.
  - Using the Configuration Manager (page **Hardware > Port configuration**)
- Configuring the PTMP connection and entering phone numbers.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > ISDN > PTMP connection**
  - **Public switched tel. networks > ISDN > PTMP connection > Configure**
- Configuring the PTP connection and entering phone numbers.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles**
  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles > Configure**
  - **Public switched tel. networks > ISDN > Extension numbers PTP connection**
- Creating a call distribution for the ISDN connection.
  - Using the Configuration Manager (page **Public switched tel. networks > Call Distribution**)

- If necessary, entering additional numbers for the PTP connection.
  - Using the Configuration Manager (page **Public switched tel. networks > ISDN > Additional PBX base numbers**)

Calendar

Holidays and other days with special uses (e. g., vacation times) are managed in the PBX calendar. These days are taken into account by the PBX during automatic configuration switchovers and, if desired, during Least Cost Routing (LCR) and when executing wake-up calls.

Proceed as follows for configuration:

- Creating holidays.
  - Using the Configuration Manager (page **Functions > Calendar**)

Ringing Sequence

Explanations of the Ringing Sequence
Examples

Explanations of the Ringing Sequence

Different ringing sequences can be set, as it does not always make sense that all phones logged in a group (e. g. in hotline mode) ring at the same time.

The following ringing sequences are possible:

All	• All subscribers logged into the group ring simultaneously (except fixed phones) • Call waiting is applied to all logged-in phones when they are busy • The call is ended by the network provider (after approx. 2 minutes) • In the case of call forwarding on no reply, the group call is forwarded to the destination already after the configured delay time
Linear	• All phones logged into a group ring directly one after the other • Only one phone rings at a time • Busy subscribers are skipped • In the case of call forwarding on no reply, the group call is forwarded to the destination already after the configured delay time
Adding	• All subscribers logged into a group ring one after the other delayed by the specified time • Busy subscribers are skipped

- In the case of call forwarding on no reply, the group call is forwarded to the destination already after the configured delay time

The following properties can additionally be selected for the ringing sequence, whereby the ringing sequence **All** restricts the selection for further properties.

fixed	The order phones are called depends on the assigned priority.
balancing	The order of the called phones depends on a dynamic priority list. After each call, the phone involved is placed last.

rotating	After the first loop, another one starts. In the second loop, call waiting is applied to the busy subscribers. The call is ended by the network provider (after approx. 2 minutes).
without rotating	After the first loop, the call is disconnected.
Split Group	During the ringing pahse, the group must be available for other callers. The group is split up so that as many subscribers as possible are available for each call.
without Split Group	A second caller hears the busy signal while the first caller has not been accepted yet (only possible for **All**).

Proceed as follows for configuration:

- Configuring ringing sequence, priority and ringing delay for the existing groups.
  - Using the Configuration Manager (page **Groups > Properties > Basic settings**)

- If necessary, configuring rework time for the existing groups.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)

Examples

Below are example of calls illustrating the possible options. The following parameters are common to all examples:

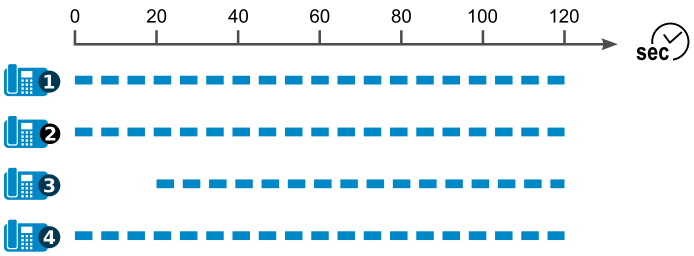
- Number of phones logged in: 4
- Configured time: 20 seconds
- Priority order: 1, 2, 3, 4
- Ringing delay for ringing sequence **All**: enabled for 3
- CF delay time for CF on no Reply: 20 seconds

All

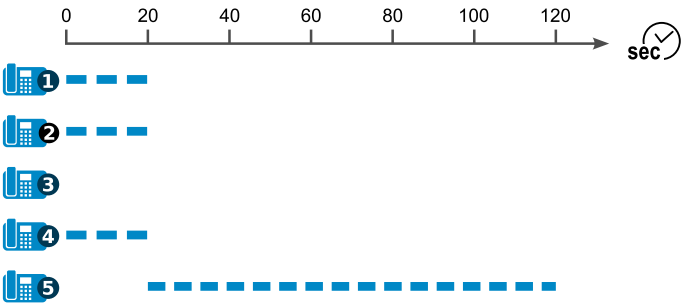
Like **All - Split Group**, but a second caller hears the busy signal while the first caller has not been accepted yet.

All - Split Group

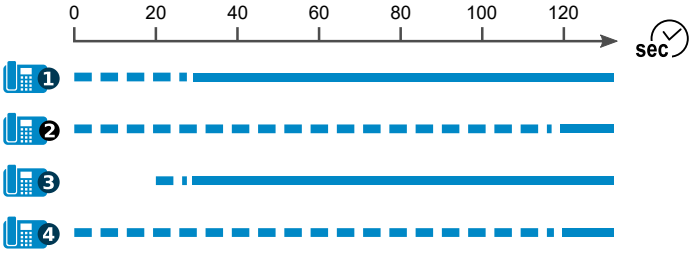
One call:



A call (call forwarding on no reply active on destination phone 5):

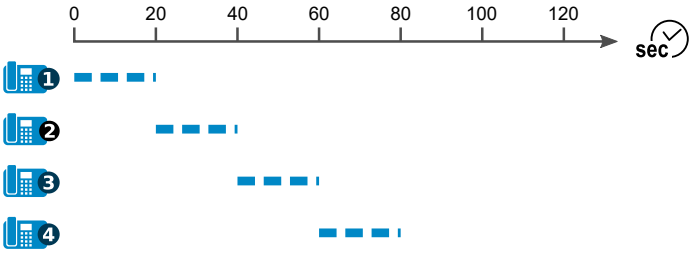


Two calls A and B (call B begins 30 seconds later):

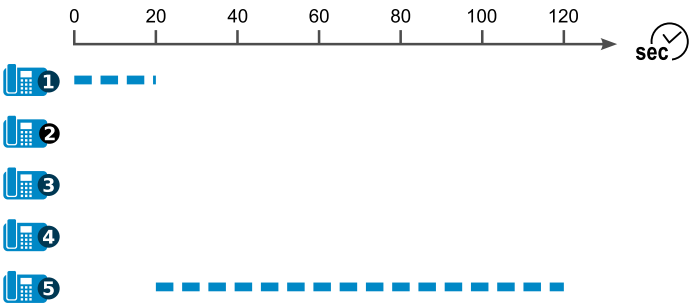


Linear - fixed - split group

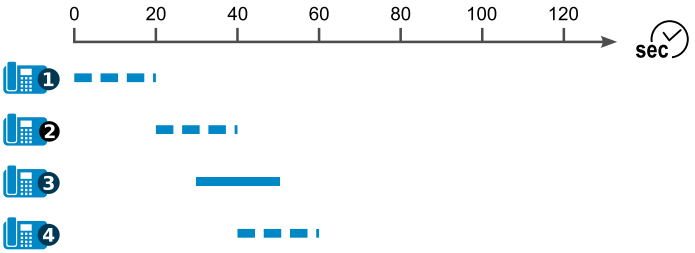
One call:



A call (call forwarding on no reply active on destination phone 5):



Two calls A and B (call B begins 30 seconds later):

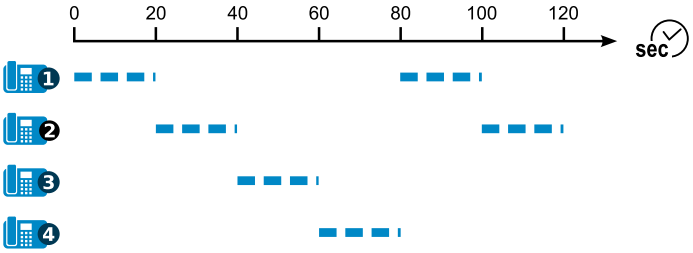


Linear - balancing - split group

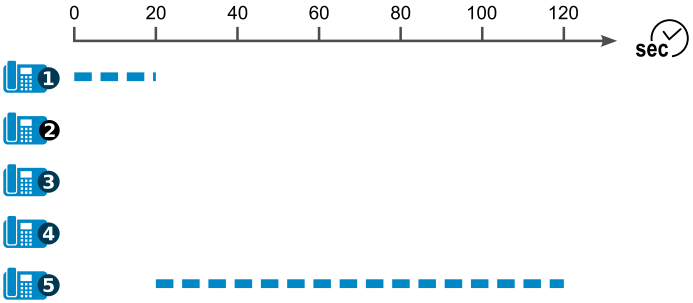
(This is different than **Linear - fixed - split group** due to the changing priorities only.)

Linear - fixed - rotating - split group

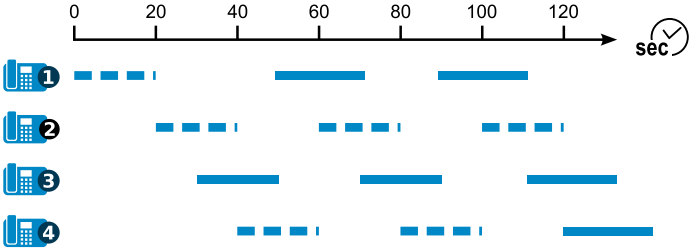
One call:



A call (call forwarding on no reply active on destination phone 5):



Two calls A - - - and B ——— (call B begins 30 seconds later):

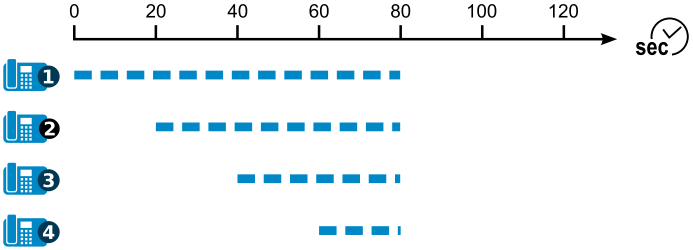


Linear - balancing - rotating - split group

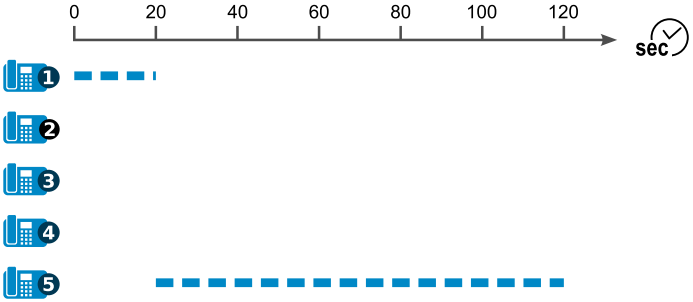
(This is different than **Linear - fixed - rotating - split group** due to the changing priorities only.)

Adding - fixed - Split Group

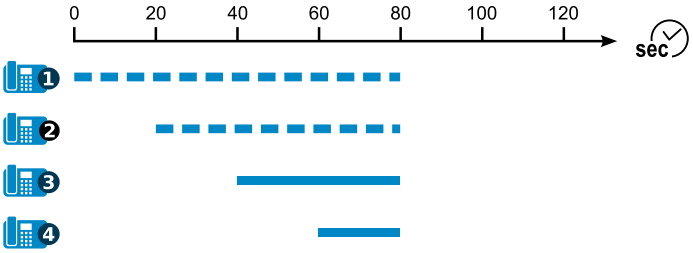
One call:



A call (call forwarding on no reply active on destination phone 5):



Two calls A - - - and B ——— (call B begins 30 seconds later):

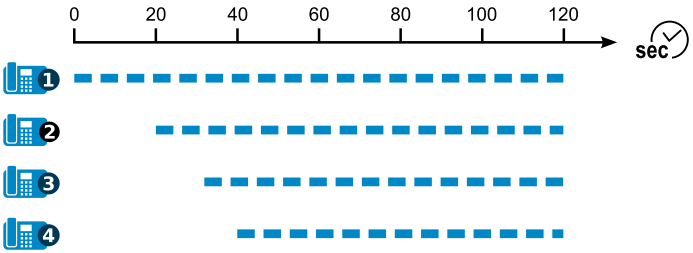


Adding - balancing - Split Group

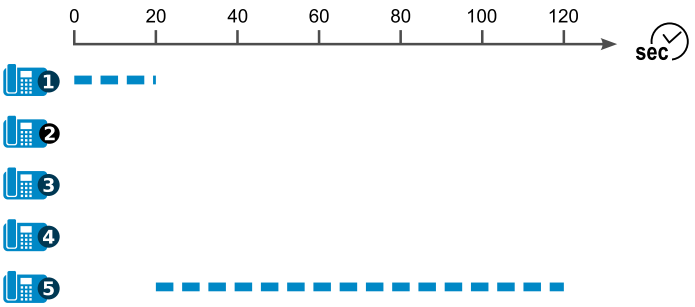
(This is different than **Adding - fixed - Split Group** due to the changing priorities only.)

Adding - fixed - rotating - Split Group

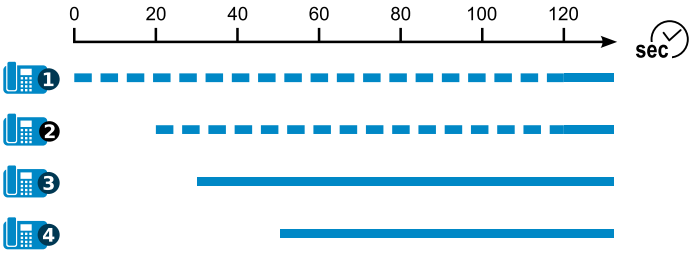
One call:



A call (call forwarding on no reply active on destination phone 5):



Two calls A - - - and B ——— (call B begins 30 seconds later):



Adding - balancing - rotating - Split Group

(This is different than **Adding - fixed - rotating - Split Group** due to the changing priorities only.)

Ringer tones

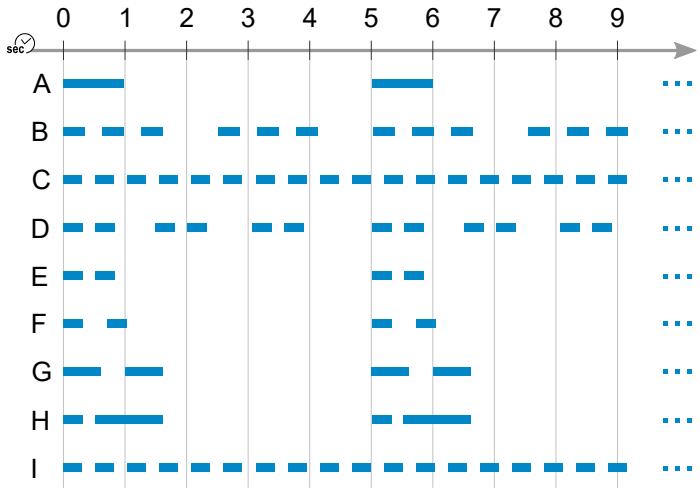
The PBX supports different ringtones for signalling calls on the internal subscribers. They can be used for example to differentiate internal calls, external calls via different external numbers, door calls and alarm calls using the ringtone.

Note: Signalling is carried out by analogue devices as described (depending on the end device used).

Note: Some end devices (e.g., ISDN phones and wireless, analogue phones) generate their own call tone/ringtone. For more information, refer to the description of your end device.

Note: On some system phones the ringtones can be configured based on the call type or the phone numbers (optional, see phone manual).

Ringtones are displayed on analogue phones as follows:



A. 1 x long
B. 3 x short
C. 1st special tone
D. 2nd special tone
E. 3rd special tone
F. 4th special tone
G. 5th special tone
H. 6th special tone
I. Alarm

Configuration procedure:

- Configuring the ringing tone for external calls.
  - Using the Configuration Manager (on the pages that follow)
  - **Public switched tel. networks > ISDN > PTMP connection**

  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles > Configure**
  - **Public switched tel. networks > ISDN > Extension numbers PTP connection**
  - **Public switched tel. networks > ISDN > Additional PTP connection numbers**
  - **Public switched tel. networks > VoIP > Accounts > Configure**
- Configuring the ringing tone for internal calls.
  - Using the Configuration Manager (**Subscriber (scr.) > Properties > Analogue settings** page)
- Configuring the ringing tone for door calls.
  - Using the Configuration Manager (**Devices > Door terminals** page)
- Configuring the ringer frequency.
  - Using the Configuration Manager (**Subscriber (scr.) > All subscribers** page)

Note: For alarm calls the ringing tone is permanently set as **Alarm**.

Backing Up and Recovering Configuration Data

Information on the Chapter

Saving the Configuration for Automatic Upload onto a USB Memory Stick

Uploading the Configuration Automatically

Information on the Chapter

The current configuration data on the PBX can be backed up in a configuration file on the PC or on a data storage device. If necessary, this configuration file can be saved in the PBX again in order to restore previous configuration data.

In addition to the configuration, the configuration data also include the network settings, PBX data, the tool COMfortel IP Editor as well as the music on hold and all annoucements saved in the PBX. When the configuration file is created, there is the option of saving the configuration data without any network settings, PBX data or music on hold/announcements so that the configuration file can also be saved in other PBXs.

The network settings include the following data:

- Ethernet configuration (DHCP client setting, IP address, Sub-network mask, Gateway)
- DNS configuration (First DNS server, Second DNS server)
- HTTP proxy configuration (HTTP proxy settings)
- PPP configuration (Local IP address, Remote IP address)
- Port of the web server

The PBX data include the following data:

- PBX name
- Admin user name
- Admin PIN
- Admin passwort
- Dealer data
- Operator data

- Country

Variants for backing up and recovering:

- Saving and reading out the configuration data
  - Using the Configuration Manager (page **Administration > Data backup**)
- Resetting the configuration
  - Using the Configuration Manager (page **Administration > Data backup**)
- Regenerating the configuration
  - Using the Configuration Manager (page **Administration > Data backup**)
- Inserting and restoring from restoration points
  - Using the Configuration Manager (page **Administration > Data backup**)
- Automatic upload
  - Using the PBX

Saving the Configuration for Automatic Upload onto a USB Memory Stick

Requirements:

- A configuration file saved on the PC

Note: Any network settings saved in the configuration file will be taken over during automatic uploading. If the network settings were not saved in the configuration file, the factory network settings will be retained.

1. Rename the configuration file saved on the PC to auconfig.cfg.
2. Save the configuration file in the root directory of a USB memory stick.

Further steps:

- Perform the automatic upload of the configuration.

Uploading the Configuration Automatically

Proceed as follows:

- Using the PBX

Requirements:

- PBX set to factory settings
- Configuration file named **auconfig.cfg** saved on a USB memory stick

Warning:

- For some installation and maintenance work, it is necessary to open the PBX while it is in operation (qualified electrician only). Make sure that the PBX is never left unattended while working with an open casing. Close the casing after completing this work.
- Only a qualified electrician may perform installation work within an open casing or service work using the buttons inside the casing. If necessary, commission a specialised dealer to perform this work.

Touching voltage-carrying conductors or phone connections can cause life-threatening electric shocks.

Warning: Power surges, which can occur during electrical storms, can cause life-threatening electric shocks and damage or destroy the PBX.

- Do not open the PBX during an electrical storm.

1. Turn the PBX off.

2. Insert the USB memory stick into the USB port (USB host) on the CPU module.
3. Turn the PBX on.

   The configuration data saved on the USB memory stick is automatically saved in the PBX. Finally, the PBX restarts.

   If the **Power** LED lights up in green again, the restart is complete and the PBX is ready for operation.

Note: If the **Power** LED remains continuously lit in red, an error has occurred. Please contact your specialised dealer or the manufacturer directly.

Note: Any network settings saved in the configuration file will be taken over during automatic uploading. If the network settings were not saved in the configuration file, the factory network settings will be retained.

Configuration Switchover

Some functions can, e.g., be configured differently according to the day, night, weekend, vacation or holidays. They are also assigned to different configurations and switched depending on the configuration.

The following functions are configuration-dependent:

- Profile assignment for subscriber (and with it e.g.: exchange line authorisation, phone number presentation and subscriber call forwarding)
- Profile assignment for groups (and with it e.g.: announcement before answering, exchange line authorisation, phone number presentation and group call forwarding)
- Profile assignment for Voice mail and fax boxes (and with it e.g.: readiness)
- Fallback options of the external ports (Fallback number and delayed fallback)
- Call distribution and door terminal distribution
- Relay settings in the configuration dependent mode
- VoIP and GSM Routing
- Call Forwarding for own connections
- Power save mode of the system phones (optional, see the phone manual)

Switching over from one configuration to another can be carried out in two different manners:

- manually
- time controlled (automatic configuration switchover)

For situations where your settings are supposed to apply even after configuration switchovers, you can use the so-called permanent configuration. If permanent configuration is enabled, the configuration-dependent settings will be overridden by the permanent configuration settings.

The following functions can use the permanent configuration:

- Call forwarding of one's own external number (CF for MSNs, DDI numbers, or additional numbers)
- Readiness of the Voice mail and fax boxes

Proceed as follows for configuration:

- Creating sonfigurations.
  - Using the Configuration Manager (page **Time control > Configurations**)
- Making settings configuration dependent.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > Call distribution**
  - **Public switched tel. networks > Call forwarding**
  - **Subscriber (scr) > Profiles**
  - **Devices > Voice mail and fax boxes > Profiles**
  - **Devices > Door terminals**
  - **Devices > Relays (Actuators)**
  - **Routing > VoIP and GSM Routing > Routing is configuration dependent**
- Switching the authorisation for configuration switching for the involved internal phones on.

  - Using the Configuration Manager (page **Subscriber (scr) > Properties > Authorisations**)

- Switching configuration on.
  - Using the Configuration Manager (page **Time control > Configurations**)
  - Using a character string on the internal or external phone (see Configuration Switchover)
  - Using the menu/function key on a system phone (optional, see the phone manual)

Proceed as follows for additional configuration of the automatic configuration switchover:

- Switching automatic configuration switchover on/off.
  - Using the Configuration Manager (page **Time control > Configurations**)
  - Using a character string on the internal or external phone (see Relay)
  - Using the menu/function key on a system phone (optional, see the phone manual under Relay)

Note: Using a phone switch the system relay with operation mode automatic configuration switchover and the Identification number 900 (factory settings).

- Creating switching times for automatic configuration switchover.
  - Using the Configuration Manager (page **Time control > Switching times**)

Proceed as follows for additional configuration of the permanent configuration:

- Configuring the permanent configuration.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > Call forwarding**
  - **Devices > Voice mail and fax boxes > Properties > Call acceptance/Announcement**

Note: The permanent configuration for CF and the voice mailboxes/ fax boxes are set independent of each other and do not depend on each other.

LAN-TAPI

Explanations and Setting Options
Installing TAPI driver (TSP)
Checking the Function of the LAN-TAPI

Explanations and Setting Options

For a CTI solution, the PBX can be connected to a local network (CTI server and CTI clients). This requires CTI software, e. g. PBX Call Assist / 2. Alternatively, we recommend the following software manufacturers:

- ESTOS (estos page)
- ilink (ilink page)

The LAN-TAPI provides the TAPI 2.1 interface over the Ethernet port on the PBX.

A Microsoft server then distributes the functionality throughout the network (third party network connection) onto the separate Microsoft computers so that telephony software, e.g., PBX Call Assist / 2 or other TAPI applications can use the TAPI functions. The PBX is addressed by the server over the Ethernet interface. When a user wants to dial from the client PC, the command is forwarded to the server. The server signals this to the PBX which then establishes the connection.

LAN-TAPI supports the following phones connected to the PBX:

- IP and ISDN system phones
- ISDN Telephones
- Analogue phones
- Standard VoIP phones (SIP)

LAN-TAPI supports the following functions:

- Display of the called MSN (multi-company operation)
- Call forwarding (CFU, CFB, CFNR)
- Call deflection (forwarding before call acceptance)
- Transferring calls

- Conference call
- Splitting a call
- Monitoring (status display of the extensions)
- Pick-Up
- Do not disturb
- Blind transfer using ESTOS

Important: The possible functions are dependent on the end devices/ applications used.

Note: For controlling a phone of the COMfortel D-series via LAN TAPI, the **CTI authorization code** is necessary (from firmware version 1.4 of the phone). The code set for each subscriber in the PBX can be adjusted individually.

Configuration procedure:

- Making settings for LAN-TAPI.
  - Using the Configuration Manager (**Functions > LAN TAPI** page)
- Installing TAPI driver (TSP).
- If necessary, change the **CTI authorization code** for a phone of the COMfortel D-series.
  - Using the Configuration Manager (**Subscriber (scr.) > Function overview > Basic settings** page)

Installing TAPI driver (TSP)

The PBX must be connected with a server PC via a TCP/IP network. The TAPI driver (TSP) that belongs to the PBX must be installed on this server PC from the Auerswald Mega Disk.

Important: If you want to integrate the LAN-TAPI into an existing network, please contact the system administrator responsible for this. Making changes to an existing network may cause considerable functional disturbances.

Note: You can find the current version of the driver on the Internet (see Auerswald: Auerswald Service Portal, FONtevo: FONtevo Support).

Requirements:

- A driver file (*.exe) in its current version saved on a data storage device (PC)
- Server PC that fulfils one of the following minimum requirements:

- Windows 7 (32-/64-Bit)
- Windows Server 2008 und Windows Server 2008 R2
- Windows 8/8.1 (32-/64-Bit)
- Windows 10 (32-/64-Bit)
- Windows Server 2012 und Windows Server 2012 R2
- Windows Server 2016

1. Start driver file (*.exe) on the server PC.

   The TAPI driver installation is started.

2. Follow the installation by clicking Next.

3. Depending on the driver version, the driver may need to be activated at this point. If you do not want to activate it yet, click the Activate later button.

4. In the IP address of the PBX entry field, enter the IP address of the PBX. Enter the network port of the TAPI server in the Port entry field. Enter the password for the network connection of the TAPI server in the Password and Repeat password entry fields. Click OK.

5. Click Close.

6. Restart the PC.

Checking the Function of the LAN-TAPI

To perform a simple function check of the TAPI driver (TSP) as well as the settings configured in the PBX, use the dialling help on Windows on the server PC.

Proceed as follows:

- Using the server PC

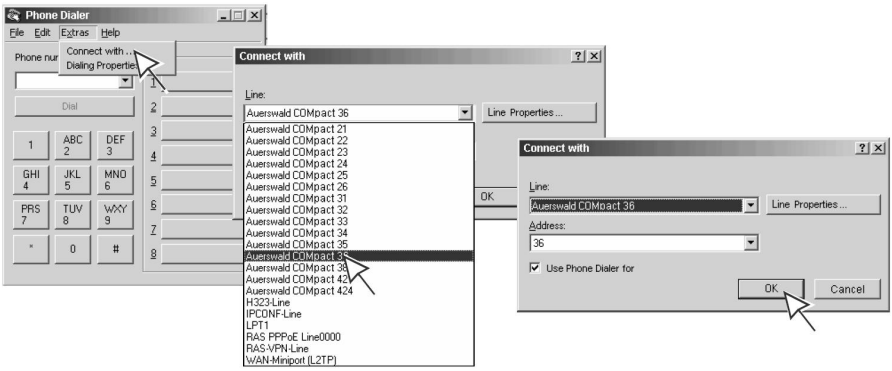
Requirements:

- A completed TAPI drive installation on the server PC
- Operating system: Windows 10

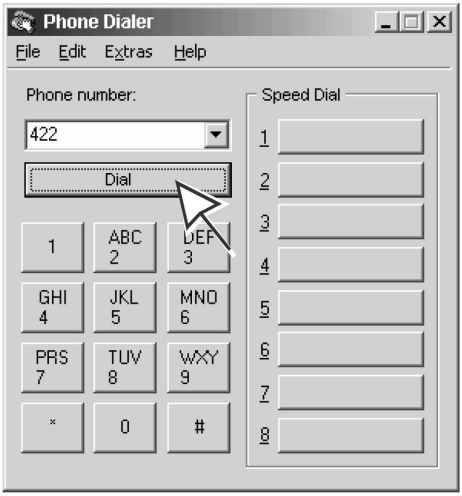
1. Start Windows dialling help. To do this, click Start, enter dialer.exe in the entry field and press Enter.
2. Click Extras > Connect with....

   In the **Line** list field, select one of the phones offered for controlling.

   Click **OK**.



3. In the Phone number entry field, entry an internal or external number. Enter the phone number exactly as you would dial it and click Dial.



The controlled phone is called. As soon as you pick up the receiver, the destination number dialled (in this example: 422) is called.

Loudspeaker Announcement

This function enables loudspeaker announcements over an active loudspeaker (for example, a loudspeaker system in a waiting room). In addition, a previously specified phone number is called by an internal phone. The call connection is immediately established after dialling the phone number.

In order to make loudspeaker announcements, a device that can be used as announcement output (e. g. an a/b-Audiobox at a loudspeaker system) must be connected to the PBX. This announcement output must be created in the configuration of the PBX and assigned an internal number to call it.

Proceed as follows for configuration:

- Creating an announcement output with internal number.
  - Using the Configuration Manager (page **Devices > Device wizard**)
- Creating a relais and assigning it to an announcement output.
  - Using the Configuration Manager (page **Devices > Announcement outputs**)
- If necessary, making additional settings for the announcement output.
  - Using the Configuration Manager (page **Devices > Announcement outputs**)
- Switching the authorisation for speaker announcements for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)

LCR

LCR (Least Cost Routing) is a function of the PBX that enables the subscriber to automatically make calls using the most cost-effective phone service provider. The PBX determines the phone service provider to be used by the dialled phone number and the configured tariff tables.

Note: Automatic LCR is not carried out for the entered emergency numbers (**Public switched tel. networks > Emergency calls**).

Note: If a call is initiated via a targeted exchange line access, automatic LCR is not carried out.

Requirements:

- Support of dial-around service by the network provider

Proceed as follows for configuration:

- Switching LCR at the external connection on.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > ISDN > PTMP connection**
  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles > Configure**
  - **Public switched tel. networks > VoIP > Accounts > Configure**
- Selecting the subscribers on which automatic LCR should be carried out.
  - Using the Configuration Manager (page **Routing > Soft-LCR 4.0 > LCR subscriber**)

- If necessary, switching LCR for Call Through calls on.
  - Using the Configuration Manager (page **Functions > Call Through**)
- Creating LCR data and keeping them up to date.
  - Using the Configuration Manager (on the following pages)
  - **Routing > Soft-LCR 4.0 > Provider**
  - **Routing > Soft-LCR 4.0 > Networks and LCR import**
  - **Routing > Soft-LCR 4.0 > LCR tables**
- Creating the holidays in the calendar of the PBX and switching them on for LCR.
  - Using the Configuration Manager (page **Functions > Calendar**)

Macro numbers

With the aid of macro numbers, longer digit sequences and macro commands can be dialled more easily.

The Makrokurzwahlnummers are part of the internal phone numbering plan and therefore must at least have 2 digits.

Proceed as follows for configuration:

- Configure shorte-code macros.
  - Using the Configuration Manager (on the following pages)
  - **Functions > Short-code macros**

MWI

MWI (Message Waiting Indication) is a SIP function which allows to signal the presence of new voice and fax messages on internal VoIP end devices. Signalling is possible for all voice mail and fax boxes for which the subscriber in question is entered as user.

You can use this function with the following VoIP end devices:

Auerswald VoIP system phones and standard VoIP phones

On standard VoIP phones, the LED of a function key assigned to MWI is used for signalling (optional, see the phone manual).

If operated as a system phone, the LED of the message key is used for signalling (optional, see the phone manual).

Auerswald IP DECT phones

The symbol on the display is used for signalling on COMfortel M-100, M-200, M-210, M-300 and M-310 handsets.

The function can be configured using the Configuration Manager of the IP DECT servers COMfortel WS-400 IP or COMfortel WS-650 IP, on th page **Configuration > SIP** under **Message waiting indication** (see the IP DECT server manual).

Standard VoIP end decives of other manufacturers

Requirement for the support of the MWI function is the activated **SIP convenience package** / **SIP convenience package brand plus** enabled (optional, see activation).

Signalling and configuration depend on the end device (see the end device manual).

Restart

A restart (reboot) is when you cause the PBX to reboot while it is already functioning. The configuration of the PBX is not deleted.

A restart is required, for example, in the following situations:

- After the operating software on the PBX has been updated (firmware update)
- After the operating software on the PBX crashes
- After a change to the configuration that cannot be applied when the PBX is in operation

An automatic restart is executed after the following functions have been carried out:

- An instant firmware update
- Restoring firmware (reboot)
- A factory settings restoration
- Restoring Configuration Data from a Restoration Point
- A configuration reset

Note: Resetting the configuration requires only a limited restart. Only some of the PBX functions are restarted. All calls are ended, and all data services are closed. The web server remains unaffected by this.

There are also various options for performing a restart yourself when necessary.

Proceed as follows for a manual restart:

Note: All connections to the web server, all calls and data services will be disconnected this way. All unstored information will be lost!

- Immediate restart.
  - Using the Configuration Manager (page **Administration > Firmware update and restart**)
  - By pressing a button on the PBX

Note: During an immediate restart, no consideration is made of existing calls. This means that all connections to the web server, all calls and all data services closed immediately. Unsaved data is lost.

- Delayed restart.
  - Using the Configuration Manager (page **Administration > Firmware update and restart**)

Note: A delayed restart runs just like an immediate restart, except that current calls are taken into consideration. As soon as all of the existing calls have ended (also before the expiration of the time specified), the PBX restarts. During the delay, the PBX prevents new calls from being established. If a call is still active when the timeout occurs, the call is interrupted in order to carry out the restart.

Emergency Calls

The emergency numbers set up on the PBX are used for emergency calls. These calls are handled in a special way.

Depending on the configuration, emergency calls can be made either with or without dialling the exchange line access number. This has the benefit that even people who don't know how to use exchange line access numbers can make emergency calls.

A system phone can also be used to make emergency calls if the screen lock is switched on (optional, see phone's instructions).

Users who cannot usually make external calls can make emergency calls if the appropriate exchange line authorisation is enabled.

Note: If you want to prevent emergency numbers from being dialled (e.g. to an external subscriber on the same PBX), set up a call restrictor for each individual emergency number.

Emergency calls are then only transmitted via external connections/ providers/accounts on which emergency call ability is enabled. If the telephone cannot make an emergency call, the subscriber is informed by an announcement.

Emergency calls can be configured to take priority over other calls. If necessary, change the a call can be interrupted to enable an emergency call to be made.

The phone number is always sent along with an emergency call. If number presentation suppression is switched on, it is disabled automatically when the emergency call is made.

Note: Local service numbers, such as numbers for doctors or local authorities are also increasingly being added to the emergency numbers. These service numbers can also be set up in the PBX and,

like emergency numbers, dialled either with or without the exchange line access number, depending on the configuration. However, calls to service numbers are not handled like emergency calls. If you want to prevent service numbers from being dialled (e.g. to an external subscriber on the same PBX), set up a call restrictor for each individual service number.

Configuration procedure:

- switch emergency call ability for external connections, providers and accounts on/off.
  - Using the Configuration Manager (on the pages that follow)
  - **Public switched tel. networks > ISDN > PTMP connection**
  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles > Configure**
  - **Public switched tel. networks > Analogue Connections**
  - **Public switched tel. networks > Voice over IP (VoIP page) > Provider > Configure**
  - **Public switched tel. networks > VoIP > Accounts > Configure**
- Set up emergency calls.
  - Using the Configuration Manager (**Public switched tel. networks > Emergency calls** page)

Note: Public emergency numbers are set as default settings when you select the country settings (using the configuration wizard), when commissioning the PBX. These numbers cannot be deleted. You can only enter different public emergency numbers as default settings by changing the country settings at a later point in time. Be aware of any possible conflicts between the country-specific emergency numbers and

the existing internal telephone numbering plan. If necessary, assign new "Internal numbers".

- If necessary, change the configuring the country in which the PBX is operated.
  - Using the Configuration Manager (**Administration > Dealer and operator data** page)
- Input your own emergency numbers.
  - Using the Configuration Manager (**Public switched tel. networks > Emergency calls** page)
- If necessary, change the entering service numbers.
  - Using the Configuration Manager (**Public swichted tel. networks > Emergency calls > Service numbers** page)
- Configure exchange line authorisations.
  - Using the Configuration Manager (on the pages that follow)
  - **Subscriber (scr.) > Properties > Exchange line settings**
  - **Groups > Properties > Exchange line settings**
- If necessary, change the enabling the function "Prevention of emergency calls".
  - Using the Configuration Manager (**Subscriberiber (scr.) > Properties > Exchange line settings** page)
- Manage and assign audio files for announcements.
  - Using the Configuration Manager (on the pages that follow)
  - **Functions > Music on Hold and Announcements > File management**

- **Functions > Music on Hold/announcements > File assignment > Emergency call**
- Using a character string on the internal phone (see page)

Note: The PBX is supplied with announcements in English and German. You can create other announcements in the local language at a later point in time.

Call Parking

Call parking lets a caller in a waiting position be transferred so that the desired communication partner can accept the call on another internal phone.

Call parking is carried out as follows:

- The call is first transferred to a phone number that has been previously specified for this function and then parked.
- The caller is kept on hold in the PBX and listens to hold music.
- The desired communication partner is then notified of the waiting caller. He can then take the call by dialling the phone number that had previously been transferred.

Proceed as follows for configuration:

- Making settings for call parking.
  - Using the Configuration Manager (page **Functions > Call parking**)
- Switching the authorisation for call parking for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)

Online Name Search

The online name search lets you accept names to put with existing phone numbers via a service provider on the Internet.

If a phone number is transmitted during an incoming call, the entries in the phone books (PBX, system phone) are checked and, if necessary, an associated name is displayed and saved in the call data list.

If the phone number transmitted is not found in the phone books, the function attempts an online name search of the name for the phone number on a server on the Internet that offers a search service. If the server is accessible and the desired data is stored there, this information is transferred to the phone. The name is shown on the display of a system phone (optional, see phone manual) and saved in the **External partner** column of the call data list.

Clicking in the **External partner** column of the call data list adds the name to the call data record after the fact using the online name search function. The name can also be verified and updated.

Clicking in the **External name** column of the call data list transfers the phone number and name of the external calling partner into the PBX phone book.

Proceed as follows for configuration:

- Creating service providers.
  - Using the Configuration Manager (page **Functions > Online name search > Basic settings**)
- Creating group filters for service providers.
  - Using the Configuration Manager (page **Functions > Online name search > Basic settings > Configure**)

- Configuring the text converter.
  - Using the Configuration Manager (page **Functions > Online name search > Text converter**)
- Switching online name search on.
  - Using the Configuration Manager (page **Functions > Online name search > Basic settings**)

Multi-path Call Forwarding

Multi-path call forwarding is used to signal internal and external calls, in addition to call distribution for an internal or an external number, e.g., on a mobile phone. Multi-path call forwarding is also executed if the "main phone number" is, for example, busy. If call forwarding for external numbers is additionally enabled, multi-path call forwarding might work only limited.

Proceed as follows for configuration:

- Switching multi-path call forwarding on.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) > Properties > Reachability**
  - For groups: **Groups > Properties > Reachability**
  - Using a character string on the internal or external phone (see Follow-me)
- Configuring multi-path call forwarding configuration dependent.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) > Profiles > ... > Reachability**
  - For groups: **Groups > Profiles > ... > Reachability**

Pick-up and Call Take-over

Using Pick-up, you can take a call on your own phone that was originally sent to another internal phone.

If the call has already been taken, the call can be taken using same procedure if the call take-over has been allowed by the subscriber being called. This function is useful, for example, in the case of answering machines.

Proceed as follows for configuration:

- Switching authorisation for pick-up for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)
- Switching call take-over for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings**)

Preliminary Pickup Query and Monitoring

The PBX supports the SIP function BLF (in accordance with RFC 4235). BLF allows VoIP end devices to monitor other subscribers and receive information about their calls.

Support of the SIP function BLF provides the following functions:

- **Preliminary pick-up query**: A preliminary pick-up query allows a subscriber, even before a pick-up, to see which caller he would be picking up. Based upon the phone number of the caller, the subscriber can then decide whether to pick up the call or not. As precondition, the called subscriber or group must have enabled "preliminary pick-up query and monitoring“.
- **Monitoring or BLF (Busy Lamp Field)**: BLF allows VoIP end devices to monitor other internal subscribers, groups, waiting positions of call parking, voice mail/ fax boxes or door terminals and receive information about their calls. As precondition for the display of phone numbers, the monitored subscriber or group must have enabled "preliminary pick-up query and monitoring". Whether voice mail and fax boxes can be monitored in this way depends on the subscribers and groups who own the boxes or have redirected their calls to them.

You can use this functions with the following VoIP end devices:

- **Auerswald/FONtevo VoIP system phones and standard VoIP phones (except COMfortel VoIP 2500 AB)**: Other subscribers are monitored with the help of the LEDs of those function keys which are assigned to speed dialling (optional, see manual for the phone).
- **Standard VoIP end devices of other vendors**: As precondition to support the SIP function BLF, the **SIP convenience package** / **SIP convenience package brand plus** must have been unlocked (optional, see activation).

Monitoring of other subscribers on the same PBX depends on the end device (see manual for the end device).

Proceed as follows for configuration:

- Switching preliminary pick-up query and monitoring on.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) >Properties > Settings**
  - For groups: **Groups > Properties > Settings**
- Configuring preliminary pick-up query and monitoring configuration dependent.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) >Profiles > ... > Settings**
  - For groups: **Groups > Profiles > ... > Settings**

Pick-up of All Calls to Group Members

If a subscriber has assigned a speed-dialling key to a group, the speed-dialling key blinks red to signalise an incoming call to the phone number of the group so that he can take over the call.

If the function Pick-up of all calls to group members is enabled, not only calls to the phone number to the group, but also calls directly to the phone number of a group member will be signalised.

Proceed as follows for configuration:

- Switching pick-up of all calls to group members on.
  - Using the Configuration Manager (page **Groups > Properties > Settings**)
- Configuring pick-up of all calls to group members configuration dependent.
  - Using the Configuration Manager (page **Groups > Profiles > ... > Settings**)

Private (Personalised) Exchange Line Access

Private (personalised) exchange line access enables separate billing for business and personal calls for individual employees. Using private exchange line access, calls initiated are given a special identification while the call data is being recorded. This allows the charges to be billed to the employee.

Proceed as follows for configuration:

- Switching differently configurable functions for private and business calls on.
  - Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)
- If necessary, switching private exchange line access without PIN entry for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings**)

Profiles

Profiles can be created for subscribers, groups, and Voice mail and fax boxes. These profiles control the behaviour by determining the properties. In this way, analogue behaviour can be reached via the configuration-dependent assignment of profiles. When creating the profiles, it can be determined whether the individual settings from the profile can be changed via the properties of the subscriber, group, or the boxes or not.

Proceed as follows for configuration:

- Create profiles for subscribers and assign them depending on the configuration.
  - Using the Configuration Manager (on the following pages)
  - **Subscriber (scr.) > Profiles > Configuration**
  - **Subscriber (scr.) > Profiles > Profile assignment**
- Create profiles for groups and assign them depending on the configuration.
  - Using the Configuration Manager (on the following pages)
  - **Groups > Profiles > Configuration**
  - **Groups > Profiles > Profile assignment**
- Create profiles for voicemail and fax boxes and assign them depending on the configuration.
  - Using the Configuration Manager (on the following pages)
  - **Devices > Voice mail and fax boxes > Profiles > Configuration**
  - **Devices > Voice mail and fax boxes > Profiles > Profile assignment**

Project Assignment

The exchange line access with project assignment allows the sorting of external calls to a specific project/customer/client (e.g., within a law firm). With call data management, the charges and the time expenditures may be assigned to different projects/customers/clients.

Note: On a system phone the project number can be selected not only before the call but also during a call (optional, see the phone manual). This makes the assignment of incoming calls (for billing the time expenditure) possible.

Requirements:

- Assigning of project numbers function enabled (optional, see activation)

Proceed as follows for configuration:

- If necessary, configuring a project list on a system phone (optional, see the phone manual).
- If necessary, configuring a function key for project assignment on the system phone (optional, see the phone manual).

Provisioning for IP Phones

Device Configurability and Characteristics and Setting Options
Provisioning files
Creating a New Provisioning File
Provisioning COMfortel D-series / 3600 IP

Device Configurability and Characteristics and Setting Options

Provisioning is the automated configuration of phones via a provisioning server from which the phones can transmit the necessary configuration data.

Further help in the Auerswald DokuWiki.

The phone accepts the configuration data directly from a file distributed pver the network.

Configurability	
COMfortel 1200 IP	Yes, but not automatically/by PBX search
COMfortel 1400 IP	Yes
COMfortel 2600 IP	Yes
COMfortel 3200	Yes, but not automatically/by PBX search
COMfortel 3500	Yes, but not automatically/by PBX search
COMfortel 3600 IP	Yes
COMfortel D-100/D-200	Yes
Yealink phones	Yes
Snom phones	Yes

Provisioning via the provisioning server can be done regularly (e.g. weekly) automatically. You can set the weekday, time and the address of the provisioning server on the phone.

Possibilities of automatic provisioning:

- Provisioning with the help of DHCP data
- Provisioning via the redirect server
- Provisioning from PBX search
- Provisioning via a set provisioning server

Note: For further information refer to the Advanced Information of the IP phones.

Specific characteristics COMfortel 1200 IP

The COMfortel 1200 IP is no system phone and cannot be configured automatically by PBX search, therefore, further settings on the phone are required (see Advanced Information of the phone). However, the phone can be identified by its MAC address.

Specific characteristics COMfortel 1400 IP/2600 IP/3600 IP

If configured as system phones, these phones will find the provisioning server automatically in factory settings state – as of firmware version 2.4 – and will automatically carry out a provisioning process in case of modifications.

Provisioning-capable PBX systems:

- COMpact 4000
- COMpact 5000/R

- COMpact 5200/5500/R
- COMmander 6000/R/RX

Specific characteristics COMfortel 3200/3500

Provisioning is only possible cyclically, in accordance with a schedule. The address of the provisioning server must be communicated via DHCP or entered manually on the phone.

Particularities COMfortel D-100/D-200

These devices are not system phones, but provisioning is possible. The devices automatically find the provisioning server in the as-delivered state and automatically carry out a new provisioning if changes are made.

You can use the templates in a provisioning file to extend the range of functions on the phone. These additional functions can be assigned to function keys (see Advanced Information of the phone).

Further help in the Auerswald DokuWiki.

Configuration procedure:

- Assigning a MAC address (page **Subscriber (scr.) > Properties > VoIP settings**
- Creating a file as header file (page **Subscriber (scr.) > IP provisioning > File Management** page)
- If required, selecting a provisioning file as template (page **Subscriber (scr.) > IP provisioning > File Management** page)
- Creating provisioning files (see Auerswald DokuWiki page)
- Making provisioning settings on the phone:

- Setting the provisioning type (see manual of the phone)
- If necessary, change the entering the address of the provisioning server, starting with https://, (see manual of the phone)

Note: For COMfortel 3200/3500, you must enter **https://IP-Anlage:4433/provdata?mac=<MACADR> under Settings > Update and Provisioning > Automatic Provisioning > Provisioning Server** .

- Proceed as follows for configuration (page **Subscriber (scr.) > IP provisioning > Settings**):
  - Switching provisioning on/off
  - Switching access without client certificate on/off
  - Switching provisioning only with local network on/off
  - Setting the language of the phone after configuration
- If necessary, change the remove unassigned phones (page **Subscriber (scr.) > Unassigned phones** page)

Provisioning files

Provisioning files contain configurations for phones. For Auerswald/ FONtevo IP phones, these provisioning files are *.xml files, built after a defined scheme. They contain e.g. account, provider or function key information in an *.xml file.

A provisioning file that was not generated in the PBX (e.g. via a backup), is typically applicable for only one subscriber, as the file contains personal data such as account information. Provisioning data is user-specific if they contain at least username and password of the subscriber at the VoIP account.

A provisioning file can also be generated from a standard provisioning file by saving it under a different file name.

Further help in the Auerswald DokuWiki.

Provisioning templates

Provisioning templates are a specific form of provisioning files. A provisioning file can be saved and selected in the PBX. If an IP phone requests a provisioning, the PBX replaces the user-specific data of the provisioning file or rather the template with suitable data content for the current subscriber. A provisioning template can be assigned to several subscribers.

Standard Provisioning Files

The standard provisioning file is a specific provisioning template type and only contains the basic structure of a provisioning file. The file cannot be deleted or exported.

Header Files

Header files do not contain configuration data but refer to one or more provisioning files which contain configuration data. Header files contain a list of URLs which are processed in order. Provisioning files can neither be provided by the PBX or by an external server. A combination of both variants is permitted. The header file is the file that the phone must read first. It can contain references to sub files. Any files can be imported, however, Auerswald/FONtevo IP phones only support *.xml files.

Further help in the Auerswald DokuWiki.

Note: For the COMfortel WS-500 device type, the use of a header file is not possible.

Variables as placeholder

Variables for the parameters to be provisioned are created as placeholders in the configuration files provided by the manufacturer.

Provisioning examines a file to be delivered to the phone. The recognised variable names are replaced by the values from the PBX configuration. The changed file is then sent to the device to be provisioned.

Creating a New Provisioning File

Via the Configuration Manager of a COMfortel 1400IP/2600IP/3600IP you can create a phone configuration and then select the configuration data which are to be distributed to other phones.

Proceed as follows:

- Using the Configuration Manager

Note: You find the account IDs for the PBX and for the exchange lines in the file editor under **Subscriber (scr.) > IP provisioning > File management > Edit > Account IDs**.

1. Open the Configuration Manager of the IP phone.
2. Open the page Settings > Personal > Backup and reset.
3. In the line Phone configuration under Create backup, click Export.
4. Accept the legal notes by entering Ok in the entry field.
5. Click Export.
6. Save the created *.xml file created on your PC.
7. Open the *xml file with a text editor, e. g. Notepad++.
8. Press Alt 0 in Notepad++ to close the sub-levels to get a better overview.
9. Select the level which you want to distribute to other phones by deleting the levels that are not required.

Note: In the case that the file shall be used as Template, the section <providersAndAccounts> </providersAndAccounts> must not be deleted as it is replaced by the subscriber-specific data by the PBX.

10. If required, open the level by clicking ⊞ in order to modify the settings made there.

Note: We recommend to contact the system administrator or specialised dealer for editing provisioning files.

Provisioning COMfortel D-series / 3600 IP

Following described is the provisioning in the "zero touch" mode of a **COMfortel D Series** and a **COMfortel 3600 IP** telephone in connection with an Auerswald/FONtevo PBX. The device is in this mode

- before the initial commissioning
- after resetting the device to factory settings

Proceed as follows:

- Via the configuration manager of the PBX

Proceed as described in the following.

1. Determine the MAC address. It can be found on the package as well as on the back of the telephone.
2. Create a VoIP subscriber for the following device type (page Subscriber (scr.) > Telephone number > New):
3. Assign or enter the corresponding MAC address.
4. Create a provisioning file for the following device type (page Subscriber (scr.) > IP provisioning > File management):
5. Grant access to the subscriber.
6. Connect the device.

   It now receives the corresponding settings from the assigned provisioning file and is afterwards ready for operation.

Relays (Actuators)

Expalanations and Setting Options
Relay operating modes

Expalanations and Setting Options

The PBX supports the connection and control of various types of relays. A relay can be part of a connected door terminal or another connected device (e. g. an IP switch relay). Relays can have various functions on the PBX. For a detailed description of the operating modes, refer to the chapter .

Note: The internal system relay with the identification number 900 (factory setting) has a special position. It can only be used in the operating type automatic configuration switchover to switch it via phone on or off. For this purpose, the relay is being switched on or off. For the operation of standard phones see Relay, for system phones see manual of the phone.

Proceed as follows for configuration:

- Creating an IP switch relay.
  - Using the Configuration Manager (page **Devices > Device wizard**)
- Creating a relay.
  - Using the Configuration Manager (page **Devices > Relays (Actuators)**)
- Configuring the operating mode for the relays.
  - Using the Configuration Manager (page **Devices > Relays (Actuators)**)
- Configuring relays.
  - Using the Configuration Manager (page **Devices > Relays (Actuators) > Configure**)

- Switching authorisation for switching of relays and alarms for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)
- Switching relays.
  - Using a character string on the internal or external phone (see Relay)
  - Using the menu/function key on a system phone (optional, see the phone manual)

Relay operating modes

Relays can have various functions on the PBX.

You can configure the following operation modes:

Speaking

The relay is assigned to a door terminal or an announcement output.

The PBX switches on the relay as soon as an audio connection has been established between the PBX and the assigned device (e.g. an a/b-Audiobox).

The PBX switches the relay off as soon as the audio connection is terminated.

Possible use:

- Switching an LED of the door terminal
- Switching the loudspeaker area of the loudspeaker system for the announcement output

Door opener

The relay is assigned to a door terminal.

The PBX switches on the relay as soon as an internal subscriber triggers door opening for the assigned door terminal.

Alternatively, switching can be triggered manually by phone.

The PBX switches off the relay after the defined switching duration.

Possible use:

- Switching a door opener

Light

The relay is assigned to a door terminal.

The PBX switches on the relay as soon as an internal subscriber triggers light switching for the assigned door terminal.

Alternatively, switching can be triggered manually by phone.

The PBX switches off the relay after the defined switching duration.

Possible use:

- Switching a staircase light

Alarm

The relay is assigned to an alarm.

The PBX switches on the relay as soon as the assigned alarm is triggered.

Alternatively, switching can be triggered manually by phone.

The PBX switches off the relay after the defined switching duration.

Alternatively, switching off can be triggered by phone after the alarm has been acknowledged.

Possible use:

- Switching a siren or an indicator light

Music on hold

The relay is assigned to music on hold.

The PBX switches on the relay when a call is to be transferred or in the case of a query call, when a communication partner waiting in the background is to hear music on hold.

The PBX switches off the relay after the defined switching duration.

Possible use:

- Playback of music on hold during a transfer or a query.

PC relay

The relay is assigned to a subscriber.

The PBX switches on the relay as soon as the assigned subscriber is being called.

Alternatively, switching can be triggered manually by phone.

The PBX switches off the relay after the defined switching duration.

Possible use:

- Switching a PC

Second ringer

The relay is assigned to a subscriber, a group or a door call.

The PBX switches on the relay as soon as the assigned subscriber or group is being called, or at the start of a door call.

The PBX switches off the relay at the end of the call.

Possible use:

- Switching a second ringer

Depending on configuration

The relay is assigned to one or more configuration/s (day, night, etc.).

The PBX switches on the relay as soon as an assigned configuration is switched on.

The PBX switches off the relay as soon as a configuration to which it is not assigned is switched on.

Alternatively, switching on/off can be triggered manually by phone.

Auxiliary relay (secondary)

The secondary relay is assigned to a primary relay.

The PBX switches the secondary relay together with the assigned relay.

Manual relay

The PBX does not automatically switch the relay on and off.

Switching on/off can only be triggered manually by phone.

Call Channel Reservation

If a group of subscribers should have the option of making an external call at any time, the call channels on an external ISDN connection or a VoIP account can be reserved for the group. This call channel is only available for subscribers who are logged in this group outgoing.

Proceed as follows for configuration:

- Switching call channel reservation on.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)
- Configuring call channel reservation configuration dependent.
  - Using the Configuration Manager (page **Groups > Profiles > ... > Reachability**)

Phone Number Display (CLIP)

The phone number display (CLIP) enables the phone number of the caller to be shown on the display of a system phone, an ISDN phone or on a CLIP-capable analogue phone. The prerequisite is that the caller transmits a phone number.

A few CLIP-capable analogue phones support the recording of the date and time in conjunction with the caller list. On some, it is also possible to display the name instead of or in addition to the phone number.

Proceed as follows for configuration:

- Switching CLIP information for the internal analogue phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Analogue settings**)

Number presentation

With the number presentation (outgoing), it is possible to select the number presented to the partner called. A different number may be presented for a business or a private call or for a call as a member of a group (logged into the group as outgoing).

The suppression of the number presentation (outgoing) can prevent the presentation of your own number to the partner called (anonymous call). This feature allows you to decide before starting a call if the number should be presented to the partner called or not. The number from your own connection can still be transmitted to the network provider so that the charges can be calculated correctly.

The number presentation (incoming) lets you present the actual number reached to the incoming caller. The number presented back to you may be one of the numbers from the exchange line connection reached. This number may be different than the number actually dialled by the caller. This is useful if, for example, another subscriber takes the call via pick-up or call forwarding is active. The number to be presented back may also be suppressed (COLR).

For outgoing calls from a call list of a system telephone COMfortel 1400 IP/2600 IP/3600 IP or COMfortel 1200/1400/1600/2600 (ISDN) the phone number that originally received the incoming call is always transmitted. It is therefore not the phone number of the subscriber who received this call through active call forwarding that is transmitted.

With the number presentation (internal) for group calls, it is possible to select the number which is presented when a subscriber makes an internal call as a group member.

Note: The function can be configured via keypad codes (depending on the provider) via the keypad of the phone.

Configuration procedure:

- Configuring the number presentation (outgoing) and (incoming) and number suppression.
  - Using the Configuration Manager (on the pages that follow)
  - For subscriber: **Subscriber (scr.) > Properties > Exchange line settings**
  - For groups: **Groups > Properties > Exchange line settings**
  - For the exchange line transfer function: **Functions > Exchange line transfer**
  - For the Call Through function: **Functions > Call Through**
- Configuring the number presentation (outgoing) and (incoming) and number suppression configuration dependent.
  - Using the Configuration Manager (on the pages that follow)
  - For subscriber: **Subscriber (scr.) > Profiles > ... > Exchange line settings**
  - For groups: **Groups > Profiles > ... > Exchange line settings**
- Configuring the number presentation (internal) for group calls.
  - Using the Configuration Manager (**Groups > Properties > Settings** page)
- Configuring the number presentation (internal) for group calls configuration dependent.
  - Using the Configuration Manager (**Groups > Profiles > ... > Settings** page)

Call Forwarding

Call Forwarding Types
Call Forwarding for Subscribers
Call Forwarding for Groups
Call Forwarding for External Numbers

Call Forwarding Types

Call forwarding allows calls to be forwarded, e.g., to a mobile phone.

The PBX support three different call forwarding types:

- Call forwarding for subscribers
- Call forwarding for groups
- Call forwarding for external numbers

As there can be many different reasons for not accepting a call, for example, the subscriber in question is not present or is on another line, three call forwarding types are provided:

CF unconditional: The phone being called does not ring. The call is forwarded immediately.

CF on busy: The call is forwarded immediately but only if the phone being called is busy.

CF on no reply: The phone being called rings for a certain amount of time. If the call is not accepted, it is forwarded.

Note: A different destination can be configured for each of the three call forwarding types.

Note: If **CF on busy** and **CF on no reply** are enabled at the same time, both types apply. Depending on which case occurs – the phone is busy or nobody answers it – the call is forwarded to their respective phone numbers.

Note: If **CF unconditional** is enabled in addition to the **CF on busy** and/or **CF on no reply**, all of the calls are forwarded immediately. In this case, the other call forwarding types are overridden, but remain enabled.

As soon as **CF unconditional** is disabled, the other types that are still enabled are again applicable.

Call Forwarding for Subscribers

When call forwarding (CF) for subscribers is used, internal and external calls to a single phone can be rerouted to other internal phones or external connections. In this way, the subscriber in question or his replacement can take these calls on another phone.

Call forwarding for subscribers makes sure that the subscriber in question or his replacement is always available at his own phone number, even if the subscriber cannot take a call on his own phone.

Note: To avoid having to use a second call channel in the case of call forwarding to an external destination, the PBX tries to reroute the call directly at the network provider using call deflection. The following requirements must be fulfilled for this purpose: 1. The incoming call was made via an ISDN or a VoIP network provider supporting call deflection. 2. The rerouted phone number is neither called as a group member, nor is it required to consider parallel calling or fallback on busy.

Proceed as follows for configuration:

- Switching authorisations for the involved internal subscribers on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)

Note: The forwarding subscriber needs an authorisation e.g. for external destinations. For configuring using a character string on an internal phone the subscriber needs an authorisation for controlling of the PBX.

- Configuring sufficient exchange line authorisations for the involved internal subscribers.

- Using the Configuration Manager (page **Subscriber (scr.) > Properties > Exchange line settings**)

Note: The destination subscriber needs a sufficient exchange line authorisation for taking external calls.

- Switching call forwarding on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Reachability**)
  - Using a character string on the internal phone (see Call Forwarding for Subscribers)
  - Using the menu/function key on a system phone (optional, see the phone manual)
  - Using the menu/T-Net function key on an analogue T-Net phone (optional, see the phone manual)
- Making additional settings for call forwarding.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Reachability**)
- Configuring the call forwarding configuration dependent.
  - Using the Configuration Manager (page **Subscriber (scr.) >Profiles > ... > Reachability**)

Call Forwarding for Groups

When call forwarding for groups is used, internal and external calls to a group can be rerouted to other internal phones or external connections. In this way, a person that is not a member of this group and therefore cannot log in, can receive these calls.

Call forwarding for groups makes sure that someone is always available on the group number, even if the calls cannot be taken by a member of the group.

Note: LCR will be used for call forwarding for groups to an external destination (e.g., mobile phone) if at least one member of the group has LCR enabled. It does not matter if the corresponding subscriber is logged in or not.

Note: When forwarding a group the option **As subscriber** in the exchange line settings of the group is not considered. For call forwarding instead exchange line authorisation is not restricted and no call restrictors and call deblockers are used.

Note: Preferred exchange line/Number Presentation: The number of the group is always presented.

Note: To avoid having to use a second call channel in the case of call forwarding to an external destination, the PBX tries to reroute the call directly at the network provider using call deflection. The following requirements must be fulfilled for this purpose: 1. The incoming call was made via an ISDN or a VoIP network provider supporting call deflection. 2.

The rerouted phone number is neither called as a group member, nor is it required to consider parallel calling or fallback on busy.

Proceed as follows for configuration:

- Switching authorisations for the involved internal subscribers on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)

Note: The forwarding subscriber needs an authorisation e.g. for group call forwardings. For configuring using a character string on an internal phone the subscriber needs an authorisation for controlling of the PBX.

- Configuring sufficient exchange line authorisations for the involved internal subscribers.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Exchange line settings**)

Note: The destination subscriber needs a sufficient exchange line authorisation for taking external calls.

- Switching call forwarding on.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)
  - Using a character string on the internal phone (see Call Forwarding for Groups)
- Making additional settings for call forwarding.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)

- Configuring the call forwarding configuration dependent.
  - Using the Configuration Manager (page **Groups >Profiles > ... > Reachability**)

Call Forwarding for External Numbers

When call forwarding for external numbers is used, calls made to an external number (dialled by an external caller) are rerouted to other external connections. In this way, these calls can be taken on another phone, for example, on a mobile phone if no one can take the call internally.

Call forwarding for external numbers can be configured for the current configuration or for the permanent configuration.

Note: If you want to also forward internal calls, use call forwarding for subscribers or for groups instead of call forwarding for external numbers. This also allows forwarding to internal phones.

Note: The function can be configured with a voip exchange line via keypad codes (depending on the provider) via the keypad of the phone. To do this, you must first switch to internal if you have direct exchange access.

Proceed as follows for configuration:

- Switching authorisation for configuring features configured by the network provider for the involved internal phones on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)
- Switching call forwarding on.
  - Using the Configuration Manager (page **Public switched tel. networks > Call forwarding**)

  - Using a character string on the internal phone (see Call Forwarding for External Numbers)
  - Using the menu/function key on a system phone (optional, see the phone manual)
- Configuring the call forwarding configuration dependent.
  - Using the Configuration Manager (page **Public switched tel. networks > Call forwarding**)
- Making additional settings for call forwarding.
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > ISDN > PTMP connection**
  - **Public switched tel. networks > ISDN > PBX base numbers/ trunk bundles > Configure**
  - **Public switched tel. networks > VoIP > Accounts > Configure**

Call Distribution

In call distribution, you specify which internal subscribers should be given the signal for an incoming external call. This means that in call distribution, you specify, e.g., permanently, which internal phones should ring when you are called. Call distribution for each phone number is configured individually on a connection with multiple external numbers (e.g., ISDN connection).

You can configure both one phone for multiple phone numbers and multiple phones (group) for one phone number to ring.

Requirements:

- Configured external numbers:
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > ISDN**
  - **Public switched tel. networks > VoIP**
- Configured internal numbers:
  - Using the Configuration Manager (on the following pages)
  - **Subscriber (scr.) > Phone numbers**
  - **Groups > Phone numbers**

Proceed as follows for configuration:

- Creating the call distribution.
  - Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)
- Setting the fallback.

  - Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)

- Setting the modem/fax call distribution for analogue connections.
  - Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)

Switch Input

If a device that is suitable as switch input (e. g. a door terminal with door bell buttons) is connected to the PBX, it can then be used to trigger a door call or an alarm.

Proceed as follows for configuration:

- Creating a switch input for an alarm.
  - Using the Configuration Manager (page **Devices > Alarms**)
- Creating a switch input for a door call.
  - Using the Configuration Manager (page **Devices > Door terminals**)
- Configuring switch inputs.
  - Using the Configuration Manager (on the following pages)
  - **Devices > Alarms > Switch input >**)
  - **Devices > Door terminals > Door terminal calls >**)

Service Data

Setting Options
Recording the D-Channel Protocol

Setting Options

In case of a malfunction of the PBX, a log can be recorded. This can then be forwarded to the specialist departments (Serice, Development) for troubleshooting.

Proceed as follows for configuration:

- Make settings for logging.
  - Using the Configuration Manager (page **Administration > Logging**)
- Log network data stream.
  - Using the Configuration Manager (page **Monitoring > Network data stream**)

Recording the D-Channel Protocol

A D-channel protocol can be started to record and display control and management information about the connections to the PBX e.g. to display errors. You will thus be able to see the error type, and maybe even its cause, from the list of transferred information.

Proceed as follows:

- Using the D-channel decoder PC programme

Requirements:

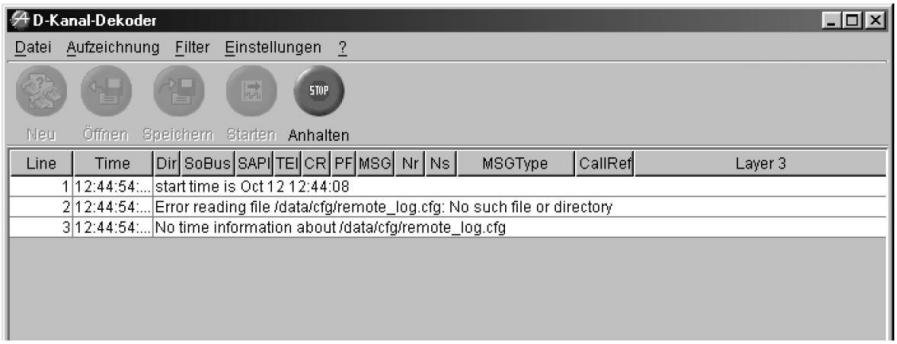
- D-channel decoder PC programme installed (see Auerswald: Auerswald Service Portal, FONtevo: FONtevo)
- **Logging enabled** function switched on in the PBX
- **D-channel** option selected in the PBX

1. Start the D-channel decoder computer programme.
2. Click Start.
3. In the Interface list field, select the IP option.
4. Enter the IPv4 address or, as of decoder version 2.6.7, the IPv6 address of the PBX in the IP Address entry field.

Note: If a port is present, omit it when you enter the IP address.

5. Click OK.

   The protocol log is displayed.



6. Click Pause to stop the protocol being recording.

Further steps:

- The technical departments (Service and Development) can use this file to perform troubleshooting. If the cause of the error cannot be found in the protocol, your specialised dealer or the Service Department (see Auerswald Service Portal) will set additional categories and record an extended protocol.

Protection from Unauthorised Access

The following authorisation levels are available:

- Administrator (access with the admin username and admin password)
- Sub-administrator (access with the username **sub-admin** and assigned sub-admin password)
- User (access with the internal subscriber number and with the assigned user password)

In order to prevent important settings from being changed accidentally or by unauthorised people, the PBX can be protected from unauthorised access.

Administrator

The administrator has unrestricted access to the Configuration Manager. The PBX can be configured completely and the PINS of the other users (internal subscribers) can be changed with this access.

Sub-administrator

A maximum of four sub-administrators can be configured. The access authorisations to the Web interface are assigned to the sub-administrators by the administrator according to the local requirements. With the exception of some pages (e.g. **Administration > Access Authorisations**) the whole Configuration Manager or alternatively only some individual pages can be released to the sub-administrators.

Each sub-administrator has the same access and modification authorisations.

Note: If the sub-administrator enters his internal number as user name, this registration is recognised as an user login and the Web interface is presented in user-mode.

Users (internal subscribers)

The user is any internal subscriber of the PBX. For each user, a user PIN is generated. The access authorisations to the Web interface are assigned to the user by the administrator according to local requirements. The possible range of page releases is limited to a very small number of own subscriber and group settings.

Each user has the same access authorisations. There modification rights may differ depending on the profiles.

Additional app passwords for subscribers

The subscriber password is used for SIP registration and the access to the Configuration Manager and API. Therefore, PBX access data must frequently be saved in external systems. This will lead to security risks, if these systems are insufficiently secured.

Therefore, additional subscriber passwords for accessing the Configuration Manager or API or for accessing Action URLs can be generated.

If an app password has been generated for the subscriber and the corresponding access authorisation has been enabled, the previous subscriber password loses the access authorisation for the correspondig application.

All generated passwords are listed in the table under **Overviews > Additional subscriber passwords**. The passwords can be generated system-wide or for subscribers, although system-wide passwords can only

be generated for Action URLs. Subscriber passwords can be generated for the Configuration Manager, API and for Action URLs.

Proceed as follows for configuration:

- Configuring protection from unauthorized access.
  - Using the Configuration Manager (page **Administration > Access authorisations**)
- Configuring administrator access.
  - Using the Configuration Manager (page **Administration > Dealer and operator data**)
- Configuring sub-administrator access.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)
- PINs and passwords for sub-administrators and users.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure**)
- Generating additional app passwords (Configuration Manager/API) for a subscriber.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure**)
- Generating passwords for accessing Action URLs.
  - Using the Configuration Manager (page **Devices > Action URLs**)
- Configuring authorization for controlling of the pbx via phone.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)

System messages

System messages are used to inform administrators and users about system events such as operating errors. Every event that happens in the system is logged. E-mails with information about the events can also be sent out.

The following different types of system event are possible:

- system information, e.g. automatic memory clean-up and memory check (Voice Mail/Fax function), switch between configurations, switch relays on/off, send system e-mails, register in Configuration Manager
- simple malfunctions, e.g. automatic memory clean-up and memory check (Voice Mail/Fax function) errors
- serious malfunctions, e.g. fan fault (only COMpact 5000R/5200R/5500R), alarm temperature exceeded
- security warnings, e.g. failed logon attempts, entries in the IP blocklist, Admin password change

Proceed as follows to set up system messages:

- Configure DNS server.
  - Using the Configuration Manager (page **Administration > IP configuration**
- Configure outgoing mail server.
  - Using the Configuration Manager (page **Administration > E-mail > Outgoing Mail Server**)
- If necessary, set up the language file.
  - Using the Configuration Manager (page **Administration > E-mail > Internationalisation**)

- If necessary, set up the logo.
  - Using the Configuration Manager (page **Administration > E-mail > Logo management**)
- Set up the sending of system e-mails and event logging.
  - Using the Configuration Manager (on the following pages)
  - **Administration > System messages**
  - **Devices > Voice mail and fax boxes**
  - **Devices > Voice mial/fax boxes > Properties > Box settings**

Subscribers

All connected end devices with telephony functions (phones, answering machines, fax machines etc.) are set in the pbx as subscribers. To the subscriber, a phone number and a series of properties are assigned. Depending on the existing end devices there are analogue, ISDN and VoIP subscribers.

Proceed as follows for configuration:

- Creating a subscriber with an internal number.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers**)
- Assigning properties to the subscribers.
  - Using the Configuration Manager (on the following pages)
  - **Subscriber (scr.) > Profiles > ...**
  - **Subscriber (scr.) > Function overview**
  - **Subscriber (scr.) > Properties**
- Configuring provisioning for IP phones.
  - Using the Configuration Manager (on the following pages)
  - **Subscriber (scr.) > IP provisioning > Settings**
  - **Subscriber (scr.) > IP provisioning > File management**
- Making common settings for all subscribers.
  - Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)
- Assigning unassigned phones to a subscriber or deleting them.

- Using the Configuration Manager (page **Subscriber (scr.) > Unassigned phones**)

Announcement before Answering

This function allows a greeting message to be played to the caller before accepting the call and/or if all group members are busy.

The call is first accepted by the PBX (the external caller is charged from now on). The caller hears the announcement before answering (e.g., with information about the company). Afterwards, the caller hears Music on Hold or the call tone again (if the line is busy, he is put into the waiting loop after the announcement before answering). As soon as one of the called phones accepts the call (or if busy, finishes the call before that), he is connected to the caller.

After the announcement before answering, the caller is put on hold by the PBX for max. 8 minutes. If the call is not taken within these 8 minutes, it is disconnected. The on hold time is reduced if internal subscribers are ringing but not taking the call. Then the caller is rejected after less than 8 minutes.

Note: If the announcement before answering is enabled, it may be necessary to use the Busy-on-Busy function to limit the number of callers that are automatically put on hold in the waiting loop. This prevents more callers being on hold than can be accepted within a reasonable time. If, for example, two phones are logged into the corresponding group and both are in a call, a maximum of four callers will be put into the waiting loop if the limit is six. Callers not put on hold in the waiting loop due to the restriction will directly hear the busy signal instead of the announcement before answering.

Proceed as follows for configuration:

- Managing and assigning audio files for announcements.
  - Using the Configuration Manager (on the following pages)

  - **Functions > Music on Hold and Announcements > File management**
  - **Functions > Music on Hold/announcements > File assignment > Ann. before answering**
  - Using a character string on the internal phone (see Announcement before Answering)
- Configuring announcement before answering.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)
- Switching announcement before answering on.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)
  - Using the menu/function key on a system phone (optional, see the phone manual)
- Switching announcement before answering for fallback on.
  - Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)
- If necessary, switching Busy-on-Busy on.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)

Text Messages (Instant Messaging)

Note: You can use the transmission of the display name to send text messages during an internal call (see Transmission of the Display Name).

The PBX supports the sending and receiving of text messages via SIP (according to RFC 3428).

You can use this function on the following VoIP end devices:

VoIP system phones and standard VoIP phones from Auerswald/ FONtevo

The sending and receiving of text messages takes place via SIP via the instant messaging app (optional, see manual of the phone).

IP DECT phones from Auerswald

The handsets COMfortel M-100, M-200, M-210, M-300 and M-310 only support the receiving of text messages via.

Standard VoIP end decives of other manufacturers

Prerequisire for the support of the text message function via SIP is the activated **SIP convenience package** / **SIP convenience package brand plus** (optional, see activation).

The sending and receiving of text messages vis SIP depends on the end device (see manual of the end device).

Sound

Signaling scenarios

The PBX supports different tones on the receiver in order to signal the various operating states on the PBX:

- External dial tone (long dial tone):
  - After dialing the exchange line access number, when a free external line is available – dialing a telephone number is possible
- Internal dial tone (private branch exchange dial tone):
  - After picking up the receiver or after pressing the Flash or Query key, when a free line is available – dialing a telephone number is possible

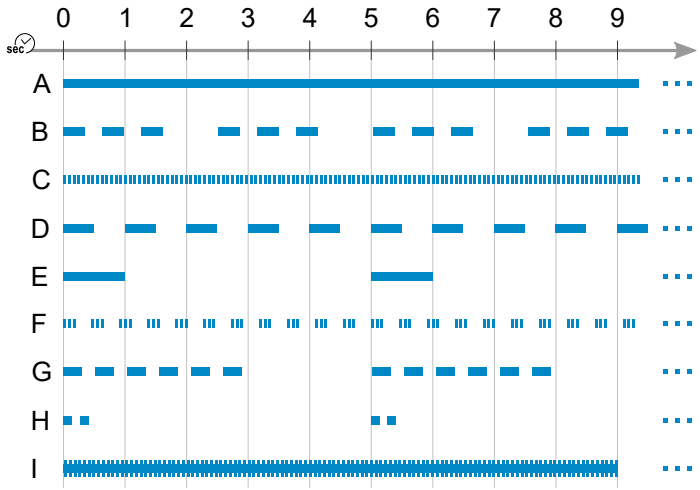
Note: On standard VoIP phones, a dial tone is generated by the phone itself, even when no free line is available.

- Confirmation tone:
  - At the completion of programming, when the programming has been successfully executed
- Busy tone:
  - After picking up the receiver, when no free line is available or an authorization is missing
  - After dialling a phone number, when the desired connection is busy or the network is overloaded or an authorisation is missing
  - At end of call

  - During programming, when the programming was unsuccessful or an authorization is missing
  - During a Call Through call, when the telephone number transferred from the mobile phone is not permitted to use Call Through
- Call tone:
  - during the call phase
- Special tone 1 and special tone 2:
  - After picking up the receiver (when a special tone has been configured as notification that call forwarding, do-no-disturb or baby call has been switched on)
  - After picking up the receiver (when a special dial tone has been enabled), when the call data memory is full
- Call waiting tone:
  - During a call (when call waiting has been enabled), when another call comes in – it is possible to accept or reject the additional call
- Changing tone:
  - During a Call Through call, when the telephone number transferred from the mobile phone is permitted to use Call Through - dialing a telephone number is possible
  - During a remote programming call, when the connection for remote programming is available – programming is possible

Display of signaling types

The tones are signaled as follows:



A. Permanent dial tone (425 Hz)
B. Private branch exchange dial tone (425 Hz)
C. Confirmation tone (425 Hz)
D. Busy tone (425 Hz)
E. Call tone (425 Hz)
F. Special tone 1 (425 Hz)
G. Special tone 2 (425 Hz)
H. Call waiting tone (425 Hz)
I. Changing tone (697/1633 Hz)

Configuration procedure:

- Making settings for tones.
  - Using the Configuration Manager (**Subscriber (scr.) > Properties > Settings** page)

Door Functions

The PBX supports the connection and control of various types of door terminals. In addition to a door bell, ringing at the door can in this way also be indicated by internal or external phones or groups. Each door bell button can be assigned an individual target number. You can also carry out various door functions such as speech connection with the door and actuation of the door opener from internal phones.

Proceed as follows for configuration:

- Creating a door terminal.
  - Using the Configuration Manager (page **Devices > Device wizard**)
- Configuring door terminals.
  - Using the Configuration Manager (page **Devices > Door terminals > Configure**)
- Creating a door call and assigning it to the door terminal.
  - Using the Configuration Manager (page **Devices > Door terminals**)
- Configuring door calls.
  - Using the Configuration Manager (page **Devices > Door terminals >**)
- Creating a relais and assigning it to the door terminal.
  - Using the Configuration Manager (page **Devices > Door terminals**)
- Switching the authorisation for opening the door for the involved internal phones on.

- Using the Configuration Manager (page **Subscriber (scr) > Properties > Authorisations**)

Transmission of the Display Name

Transmission of the display name forces the internally called phone to display the name entered in the phone itself instead of the name entered in the PBX. It makes sense to use this function in combination with special calling systems such as the GIRA calling system 834 plus, for example. It permits the transmission of text messages (entered as display name) during an internal call.

Note: In combination with the alarm device function (see Alarm Device), text messages can be sent and signallised accordingly.

Note: For text messages without a simultaneous call, you can send text messages via SIP (RFC 3428) (see Text Messages (Instant Messaging)).

Proceed as follows for configuration:

- Switching the transmission of the display name on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Settings**)

Sub-System Operation

Setting Options

Transferring a Caller of the Main PBX to Another Subscriber on the Main PBX from the Sub-System PBX

Setting Options

The PBX supports sub-system operation on a higher-level PBX, for example to connect an external office to headquarters.

Requirements:

- At least one free internal VoIP channel for the main PBX
- At least one free external VoIP channel for the subordinate PBX

Proceed as follows to configure the main PBX:

- Creating VoIP subscribers in the main PBX.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > + New**)

Note: The corresponding subscribers in main and subordinate PBX must have the same internal phone number.

  - **Module/Port**: **VoIP**
  - **Device type**: **Standard VoIP phone**
- Switching internal phone on for each subscriber.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure > Settings**)
  - **Exchange line request**: **Internal phone**

Proceed as follows to configure the subordinate PBX:

- Creating subscribers in the subordinate PBX.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > + New**)

Note: The corresponding subscribers in main and subordinate PBX must have the same internal phone number.

- Switching direct exchange line phone on for each subscriber.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure > Settings**)
  - **Exchange line request** : **Direct exchange line phone**

Note: If the PBs are to be linked via a STUN server, set **STUN server for the connection of external VoIP subscribers** (page **Administration > VoIP configuration**.

- Make the following settings for the provider „Auerswald“.
  - Using the Configuration Manager (page **Public switched tel. networks > VoIP > Provider > Configure**)
  - **Domain** : IP address of main PBX
  - **Registrar** : IP address of main PBX
  - **Sub-system operation**
  - **Format of called phone number**: **Send as dialled**
  - **Format of own phone number**: **Without country code (e. g. 05306...)**
  - **Number presentation type**: **In the display text**
  - **Method of number presentation suppression**: **Anonymous**
- Configure a VoIP account for each subscriber.
  - Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > + New**)

- **Provider name**: **Auerswald**
- **Account name** : Freely selectable name for identification of the account
- **Connection type**: **PTMP connection**
- **Use**

- Entering access data for each VoIP account subscriber.
  - Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > Configure**)
  - **Exchange line access number (account number)**: 2- to 4-digit number
  - **User name**: Internal phone number of the VoIP subscriber of the main PBX
  - **Password**: PIN of the VoIP subscriber of the main PBX

Note: We recommend to use a password for remote extensions.

  - **Account has emergency call ability** disabled:
- Assigning each subscriber to the corresponding account.
  - Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > Configure > + New**)
  - **Multiple subscriber number (MSN)**: Subscriber‘s phone number
- Switching on automatic assignment of the preferred exchange lines.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure > Exchange line settings > Preferred exchange line business**: **Automatica assignment**)
- Creating call distribution.

- Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)
- **Call distribution for**: **VoIP PTMP connections** and account name
- **Internal no. | name**: Assign internal phone numbers one-to-one to MSNs

Transferring a Caller of the Main PBX to Another Subscriber on the Main PBX from the Sub-System PBX

Requirements:

- Two free external VoIP channels for the subordinate PBX
- Authorisation **Transfer of external calls to external** assigned to the subscriber of the subordinate PBX (page **Subscriber (scr.) > Properties > Authorisations**)

Note: Set the maximum duration of the transferred call (page **Subscriber (scr.) > All subscribers**) to limit the time the two external VoIP channels are occupied.

1. Initiate a query call to the other subscriber on the main PBX.
2. Hang up the receiver while the other phone is still ringing (blind transfer) or during the query call.

   Both communication partners are connected to each other.

Transfer and Callback

The PBX requires specifications for the transfer and the callback option for external callers to be executed as desired.

Proceed as follows for configuration:

- Configuring hook flash timing.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Analogue settings**)
- If necessary, switching the authorisation for external call transfer externally on.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Authorisations**)
- Configuring sufficient exchange line authorisations for the involved internal subscribers.
  - Using the Configuration Manager (page **Subscriber (scr.) > Properties > Exchange line settings**)

Note: The transferring subscriber needs a sufficient exchange line authorisation for making external calls.

- Configuring the maximum duration of transferred calls (external to external).
  - Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)
- Switching the CCBS and CCNR option for external callers on.

- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PBX base numbers/trunk bundles > Configure**)

Voice mail and fax boxes

Setting Options
Management Voice Mail/Fax Function and Memory
Voice Mailbox
Operating and Querying a Voice Mailbox
Voice mailbox for call recordings
Fax Box
Fax Transfer

Setting Options

Voice mailboxes and fax boxes have partially the same and partially different functions. Nevertheless, they are configured in the same way.

When configuring the box, a **Phone number** and a **Name** are assigned and if required an **owner**. Further configuration options are to be found under **Configure**.

Proceed as follows for configuration:

- Creating and managing a box.
  - Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Phone numbers**).
- Switching a box for a subscriber/group on/off.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure > Voice mail and fax boxes**)
- Switching the info call for a voice mailbox on/off.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure > Voice mail and fax boxes**)

Management Voice Mail/Fax Function and Memory

Announcements for the voice mailboxes, language files, incoming faxes and voice messages are stored on the required USB data storage device.

Caution: Do not remove the USB data storage device while the system is running.

- Switch off the PBX before removing the USB data storage device.

Note: A directory structure is created on the USB data storage device. You may not change this structure.

Note: The USB data storage device must have at least 50 MB free memory.

Voice Mailbox

A voice mailbox can be used in two different ways:

- As an **answering machine** the voice mailbox accepts incoming calls and stores recorded messages as audio files (*.wav). Announcement texts can be assigned for each box.
- For **Call recording** the box records incoming and outgoing calls and stores them also as audio files (*.wav). Announcement texts cannot be assigned for this function.

Note: The function **Call recording** is available only in certain countries.

The configuration of the box can be as follows:

Both types of voice mailbox:

- Configuring the maximum recording capacity.
- Enabling the automatic clean-up.
- Configuring additional properties (Message forwarding, E-mail transfer in case of full memory or failure of automatic clean-up).

Only voice mailbox type answering machine:

- Configuring additional properties: Automatic replay of new messages, Remote access, Language of the announcements.
- Enabling Call pick-up for the voice mailbox and the **Groups(s)/Voice mailbox(es)** authorisation for the owners/users under **Pick-up** (page **Subscriber (scr.) > Properties > Authorisations**).
- Selecting further users and enabling or disabling their authorisations (**Administrate announcements and settings**).

- Recording box own announcements or saving existing announcements in the PBX.
- Assign a profile.
- Configuring **Call acceptance/announcement**, provided that no profile was assigned or settings have been made in the profile that can be switched by the user.
- Switching the readiness permanently on.
- Creating the external call distribution (page **Public switched tel. networks > Call distribution**).
- To forward calls for the owner/user (subscriber/group) to the voice mailbox, the voice mailbox has to be switched on for a subscriber/ a group and the time/requirements for an additional call have to be configured.

Proceed as follows for configuration:

Both types of voice mailbox:

- Configuration of voice mailbox.
  - Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Properties**)

Only voice mailbox type answering machine:

- Switching voice mailbox for a subscriber/group on/off.
  - Using the Configuration Manager (page **Subscriber (scr.) > Configure > Voice mail and fax boxes**)
- Switching info call on/off.
  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure > Voice mail and fax boxes**)

Operating and Querying a Voice Mailbox

Information, Notes and Method

Using an Internal or External Phone (Remote Access) to Operate and Query a Voice Mailbox

Information, Notes and Method

All voice messages are stored as audio files (*.wav). Up to 999 messages (usage as answering machine) and up to 50.000 recordings (usage for call recording) can be administrated per box.

Note: On system phones, new voice messages are signalled by the LED on the message key (optional, see phone manual).

Note: On standard phones and COMfortel 1100 system phones, recently received voice messages can also be signalled as info calls.

Note: In addition, new voice messages can be signalled to the user of the voice mailbox by e-mail or SIP MWI (optional).

Note: For a voice mailbox configured for the function **Call recording**, certain functions such as signalling via a telephone or remote access are not available, because no owner can be assigned to this box. The function **Call recording** is available only in certain countries.

Proceed as follows:

- Using the Configuration Manager
- Using a character string on an internal phone
- Using remote access on an external phone
- On a system phone using the menu/function key (optional, see the phone manual)

Using an Internal or External Phone (Remote Access) to Operate and Query a Voice Mailbox

The voice messages of a voice mailbox can be queried by means of an internal or external phone (remote access). Using various announcements, the caller is guided through the query menu. Besides accessing the messages, it is also possible to operate (e. g. switch on and off) the voice mailbox via phone.

Note: The function remote access is not available for a voice mailbox configured for the function **Call recording** because no owner can be assigned to this box. The function **Call recording** is available only in certain countries.

Requirements:

- Knowledge of the PIN for accessing the corresponding voice mailbox remotely
- DTMF dialling support on the external phone

1. Call the voice mailbox.

Note: Remote access is also possible with disabled readiness of a voice mailbox. External calls will be accepted after a waiting time of at least 50 seconds (50 seconds or **Waiting time for CF on no reply** + 10 seconds if longer than 50 seconds).

2. Dial * and then the user PIN or the PIN for remote access and dial #.

Note: If the internal phone used for the query is the owner of the voice mailbox, it is not necessary to enter a user PIN or remote access PIN.

Various announcements lead you through the remote access menu.

0	Help menu
1	Main menu
2	Plays new messages
3	Plays all messages
4	Goes to previous page
5	Repeats currently played message
6	Goes to next page
7	Deletes currently played message
8	Ends playback
***10**	Disables voice mailbox
***11**	Enables voice mailbox
***20**	Recording and changing of announcements

Note: The default announcement and own announcements are played back. Meanwhile, the following functions are available:

#	Selects the currently heard announcement as active
4	Goes to previous announcement
5	Repeats current announcement

6	Goes to next announcement
7	Selects the current or free announcement for changing or recording (**#** save announcement or * cancel)

***30**	Disables message forwarding
***31**	Enables message forwarding
***70**	Deletes messages and voice memos
***99**	Records voice memos
	Hanging up ends remote access

Additional function while a message is being replayed:

9	Plays back the phone number of the caller provided it has been transferred.

Further options after announcement of the phone number:

#	Initiates call back
*	Possible call back is cancelled

Voice mailbox for call recordings

Use the **Call recording** function to record incoming and outgoing calls and save them as audio files (*.wav) on a USB data storage device.

The calls are recorded using a voice mailbox, specially set up for this purpose, together with an assigned recording channel.

Note: The **Call recording** function is only available in specific countries, and requires activation.

Recording

The incoming call starts being recorded when the called subscriber picks up. As a result, only what the caller hears during a successful connection is recorded.

An outgoing call starts being recorded when the exchange line channel is assigned, i.e. the call progress sounds are also recorded.

Internal conversations and transfer processes (e.g. query calls) are not recorded.

This meta data is available for every recording:

- Phone number of external caller
- Phone number of the subscriber who takes the call
- Date of recording
- Time at which recording begins
- Conversation duration in the format hh.mm.ss

The meta data listed here are used as the file name for a recording.

The maximum length of a recording is 99 minutes.

Each voice mailbox can hold a maximum of 50,000 recordings.

Automatic clean-up of call recording files

If at least one voice mailbox on the PBX is configured for the **Call recording** function, the global setting option for automatically cleaning up the voice mailboxes can no longer be used. However, every voice mailbox can be configured individually, and then operates independently of the device's use as an answering machine or for call recordings.

Encryption

The files containing the recordings are stored, unencrypted, on the USB data storage device.

Encryption prevents unauthorised access to the data storage device and the files stored on it.

The encryption process sets a password for the entire partition on the data storage device. Once the data storage device is formatted, the partition is encrypted.

If the password is changed, it will no longer be possible to access existing data.

Administrator and owner of the voice mailbox and recordings

Administrator

Only the PBX's administrator is authorised to listen to and manage the files or the voice mailboxes.

Owner

It is not possible to assign an owner to a voice mailbox for call recordings.

If, at a later date, a voice mailbox for call recordings is used as a box for an answering machine, it can then be assigned to one or more owners. In

this case, after confirmation at the prompt, all existing recordings in this box are deleted.

Channels

A recording channel or an MSN must be assigned to every voice mailbox that is to be used for call recordings.

In the factory settings, 4 VoIP channels for 4 analogue exchange lines are set up for call recordings.

The channels required for this are permanently reserved on the PBX and are not available for use as voice mailboxes that function as an answering machine or for other calls.

Configuration procedure:

- Activate the Call recording function
  - Using the Configuration Manager (**Administration > Activation** page)
- Configuring and preparing a voice mailbox for call recording
  - Using the Configuration Manager (**Devices > Voice mail and fax boxes > Phone numbers** page)
- Assigning a connection to the voice mailbox and selecting the Call recording function
  - Using the Configuration Manager (**Public switched tel. networks > Analogue > Phone numbers** page)
  - Using the Configuration Manager (**Public switched tel. networks > ISDN > PTMP connection** page)
  - Using the Configuration Manager (**Public switched tel. networks > ISDN > Extension numbers PTP connection** page)

  - Using the Configuration Manager (**Public switched tel. networks > VoIP > Accounts > Configure** page)

- Make settings for saving recordings and encrypting them, if required.
  - Using the Configuration Manager (**Devices > Voice mail and fax boxes > Basic Settings** page)
- Make settings for the maximum recording capacity, if required
  - Using the Configuration Manager (**Devices > Voice mail and fax boxes > Properties** page)

Fax Box

A fax box accepts incoming faxes and saves them as *.pdf files. The maximum length of a fax is 50 pages.

The fax receiving function is supported with maximally 9.6 kbps (V.29).

The configuration of the fax box can be as follows:

- Configuring the maximum recording capacity (MB).
- Enabling the automatic clean-up).
- Configuration of further properties (**E-mail transfer** on incoming fax, in the case of full memory, in case of failure of automatic clean-up, or after fax transfer, **Fax ID** for the fax transfer).
- Selecting further users for the fax box and enable or disable their authorisations (**Administrate settings**).
- Assigning a profile.
- Configuration of **Call acceptance** (**Reject anonymous calls**), provided that no profile has been assigned or settings have been made in the profile which can be switched by the user.
- Switching the readiness permanently on.
- Creating the external call distribution (page **Public switched tel. networks > Call distribution**).
- For the fax box to accept incoming calls, the fax box has to be switched on for a subscriber/a group.

Proceed as follows for configuration:

- Switching fax box for a group on/off.

  - Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > Configure > Voice mail and fax boxes**)

- Configuration of a fax box
  - Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Properties**)

Fax Transfer

Information
Installing Printer Driver COMpact/COMmander Fax on PC
Configuring the Printer Driver COMpact/COMmander Fax
Sending a fax to a recipient
Send a fax to multiple recipients (Broadcasting file)

Information

The fax function of the PBX supports fax transfer in combination with a corresponding PC application. The required **COMpact/COMmander Fax** printer driver has to be installed on the PC used.

The **COMpact/COMmander Fax** printer driver transfers the document to be faxed to the PBX that subsequently saves the document for the fax box used and sends it in the background using the corresponding fax identification. If the destination of the fax is not available (busy or no reply), two further dial-up attempts will automatically be made by the PBX.

Installing Printer Driver COMpact/COMmander Fax on PC

You the fax printer driver in the Internet (see Auerswald: Auerswald Service Portal, FONtevo: FONtevo Support).

1. Download the printer driver and start the driver installation.
2. Select the desired language and click OK and Next.
3. Read the license agreement and subsequently click I accept the terms of the license agreement. Click Next.
4. Click Continue installation.
5. Click Finish.

Configuring the Printer Driver COMpact/COMmander Fax

Before transferring the first fax using the **COMpact/COMmander Fax** printer driver, at least one fax box of the PBX has to be configured in the printer driver.

Requirements:

- Installed printer driver **COMpact/COMmander Fax**

1. To open the printer driver, print a test document, e. g. a Word document, using the print function of the PC programme. Via the printer selection, select the printer driver **COMpact/COMmander Fax**.
2. If necessary, enter a name for your identification in the entry field Name of the fax box.

Note: It is appropriate to use the same name that has been assigned for the fax box in the Configuration Manager of the PBX.

3. In the Subscriber / Group number entry field, enter the internal number of the owner of the fax box in question.
4. In the User Password entry field, enter the password of the subscriber or of the group member in question.

Important: By enabling the check box **Display PIN**, the entered password will be displayed uncoded. In order to ensure that a password cannot be viewed by unauthorised persons, a user should always lock the

access to his/her PC or completely log off his/her PC when leaving the room.

5. In the Network address entry field, enter the IP address of the PBX.

Note: By clicking **Check** the login data are checked. A window opens displaying the check result. Click **OK** to close this window.

6. Select or clear the check box Prefix automatically for the exchange line access number of the fax number of the recipient.
7. In the Exchange line access number entry field, enter the exchange line access number of the PBX.
8. If necessary, click New and repeat steps 2 to 7 in order to configure another fax box of the PBX in the printer driver.
9. Click OK.
10. Click Close.

Sending a fax to a recipient

Requirements:

- At least one channel set for outgoing calls of the voice mail/fax function (**Channels used simultaneously for outgoing calls**)
- Created fax box
- Installed and configured **COMpact/COMmander Fax** printer driver

1. Open the document you want to send as a fax.
2. Print the file and therefor select COMpact/COMmander Fax in the printer selection dialogue.
3. Under Fax number of the recipient, enter the corresponding phone number (special characters: +, (,), _, -).

Important: Depending on the configuration of the fax box in the printer driver, the exchange line access number of the PBX automatically precedes the fax number or has to be entered in front of the **Fax number of the recipient**.

Note: The entered phone number is automatically converted into a pure digit string before sending.

Note: If a phone number is stored in the clipboard, it is automatically transferred to the entry field.

4. Under Fax box for transfer select the desired fax box.

5. Click Send.
6. Click Close.

Note: To cancel the fax transfer, delete the fax message under **User data > Messages** (administrator) or **Messages (Voice mail and fax boxes)** (user) in the Configuration Manager.

Send a fax to multiple recipients (Broadcasting file)

Requirements:

- At least one channel set for outgoing calls of the voice mail/fax function (**Channels used simultaneously for outgoing calls**)
- Created fax box
- Installed and configured **COMpact/COMmander Fax** printer driver

1. Create the text file (*.csv, *.txt) with the fax recipient‘s phone number (max. 18 digits per phone number (special characters: +, (,), /, , , -, :))

Important: Depending on the configuration of the fax box in the printer driver, the exchange line access number of the PBX automatically precedes the fax number or has to be entered in front of the fax number of the recipient.

2. Open the document you want to send as a fax.
3. Print the file and therefor select COMpact/COMmander Fax in the printer selection dialogue.

Note: Entries under **Fax number of the recipient** are not considered in case of a broadcasting file.

4. Under Fax box for transfer, select the desired fax box.
5. Click Broadcasting file.
6. Select the text file (*.csv, *.txt) created in step 1 and click Open.

Note: If the error message is 507, fax sending is interrupted for 3 minutes and then restarted. After 10 unsuccessful attempts, the transmission is stopped.

7. Click Close.

Note: To cancel the fax transfer, delete the fax message under **User data > Messages** (administrator) or **Messages (Voice mail and fax boxes)** (user) in the Configuration Manager.

VoIP

Note
Configuring External Internet Telephony
Configuring Internal IP Telephony
Configuring Remote Extensions

Note

The transmission quality and the reliability of Voice over IP depends to a great extent on the quality of the Internet connection used.

Configuring External Internet Telephony

Requirements:

- External VoIP channel (optoptional, see Technical Data in the Instructions)
- An account with a VoIP provider
- Configured internal subscribers

Configure the PBX as follows:

1. Configure at least one VoIP channel as an external VoIP channel.
2. Make sure you have current VoIP provider data:
   - Check previously configured provider data.
   - Download and import provider data from the Internet.
   - Enter provider data manually.
3. Configure the DNS server.
4. Create at least one VoIP account in the PBX and enter the account and access data.
5. Enter your own VoIP phone numbers.
6. Set up call distribution (for incoming calls).
7. If necessary, enable DiffServ to preferably treat the voice data over other data by menas of the router and to increase the VoIP call quality.
8. Check if the registration of the VoIP account was successful (Monitoring > Status VoIP accounts).

Note: If no calls are possible via VoIP, although account registration and perhaps even an initial call were successful, this might have something to do with a very short timeout configured in the firewall. Reduce the **Interval for NAT Keep Alive** for all of the providers used (**Public switched tel. networks > VoIP > Provider > Configure**).

Configuring Internal IP Telephony

Requirements:

- Internal VoIP channel (optional, see Technical Data in the Instructions)

Proceed as follows:

1. Configure a VoIP channel as an internal VoIP channel for each planned internal VoIP subscriber.
2. If necessary, configure the SIP port of the internal VoIP registrar.
3. Create internal numbers for the VoIP subscribers.
4. If required, change the automatically generated PINs and passwords.

Note: The user passwords are needed to authenticate the VoIP phones when they are registered with the PBX.

5. Configure the VoIP end devices. To do this, you need the following information:

   IP address or URL of the PBX

   SIP port of the internal VoIP registrar

   Internal number of the VoIP subscriber

   User password of the VoIP subscriber

Note: You will find notes about how to configure different VoIP end devices on the Internet (see Auerswald Service Portal).

6. Check whether the VoIP subscriber was successfully registered on the PBX (page Monitoring > Status int. VoIP subscribers).

7. If necessary, enable DiffServ, to increase VoIP call quality (Quality of Service (QoS)).
8. If necessary, make the VoIP settings (Jitter buffer, Echo cancellation) for the VoIP subscribers to improve the call quality.

Configuring Remote Extensions

In order to operate a VoIP phone as a remote extension, it must either be connected to the network via a VPN tunnel or the phone and the PBX must establish the connection via a publicly accessible STUN server on the Internet and a DynDNS service as described in the following.

Caution: Each port forwarding ares a security risk.

- You should use as few forwarding configurations as possible.

Important: As a protection measure against the access by third parties, regularly check the call data recording of your PBX and the LOGs of your NAT router for inconsistencies.

Important: For security reasons, the connection of remote extensions by means of a VPN tunnel should be preferred.

Important: Note that when using a remote extension, emergency calls can only be traced backed to the location of the PBX (localisation). Localisation is necessary if the caller is no longer able to give his name and address. Therefore, for emergency calls from a remote extension, a mobile phone or a phone connected to the local VoIP or public switched network is required.

Note: In order to avoid that emergency calls are made from a remote extension, the prevention of emergency calls can be enabled under Subscriber (scr.) > Properties > Exchange line settings. When the subscriber tries to dial an emergency number, he will hear the

announcement “This phone does not allow emergency calls. Please use an alternative.“

Requirements:

- Portforwarding configured on router

  Configure the PBX as follows if at least one remote extension is not coupled over a VPN tunnel:

Caution: The generation of user passwords and step 4 to 6 are essential protection measures against the access by third parties

1. Configure a VoIP channel as an internal VoIP channel for each remote extension planned.
2. Create the internal numbers for the external private branch exchanges (a maximum of two VoIP subscribers is recommended).
3. If required, change the automatically generated PINs and passwords.

Note: The user passwords are needed to authenticate the VoIP phones when they are registered with the PBX.

4. In dependence on the phone tariffs, restrict the exchange line authorisation for remote extensions (e.g. National). At times of the day when phone calls are usually not made, e.g. in the night or outside of business hours, it is possible to restrict the exchange line authorisation to a minimum (under Subscriber (scr.) > Properties > Exchange line settings).
5. Configure a call restrictor (e.g. for added value services or mobile radiocommunications networks – 0900, 0180, 01…) and assign it

to the remote extensions (under Subscriber (scr.) > Properties > Exchange line settings).

6. Enable data encryption (see SIPS and SRTP).
7. Configure the STUN server for connecting external VoIP subscribers.
8. Configure the size of the jitter buffer for the remote extensions.
9. Configure the remote extensions. To do this, you need the following information:

   The external IP address or URL of the PBX (as registrar and domain)

   The internal number of the VoIP subscribers (as MSN or user name)

   The user password of the VoIP

   STUN server (the same as in Step 6)

   Size of the jitter buffer (the same as in Step 7)

Note: If the Internet connection on the PBX does not have a permanent IP address, an account with a provider for dynamic DNS is also required (e.g., dyndns.org). Enter the associated URL in the VoIP phone as registrar and domain (e.g., pbx.dyndns.org).

10. Check whether the VoIP subscriber was successfully registered on the PBX (page Monitoring > Status internal subscriber).
11. If necessary, enable DiffServ in order to increase VoIP call quality (Quality of Service (QoS)).

VoIP/GSM Routing

VoIP/GSM routing precedes Least Cost Routing (LCR) and can be used to specify an order for exchange line assignment to certain prefixes or complete phone numbers. The LCR provider will then be dialled in advance at the selected exchange lines (except for GSM gateways). A table with exception numbers must be set up for this and saved in the PBX.

VoIP/GSM routing proceeds as follows:

- A user dials an external number.
- The PBX checks whether the phone number or the phone number range is specified in the exception numbers and switched on for the current configuration.
- If the phone number is found, the PBX automatically forwards the dial attempt to the first selected connection.
- If the selected connection is busy when dialled, the next connection selected is attempted (up to three are possible).
- After a dial attempt with the third connection, if no connection is established, you hear the busy signal.

Note: If a call is initiated over a targeted exchange line access point or VoIP access point, VoIP/GSM routing is not performed.

Note: The VoIP/GSM routing is performed depending on the destination number, if the exchange line settings (subscriber, group) do not prevent this (e.g. exchange line authorisation only for selected connections, preferred exchange line, call restrictor).

Note: The PBX uses fixed network connections to handle emergency calls as a priority (i.e. calls made to the phone numbers specified in **Public switched tel. networks > Emergency calls** as well as the automatic alarm calls made by the PBX.

Note: The special numbers provided by the VoIP provider, for checking one's account status or mailbox, for example, usually do not include a local code. It is not possible to route numbers via the exception table. In order to correctly dial these phone numbers, use targeted exchange line assignment with the account number (targeted VoIP access point). Alternatively, you can enter such numbers as service numbers under **Public swichted tel. networks > Emergency calls > Service numbers**.

Note: If no particular connection has been selected for door-to-external calls the routing is performed depending on the destination number.

Note: For Call Through calls, the routing performed is dependent on the destination number, if the special exchange line settings do not prevent this (e.g. exchange line authorisation only to selected connections, preferred exchange line, call restrictor).

Note: If routing has been enabled for call forwarding for external numbers, the routing performed is dependent on the destination number.

Note: For call forwarding for subscribers and groups, the routing performed is dependent on the destination number.

Note: When you dial one of the local numbers (phone number without a prefix) listed in the exception number table over a VoIP access point, the PBX automatically converts the phone number as specified under **Number presentation (outgoing)** for the VoIP provider.

Note: When you dial one of the local numbers (phone number without a prefix) listed in the exception number table over a GSM connection, the phone number is automatically prefixed with the area code specified under **Administration > Exchange line request**.

Proceed as follows for configuration:

- Creating exception numbers.
  - Using the Configuration Manager (page **Routing > VoIP and GSM routing > Exception numbers**)
- Configuring the VoIP/GSM routing configuration dependent.
  - Using the Configuration Manager (page **Routing > VoIP and GSM Routing > Routing configuration dependent**)

Preferred Exchange Line

If a phone should use primarily specific external connections, several preferred exchange lines for business and/or private calls may be configured for the corresponding group or subscriber.

Proceed as follows for configuration:

- Configuring the preferred exchange line.
  - Using the Configuration Manager (on the following pages)
  - For subscriber **Subscriber (scr.) > Properties > Exchange line settings**
  - For groups **Groups > Properties > Exchange line settings**
  - For the exchange line transfer function **Functions > Exchange line transfer**
  - For the Call Through function **Functions > Call Through**
- Configuring the preferred exchange line configuration dependent.
  - Using the Configuration Manager (on the following pages)
  - For subscribers: **Subscriber (scr.) > Profiles > ... > Exchange line settings**
  - For groups: **Groups > Profiles > ... > Exchange line settings**

Waiting Field Function

Callers can be parked in the waiting field either automatically or by pressing a key and then be retrieved again or transferred. The waiting field is operated using one or more waiting field receptions.

Call acceptance in the waiting field is carried out as follows:

- If all of the waiting field receptions have been switched off, the phones ring according to normal call distribution.
- After switching on one or more waiting field receptions, the calls in the respective waiting field are signalled on the waiting field keys of the enabled waiting field receptions.

Important: After switching on the waiting field reception, any call forwarding for subscribers configured for the phone, parallel calls as well as the do-not-disturb function are switched off. A rework time configured for the group does not work. Not until the waiting field reception is switched off (on the phone) will these functions (for example, call forwarding on an answering machine) become available again. The function in question (except parallel calling, rework time) must be switched on again.

Important: For existing calls, incoming exchange line calls are signalled by blinking LEDs and also by an attention tone, if enabled. For multiple active waiting field receptions, a caller placed in the waiting field is also signalled in the other waiting field receptions by the attention tone.

- The caller can be accepted normally by picking up the receiver or placed in the waiting field by pressing the associated waiting field key. If the automatic call acceptance is set to on busy or according to a time, callers are also automatically placed in the waiting field.

- A caller who has been placed in the waiting field, hears hold music. If a group is configured as waiting field reception and announcement before answering is enabled, the caller hears the announcement before answering first.
- A caller is kept a max. of 16 minutes without interruption in the waiting field. If the waiting time is not interrupted by a call to the waiting caller or the call is forwarded, the connection is disconnected.
- You are reminded that the subscriber is still waiting by a short tone once per minute.
- Press the corresponding waiting field key to initiate a call with a caller in the waiting field.
- An external caller in the waiting field can be rerouted internally as normal. If the line of the person being called is busy, the external subscriber can be placed in that person's queue. If the person being called does not accept the call, the external caller is put back in the waiting field. This is signalled by an attention tone, if enabled.
- If there is no one in the waiting field, you can make calls as normal and use the keys on the phone without limitations.
- As long as information about a call in the queue is showing on the display (LED lights up in yellow), several of the keys are without function and as soon as you picked up the receiver, you are connected to the call in the queue. You can prevent this by pressing the **Exit** key. After this, the LED lights up in red and the display is in the idle state. Now you can make a call as normal, for instance, by selecting a phone number from the phone book.

Note: If the waiting field function has been enabled, the LAN TAPI function will not be available for the enabled extensions.

Requirements:

- System phone, that supports function keys for reception phones (optional, see the phone manual)

Proceed as follows for configuration:

- Creating a group for the waiting field.
  - Using the Configuration Manager (on the following pages)
  - **Groups > Phone numbers**
  - **Groups > Properties > Basic settings**

Note: To avoid user errors, all subscribers should be **Permanently logged in**.

- Managing and assigning audio files for announcements.
  - Using the Configuration Manager (on the following pages)
  - **Functions > Music on Hold and Announcements > File management**
  - **Functions > Music on Hold/announcements > File assignment > Ann. before answering**
  - Using a character string on the internal phone (see Announcement before Answering)
- Configuring announcement before answering for the group.
  - Using the Configuration Manager (page **Groups > Properties > Reachability**)
- Configuring call distribution for the group.
  - Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)

- Selecting the group for the waiting field and making settings for the waiting field.
  - Using the Configuration Manager (page **Functions > Waiting field**)
- On each waiting field reception, assigning one or more waiting field keys and one internal waiting field key (optional, see manual for the phone).

Music on Hold and Announcement

When transferring a call or in the case of a query call, a communication partner waiting in the background hears music on hold from the PBX, perhaps with an announcement played over the music.

Configuration procedure:

- Configuring the source of the music on hold and the use of the announcement during the music on hold.
  - Using the Configuration Manager (**Functions > Music on Hold/ announcements > File assignment > Music on Hold** page)
  - Using a character string on the internal phone (see page)
- Managing and assigning audio files for announcement and music on hold.
  - Using the Configuration Manager (on the pages that follow)
  - **Functions > Music on Hold and Announcements > File management**
  - **Functions > Music on Hold/announcements > File assignment > Music on Hold**
  - Using a character string on the internal phone (see page)
- Switching music on hold during the ringing period on/off.
  - Using the Configuration Manager (on the pages that follow)
  - **Functions > Music on Hold/announcements > File assignment > Music on Hold**
  - **Functions > Automatic Receptions > Configure**

Wake-up Functions

To be reminded of one-time or recurring appointments, every user can configure different wake-up times for his phone.

Proceed as follows for configuration:

- Making settings for wake-up calls.
  - Using the Configuration Manager (page **Functions > Wake-up times**)
- Configuring wake-up times and switching wake-up on/off.
  - Using the Configuration Manager (page **Functions > Wake-up times**)
  - Using a character string on the internal phone (see Wake-up Functions)
- Managing and assigning audio files for announcements.
  - Using the Configuration Manager (on the following pages)
  - **Functions > Music on Hold and Announcements > File management**
  - **Functions > Music on Hold/announcements > File assignment > wake-up announcements**
  - Using a character string on the internal phone (see Wake-up Functions)
- Creating switching times for switching the wake-up announcement.
  - Using the Configuration Manager (page **Time control > Switching times**)

Central Caller List

The phone numbers of unaccepted calls are stored in the caller list on the system phone (optional, see phone manual). As soon as a connection (incoming or outgoing) has been made to a caller, the corresponding phone number is automatically deleted from the caller list.

If a group call occurred and if the central caller list for groups is enabled, the phone number is also deleted from the caller lists of the other members of the group.

Note: Group-wide deletion does not occur if the person calling back has already deleted his caller list or has been called again by the same caller on his subscriber number.

Requirements:

- All group members are system phones, that support the central caller list (optional, see the phone manual)

Proceed as follows for configuration:

- Switching the central caller list for the involved groups on.
  - Using the Configuration Manager (page **Groups > Properties > Settings**)

Phoning

Using Different Phone Types
Incoming Calls
Call Waiting Calls
Outgoing Calls
Unsuccessful Outgoing Calls
Query Calls
Transferring
Conference Calls

Using Different Phone Types

Depending on the used phone type, there are differences in the operation and programming of the PBX. The following phones can be operated at the PBX:

Analogue Phones (PD and DTMF)

If you are already in a call state with another subscriber (internal or external), you have to press the **Flash key (signal key, R key)** first on a DTMF phone before dialling a digit, e.g. for a call transfer.

Due to the difficulties concerning the transferring with PD phones, this function is not available for this kind of phones. If your phone offers both dialling modes, you should therefore set it to DTMF.

Analogue T-Net Phones

On some analogue DTMF phones, specific T-Net functions can be performed in the analogue T-Net via the menu or via specific function keys. If you are operating such a phone as an internal subscriber, you can also use these comfortable operating modes for some functions of the PBX. Please note that the T-Net functions of the phone are limited because the PBX doesn‘t support some functions. If you want to use the T-Net functions of your phone, read the manual for the phone. When entering phone numbers, take care to enter the leading exchange line access number. Instead of an "announcement of the public exchange", you will hear the confirmation tone for successful programming.

ISDN Phones

To be able to operate your ISDN phone, you have to make yourself acquainted with its functions. To do this, read the manual for the phone. It is vital to know which key (e.g. **hold** or **R key** or menu you have to use to initiate a query call. To use some functions, you have to rely completely

on the menus/keys of the phone (see notes). The digits to be dialled for analogue phones are not required here.

System Phones

As described in this manual, you can use the major part of the PBX functions with all connected phones. With the system phones, the operation is more comfortable, since the most important functions can be controlled via a menu/function key (optional). In this case, the operation is described in the manual. Some of the PBX functions described can only be used on the system phones specified in the description.

Standard VoIP Phones

On some standard VoIP phones, functions such as **Call waiting**, **Conference**, **Query calling**, and **Hold** are not carried out via the PBX, but rather on the phone itself. Therefore, the instructions for operating these functions may deviate from the descriptions in this guide. For more information on this, read the manual for the phone.

Normally, pressing the pound key # on a standard VoIP phone initiates a dialling procedure. Since this key is required for numerous functions, as well as for all settings (programming), using the pound key for these actions is not possible on most standard VoIP phones. For more information on this, read the manual for the phone.

Incoming Calls

Accepting Incoming Calls
Taking over a Call (Pick-up)
Taking over a Call (Call Take-over)
Accepting Door Call and Opening the Door
Accepting an Alarm Call and Confirming the Alarm

Accepting Incoming Calls

When you receive a call, proceed as follows:

- Pick up the receiver.

  You are connected with the caller.

Taking over a Call (Pick-up)

Using Pick-up, you can take a call on your own phone that was originally sent to another internal phone.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- Authorisation **Pick-up** assigned to the subscriber
- To pick up an external call: **Exchange line authorisation** assigned to the subscriber for incoming calls (**International**, **National**, **City** or **only incoming, with emergency call**)

If an internal phone rings, proceed as follows:

1. Pick up the receiver of your phone.

Note: During an existing call press the **Flash key** or the **Query key**.

2. Dial ##06.
3. Enter the internal number of the phone that is ringing. Possible entries:

   Internal number of the individual phone (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Taking over a Call (Call Take-over)

If the answering machine has already taken the call, you can perform a call take-over to take the call on your own phone.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- **Call take-over** enabled for the subscriber
- To perform call take-over of an external call: **Exchange line authorisation** assigned to the subscriber for incoming calls (**International**, **National**, **City** or **only incoming, with emergency call**)

If the answering machine has already taken the call, proceed as follows:

1. Pick up the receiver.

Note: During an existing call press the **Flash key** or the **Query key**.

2. Dial ##06.
3. Enter the internal number of the answering machine that is conducting the call to be picked up. Possible entries:

Internal number of the individual answering machine (for exchange line request type Direct exchange line phone: with preceding **)

Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Accepting Door Call and Opening the Door

The PBX supports the connection and control of various types of door terminals. In addition to a door bell, ringing at the door can in this way also be indicated by internal or external phones or groups. Each door bell button can be assigned an individual target number. You can also carry out various door functions such as speech connection with the door and actuation of the door opener from internal phones.

Proceed as follows:

- Using an internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- Configured door terminal
- Configured door call
- Configured relay with operating mode **Door opener**
- Authorisation **Open door** assigned to subscriber or group

1. Pick up the receiver.

Note: During an existing call press the **Flash key** or the **Query key** and dial **2**.

   You are connected with the door terminal.

2. Press the **Flash key** or the **Query key**.

3. Dial #.

Accepting an Alarm Call and Confirming the Alarm

An alarm can be signalled by the ringing of an internal or external phone. To terminate the alarm loop and to switch off assigned relays, the alarm can be confirmed.

Proceed as follows:

- Using an internal or external phone

Requirements:

- DTMF-capable phone or DTMF dialler

If an alarm is signaled at an internal or external phone, proceed as follows:

1. Pick-up the receiver.

   You will hear the announcement.

2. Within the next 60 seconds dial 0.

   You will hear the confirmation tone.

Note: You can confirm the alarm already during the announcement. If you do not confirm the alarm (or dial a wrong digit), you will hear the busy signal and the call will be automatically terminated by the PBX. A further alarm call may be initiated after some time if the other alarm subscribers do not confirm the alarm.

Call Waiting Calls

Rejecting a Call Waiting Call
Accepting the Call Waiting Call
Disconnecting the Current Call and Accepting the Call Waiting Call

Rejecting a Call Waiting Call

If you receive another call while you are already making a call and you do not want to speak with the second caller, or you do not want to interrupt the first call, you can reject the call waiting call. The caller receives a busy signal as long as no other phones are called.

If a call waiting call is signalled, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 0.

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On ISDN and system phones, this function is performed via an existing function key or via the menu (optional, see manual for the phone).

Accepting the Call Waiting Call

If you receive another call while you are already making a call and you want to speak to the second caller, or you want to end the current call, you can accept the call waiting call. After accepting, you are connected with the other caller. Your previous communication partner hears the hold music.

If a call waiting call is signaled, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 2.

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On ISDN and system phones, this function is performed via an existing function key or via the menu (optional, see manual for the phone).

Note: If the waiting call is an alarm call, the currently existing call will be terminated when accepting the waiting alarm call.

Disconnecting the Current Call and Accepting the Call Waiting Call

A call received during a call can be accepted. At the same time, the ongoing call is ended.

If a call waiting call is signalled, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 1.

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: Note. On ISDN and system phones, this function is performed via an existing function key or via the menu (optional, see manual for the phone).

Outgoing Calls

Initiating an Internal Call
Initiating an External Call
Initiating an External Call with Calling Line Identity Restriction
Initiating Private Exchange Line Access
Initiating Private Exchange Line Access without a PIN
Initiating Targeted Exchange Line Access
Force Public Exchange Call for Automatic Exchange Line Request
Initiating Targeted VoIP Access
Initiating Calls with Project Assignment
Calling a Door and Opening the Door
Initiating InterCom Announcement/Hands-Free
Initiating Loudspeaker Announcement

Initiating an Internal Call

Internal calls are free of charge.

1. Pick up the receiver.
2. Enter an internal number. Possible entries:

   Internal number of the individual phone (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

   Once you have finished dialling the phone number, the subscriber will be called.

Note: If the internally called subscriber is busy, you can knock on his phone after a short waiting (if call waiting is allowed on his phone).

Initiating an External Call

1. Pick up the receiver.
2. Enter an external number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   Once you have finished dialling the phone number, the subscriber will be called.

Initiating an External Call with Calling Line Identity Restriction

If you want to prevent your next communication partner from seeing your phone number on his phone display, you can suppress the phone number display to your communication partner.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- The CLIR (calling line identity restriction) service feature has been enabled by the network operator.

1. Pick up the receiver.
2. Dial *31#.
3. Enter an external number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   Once you have finished dialling the phone number, the subscriber will be called.

   The phone number display suppression has been enabled for this call.

Note: Note. You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On some ISDN phones and system phones, this function can also be called via an existing function key or via the menu (optional, see manual for the phone).

Note: If an emergency number is called while calling line identity restriction is enabled, calling line identity restriction is disabled so that the emergency service can identify the caller.

Note: The feature code (keypad) to execute the function, as defined in the PBX, may differ from the requirements of the network provider. In case of malfunctions, ask your network provider for the required feature code (keypad) and change the setting. If you want to use the feature directly on the exchange line instead, the subscriber needs the authorisation for configuring features to be configured by the network provider.

Initiating Private Exchange Line Access

Private (personalised) exchange line access enables separate billing for business and personal calls for individual employees. The subscriber that want to make personal calls can initiate private exchange line access on any internal phone by entering the user PIN. Personal calls are assigned to the subscriber in question in the call data record based on the user PIN.

Requirements:

- User PIN configured for the subscriber

1. Pick up the receiver of any internal phone.
2. Dial ##92.
3. Enter the user PIN.
4. Dial *.
5. Enter the external number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   You are making a personal call.

Initiating Private Exchange Line Access without a PIN

Private (personalised) exchange line access enables separate billing for business and personal calls for individual employees. Authorisation granted specifically to the subscriber allows private exchange line access without requiring a user PIN to be initiated on the subscriber's own phone. The charges are billed to the phone used.

Requirements:

- Authorisation **Private exchange line access without PIN** assigned to the subscriber

1. Pick up the receiver.
2. Dial ##91.
3. Enter the external number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   You are making a personal call.

Initiating Targeted Exchange Line Access

Using targeted exchange line access, you can define which external exchange line and own external number is transmitted for an external call.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- External S_0 port, external analogue port (optional, see Technical Data in the Instructions)

1. Pick up the receiver.
2. Dial ##96.
3. Enter the separate external number to be transmitted. Possible entries:

   External numbers
4. Dial *.
5. Enter an external number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   The subscriber is called.

Force Public Exchange Call for Automatic Exchange Line Request

1. Pick up the receiver.
2. Select ##94.
3. Enter an external number. Possible entries:

   External number

   Once you have finished dialling the phone number, the subscriber will be called.

Initiating Targeted VoIP Access

An external call can be initiated via a specific VoIP account using targeted VoIP access point.

The Internet service provider's network can only be accessed via targeted VoIP access, e.g., when querying the mailbox.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

1. Pick up the receiver.
2. Enter the account number. Possible entries:

   Internal number of the VoIP account (for exchange line request types Direct exchange line phone **and Automatic exchange line request**: with preceding **)

3. Enter an external number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

Note: In any case, the external number must be dialled with the corresponding local area code, even if it is your own local area code. The special numbers provided by the VoIP provider for checking one's account status or dialling usually do not include a local area code.

The subscriber is called.

Note: No callback is possible (CCBS or CCNR) for outgoing calls over VoIP.

Initiating Calls with Project Assignment

The exchange line access with project assignment allows the sorting of external calls to a specific project/customer/client. With call data management, the charges and the time expenditures may be assigned to different projects/customers/clients.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- Assigning of project numbers function enabled (optional, see activation)

1. Pick up the receiver.
2. Dial ##93.
3. Enter the 2- to 6-digit project number.
4. Dial *.
5. Enter an external number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

Calling a Door and Opening the Door

The PBX supports the connection and control of various types of door terminals. In addition to a door bell, ringing at the door can in this way also be indicated by internal or external phones or groups. Each door bell button can be assigned an individual destination number. You can also carry out various door functions such as speech connection with the door and actuation of the door opener from internal phones.

Proceed as follows:

- Using an internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- Configured door terminal
- Configured relay with operating mode **Door opener**
- Authorisation **Open door** assigned to subscriber or group

1. Pick up the receiver.

Note: During a call, press the **Flash key** or the **Query key**.

2. Enter the internal number of the door terminal. Possible entries:

   Internal number of the door terminal (for exchange line request type Direct exchange line phone: with preceding **)

   The call connection is immediately established after dialling the phone number.

3. Press the **Flash key** or the **Query key**.
4. Dial #.

Initiating InterCom Announcement/Hands-Free

The InterCom function enables an announcement to system phones (one phone or group) from any internal phone without someone having to actively receive a call (for example, in a doctor's office).

Furthermore, the caller's system phone can be programmed to switch on the microphone in addition to the loudspeaker (hands-free) so that a person nearby can speak with the caller using this intercom system.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- Enabled InterCom authorisation at the InterCom destination

1. Pick up the receiver.
2. Dial one of the following character strings:

   ##011: Initiates an InterCom announcement.

   ##012: Initiates InterCom hands-free.

3. Enter the internal number of the system phone. Possible entries:

   Internal number of the individual system phone (for exchange line request type Direct exchange line phone: with preceding **)

   The system phone automatically takes the call after a single ring.

Initiating Loudspeaker Announcement

This function enables loudspeaker announcements over an active loudspeaker.

Proceed as follows:

- Using a character string on the internal phone
- Using the menu/function key on a system phone (optional, see the phone manual)

Requirements:

- Created announcement output
- Authorisation **Loudspeaker announcement** assigned to the subscriber

1. Pick up the receiver.
2. Enter the internal number of the announcement output (for exchange line request type Direct exchange line phone with preceding **). The connection is immediatley established.

Unsuccessful Outgoing Calls

Initiating Internal Callback on Busy
Initiating External Callback on Busy
Deleting Callback on Busy
Initiating Reservation of Exchange Line on Busy
Initiating Internal Callback on No Response
Initiating External Callback on No Response
Initiating Priority Call for Do-Not-Disturb

Initiating Internal Callback on Busy

If you have initiated a callback, as soon as the other subscriber has hung up the receiver to end the call, the PBX calls you. If you then pick up the receiver, the other subscriber will be called. If the other subscriber picks up the receiver, the connection for the call will be established. This deletes the callback procedure in the PBX.

Important: On system phones, this function is called via the menu/phone app (optional, see manual for the phone).

Important: To use this function at ISDN phones or to delete it again, this function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

If the connection you are calling is busy, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial *37#.

   You will hear the confirmation tone.
3. Hang up the receiver.

   As soon as the connection is no longer busy, you are called by the public exchange for a period of 20 seconds.
4. Pick up the receiver to the make the callback.

Note: If you do not pick up the receiver within the 20 seconds during which the PBX calls you, the callback request will be deleted.

If you pick up the receiver, the subscriber will be called.

Initiating External Callback on Busy

If you have initiated a callback, as soon as the other subscriber has hung up the receiver to end the call, the public exchange calls you. If you then pick up the receiver, the other subscriber will be called. If the other subscriber picks up the receiver, the connection for the call will be established. This deletes the callback procedure in the public exchange.

Requirements:

- Fulfilling the technical requirements for a callback on busy (for instance, not all network providers provide callback on a PTP connection)
- Service feature CCBS (callback on busy) is enabled by the network provider.

Important: On system phones, this function is called via the menu/phone app (optional, see manual for the phone).

Note: To use this function at ISDN phones or to delete it again, this function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone). Instead of the announcement by the public exchange, you will hear the confirmation tone.

If the connection you are calling is busy, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial *37#.

You will hear the confirmation tone.

Note: If you do not hear a confirmation tone, the callback request has been rejected by the public exchange.

3. Hang up the receiver.

   As soon as the connection is no longer busy, you will be called by the public exchange for a period of 20 seconds.

Note: A callback will be stored in the public exchange for a duration of up to 45 minutes. If the called subscriber has not terminated his call until then, the callback will automatically be deleted.

4. Pick up the receiver to the make the callback.

Note: If you do not pick up the receiver within the 20 seconds during which the public exchange calls you, the callback request will be deleted.

   If you pick up the receiver, the subscriber will be called.

Note: You also have the option of letting the public exchange call you as soon as one of the B channels on the previously called ISDN connection is available. If the subscriber for which you have set up the callback is phoning on the other B channel, this callback will remain unsuccessful.

Note: The feature code (keypad) to execute the function, as defined in the PBX, may differ from the requirements of the network provider. In case of malfunctions, ask your network provider for the required feature code (keypad) and change the setting. If you want to use the feature directly

on the exchange line instead, the subscriber needs the authorisation for configuring features to be configured by the network provider.

Deleting Callback on Busy

If you want to delete the callback on busy initiated by you ahead of time, proceed as follows:

1. Pick up the receiver.
2. Dial #37#.
3. Hang up the receiver.

   The callback on busy will be deleted.

Important: If you use an analogue access for the external call, you will not be able to delete the callback.

Initiating Reservation of Exchange Line on Busy

If you hear a busy signal before you have continued dialling the phone number, all exchange lines (first and second B channel of the available external S_0 ports) are currently busy. You can initiate the reservation of an exchange line if you want your PBX to call you back as soon as a free line is available.

If you hear the busy signal before finishing the dialling process, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial *37#.

   You will hear the confirmation tone.
3. Hang up the receiver.

   As soon as a line is available, you will be called back by the PBX.
4. Pick up the receiver.

   You will hear the external dial tone.
5. Dial an external number without an exchange line access number.

Note: When using ISDN phones, you cannot dial directly after picking up the receiver of the ringing phone (callback of the PBX). After picking up the receiver when the phone rings, you will hear the confirmation tone instead of the external dial tone. Put down the receiver and pick it up again immediately. You can now dial as usual.

Initiating Internal Callback on No Response

If you initiated a callback, you call the subscriber you did not reach previously as soon as he is available again.

Important: On the system phones, this function is called via the menu/phone app (optional, see manual for the phone).

Important: To use this function at ISDN phones or to delete it again, this function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

Note: Some analogue phones permit this function via the T-Net function key/menu (optional, see manual for the phone).

If the subscriber does not answer your call proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial *37#.
3. Hang up the receiver. You are called as soon as the subscriber is available.
4. Pick up the receiver to the make the callback. If you do not pick up within 20 seconds, the callback is deleted.

Initiating External Callback on No Response

If you initiated a callback, you call the subscriber you did not reach previously as soon as he is available again.

Requirements:

- Service feature CCNR (callback on no response) is enabled by the network provider.
- Use of an external S_0 port for the call (optional, see Technical Data in the Instructions)

Important: On system phones, this function is called via the menu/ phone app (optional, see manual for the phone).

Note: To use this function at ISDN phones or to delete it again, this function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

Note: Some analogue phones permit this function via the T-Net function key/menu (optional, see manual for the phone).

If the subscriber does not answer your call proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial *37#.

Note: If you do not hear a confirmation tone, the callback request was not successful.

3. Hang up the receiver. You are called as soon as the subscriber is available.

Note: A callback will be stored for up to 45 minutes.

4. Pick up the receiver to the make the callback. If you do not pick up within 20 seconds, the callback is deleted.

Initiating Priority Call for Do-Not-Disturb

If, for example, a person must absolutely be reached in an emergency despite the do-not-disturb function, the do-not-disturb function can be circumvented by a priority call from an internal phone.

If the internal phone you are trying to call is busy because it is set to do-not-disturb, proceed as follows:

- Wait ten seconds.

  The subscriber is now called.

Query Calls

Information on the Chapter
Initiating a Query Call
Ending a Query Call
Toggling between Conversations
Ending One of the Two Calls
Connecting Both Communication Partners
Initiating Query Call to Third Subscriber

Information on the Chapter

During a query call, you are talking with one subscriber while your previous communication partner is on hold on the PBX in the background listening to hold music.

Initiating a Query Call

Requirements:

- For a query call with two external communication partners: at least two external call channels

During a call, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Enter a phone number. Possible entries:

   Internal number of the individual phone (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the door (for exchange line request type Direct exchange line phone: with preceding **)

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   The subscriber is called.

Note: After pressing the **Flash key** or the **Query key**, you can also initiate other call types (e. g. private calls), an InterCom announcement, or a pick-up. Exception: During a door call only the dialling of one phone number is possible.

Note: Note. You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On ISDN and system phones, this function is performed via an existing function key or via the menu (optional, see manual for the phone).

Ending a Query Call

If you make a query call and the number is busy but there is no response, proceed as described in one of the following steps.

- The subscriber called for the query call does not respond: Press the **Flash key** or the **Query key** and dial 1.

  You are now reconnected with the person waiting on the other line.

- The subscriber called for the query call is busy. Wait.

  You will automatically be reconnected with the communication partner waiting on the other line.

Note: On ISDN and system phones, this function is performed via a function key or via the menu (optional, see manual for the phone).

Toggling between Conversations

You can toggle between your two communication partners.

During a query call, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 2.

   You continue to conduct a query call. The other communication partner is now active.

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On ISDN and system phones, this function is performed via an existing function key or via the menu (optional, see manual for the phone).

Ending One of the Two Calls

When you end a query call and want to continue talking with one of the communication partners, you can specifically end the call with the other communication partner separately.

If one of your two communication partners hangs up the receiver, you will remain connected with the other one.

Alternatively, during a query call, proceed as described in one of the following steps.

- Press the **Flash key** or the **Query key** and dial 1.

  The current call is disconnected.

  You will then make an one-to-one call with the communication partner who had previously been on hold.

- Press the **Flash key** or the **Query key** and dial 0.

  The call in the background will be disconnected.

  You will then make an one-to-one call with the communication partner who had previously been active.

Note: Note. You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On ISDN and system phones, this function is performed via an existing function key or via the menu (optional, see manual for the phone).

Connecting Both Communication Partners

If you want to let your communication partners speak with each other, you can connect them.

Requirements:

- To connect two external communication partners, the following is required: Authorisation **Transfer of external calls to external** assigned to the subscriber

During a query call, proceed as follows:

- Hang up the receiver.

  Both communication partners are connected with each other.

Initiating Query Call to Third Subscriber

If you want to connect a further communication partner during a query call, you can initiate a third call for this purpose. Both the first and second communication partner will be on hold in the background during this time.

Requirements:

- Initiated query call

During a query call, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 5.
3. Enter a phone number. Possible entries:

   Internal number of the individual phone (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   As soon as to the subscriber being called picks up the receiver, you will make a third call.

Important: To use this function on ISDN and system phones, the function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

4. Proceed as required.

   - To disconnect the third call and go back to the query call with the previously active communication partner, press the **Flash key** or the **Query key** and dial **1**.

     •You then make a query call with the communication partner who had previously been active. The call with the third communication partner is disconnected.

   - To disconnect the third call and go back to the query call with the communication partner who had previously been on hold, press the **Flash key** or the **Query key** and dial **2**.

     •You will then make a query call with the communication partner who had previously been on hold. The call with the third communication partner is disconnected.

   - To disconnect the previous calls and make an one-on-one call with the third communication partner, press the **Flash key** or the **Query key** and dial **0**.

     •You are making an one-on-one call with the third communication partner. The previous calls have been disconnected.

   - To connect the third communication partner to the second communication partner, press the **Flash key** or the **Query key** and dial **4**.

     •The second and the third communication partner are now connected with each other. They are making an one-on-one call with the first communication partner.

Note: If the third communication partner goes off-hook beforehand, you will get back to the previously active communication partner.

Subsequently press the **Flash key** or the **Query key** and dial **1**. Also use this function to terminate the call to this communication partner.

Transferring

Transferring with Announcement
Blind Transfer (Transferring without Announcement)
Placing Communication Partner in Waiting Loop
Calling a Third Subscriber and Transferring to an Active Communication Partner
Parking the Call Internally (Call Parking)
Unparking the Call (Call Parking)

Transferring with Announcement

In order to transfer a call, first initiate a query call, announce the call and then connect both communication partners.

Requirements:

- Initiated query call
- To connect two external communication partners, the following is required: Authorisation **Transfer of external calls to external** assigned to the subscriber

- Hang up the receiver during a query call.

  Both communication partners are connected with each other.

Note: Some ISDN phones may ring after hanging up the receiver and you may then be reconnected with your first communication partner after picking up the receiver. Use the manual for the phone to check whether the **Transfer at a PBX** function has been enabled. If this has not been done, do this subsequently.

- If you want to initiate a new call immediately after the transfer, proceed during the query call as follows.

  Both communication partners are connected with each other.

  You will hear the internal dial tone.

Important: A call transferred between two external communication partners is limited to a defined period of time. This is a security feature, since the transfer of the time announcement or of the weather forecast made by mistake, for instance, would lead to a very long call.

Note: You will be charged the fees for a transferred call by your network provider.

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On some ISDN phones, this function can be called via an existing function key or via the menu (optional, see manual for the phone).

Blind Transfer (Transferring without Announcement)

If you want to transfer an external communication partner to another internal subscriber, you do not have to wait until the subscriber being called via the query call accepts the call; you can hang up the receiver while this call is being made. The internal subscriber will continue to be called.

Requirements:

- Internal transfer destination

During a call, proceed as follows.

1. Press the **Flash key** or the **Query key**.
2. Enter the internal number. Possible entries:

   Internal number of the individual phone (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)
3. Wait until you hear the ringtone.
4. Hang up the receiver.

   Once the called subscriber accepts the call, he is connected to the external subscriber.

Note: If the subscriber being called does not accept the call, the call will be transferred back to your phone 60 seconds later (your phone rings). If you do not accept the call within 60 seconds the call will be disconnected. If your own phone has become busy in the meantime, the

procedure will already be cancelled after the first 60 seconds and the external call will be disconnected.

Placing Communication Partner in Waiting Loop

If the internal subscriber to which you want to transfer an communication partner is busy, you can place the communication partner in the waiting loop of this subscriber.

Requirements:

- Internal transfer destination is a group (at least one subscriber logged in as incoming) or a subscriber

If the internal phone to which you are trying to transfer a call is busy, proceed as follows:

1. Wait.

   You will automatically be reconnected with the waiting communication partner.

Note: On many ISDN phones, the termination of a query call on busy is not performed automatically, but via a function key or via the menu (optional, see manual for the phone).

2. Press the **Flash key** or the **Query key**.
3. Dial ##07.

   You will hear the confirmation tone.

4. Hang up the receiver.

   The communication partner continues to listen to hold music.

   As soon as the internal subscriber hangs up the receiver or one subscriber logged in to the group becomes available, this

communication partner is called. Once he accepts the call, he is connected with the subscriber in the waiting loop.

Note: If the internal subscriber does not answer the call within 2 minutes or the line remains busy during the 15-minute waiting time, the call will be transferred back to you. If you pick up the receiver, you will be reconnected with the communication partner. You can then place him in the waiting loop again. If you do not pick up the call within 2 minutes, the connection is disconnected completely (for example, if your line was busy even for a short time).

Calling a Third Subscriber and Transferring to an Active Communication Partner

Assuming that you have gained an additional communication partner from a call, e. g. from accepting a call signaled by call waiting, and this second communication partner want to be transferred now to somebody else. You can now initiate a third call that connects the second and third communication partners with each other and then continue phoning with your first communication partner.

During a query call, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 5.
3. Enter a phone number. Possible entries:

   External number (for exchange line request type Internal phone: with preceding exchange line access number)

   Short code number (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the individual phone (for exchange line request type Direct exchange line phone: with preceding **)

   Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

   As soon as the subscriber being called picks up the receiver, you will make a third call.

Important: To use this function on ISDN and system phones, the function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

4. Proceed as required.

   - To connect the second and third communication partner and to perform a one-to-one call with the first communication partner, press the **Flash key** or the **Query key** and dial **4**.

     •The second and the third communication partner are now connected with each other.
     •Then return to conducting an individual call with your first communication partner.

   - To connect the second and third communication partner and to terminate the call with the first communication partner, hang up the receiver.

     •The second and the third communication partner are now connected with each other.
     •The first call will be signaled by the PBX if it was an external call. The internal call is disconnected.

Note: If the third communication partner is not ready to be transferred to the second communication partner, press the **Flash key** or the **Query key** and dial **1**. You will then get back to the normal query call.

Parking the Call Internally (Call Parking)

Call parking lets a caller in a waiting position be transferred so that the desired communication partner can accept the call on another internal phone.

Requirements:

- Authorisation **Call Parking** assigned to the subscriber
- Configured base number

During an external call, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Enter the internal base number. Possible entries:

   Internal base number for the parking zone (for exchange line request type Direct exchange line phone: with preceding **)

3. Dial the number of the park position. Possible entries:

   0 to **9**

Note: If you hear a busy signal when you dial the park position, this position is already in use. Pressing the **Flash key** or the **Query key** takes you back where you can redial the internal base number with another park position.

4. Wait until you hear the ringtone.
5. Hang up the receiver.
6. Notify the desired communication partner of the waiting caller (e.g., by making an announcement over the loudspeaker).

Unparking the Call (Call Parking)

Call parking lets a caller in a waiting position be transferred so that the desired communication partner can accept the call on another internal phone.

Requirements:

- Authorisation **Call Parking** assigned to the subscriber
- Parked call

1. Pick up the receiver.

Note: During an existing call press the **Flash key** or the **Query key** and dial **##06**.

2. Enter the internal base number. Possible entries:

   Internal base number for the parking zone (for exchange line request type Direct exchange line phone: with preceding **)

3. Dial the number of the park position. Possible entries:

   0 to **9**

   You are talking with the waiting caller.

Conference Calls

Initiating Conference Calls
Initiating Toggling between Conversations during a Conference Call
Connecting Both Conference Call Partners
Ending Conference Call

Initiating Conference Calls

Requirements:

- A query call you initiated yourself

During a query call, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 3.

   You are making a conference call.

Note: If a hands-free phone is involved in a conference with 3 internal subscribers, echo or singing effects may occur.

Note: You can also use this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Note: On ISDN and system phones, this function is performed via an existing function key or via the menu (optional, see manual for the phone).

Note: A conference is not possible with a door terminal.

Initiating Toggling between Conversations during a Conference Call

When you end a conference call and want to talk to each of the conference call partners separately one after the other, you can specifically put one of the conference call partners on hold in the background.

Requirements:

- A conference call that you initiated yourself

During a conference call, proceed as follows:

- Press the **Flash key** (when using a phone with DTMF dialling) or the **Query key** (when using an ISDN phone) and dial 2.

  You then make an individual call with the communication partner who had previously been active. The other subscriber of the conference hears the hold music.

- Press the **Flash key** or the **Query key** and dial 1.

  You then make a call with the communication partner who had previously been on hold in the background. The other subscriber of the conference hears the hold music.

Important: To use this function on ISDN and system phones, the function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

Note: You also can use this function this function on analogue phones, where it can be performed via the T-Net function key/menu (optional, see manual for the phone).

Connecting Both Conference Call Partners

If you want to leave a conference call that you initiated yourself and your conference call partners want to continue talking with each other, you can connect both parties.

Requirements:

- A conference call that you initiated yourself
- To connect two external communication partners, the following is required: Authorisation **Transfer of external calls to external** assigned to the subscriber

During a conference call, proceed as follows:

1. Press the **Flash key** or the **Query key**.
2. Dial 4.
3. Hang up the receiver.

   Both conference call partners are connected with each other.

Important: To use this function at ISDN phones and system phones, the function must be supported via a function key or via the menu by the respective phone (optional, see manual for the phone).

Ending Conference Call

If you initiated the conference call yourself, you can end the conference call completely.

Requirements:

- A conference call that you initiated yourself

During a conference call, proceed as follows:

- Hang up the receiver.

  The conference call is disconnected completely.

Short Reference

Information on the Chapter
Alarm
Reservation of Exchange Line on Busy
Force Public Exchange Call for Automatic Exchange Line Request
Call Waiting
Time and Date (of the PBX)
Do-not-Disturb
Automatic Reception
Baby Call/Hotline
Call Through
Remote Configuration (Releasing)
Hook Flash Timing
Follow-me
Call Deblocker (Incoming)
Call Allowance Account
Outgoing Calls
Call Data (Single Call Record)
Targeted Exchange Line Access
Targeted VoIP Access
Groups
Hotel Functions
Identification (IPv4 Address, Subnet Mask, Gateway, DHCP Client)
InterCom Announcement/Handsfree Speaking
Conference

Configuration Switchover
Loudspeaker Announcement
Emergency Call Announcement
Call Parking
Multi-Path Call Forwarding
Pick-up and Call Take-over
PIN
Private (Personalised) Exchange Line Access
Project Assignment
Relay
Query and 3rd Call
Query and Transfer
Callback on Busy
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Call Forwarding for Groups
Call Forwarding for Subscribers
Call Restrictor (Incoming)
Announcement before Answering
Door Functions
Voice Mailbox
Music on Hold and Announcement
Wake-up Functions

Information on the Chapter

This section includes an alphabetical list of functions and settings of the PBX you can configure using an internal or, in some cases, an external phone. The functions available in the menu of the system phones as well as the menu functions or keys used by individual ISDN phones are not listed (see manual for the phone).

The headlines explain which device may be used for the named character string. If the column is limiting the character string (e.g. for starting a conference) to analogue phones only it may be available to ISDN phones via menu (see manual of the phone).

Is the function also available to external devices a remote switching number configured in the PBX has to be dialled first.

Alarm

Confirming an Alarm within 60 sec (on an Internal or External Phone)

0

additional information	Alarm enabled once: Alarm input is disabled after triggering and alarm must be enabled again. Alarm enabled always: Alarm input remains enabled after triggering and another alarm can be triggered.

Activating an Alarm Once (on an Internal Phone)

8 PIN * 24 Alarm number * 1

PIN	Admin or sub-admin PIN
Alarm number	Alarm number (for exchange line request type Direct exchange line phone: with preceding **)
additional information	Alarm input is disabled after triggering.

Activating an Alarm Once (on an External Phone)

Remote switching number ## 8 PIN * 24 Alarm number * 1 #

Remote switching number	Remote switching number for remote programming
PIN	Scr. PIN
Alarm number	Alarm number

additional information	Alarm input is disabled after triggering.

Activating an Alarm Always (on an Internal Phone)

8 PIN * 24 Alarm number * 2

PIN	Admin or sub-admin PIN
Alarm number	Alarm number (for exchange line request type Direct exchange line phone: with preceding **)
additional information	Alarm input stays enabled even after triggering.

Activating an Alarm Always (on an External Phone)

Remote switching number ## 8 PIN * 24 Alarm number * 2

Remote switching number	Remote switching number for remote programming
PIN	Scr. PIN
Alarm number	Alarm number
additional information	Alarm input stays enabled even after triggering.

Deactivating an Alarm (on an Internal Phone)

8 PIN * 24 Alarm number * 0

PIN	Admin or sub-admin PIN
Alarm number	Alarm number (for exchange line request type Direct exchange line phone: with preceding **)
additional information	Alarm input is disabled and will not react to triggering.

Deactivating an Alarm (on an External Phone)

Remote switching number ## 8 PIN * 24 Alarm number * 0

Remote switching number	Remote switching number for remote programming
PIN	Scr. PIN
Alarm number	Alarm number
additional information	Alarm input is disabled and will not react to triggering.

Recording an Announcement (on an Internal Phone)

8 PIN * 56 A * *

PIN	Admin or sub-admin PIN
A	36...39 (no. 1-4 of the announcement) 71...86 (no. 1-16 of the announcment)
additional information	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #. Afterwards the recording is played back for checking. Both number ranges can be used for A (also at the same time) but a maximum of 16 announcements is configurable.

Deleting the Announcement (on an Internal Phone)

8 PIN * 55 A

PIN	Admin or sub-admin PIN
A	36...39 (no. 1-4 of the announcement) 71...86 (no. 1-16 of the announcment)
additional information	Both number ranges can be used for A (also at the same time) but a maximum of 16 announcements is configurable.

Listening to the Announcement (on an Internal Phone)

8 PIN * 57 A

PIN	Admin or sub-admin PIN
A	36...39 (no. 1-4 of the announcement) 71...86 (no. 1-16 of the announcment)
additional information	Both number ranges can be used for A (also at the same time) but a maximum of 16 announcements is configurable.

Reservation of Exchange Line on Busy

Initiating the Reservation of an Exchange Line on Busy (on an Internal Phone)

R * 37 #

R	**Flash key** or **Query key**
Additional information	You will hear a confirmation tone. Hang up the receiver. As soon as a line is available, you will be called back by the PBX. Pick up the receiver. You will hear the external dial tone. Dial an external number without an exchange line access number.

Force Public Exchange Call for Automatic Exchange Line Request

Force Public Exchange Call for Automatic Exchange Line Request ## 94

Additional information	Pick up the receiver. Enter an external number. Once you have finished dialling the phone number, the subscriber will be called.

Call Waiting

Rejecting a Knocking Caller (on an Analogue Internal Phone)

R 0

R	**Query key** or **Flash key**)

Accepting a Knocking Caller and Disconnecting the Active Call (on an Analogue Internal Phone)

R 1

R	**Query key** or **Flash key**

Accepting a Knocking Caller and Putting the Active Call on Hold (on an Analogue Internal Phone)

R 2

R	**Query key** or **Flash key**

Time and Date (of the PBX)

Setting the Time (on an Internal Phone)

8 PIN * 300 hh mm ss

PIN	Admin or sub-admin PIN
hh	00...23 (hour)
mm	00...59 (minute)
ss	00...59 (second)

Setting the Date (on an Internal Phone)

8 PIN * 301 dd mm yy

PIN	Admin or sub-admin PIN
dd	01...31 (day)
mm	01...12 (month)
yy	00...99 (year)

Do-not-Disturb

Enabling Do-not-Disturb (on an Internal Phone)

8 * 21 1

Disabling Do-not-Disturb (on an Internal Phone)

8 * 21 0

Automatic Reception

Recording an Announcement (on an Internal Phone)

8 PIN * 564 A * *

PIN	Admin or sub-admin PIN
A	1...9, 0 (no. 1-10 of the announcement)
Additional information	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #. Afterwards the recording is played back for checking.

Deleting an Announcement (on an Internal Phone)

8 PIN * 554 A

PIN	Admin or sub-admin PIN
A	1...9, 0 (no. 1-10 of the announcement)

Listening to an Annoucement (on an Internal Phone)

8 PIN * 574 A

PIN	Admin or sub-admin PIN
A	1...9, 0 (no. 1-10 of the announcement)

Baby Call/Hotline

Configuring Baby Call/Hotline (on an Internal Phone)

*** 53 * Destination number #**

Destination number	Internal or external phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number)

Enabling Baby Call/Hotline without Re-entering the Number (on an Internal Phone)

*** 53 #**

Disabling Baby Call/Hotline (on an Internal Phone)

 # 53 #

Checking Activation (on an Internal Phone)

*** # 53 #**

Additional information	Busy tone = baby call/hotline is disabled

Call Through

Initiating a Call (on an External Phone)

Call Through number Destination number

Call Through number	External number defined for Call Through on the PBX
Destination number	Internal or external phone to be called ** + internal number of the group or subscriber or ** + short-code dialling number

Remote Configuration (Releasing)

Initiating/Releasing Remote Configuration (on an Internal Phone)

8 PIN * 91

PIN	Admin, sub-admin or user PIN

Initiating/Releasing Remote Configuration (on an External Phone)

Remote switching number ## 8 PIN * 91 #

Remote switching number	Remote switching number for remote programming
PIN	ARA PIN

Hook Flash Timing

Learning the Flash Timing (on an Analogue Internal Phone)

8 * 98 * R R

R	**Query key** or **Flash key**

Follow-me

Requirements:

- Authorisation for **CF (groups) / Follow-me**
- Authorisation for **Controlling of the PBX via phone**
- For internal destination: Minimum requirement - exchange line authorisation **Only incoming, with emergency call** to accept forwarded external calls
- For call forwarding to an external number: Authorisation **Configuring CF (sub) to ext. number**
- For call forwarding to an external number: The external destination must be reachable for the subscriber (minimum requirement: Exchange line authorisation **National**, selected call restrictor under Exchange line settings **Call restrictor (outgoing)** business does not restrict the selection of the external destination)
- For configuring Follow-me externally: Knowledge of the Remote switching number for remote programming and PIN (Scr. PIN)
- For configuring Follow-me externally: DTMF dialling at the configuring device

Enabling Follow-me (on the Internal Destination Phone)

8 * 19 1 Subscriber/group number

Subscriber/group number	Phone/group to be redirected Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **)

Disabling Follow-me (on the Internal Destination Phone)

Deletes all redirections via Follow-me with this phone as destination.

8 * 19 0

Enabling Follow-me (on Any Internal Phone)

8 * 19 1 Subscriber/group number * Destination number

Subscriber/group number	Phone/group to be redirected Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **)
Destination number	Internal or external phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number)

Disabling Follow-me (on Any Internal Phone)

8 * 19 0 Subscriber/group number

Subscriber/group number	Redirected phone/group Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **)

Enabling Follow-me (on an External Phone)

Remote switching number ## 8 PIN * 19 1 Subscriber/group number * Destination number

PIN	Scr. PIN
Remote switching number	Remote switching number for remote programming
Subscriber/group number	Phone/group to be redirected Internal number of the group or subscriber
Destination number	Internal or external phone to be called Internal number of the group or subscriber or External number
Additional information	After entering the remote switching number, wait for external confirmation signal (1 second) before continuing DTMF entry.

Disabling Follow-me (on an External Phone)

Remote switching number ## 8 PIN * 19 0 Subscriber/group number

PIN	Scr. PIN
Remote switching number	Remote switching number for remote programming
Subscriber/group number	Phone/group to be redirected Internal number of the group or subscriber

Additional information	After entering the remote switching number, wait for external confirmation signal (1 second) before continuing DTMF entry.

Call Deblocker (Incoming)

Enabling Call Deblocker (Incoming) for Subscribers (on an Internal Phone)

8 PIN * 22 1

PIN	User PIN

Disabling Call Deblocker (Incoming) for Subscribers (on an Internal Phone)

8 PIN * 22 0

PIN	User PIN

Call Allowance Account

Configuring the Call Allowance Account of a Subscriber (on an Internal Phone)

8 PIN * 74 Subscriber number * E

PIN	Admin or sub-admin PIN
Subscriber number	Internal number of the individual phone (for exchange line request type Direct exchange line phone: with preceding **)
E	0...99998 (setting the account to a specific amount: 200, for example, equals EUR 2.20.) or 99999 (unlimited account)

Outgoing Calls

Initiating an Internal Call (on an Internal Phone)

Destination number

Destination number	Internal phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **)

Initiating an External Call (on an Internal Phone)

Destination number

Destination number	Internal phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Initiating an External Call with Suppressed Number Presentation (on an Internal Phone)

* 31 # Destination number

Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number) or

	Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Call Data (Single Call Record)

Deleting Own Call Data Records (Works Council Function - on an Internal Phone)

8 PIN * 51

PIN	User PIN
Additional information	Authorisation required

Targeted Exchange Line Access

Initiating a Call with a Specific MSN (on an Internal Phone)

96 MSN * Destination number

MSN	MSN to be sent Own phone number (number without exchange line access number (unaffected by a direct exchange line phone) or, for a PTP connection, PBX base number and extnsion number)
Destination number	External phone to be called External number (for exchange line request type internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type direct exchange line phone: with preceding **)

Targeted VoIP Access

Initiating an External Call Using Targeted VoIP Access (on an Internal Phone)

Account number Destination number

Account number	Used VoIP account VoIP Account number (for exchange line request types Direct exchange line phone **and Automatic exchange line request**: with preceding **)
Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number)

Groups

Logging in to a Group (Incoming and Outgoing - on an Internal Phone)

8 * 40 1 Group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Logging out of a Group (Incoming and Outgoing - on an Internal Phone)

8 * 40 0 Group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Logging out of All Groups (Incoming and Outgoing - on an Internal Phone)

8 * 40 0

Logging in to a Group (Only Incoming - on an Internal Phone)

8 * 41 1 Group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Logging in to All Groups (Only Incoming - on an Internal Phone)

8 * 41 1

Logging out of a Group (Only Incoming - on an Internal Phone)

8 * 41 0 Group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Logging out of All Groups (Only Incoming - on an Internal Phone)

8 * 41 0

Logging in to a Group (Only Outgoing - on an Internal Phone)

8 * 42 1 Group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Logging out of the Group (Only Outgoing - on an Internal Phone)

8 * 42 0

Logging All Subscribers out of a Group (Incoming and Outgoing - on an Internal Phone)

8 * 48 0 Group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)

Additional information	Membership of the group required

Hotel Functions

Entering the wake-up Time for One-time wake-up (internal phone)

*** hh mm #**

hh	00...23 (hour)
mm	00...59 (minute)

Deleting the Wake-up Time (internal phone)

*** #**

Setting the Room Status (internal phone)

*** 99 Z #**

Z	1 (room cleaned) 2 (room not cleaned) 3 (room blocked)

Identification (IPv4 Address, Subnet Mask, Gateway, DHCP Client)

Note: Only IPv4 addresses can be entered using a character string on the phone. IPv6 addresses must be entered using the Configuration Manager of the PBX.

Configuring the IPv4 Address of the PBX (on an Internal Phone)

8 PIN * 931 * E

PIN	Admin or sub-admin PIN
E	IPv4 address of the PBX (12 digits)

Querying the IPv4 Address of the PBX (on an Internal Phone)

8 * 941

Additional information	Display of the IPv4 address of the PBX on the phone display after callback by the PBX

Configuring the Subnet Mask of the PBX (on an Internal Phone)

8 PIN * 932 * E

PIN	Admin or sub-admin PIN
E	Subnet mask (12 digits)

Querying the Subnet Mask of the PBX (on an Internal Phone)

8 * 942

Additional information	Display of the subnet mask on the phone display after callback by the PBX

Configuring the Gateway of the PBX (on an Internal Phone)

8 PIN * 933 * E

PIN	Admin or sub-admin PIN
E	IPv4 address of the gateway (12 digits)

Querying the Gateway of the PBX (on an Internal Phone)

8 * 943

Additional information	Display of the IPv4 address of the gateway on the phone display after callback by the PBX

Switching the DHCP Client of the PBX on (on an Internal Phone)

8 PIN * 930 * 1

PIN	Admin or sub-admin PIN

Switching the DHCP Client of the PBX off (on an Internal Phone)

8 PIN * 930 * 0

PIN	Admin or sub-admin PIN

Querying the DHCP Client Status of the PBX (on an Internal Phone)

8 * 940

Additional information	Display of the DHCP client status of the PBX on the phone display after callback by the PBX 0 = off 1 = on

InterCom Announcement/Handsfree Speaking

Announcement to a System Phone (on an Internal Phone)

011 Subscriber/group number

Subscriber/group number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **)

Handsfree Speaking to a System Phone (on an Internal Phone)

012 Subscriber/group number

Subscriber/group number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **)

Conference

Initiating a Conference from a Query (on an Analogue Internal Phone)

R 3

R	**Query key** or **Flash key**

Back to Toggling Between Conversations, as before Conference (on an Analogue Internal Phone)

R 2

R	**Query key** or **Flash key**
Additional information	You are talking to the previously active call partner. The other conference partner is hearing music on hold.

Back to Toggling Between Conversations (on an Analogue Internal Phone)

R 1

R	**Query key** or **Flash key**
Additional information	You are talking to the call partner previously held in the background. The other conference partner is hearing music on hold.

Connecting Call Partners (on an Analogue Internal Phone)

R 4

R	**Query key** or **Flash key**

Configuration Switchover

Switching the Configuration (on an Internal Phone)

8 * 81 K

K	2- to 4-digit ID no. of the configuration
Additional information	Authorisation required

Switching the Configuration (on an External Phone)

Remote switching number ## 8 PIN * 81 K

Remote switching number	Remote switching number for remote programming
PIN	Scr. PIN
K	2- to 4-digit ID no. of the configuration
Additional information	Authorisation required

Loudspeaker Announcement

Announcement to Announcement Output (on an Internal Phone)

Number of the announcement output

Number of the announcement output	Internal number of the announcement output (for exchange line request type Direct exchange line phone: with preceding **)

Emergency Call Announcement

Recording an Announcement (on an Internal Phone)

8 PIN * 5650 * *

PIN	Admin or sub-admin PIN
Additional information	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #. Afterwards the recording is played back for checking.

Listening to the Announcement (on an Internal Phone)

8 PIN * 5750

PIN	Admin or sub-admin PIN

Deleting the Announcement (on an Internal Phone)

8 PIN * 5550

PIN	Admin or sub-admin PIN

Call Parking

Putting a Call into the Parking Zone (on an Internal Phone)

R Parking number P

R	**Query key** or **Flash key**
Parking number	Internal number of the parking zone (for exchange line request type Direct exchange line phone: with preceding **)
P	0...9 (no. 0-9 of the park position)
Additional information	Authorisation required

Picking up a Call/Conversation (on an Internal Phone)

Parking number P

Number	Internal number of the parking zone (for exchange line request type Direct exchange line phone: with preceding **)
P	0...9 (no. 0-9 of the park position)
Additional information	Authorisation required

Taking over a Call/Conversation in an Ongoing Call (on an Internal Phone)

R ## 06 Parking number P

R	**Query key** or **Flash key**
Parking number	Internal number of the parking zone (for exchange line request type Direct exchange line phone: with preceding **)

P	0...9 (no. 0-9 of the park position)
Additional information	Authorisation required

Multi-Path Call Forwarding

Configuring Multi-Path Call Forwarding for Subscribers (on an Internal Phone)

*** 481 * Destination number #**

Destination number	Internal or external phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External telepohone number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Enabling Multi-Path Call Forwarding for Subscribers (on an Internal Phone)

*** 481 #**

Disabling Multi-Path Call Forwarding for Subscribers (on an Internal Phone)

 # 481 #

Checking Activation (Subscriber – on an Internal Phone)

*** # 481 #**

Additional information	Busy signal = multi-path call forwarding is disabled

Configuring Multi-Path Call Forwarding for Groups (on an Internal Group Phone)

97 Group number # * 481 * Destination number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)
Destination number	Internal or external phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)
Additional information	Membership of the group required

Enabling Multi-Path Call Forwarding for Groups (on an Internal Group Phone)

97 Group number # * 481

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)
Additional information	Membership of the group required

Disabling Multi-Path Call Forwarding for Groups (on an Internal Group Phone)

97 Group number ## 481

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)
Additional information	Membership of the group required

Checking Activation (Group – on an Internal Phone)

97 Group number # * # 481

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **)
Additional information	Busy signal = multi-path call forwarding is disabled

Pick-up and Call Take-over

Taking over a Call/Conversation (on an Internal Phone)

06 Subscriber/group number

Subscriber/group number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **)
Additional information	Authorisation required

PIN

Changing the PIN (on an Internal Phone)

*** 99 * PIN * PIN * PIN #**

1st PIN	Old PIN to be changed = Admin, sub-admin, user or ARA PIN
2nd PIN	New PIN = Admin, sub-admin, user or ARA PIN
3rd PIN	New PIN = Admin, sub-admin, user or ARA PIN

Note: If, at the end, you hear a busy signal instead of a confirmation tone, the new PIN may already be in use in the PBX and no change takes place. Re-attempt with another digit sequence.

Private (Personalised) Exchange Line Access

Initiating an External Call with Private Exchange Line Access (on an Internal Phone)

92 PIN * Destination number

PIN	User PIN
Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Initiating an External Call with Private Exchange Line Access without PIN (on Own Internal Phone)

91 Destination number

Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)
Additional information	Only on own phone Authorisation required

Project Assignment

Initiating an External Call with Assigned Project (on an Internal Phone)

93 P * Destination number

P	2- to 6-digit project no.
Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Relay

Switching a Relay on – Switches off Automatically after Configured Switching Duration (on an Internal Phone)

8 * 26 R

R	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes light, door opener, alarm and PC relay Authorisation required

Switching a Relay on – Switches off Automatically after Configured Switching Duration (on an External Phone)

Remote switching number ## 8 PIN * 26 R #

Remote switching number	Remote switching number for remote programming
PIN	Scr. PIN
R	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes light, door opener, alarm and PC relay Authorisation required

Switching a Relay on (on an Internal Phone)

8 * 26 R * 1

R	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes manual relay, configuration-dependent and automatic configuration switchover Authorisation required

Switching a Relay on (on an External Phone)

Remote switching number ## 8 PIN * 26 R * 1

Remote switching number	Remote switching number for remote programming
PIN	Scr. PIN
R	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes manual relay, configuration-dependent and automatic configuration switchover Authorisation required

Switching a Relay off (on an Internal Phone)

8 * 26 R * 0

R	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes manual relay, configuration-dependent and automatic configuration switchover Authorisation required

Switching a Relay off (on an External Phone)

Remote switching number ## 8 PIN * 26 R * 0

Remote switching number	Remote switching number for remote programming
PIN	Scr. PIN
R	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes manual relay, configuration-dependent and automatic configuration switchover Authorisation required

Query and 3rd Call

Initiating a 3rd Call (on an Analogue Internal Phone)

R 5 Destination number

R	**Query key** or **Flash key**
Destination number	Internal or external phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Disconnecting the 3rd Call and Returning to the Last Active Call (on an Analogue Internal Phone)

R 1

R	**Query key** or **Flash key**

Disconnecting the 3rd Call and Returning to the Last Call on Hold (on an Analogue Internal Phone)

R 2

R	**Query key** or **Flash key**

Connecting 2nd and 3rd Call Partner and Continuing to Talk to the 1st Call Partner (on an Analogue Internal Phone)

R 4

R	**Query key** or **Flash key**

Continuing the 3rd Call and Disconnecting the Other Calls (on an Analogue Internal Phone)

R 0

R	**Query key** or **Flash key**

Query and Transfer

Initiating a Query (on an Analogue Internal Phone)

R Destination number

R	**Query key** or **Flash key**
Destination number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number) or Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Cancelling a Ringing Call (on an Analogue Internal Phone)

R 1

R	**Query key** or **Flash key**

Splitting (on an Analogue Internal Phone)

R 2

R	**Query key** or **Flash key**

Disconnecting a Call on Hold in the Background (on an Analogue Internal Phone)

R 0

R	**Query key** or **Flash key**

Disconnecting the Active Call and Returning to the Call Partner on Hold (on an Analogue Internal Phone)

R 1

R	**Query key** or **Flash key**

Connecting Both Call Partners (on an Analogue Internal Phone)

R 4

R	**Query key** or **Flash key**

Transferring a Call Partner into Waiting Loop on Busy (on an Analogue Internal Phone)

R ## 07

R	**Query key** or **Flash key**

Callback on Busy

Initiating Callback on Busy (on an Internal Phone)

R * 37 #

R	**Query key** or **Flash key**

Cancelling Callback on Busy (on an Internal Phone)

 # 37 #

Callback on No Reply

Initiating Callback on No Reply (on an Internal Phone)

R * 37 #

R	**Query key**or **Flash key**

Cancelling Callback on No Reply (on an Internal Phone

 # 37 #

Call Forwarding for External Numbers

Configuring Unconditional Call Forwarding (on an Internal Phone)

96 Own number # * 21 * Destination number

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone) The 1st number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation required Authorisation for external destination required Permanent configuration is enabled.

Enabling Unconditional Call Forwarding without Re-entering the Number (on an Internal Phone)

96 Own number # * 21

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required The destination number may only be an external number.

	Permanent configuration is enabled.

Disabling Unconditional Call Forwarding (on an Internal Phone)

96 Own number ## 21

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required The destination number may only be an external number. Permanent configuration is enabled.

Configuring Call Forwarding on Busy (on an Internal Phone)

96 Own number # * 67 * Destination number

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The 1st number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Destination number	External number to be called External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation required Authorisation for external destination required Permanent configuration is enabled.

Enabling Call Forwarding on Busy without Re-entering the Number (on an Internal Phone)

96 Own number # * 67

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required The destination number may only be an external number. Permanent configuration is enabled.

Disabling Call Forwarding on Busy (on an Internal Phone)

96 Own number ## 67

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required The destination number may only be an external number. Permanent configuration is enabled.

Configuring Call Forwarding on No Reply (on an Internal Phone)

96 Own number # * 61 * Destination number

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The 1st number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Destination number	External number to be called External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation required Authorisation for external destination required Permanent configuration is enabled.

Enabling Call Forwarding on No Reply without Re-entering the Number (on an Internal Phone)

96 Own number # * 61

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required The destination number may only be an external number. Permanent configuration is enabled.

Disabling Call Forwarding on No Reply (on an Internal Phone)

96 Own number ## 61

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone)) The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required The destination number may only be an external number. Permanent configuration is enabled.

Enabling Time Control for the External Number (MSN) of the Dialling Phone (on an Internal Phone))

96 # * 69

Additional information	Permanent configuration is disabled.

Enabling Time Control for a Special Phone Number (on an Internal Phone)

96 Own number # * 69

Own number	Own phone number (number without exchange line access number (unaffected by a direct exchange line phone))
Additional information	Permanent configuration is disabled.

Call Forwarding for Groups

Configuring Unconditional Call Forwarding (on an Internal Phone)

97 Group number # * 21 * Subscriber/group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The 1st number can be omitted if the phone is logged in to the group as outgoing.
Subscriber/group number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required Authorisation and membership of the group necessary

Enabling Unconditional Call Forwarding without Re-entering the Number (on an Internal Phone)

97 Group number # * 21

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Disabling Unconditional Call Forwarding (on an Internal Phone)

97 Group number ## 21

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Configuring Call Forwarding on Busy (on an Internal Phone)

97 Group number # * 67 * Subscriber/group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The 1st number can be omitted if the phone is logged in to the group as outgoing.
Subscriber/group number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required Authorisation and membership of the group required

Enabling Call Forwarding on Busy without Re-entering the Number (on an Internal Phone)

97 Group number # * 67

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Disabling Call Forwarding on Busy (on an Internal Phone)

97 Group number ## 67

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Configuring Call Forwarding on No Reply (on an Internal Phone)

97 Group number # * 61 * Subscriber/group number

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The 1st number can be omitted if the phone is logged in to the group as outgoing.

Subscriber/group number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required Authorisation and membership of the group required

Enabling Call Forwarding on No Reply without Re-entering the Number (on an Internal Phone)

97 Group number # * 61

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Disabling Call Forwarding on No Reply (on an Internal Phone)

97 Group number ## 61

Group number	Internal number of the group (for exchange line request type Direct exchange line phone: with preceding **) The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Call Forwarding for Subscribers

Configuring Unconditional Call Forwarding (on an Internal Phone)

*** 21 * Destination number #**

Destination number	Internal or external phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required

Enabling Unconditional Call Forwarding without Re-entering the Number (on an Internal Phone)

*** 21 #**

Disabling Unconditional Call Forwarding (on an Internal Phone)

 # 21 #

Configuring Call Forwarding on Busy (on an Internal Phone)

*** 67 * Destination number #**

Destination number	Internal or external phone to be called Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or

	External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required

Enabling Call Forwarding on Busy without Re-entering the Number (on an Internal Phone)

*** 67 #**

Disabling Call Forwarding on Busy (on an Internal Phone)

 # 67 #

Configuring Call Forwarding on No Reply (on an Internal Phone)

*** 61 * Destination number #**

Number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required

Enabling Call Forwarding on No Reply without Re-entering the Number (on an Internal Phone)

*** 61 #**

Disabling Call Forwarding on No Reply (on an Internal Phone)

 # 61 #

Call Restrictor (Incoming)

Enabling the Call Restrictor (Incoming) for Subscribers (on an Internal Phone)

8 PIN * 23 1

PIN	User PIN

Disabling the Call Restrictor (Incoming) for Subscribers (on an Internal Phone)

8 PIN * 23 0

PIN	User PIN

Announcement before Answering

Recording an Announcement (on an Internal Phone)

8 PIN * 56 AA * *

PIN	Admin or sub-admin PIN
AA	01...10 (no. 1-10 of the announcment)
Zusatzinformationen	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #. Afterwards the recording is played back for checking.

Deleting an Announcement (on an Internal Phone)

8 PIN * 55 AA

PIN	Admin or sub-admin PIN
AA	01...10 (no. 1-10 of the announcment)

Listening to an Announcement (on an Internal Phone)

8 PIN * 57 AA

PIN	Admin or sub-admin PIN
AA	01...10 (no. 1-10 of the announcment)

Door Functions

Initiating a Door Call (on an Internal Phone)

Door number

Door number	Internal number of the door (for exchange line request type Direct exchange line phone: with preceding **)

Opening a Door (on an Internal Phone)

R

R	**Query key** or **Flash key**
Additional information	Authorisation required

Initiating a Query Call from a Door Call (on an Analogue Internal Phone)

R Door number

R	**Query key** or **Flash key**
Door number	Internal number of the group or subscriber (for exchange line request type Direct exchange line phone: with preceding **) or External number (for exchange line request type Internal phone: with preceding exchange line access number) or

	Short-code dialling number (for exchange line request type Direct exchange line phone: with preceding **)

Voice Mailbox

Disabling the Readiness (on an Internal Phone If the Internal Phone Is the Owner of the Voice Mailbox)

8 * 28 * 0

Enabling the Readiness (on an Internal Phone If the Internal Phone Is the Owner of the Voice Mailbox)

8 * 28 * 1

Enabling the Time Control for Readiness (on an Internal Phone If the Internal Phone Is the Owner of the Voice Mailbox)

8 * 28 * 9

Recording an Announcement for a Voice Mailbox with a Phone (on an Internal Phone)

8 PIN * 566 A Voice mailbox number * *

PIN	Admin, sub-admin or user PIN
A	1...9, 0 (no. 1-10 of the announcement)
Voice mailbox number	Internal number of the voice mailbox (for exchange line request type Direct exchange line phone: with preceding **)
Additional information	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #. Afterwards the recording is played back for checking.

Listening to the Announcement for a Voice Mailbox (on an Internal Phone)

8 PIN * 576 A Voice mailbox number

PIN	Admin, sub-admin or user PIN
A	1...9, 0 (no. 1-10 of the announcement)
Voice mailbox number	Internal number of the voice mailbox (for exchange line request type Direct exchange line phone: with preceding **)

Deleting the Announcement for a Voice Mailbox (on an Internal Phone)

8 PIN * 556 A Voice mailbox number

PIN	Admin, sub-admin or user PIN
A	1...9, 0 (no. 1-10 of the announcement)
Voice mailbox number	Internal number of the voice mailbox (for exchange line request type Direct exchange line phone: with preceding **)

Querying a Voice Mailbox (on an Internal Phone If the Internal Phone Is the Owner of the Voice Mailbox)

Voice mailbox number F

Voice mailbox number	Internal number of the voice mailbox (for exchange line request type Direct exchange line phone: with preceding **)
F	0 (Opens the Help menu.) 1 (Opens the main menu.)

2 (Plays back new messages.)

3 (Plays back all of the messages.)

4 Jumps back to the previous message.)

5 (Repeats the message currently being played back.)

6 (Jumps to the next message.)

7 (Deletes the message currently being played back.)

8 (Ends playback.)

* 10 (Disables the voice mailbox.)

* 11 (Enables the voice mailbox.)

* 20 (Enables recording and changing announcements.)

Note: The default announcement and your own announcements, if any have been recorded, will be played back. The following functions are available while your own announcements are being played back:

Note: # (Selects the currently heard announcement as active announcement.)

Note: 4 (Jumps to the previous announcement.)

Note: 5 (Repeats the announcement you have just heard.)

Note: 6 (Jumps to the next announcement.)

Note: 7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)

F	* 30 (Disables message forwarding.) * 31 (Enables message forwarding.) * 70 (Deletes all messages and voice memos.) * 99 (Records a voice memo.) Hang-up (Ends remote access.) **Additional function while a message is being replayed:** 9 (Announces the phone number of the caller – provided it has been transferred.) **Further options after announcement of the phone number:** # (Initiates a callback.) * (Back to the message playback.)
Additional information	After having dialled the internal number, wait until the announcement has ended

Querying a Voice Mailbox (on an Internal Phone If the Internal Phone Is not the Owner of the Voice Mailbox and Readiness of the Voice Mailbox Has Been Enabled)

Voice mailbox number * PIN # F

Voice maibox number	Internal number of the voice mailbox (for exchange line request type Direct exchange line phone: with preceding **)
PIN	User or remote access PIN

F	0 (Opens the Help menu.) 1 (Opens the main menu.) 2 (Plays back new messages.) 3 (Plays back all of the messages.) 4 Jumps back to the previous message.) 5 (Repeats the message currently being played back.) 6 (Jumps to the next message.) 7 (Deletes the message currently being played back.) 8 (Ends playback.) * 10 (Disables the voice mailbox.) * 11 (Enables the voice mailbox.) * 20 (Enables recording and changing announcements.)

Note: The default announcement and your own announcements, if any have been recorded, will be played back. The following functions are available while your own announcements are being played back:

Note: # (Selects the currently heard announcement as active announcement.)

Note: 4 (Jumps to the previous announcement.)

Note: 5 (Repeats the announcement you have just heard.)

Note: 6 (Jumps to the next announcement.)

Note: 7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)

F	* 30 (Disables message forwarding.) * 31 (Enables message forwarding.) * 70 (Deletes all messages and voice memos.) * 99 (Records a voice memo.) Hang-up (Ends remote access.) **Additional function while a message is being replayed:** 9 (Announces the phone number of the caller – provided it has been transferred.) **Further options after announcement of the phone number:** # (Initiates a callback.) * (Back to the message playback.)
Additional information	After having dialled the internal number, press * while the announcement is being played, After having pressed #, wait until the announcement has ended

Remote Access (on an External Phone If Readiness of the Voice Mailbox Is Enabled)

Voice mailbox number * PIN # F

Voice mailbox number	External number of the voice mailbox or External number of the phone which is owner/user of the voice mailbox

PIN	User or remote access PIN
F	0 (Opens the Help menu.) 1 (Opens the main menu.) 2 (Plays back new messages.) 3 (Plays back all of the messages.) 4 Jumps back to the previous message.) 5 (Repeats the message currently being played back.) 6 (Jumps to the next message.) 7 (Deletes the message currently being played back.) 8 (Ends playback.) * 10 (Disables the voice mailbox.) * 11 (Enables the voice mailbox.) * 20 (Enables recording and changing announcements.)

Note: The default announcement and your own announcements, if any have been recorded, will be played back. The following functions are available while your own announcements are being played back:

Note: # (Selects the currently heard announcement as active announcement.)

Note: 4 (Jumps to the previous announcement.)

Note: 5 (Repeats the announcement you have just heard.)

Note: 6 (Jumps to the next announcement.)

Note: 7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)

F	* 30 (Disables message forwarding.) * 31 (Enables message forwarding.) * 70 (Deletes all messages and voice memos.) * 99 (Records a voice memo.) Hang-up (Ends remote access.) **Additional function while a message is being replayed:** 9 (Announces the phone number of the caller – provided it has been transferred.) **Further options after announcement of the phone number:** # (Initiates a callback.) * (Back to the message playback.)
Additional information	After having dialled the inter number, press * while the announcement is being played, After having pressed #, wait until the announcement has ended

Remote Access (on an External Phone If Readiness of the Voice Mailbox Is Disabled)

Voice mailbox number PIN # F

Voice mailbox number	External number of the voice mailbox or External number of the phone which is owner/user of the voice mailbox

PIN	User or remote access PIN
F	0 (Opens the Help menu.) 1 (Opens the main menu.) 2 (Plays back new messages.) 3 (Plays back all of the messages.) 4 Jumps back to the previous message.) 5 (Repeats the message currently being played back.) 6 (Jumps to the next message.) 7 (Deletes the message currently being played back.) 8 (Ends playback.) * 10 (Disables the voice mailbox.) * 11 (Enables the voice mailbox.) * 20 (Enables recording and changing announcements.)

Note: The default announcement and your own announcements, if any have been recorded, will be played back. The following functions are available while your own announcements are being played back:

Note: # (Selects the currently heard announcement as active announcement.)

Note: 4 (Jumps to the previous announcement.)

Note: 5 (Repeats the announcement you have just heard.)

Note: 6 (Jumps to the next announcement.)

Note: 7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)

F	* 30 (Disables message forwarding.) * 31 (Enables message forwarding.) * 70 (Deletes all messages and voice memos.) * 99 (Records a voice memo.) Hang-up (Ends remote access.) **Additional function while a message is being replayed:** 9 (Announces the phone number of the caller – provided it has been transferred.) **Further options after announcement of the phone number:** # (Initiates a callback.) * (Back to the message playback.)
Additional information	After having dialled the inter number, press * while the announcement is being played, The voice mailbox accepts the call after up to 55 seconds. After having pressed #, wait until the announcement has ended

Music on Hold and Announcement

Deleting Music on Hold (on an Internal Phone)

##8 PIN * 5532

PIN	Admin or sub-admin PIN

Recording an Announcement (on an Internal Phone)

##8 PIN * 5631 * *

PIN	Admin or sub-admin PIN
Additional information	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #.

Deleting the Announcement (on an Internal Phone)

##8 PIN * 5531

PIN	Admin or sub-admin PIN

Switching Music on Hold and Announcement on/off and Listen to It (on an Internal Phone)

##8 PIN * 573 Option

PIN	Admin or sub-admin PIN
Option	0 (no music on hold/announcement)

1 (announcement only)

2 (internal music on hold only)

3 (internal music on hold and announcement)

4 (external music on hold only)

5 (external music on hold and announcement)

Wake-up Functions

Entering the wake-up Time and Activating One-time wake-up (on an Internal Phone)

8 PIN * 31 hh mm w

PIN	User PIN
hh	00...23 (hour)
mm	00...59 (minute)
w	1...7 (single weekday: Mon...Sun) 8 (weekend: Sat and Sun) 9 (working days: Mon to Fri) 0 (whole week: Mon to Sun)

Entering the wake-up Time and Activating Repetitive wake-up (on an Internal Phone)

8 PIN * 32 hh mm w

PIN	User PIN
hh	00...23 (hour)
mm	00...59 (minute)
w	1...7 (single weekday: Mon...Sun) 8 (weekend: Sat and Sun) 9 (working days: Mon to Fri)

	0 (whole week: Mon to Sun)

Deleting All Configured wake-up Times for the Phone (on an Internal Phone)

8 PIN * 33

PIN	User PIN

Deleting the wake-up Times of All Subscribers (on an Internal Phone)

8 PIN * 34

PIN	Admin or sub-admin PIN

Recording an Announcement (on an Internal Phone)

8 PIN * 562 A * *

PIN	Admin or sub-admin PIN
A	1...3 (no. 1-3 of the announcement)
Additional information	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #. Afterwards the recording is played back for checking.

Deleting an Announcement (on an Internal Phone)

8 PIN * 552 A

PIN	Admin or sub-admin PIN

A	1...3 (no. 1-3 of the announcement)

Listening to an Announcement (on an Internal Phone)

##8 PIN * 572 A

PIN	Admin or sub-admin PIN
A	1...3 (no. 1-3 of the announcement)

Configuration Wizard

Basic Knowledge
Language
Network Configuration
Dealer and operator data
Port configuration
Basic Settings
Public switched tel. networks
Subscriber
Call distribution
Data backup
Display and printout
End

Basic Knowledge

The configuration wizard runs through all pages of the Configuration Manager that are required for quick configuration of the PBX.

Symbols of the toolbars and navigation bars are identical to those of the Configuration Manager. New symbols or different symbols are listed below.

Entered data is adopted as soon as another page is opened.

Note: Changes in the network configuration are saved but not activated until you leave the configuration wizrad via **End** or .

Opening/closing the Configuration Wizard

Requirements:

- Logged in to the PBX as an administrator

Opens or closes the configuration wizard.

Navigating in the Configuration Wizard

Opens the next or previous page.

Note: Port configuration to **End** and **Continue** are only available if under **Dealer and operator data** an admin user password, admin PIN, admin password and an activation code was entered.

Delete

Via this symbol, marked entries are deleted.

Language

Proceed as follows:

- Using the configuration wizard (**> Language**)

Language selection > **Continue**.

Network Configuration

Ethernet Configuration
DNS Configuration

Ethernet Configuration

Proceed as follows:

- Using the configuration wizard (**> Network configuration**)

Receive the IP address automatically (DHCP client)

Note: In the factory settings, DHCP is enabled.

The PBX receives its IP address automatically from the DHCP server.

Assigning IP address manually

Requirements:

- Disabled DHCP client

IP address

IPv4 address in the range 0-255 (example: 192.168.0.240).

Apply immediately

The IPv4 address is immediately applied and the PBX is routed to the new IPv4 address. Renewed registration required.

Apply at restart

The IPv4 address is not applied until the next time the PBX is restarted.

Subnet mask

IPv4 address of the subnet mask in the range 0-255 (example: 255.255.255.0).

Gateway

IPv4 address of the gateway in the range 0-255 (example: 192.168.0.2).

DNS Configuration

Proceed as follows:

- Using the configuration wizard (**> Network configuration**)

Permanent IP address from the APIPA address range

The PBX can be reached via the IP address 169.254.1.240, if

- it is directly connected to an individual computer which automatically receives its IP address.
- it is connected to a router that automatically assigns IP addresses.

Use own DNS server

Requirements:

- DHCP enabled

Despite the enabled DHCP client, the DNS server set here is used.

Note: When switched off, the PBX receives the DNS server address automatically from a DHCP server.

Dealer and operator data

Administrator Access
PBX Data

Administrator Access

Proceed as follows:

- Using the configuration wizard (**> Dealer and operator data**)

Administrator access > Change

Requirements:

- Knowledge of the current admin password

The user data for the administrator access are used for administrator registration on the PBX.

Note: Changes at the administrator access are only possible with an encrypted HTTPS connection.

Note: After a change, a new regisration is required.

Admin user name

Consists of up to 32 characters containing digits, letters (upper and lower case, but no umlauts and ß).

Change admin PIN

Enables the input option (6 characters (digits)) for **Admin PIN** and **Repeat admin PIN** (e. g. for access to COMfortel Set).

Note: All PINs in the PBX are unambiguous. Therefore, you cannot assign the same PIN twice.

Note: Avoid using PINs that can easily be found out (e. g. birthdays, specific dates, 123456, etc.)

Change admin password

Enables the input option **Admin password** and **Repeat admin password** (8-32 characters containing digits, letters (upper and lower case, but no umlauts and ß), special characters **- _ . ! ~ * ' ( ) & = + $,**).

Note: A coloured bar under the entry field shows the security level of the password (red = low level, green = high level).

Note: Passwords with a low security level cannot be saved.

Caution: A high level does not necessarily indicate a secure password as this can also be reached by entering names and birth dates in connection with special characters. However, names and birth dates should not be used as passwords.

- To create a secure password you can consult, for example, the guidelines of the German Federal Office for Information Security (BSI, in German).

Please confirm your changes with the current admin password

Protection against unauthorised changes.

PBX Data

Proceed as follows:

- Using the configuration wizard (**> Dealer and operator data**)

PBX name

For identification of the PBX. It is displayed in the status line and the PBX data overview (click on the logo).

PBX description

This is used to enter any additional PBX information that may be required.

Buy activation/voucher code Open shop

Button to open the online shop.

Buy activation/voucher code

Entry field for an unblocking code for system activation.

Important: The steps **Port configuration** to **End** of the configuration wizard and the button **Continue** are only available if an unblocking code/voucher code has been entered.

PBX is used in the following country

When you configure the country the PBX is automatically assigned the default settings that are required in the selected country (e.g. settings for analogue ports, and emergency numbers).

Note: Emergency numbers preset in this way cannot be deleted. You can only preset different public emergency numbers by subsequently changing the country settings.

Note: Make sure that there are no conflicts between country-specific emergency numbers and your internal phone numbering plan. Assign new internal phone numbers, if necessary.

Port configuration

Proceed as follows:

- Using the configuration wizard (**> Port configuration**)

Connection type

The connection type for the ISDN connection that the network provider has provided for the NT needs to be configured for the external S_0 port.

Note: No selection of the connection type is possible for internal S_0 ports. They are fixed on PTMP connection.

PTMP connection

ISDN PTMP connection (Point-to-Multipoint) with up to 10 multiple subscriber numbers (MSNs).

PTP connection

ISDN PTP connection (Point-to-Point) with a PBX base number and Direct Dial In numbers (DDI numbers).

Basic Settings

Exchange Line Request/Exchange Line Access Number Prefixes

Exchange Line Request/Exchange Line Access Number

Proceed as follows:

- Using the configuration wizard (**> Basic settings**)

System-wide Automatic Exchange Line Request

Requirements:

- Emergency call only with disabled exchange line access number
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > Emergency calls**

The PBX automatically distinguishes for each subscriber between internal and external numbers by analysing the digit sequence and the adjustment of the dialled phone number with the internal numbering plan.

Note: When switched off, the exchange line request type can be set individually for each subscriber.

Exchange Line Access Number

Requirements:

- Configured prefixes (country prefix and local area code)
- Automatic exchange line request disabled

Is required to request an external line (exception: direct exchange access and automatic exchange line request).

Note: An available external line is indicated by the external dial tone after the exchange line access number has been dialled. The external number can then be dialled.

Note: The factory setting for the exchange line access number on the PBX is 0. This number is not effective at first because automatic exchange line request is active in the factory settings.

Note: Changing the exchange line access number is only meaningful if another exchange line access number is normally used (e.g., in England).

Note: Note that changing the exchange line access number restricts the phone numbering plan.

Prefixes

Proceed as follows:

- Using the configuration wizard (**> Basic settings**)

PBX is used in the following country

When you configure the country the PBX is automatically assigned the default settings that are required in the selected country (e.g. settings for analogue ports, and emergency numbers).

Note: Emergency numbers preset in this way cannot be deleted. You can only preset different public emergency numbers by subsequently changing the country settings.

Note: Make sure that there are no conflicts between country-specific emergency numbers and your internal phone numbering plan. Assign new internal phone numbers, if necessary.

Country Code

International phone prefix (e. g. 0044 for the UK). These are necessary for some functions of the PBX (e.g. Call Through, Call restrictors/Call deblockers).

Area Code

Phone number prefix (e. g. 0208 for Greater London). These are necessary for some functions of the PBX (e.g. Call Through, Call restrictors/Call deblockers).

Note: Some countries (e.g. Denmark, Greece, Luxembourg) do not have a fixed number sequence for their area codes. If this also applies to the location at which the PBX is installed, you do not need to enter an area code.

Dial level

The selection of the dial level (Location, National, International) is country-specific. Depending on the country and the settings for the local area code above, a dial level must be selected (example: In Belgium, local area codes do not exist. Here, the dial level **City** must not be selected).

Public switched tel. networks

Proceed as follows:

- Using the configuration wizard (**> Public switched tel. networks**)

Name of the PTMP connection

For identification of the connection within the PBX.

Multiple subscriber number (MSN)

The PTMP connection has up to 10 multiple subscriber numbers. The network provider assigns the phone numbers for the connection. (Select + **New** to enter.)

Name

For display purposes on the system phone.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

PBX base number

A PTP connection has a PBX base number (base number, header, base phone number) and (Direct Dial In - DDI) extension numbers. The network provider assigns the phone numbers for the connection. The

phone number to be dialled is formed from the PBX base number + (DDI) extension number.

Name

For identification of the connection within the PBX.

Extensions block (DDI numbers) | From | To

A PTP connection has a PBX base number (base number, header, base phone number) and (Direct Dial In - DDI) extension numbers. The network provider assigns the phone numbers for the connection. The phone number to be dialled is formed from the PBX base number + (DDI) extension number.

Note: you can enter a zero both as a single number (e.g. 0 to 99, as is usually the case) and as a digit in the set number of places in extension numbers (e.g. 00 to 99).

Note: If linear call distribution for the PTP connection is enabled (**Public switched tel. networks > Call distribution** page), and if the internal numbers match the external extension numbers, the system automatically performs a 1:1 assignment for the extension numbers in the extensions block. If an extension cannot be assigned, the fallback call distribution function is enabled.

Note: If you want an extension number to be different from the ones used in linear call distribution, or if an extension number is needed for a different function (e.g. remote configuration, CLIP no screening), you must also enter it in the phone number list (page **Public switched tel. networks > ISDN > Extension numbers PTP connection**).

Note: For example, for extensions block 0 to 99: enter extension 0 as an extension and assign it in the call distribution.

Note: For example, for extensions block 00 to 99: enter the individual extension numbers 00 to 09 as an extension number and assign them during call distribution. The PBX will then only react when the number of places is reached, for example, if the caller dials 00 as an extension number. In contrast, if the caller only dials 0, the fallback call distribution function is enabled as soon as the time for selecting a suffix dialling digit has passed.

Note: You must enter all the extension numbers for a PTP connection that is not running linear call distribution.

Name of the analogue connection

For identification of the port within the PBX.

Phone number

The analogue connection has a phone number. The phone number is assigned by the provider for the connection.

Name

Freely selectable name. This name is used to identify the phone number within the PBX.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

Configure

Opens the configuration of the VoIP provider.

Further help under Settings of the VoIP Provider

Provider name

VoIP provider with whom the account was set up.

Account name

Freely selectable name. This name is used to identify the account within the PBX.

Connection type

Connection type to be used with the account.

PTMP connection

VoIP account with multiple subscriber numbers (MSNs)

PTP connection

VoIP account with a PBX base number and extension numbers (DDI numbers)

+ New

Adds a new account to the account list. You must then make the settings for the VoIP account.

Import

IMPORT PROVIDER

Requirements:

- PC or PBX with existing connection to the Internet

A file with individual provider access data (*.XML) will be downloaded from the update server. Depending on the Internet connection, this file is then either saved to a data storage device (PC) or directly to the PBX.

Note: The file saved to a (PC) data storage device can then be imported.

Note: The access data provided for the download is for information purposes only, and is not mandatory.

Country code

International phone prefix (e. g. 0044 for the UK) for which the account's VoIP provider has provided the phone numbers.

Note: Land line connection prefixes are configured separately (**Public switched tel. networks > Exchange line request**).

Area code

Phone number prefix (e. g. 0208 for Greater London) for which the account's VoIP provider has provided the phone numbers.

Note: If there is no fixed digit sequence as the area code at the installation location of the PBX, no entry is required here.

Country

Country for which the account's VoIP provider has provided the phone numbers.

Exchange line access number (account number)

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This is then used for the specific VoIP access point.

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

User name

User name assigned by the VoIP provider when the account was set up. It is used to identify the user to the VoIP provider.

Note: Note that some providers allocate both login information for the web page and separate login information for using the Voice over IP service (e.g. a dial-in password). Find out what information is needed for the dial-in and enter it accordingly.

Note: The user name is called different things by different providers. For example, the terms Username, Authorization User, SIP code/SIP password and SIP-ID are all used to describe a user name.

Note: If your provider (in Germany) is T-Online, you must also enter the VoIP phone number you received from them under **User name** and **Display name**.

Password

Password assigned by the VoIP provider when the account was set up. It is used to verify the user name.

Note: To change an existing password, click on **Show passwords**.

Note: Note that some providers allocate both login information for the web page and separate login information for using the Voice over IP service (e.g. a dial-in password). Find out what information is needed for the dial-in and enter it accordingly.

Note: The password is called different things by different providers. For example, it can also be called a customer password.

Caution: Visible passwords are a security risk.

- After you enter your password, click on **Hide passwords**.

Authorisation ID

Authorisation ID (authentication ID) assigned by the VoIP provider for the VoIP account.

Note: To change an existing password, click on **Show passwords**.

Note: You can only input an entry in this field if the VoIP provider has assigned an authorisation ID.

Caution: Visible passwords are a security risk.

- After you enter your password, click on **Hide passwords**.

Account has emergency call ability

Requirements:

- VoIP provider has authorised and switched on emergency call ability.

This account allows emergency calls to be made.

Note: When switched off, this account cannot be used for emergency calls. When you attempt to dial an emergency number, you will hear the announcement, "This phone does not allow emergency calls. Please use an alternative."

Note: The phone allows emergency calls by default. Check you can actually use your account to make emergency calls. Not all network providers allow emergency calls.

New

Adds a new account to the account list. You must then make the settings for the VoIP account.

Edit passwords

The password is visible and can be changed.

Subscriber

Proceed as follows:

- Using the configuration wizard (**> Subscriber**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by another internal subscriber.

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Note: The internal number entered here needs to be entered in the ISDN device or in the system phone as the (first) MSN. Also on an ISDN device, make sure that the phone number is transmitted to the PBX.

Name

Freely selectable name. This name is used to identify the subscriber within the PBX and the connected system phones. The name can also be transferred to the VoIP end devices via LDAP server.

Module | Port

Module and port to which the end device (e. g. phone) is connected.

Device type

Device type depending on the end device connected.

Analogue end devices	
Phone	analogue phone, analogue modem
Answering machine	analogue answering machine
Fax machine	analogue fax machine
Door terminal	analogue door terminal, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of a door terminal
Announcement output	analogue announcement output, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of an announcement output

ISDN end devices	
ISDN system phone	System phone 1200/1400/1600/2600
ISDN phone	ISDN phone in compliance with the Euro ISDN Standard (DSS1)

ISDN end devices	
ISDN PC controller	ISDN PC controller in compliance with the Euro ISDN Standard (DSS1)
Fax machine	ISDN fax machine in compliance with the Euro ISDN Standard (DSS1)

VoIP end devices	
COMfortel IP	COMfortel 1400 IP, COMfortel 2600 IP, COMfortel 3200, COMfortel 3500, COMfortel 3600 IP
COMfortel 1200 IP	COMfortel 1200 IP
Standard VoIP phone	Standard VoIP end device (SIP) from other manufacturers, soft phone (SIP)
Handset on COMfortel WS-IP	Handset (e.g., COMfortel M-100/200/210/300/310) on a COMfortel WS-400 IP or COMfortel WS-650 IP
Handset on COMfortel WS-500	Handset COMfortel M-510/520/530 on a COMfortel WS-500S or WS-500M
COMfortel C-400	Conference phone COMfortel C-400
Door terminal	VoIP door terminal, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of a door terminal

+ New

Adds a new subscriber. You must then make the settings for this new subscriber.

Further help under Add Subscriber

Call distribution

Proceed as follows:

- Using the configuration wizard (**> Call distibution**)

...ISDN PTMP connections

...VoIP PTMP connections

Internal no. | name

Fallback

The caller is transferred to the fallback function.

Internal number: The caller will be transferred to the selected internal number (subscriber, group, voice mail/fax box or automatic reception).

Note: External numbers, e.g., those used as remote switching/ programming numbers or call through numbers, may not be used in the call distribution.

...ISDN PTP connections / ...VoIP PTP connections

Internal no. | name

For a PTP connection without linear call distribution or for extensions differing from linear call distribution, the internal destinations must be selected.

Linear/fallback

The caller will be transferred to the internal number similar to the DDI. If this phone number is not configured in the PBX, the fallback function will be used.

Rejecting a call

The caller will hear the busy signal.

Internal phone number: The caller will be transferred to the selected internal number (subscriber, group, voice mail/fax box or automatic reception).

Note: External numbers, e.g., those used as remote switching/ programming numbers or call through numbers, may not be used in the call distribution.

...Analogue

Internal no. | name

Internal number: The caller will be transferred to the selected internal number (subscriber, group, voice mail/fax box or automatic reception).

Data backup

Read Configuration Data from the PBX

Read Configuration Data from the PBX

Proceed as follows:

- Using the configuration wizard (**> Data backup**)

Store in the PBX

Important: Saving an old configuration overwrites the current configuration data.

Note: A configuration file that was created with an older firmware version, may possibly not contain all of the configuration data (for example, data for new functions that were added by a firmware update).

Select and import a configuration file (*.cfg) that is stored on a (PC) data storage device. The following data packages can be selected for import:

- **Apply configuration data** (cannot be edited)
- **Save network settings**
- **Apply PBX data**
- **Transfer music on hold and announcements**
- **Contact images**
- **COMfortel IP Editor**

Read from PBX

Important: Note the following legal notes before saving the data backup.

The data backup includes, amongst others, the following legally especially protected components:

- Configuration of the PBX
- Names of internal subscribers
- Passwords of PBX, subscribers, VoIP accounts and VoIP subscribers
- Address books of the PBX
- Connection data sets with destination, source, and time information of a telecommunication connection

Note: Before the data backup is being saved, all subscribers to the PBX must be informed that these contents will be forwarded. They must also be informed about who will use the data received.

Note: It is absolutely necessary to store the data safely and in accordance with the legal requirements.

Note: If you want to make the data backup available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

The configuration is saved to the (PC) data storage device as configuration file (*.cfg) The following data packages can be selected for export:

Display and printout

Proceed as follows:

- Using the configuration wizard (**> Display and printout**)

Print

All entries are printed.

End

Proceed as follows:

- Using the configuration wizard (**> End**)

Leaving the configuration wizard or alternatively via .

Note: When leaving via , changes on the current page are not adopted.

Using the Configuration Manager

Introduction
Overviews
Monitoring
Hardware
Public switched tel. networks
Subscriber (scr.)
Groups
Devices
Time control
Routing
Functions
User data
Administration

Introduction

Requirements and Notes
Opening the Configuration Manager
Operating Instructions for the Configuration Manager
Information on the PBX

Requirements and Notes

The Configuration Manager of the PBX is opened with a browser and allows the subdivision into three permission levels:

- Administrator (admin)
- Sub-administrator (sub-admin)
- User

Requirements:

- Recommended browsers: Mozilla Firefox, Google Chrome (current version)
- Knowledge of the IP address on the PBX
- Configured password

Note: Every login is logged (page **Monitoring > System messages**). Additionally this information can be sent as a system e-mail (page **Administration > System messages**).

Note: After the third failed login attempt, for a minute it is not possible to log in. This blocking period will be increased after another three failed login attempts.

Note: Since the administrator may block parts of the PBX, you may not be able to configure all of the settings in the Configuration Manager if you are logged in as a user or a sub-administrator (see Configuring Access Authorisation)

Opening the Configuration Manager

Open the Configuration Manager as an Administrator
Opening the Configuration Manager as Sub-Administrator
Opening the Configuration Manager as User

Open the Configuration Manager as an Administrator

1. Enter the IP address of the PBX (for example, **192.168.0.240**).
2. Enter the admin user name and the corresponding password.

Opening the Configuration Manager as Sub-Administrator

1. Enter the IP address of the PBX (for example, **192.168.0.240**).
2. Enter the user name **sub-admin** and the corresponding password.

Note: If the sub-administrator enters his own internal number as his user name, this login is recognised as a user login and the web interface is displayed for the user.

Opening the Configuration Manager as User

1. Enter the IP address of the PBX (for example, **192.168.0.240**).
2. Enter the internal number of the user (as user name) and the corresponding password.

Operating Instructions for the Configuration Manager

Configuration Manager Navigation
Logging Out
Entry and Selection Fields
Creating, Deleting, Searching for, and Sorting Table Entries

Configuration Manager Navigation

Horizontal navigation bar

From here, you reach the pages **Overviews** and **Monitoring**.

The **Search** in the horizontal navigation bar searches for terms in the **Navigation** area. The term is searched in the tree and opened at the corresponding location.

Area navigation

The available areas are displayed in a tree under **Navigation**. Most of the themes have been summarised under a main theme ( ) and are listed in the order in which to work. If you skip a theme, it may happen that the next page is not available (e.g. because no subscribers with internal numbers have been configured yet).

Area help topics

Below **Navigation** is the **Help topics** area that contains links to different help topics. It offers help topics related to the opened configuration page.

Logging Out

Via the **Logout** link in the upper area of the Configuration Manager you go back directly to the login page.

Entry and Selection Fields

Important: All settings configured on pages with the following fields, must saved before leaving the page.

	An entry field normally indicates that a name or number must be entered.
deutsch	Selection fields offer settings that are possible for this function.
	Some functions offer radio buttons for adjustments. The activation of an adjustement causes the simultaneous deactivation of all other selection options.
	A check box indicates the enabling or disabling of a function.
	Slider for making a setting. Sometimes, the exact values are displayed in front or behind the slider.

Creating, Deleting, Searching for, and Sorting Table Entries

Creating Entries

1. Click + **New**.
2. Fill in the empty entry fields.
3. Click **Save**.
4. Repeat steps 1 to 3 to create additional entries.

Note: If you want to create multiple entries simultaneously, enter the required range in the **From** and **To** entry fields, and then click **Save**.

Note: In extendible lists the lower toolbar displays information about the actual and maximum possible number of entries in the table. Dongle functions (identified by) have three numbers. In this case, the second number is the active number and the third is the maximum number that can be enabled via the dongle.

Deleting entries

1. Mark the entry and click **Delete**.

Searching for entries

1. Click in the open field in the header of the column that you want to search, such as **Number** or **Name** and enter the search term.

Sorting entries

- Click a column header.

▲	Sorted in ascending order.
▼	Sorted in descending order.

Note: If no arrow appears then the column cannot be used as a sorting criteria.

Information on the PBX

Click the logo in the Configuration Manager for opening the information about the PBX (e. g. firmware version, serial number, dealer and operator data).

Note: The dealer and operator data can be edited under **Administration > Dealer and operator data**.

Print

Prints the information about the PBX.

Overviews

Identification Number Table
Call distribution
Call Forwarding Status
Groups
Additional Subscriber Passwords
Automatic Receptions
Voice Mail/Fax Function
Wake-up Times
Switching times
Ports
Dial plan

Identification Number Table

Proceed as follows:

- Using the Configuration Manager (**Overviews > Identification number table**)

Displays all assigned identification numbers sorted by number, name or type.

Print

Prints a list of all assigned identification numbers.

Call distribution

Proceed as follows:

- Using the Configuration Manager (**Overviews > Call distribution**)

Displays on which internal subscribers a call on a separate external number is signalled.

Print

Prints a list of the configured call distributions.

Call Forwarding Status

Proceed as follows:

- Using the Configuration Manager (**Overviews > Call Forwarding Status**)

Displays the configuration-dependent status of call forwardings for all configured subscribers and groups.

- **Phone number | name**: Displays all available subscribers and groups.
- **Destination number**: Displays the configuration-dependent phone numbers of the configured (mutli-path) call forwardings.

Note: An activated (multi-path) call forwarding is highlighted in green.

Print

Prints a list of the configured call forwardings.

Groups

Proceed as follows:

- Using the Configuration Manager (**Overviews > Groups**)

Displays all created groups with corresponding phone numbers and names as well as the subscriber status.

Print

Prints a list of all configured groups.

Additional Subscriber Passwords

Proceed as follows:

- Using the Configuration Manager (**Overviews > Additional subscriber passwords**)

Displays all configured additional subscriber passwords for the Configuration Manager, API and Action URLs.

Print

Prints a list of the configured additional subscriber passwords.

Automatic Receptions

Proceed as follows:

- Using the Configuration Manager (**Overviews > Automatic Receptions**)

Displays all configured automatic receptions and the corresponding destination numbers.

Print

Prints a list of all configures automatic receptions.

Voice Mail/Fax Function

Proceed as follows:

- Using the Configuration Manager (**Overviews > Voice mail/fax function**)

Displays all created voice mail and fax boxes.

- **Type**: Voice mail or fax box
- **Phone number/Name**: Number and name of the voice mail/fax box
- **Number of messages**: Number of the stored and the maximum possible messages.
- **Memory utilisation**: Value of the actual occupied and maximum possble storage space.

Note: A **Voice mailbox** is here marked with *, if this was set for the function **Call recording**. This function is available only in certain countries.

Print

Prints a list of the configured voice mail and fax boxes.

Graphic

Displays the memory assignment in a diagram.

Wake-up Times

Proceed as follows:

- Using the Configuration Manager (**Overviews > Wake-up times**)

Displays all configured wake-up times sorted by weekdays.

Print

Prints a list of the configured wake-up times.

Switching times

Proceed as follows:

- Using the Configuration Manager (**Overviews > Switching times**)

Displays the time periods the configurations are valid (12 a.m.-11:59 p.m.) as well as manually enabled configurations.

Ports

Proceed as follows:

- Using the Configuration Manager (**Overviews > Ports**)

Displays all configured and fixed ports of the PBX and the used transport protocols.

Print

Prints a list of the configured and fixed ports.

Dial plan

Proceed as follows:

- Using the Configuration Manager (**Overviews** > **Dial plan** page)

The phone numbering plan shows all of the internal numbers already created and any additional numbers that have been taken from the internal numbering plan:

- Subscriber number
- Group numbers
- Voice mailbox numbers
- Fax box numbers
- Phone numbers for door terminals and audio outputs
- Short-code dialling numbers
- Emergency and service numbers
- Exchange line access numbers (account numbers) for VoIP accounts
- Exchange line access number of the PBX
- Automatic receptions
- Call on hold
- Internal CAPI dial-in number
- Macro short-code number

The internal numbering plan of the PBX permits the allocation of 2 to 4-digit phone numbers in the 10-9999 range. Double allocation of a number is not possible.

Using numbers with few digits (two or three-digit numbers) inevitably restricts the possible supply of numbers requiring more digits. For

example, if the phone number 12 is assigned, the phone numbers 120-129 and 1200-1299 are no longer available.

Note: For some functions a 3-digit maximum phone numbers can be assigned because the last position is needed for the function (e.g. door terminal numbers, phone numbers for call parking).

Note: The exchange line access number on the PBX is 0 in the factory settings; however, this can be changed (**Public switched tel. networks** > **Exchange line request**). Note that changing the exchange line access number restricts the number plan.

Note: The overview can be viewed and printed (**Overviews** > **Dial plan**) in the Configuration Manager by opening the page **Printing**.

Note: The phone numbers can be shown in the **Total overview** or **Grouped by type** .

Note: In the overview **Grouped by type** on

Monitoring

Network Diagnosis
External Call Channel Assignment
Status VoIP accounts
Status internal subscriber
Status IP relay
System messages
Network data stream
Status Messages

Network Diagnosis

Proceed as follows:

- Using the Configuration Manager (**Monitoring > Network diagnosis**)

When sending a ping to a remote host (e. g. computer, PBX) in the network, its reachability and the routing time is checked.

External Call Channel Assignment

Proceed as follows:

- Using the Configuration Manager (**Monitoring > External call channel assignment**)

Displays the status of the external call channels and if a GSM gateway has been connected to the external port.

Note: When opening the page it first shows the status which is automatically updated every 30 seconds.

	available.
	used.
	not available.
	emergency call ability.
	no emergency call ability.

Status VoIP accounts

Proceed as follows:

- Using the Configuration Manager (**Monitoring > Status VoIP accounts**)

Requirements:

- Configured VoIP account (**Public switched tel. networks > VoIP > Accounts**)

It is displayed:

- **STUN request**
- **Registration**
- **last outgoing call**

Note: When opening the page it first shows the status which is automatically updated every 30 seconds.

The first steps are the DNS request and an evaluation of the DNS server response.

	Registration in progress
	Registration successful
	Error, for example caused by: • DSL modem defective • Faulty network configuration of the PBX

If this is not effective, please check the status of the individual areas.

STUN request

	Flawless STUN request. The firewall permits traffic with the SIP transport protocol.
	No STUN server entered for the VoIP provider.
	Error, for example caused by: • STUN server offline • Faulty STUN server IP address/URL • Provider-specified port not specified • STUN server reports faulty responses • Firewall blocks SIP transport protocol or otherwise does not allow Voice over IP

Registration

	REGISTER request successful
	Faulty STUN request. Current status must be checked again. **Note:** Some providers have STUN disbaled.
	Error, for example caused by: • Registrar offline. Outgoing calls might be possible (STUN request and outbound proxy successful). Incoming calls not possbile (no phone number at the registrar of the provider). • Faulty domian settings • Faulty account data (check user name, password and authentication ID) • VoIP provider blocks multi registrations caused by multiple

Status internal subscriber

Proceed as follows:

- Using the Configuration Manager (**Monitoring > Status internal subscriber**)

Display of registration (IP and ISDN phones), connection (analogue phones) and provisioning of subscribers, door terminals and (not yet) assigned devices by a coloured dot.

Note: The page initially displays the status when the page is opened and will automatically be updated every 30 seconds.

Note: Not all status information can be updated automatically (see following sections).

Status analogue phones	
	Unknown analogue phone.
	Connected analogue phone with date of last activity time.

Note: After connecting an analogue phone, the receiver must be picked up for correctly displaying the phone status. The monitoring page should then be updated.

Note: The PBX cannot automatically recognize the removal of a device. It's hardware status can be reset under **Subscriber > Properties > Basic Settings > Reset hardware status**.

Status ISDN phones	
	Unkown ISDN phone.
	Connected ISDN system phone.
	Connected standard ISDN phone with date of last activity time.

Note: After connecting a standard ISDN phone, the MSN must be configured and then the receiver must either be picked up or the phone must be called for correctly displaying the phone status. The monitoring page should then be updated.

Note: The PBX cannot automatically recognize the removal of a device. It's hardware status can be reset under **Subscriber > Properties > Basic Settings > Reset hardware status**.

Registration VoIP phones	
	No registration.
	Successful encrypted registration of the VoIP subscriber.
	Successful unencrypted registration of the VoIP subscriber.
	Registration error with error message.

Provisioning VoIP phones	
	No provisioning file assigned yet.

Provisioning VoIP phones	
	Provisioning file assigned, however, not yet requested by the device.
	Provisioning file assigned and requested by the device.
	MAC address of the device has been determined.

Note: Provisioning without MAC address is not possible.

System-/standard phone VoIP and ISDN phones	
	The device was registered as system phone on the PBX.

Distributing firmware VoIP system phones	
	Distributing firmware is enabled in the phone, however, currently inactive. Either no firmware file is saved in the phone or the PBX has determinded another phone of the phone family as firmware distributing phone.
	Firmware distributing phone.

Subscribers/Door terminals

Under **Subscribers** and **Door terminals**, all known devices that are assigned to a subscriber or a door terminal are displayed. On the right there is additional information about the individual devices (IP/MAC address, type name, firmware version, status).

Note: Not all information is available for each device.

Note: By double-clicking on the additional information, these can be copied to the clipboard for further processing.

Not assigned

Under **Not assigned** the phones that are known to the PBX but not assigned to a subscriber or a door terminal are displayed.

Requirements for automatic assignment:

- Analogue phones: Slot and port of the device must correspond to the subscriber.
- ISDN phones: MSN, slot and port of the device must correspond to the subscriber
- VoIP phones: The MAC address entered for the subscriber must correspond to the MAC address sent by the device.

If these devices shall no longer be displayed, they can be deleted (page **Subscriber (scr.) > Unassigned phones**).

Status IP relay

Proceed as follows:

- Using the Configuration Manager (**Monitoring > Status IP relay**)

Displays information about the switching status of the IP relays.

Note: When opening the page it first shows the status which is automatically updated every 30 seconds. However, not all status information can be automatically updated.

System messages

Proceed as follows:

- Using the Configuration Manager (**Monitoring > System messages**)

Each event (max. 2000 entries per category) is logged and displayed in chronological order with the date and time, category (**SELECTED FILTER**) and detailled information in English.

Note: The page is not updated automatically.

Note: When the maximum number of entries is exceeded, old entries in the same category are automatically deleted.

Note: If you update your firmware from Version 6.4 to 6.6, old log files are grouped to form a *.tar archive in the SFTP range.

Print

Prints a list of the system events.

Network data stream

To log the network data stream, a network log (*.pcap) can be recorded. This log can be evaluated (e.g.,with Wireshark/Ethereal), for instance, to determine the cause of malfunctions.

Important: Note the following legal notes prior to recording a network data stream.

The recorded network data stream may contain the following components:

- Data streams of other computers in the network
- Connection data sets with destination, source, and time information of a telecommunication connection
- The spoken word of both ends of a telecommunication connection
- Configuration data of the PBX and connected phones, as well as the corresponding PINs and passwords

Important: Before any recording, all participants in the telecommunication connection must be informed that these contents will be recorded and must also be informed who will use the recorded data.

Important: If you want to make the recorded data available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

Proceed as follows:

- Using the Configuration Manager

Requirements:

- Disabled pop-up blocker in the browser or defined exception rule

1. Open the page Monitoring > Network data stream.
2. Accept the legal notes by entering ok in the entry field.
3. Click Start Trace.
4. If the error can be reproduced easily, try to repeat it now.
5. To stop the recording of the network log, click Stop Trace.
6. Save the file.

Status Messages

Proceed as follows:

- Using the Configuration Manager (**Monitoring > Status messages**)

Displays and describes those errors in the PBX that can be corrected via the Configuration Manager or in the hardware.

Note: If an error occurs, the status messages appear directly after you log into the Configuration Manager.

Note: When opening the page it first shows the status which is automatically updated every 30 seconds.

Hardware

Selection of modules
Port configuration

Selection of modules

Settings of the Modules

Settings of the Modules

Proceed as follows:

- Using the Configuration Manager (page **Hardware > Selection of modules**)

Note: The specifications here have to match the actual hard ware configurations of your PBX. If this is the case, the settings will be marked in black. If it is not the case, the settings will be marked in red.

Note: If an inserted module should not be used, select the option **not defined** in the corresponding list field.

Configuration

Select the modules that are inserted in the corresponding slot.

move

Moves a module to another slot. The settings already made for this module will not be lost by this action.

Port configuration

Settings of the Ports

Settings of the Ports

2BRI
2FXO

2BRI

Proceed as follows:

- Using the Configuration Manager (page **Hardware > Port configuration**)

Note: If the settings for the module are marked in red, the module is selected but not inserted.

Note: The numbers under **Port** correspond to the numbers used on the board.

Use

Set the use of the ISDN ports.

free

Port is not used and port is not taken into account for the calculation of the total number of existing ports/call channels.

internal S0

Port is used as internal S_0 port.

internal UP0

Port is used as internal U_{P0} port.

external

Port is used as external port.

Connection type

The connection type for the ISDN connection that the network provider has provided for the NT needs to be configured for the external S_0 port.

Note: No selection of the connection type is possible for internal S_0 ports. They are fixed on PTMP connection.

PTMP connection

ISDN PTMP connection (Point-to-Multipoint) with up to 10 multiple subscriber numbers (MSNs).

PTP connection

ISDN PTP connection (Point-to-Point) with a PBX base number and Direct Dial In numbers (DDI numbers).

2FXO

Proceed as follows:

- Using the Configuration Manager (page **Hardware > Port configuration**)

Note: If the settings for the module are marked in red, the module is selected but not inserted.

Note: The numbers under **Port** correspond to the numbers used on the board.

Requirements:

- Selected COMpact 2FXO module

Use

Set the use for the external analogue ports.

free

Port is not used and port is not taken into account for the calculation of the total number of existing ports/call channels.

external

Port is used as external port.

Configure

Opens the page **ANALOGUE CONNECTION** (see Analogue Connection).

Public switched tel. networks

Exchange line request
Emergency calls
Analogue
ISDN
VoIP
Call Distribution
Call Forwarding

Exchange line request

Settings of the Exchange Line Request

Settings of the Exchange Line Request

Exchange Line Request/Exchange Line Access Number
Prefixes
Signalling Tones

Exchange Line Request/Exchange Line Access Number

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > Exchange line request**)

Note: Some of the following settings can also be configured seperately for connections/accounts.

System-wide Automatic Exchange Line Request

Requirements:

- Emergency call only with disabled exchange line access number
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > Emergency calls**

The PBX automatically distinguishes for each subscriber between internal and external numbers by analysing the digit sequence and the adjustment of the dialled phone number with the internal numbering plan.

Note: When switched off, the exchange line request type can be set individually for each subscriber.

Exchange Line Access Number

Requirements:

- Configured prefixes (country prefix and local area code)
- Automatic exchange line request disabled

Is required to request an external line (exception: direct exchange access and automatic exchange line request).

Note: An available external line is indicated by the external dial tone after the exchange line access number has been dialled. The external number can then be dialled.

Note: The factory setting for the exchange line access number on the PBX is 0. This number is not effective at first because automatic exchange line request is active in the factory settings.

Note: Changing the exchange line access number is only meaningful if another exchange line access number is normally used (e.g., in England).

Note: Note that changing the exchange line access number restricts the phone numbering plan.

Prefixes

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > Exchange line request**)

Note: Some of the following settings can also be configured seperately for connections/accounts.

PBX is used in the following country

When you configure the country the PBX is automatically assigned the default settings that are required in the selected country (e.g. settings for analogue ports, and emergency numbers).

Note: Emergency numbers preset in this way cannot be deleted. You can only preset different public emergency numbers by subsequently changing the country settings.

Note: Make sure that there are no conflicts between country-specific emergency numbers and your internal phone numbering plan. Assign new internal phone numbers, if necessary.

Country Code

International phone prefix (e. g. 0044 for the UK). These are necessary for some functions of the PBX (e.g. Call Through, Call restrictors/Call deblockers).

Area Code

Phone number prefix (e. g. 0208 for Greater London). These are necessary for some functions of the PBX (e.g. Call Through, Call restrictors/Call deblockers).

Note: Some countries (e.g. Denmark, Greece, Luxembourg) do not have a fixed number sequence for their area codes. If this also applies to the location at which the PBX is installed, you do not need to enter an area code.

Dial level

The selection of the dial level (Location, National, International) is country-specific. Depending on the country and the settings for the local area code above, a dial level must be selected (example: In Belgium, local area codes do not exist. Here, the dial level **City** must not be selected).

Global LCR provider digit-recognition, Digit number, Digits begin with

Requirements:

- Disabled **Adopt settings from PBX location (Germany)**

The digits in the table below are the common LCR number prefixes in Germany. For other countries, more LCR number prefixes (a total of 10) can be entered.

Digit number	Digits begin with
5	010
6	0100

Signalling Tones

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > Exchange line request**)

Note: Some of the following settings can also be configured seperately for connections/accounts.

Dial tone

Ringer

Busy tone

Depending on the location of the PBX, the signalling tones are correspondingly preconfigured but they can also be changed. For these tones, only the frequencies 400, 425, 440 and 450 Hz are possible. Tones differing from these frequencies are rounded to the nearest frequency.

Note: Do not change the preconfigured values for dial tone, ringer and busy tone unless you are told to do so by our support staff.

Emergency calls

Settings of the Emergency Calls

Settings of the Emergency Calls

Emergency Call has Priority
Emergency Numbers
Service Numbers

Emergency Call has Priority

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > Emergency calls**)

Emergency Call Priority

An external line is definitely made available from the PBX when dialling the emergency number or if an alarm is detected. If necessary, a call can be interrupted to enable an emergency call to be made.

Important: If **Emergency Call Priority** is disabled and other calls are being made during an emergency, there is a risk that the emergency call cannot be made immediately.

Note: The emergency call priority switch is enabled for the entire PBX.

Emergency Call Only with Exchange Line Access Number

Requirements:

- Automatic exchange line request (also for individual subscribers) disabled
  - Using the Configuration Manager (on the following pages)
  - **Public switched tel. networks > Exchange line request**
  - **Subscriber > Properties > Settings > Exchange line request**
  - **Subscriber (scr.) > Function overview > Settings > Exchange line request > Configure**

The emergency number must be dialled with the exchange line access number entered under **Public switched tel. networks > Exchange line request** (e. g. 0112 instead of only 112). In this way, numbers beginning with 11... will remain available to be assigned as internal numbers.

Important: Since the exchange line access number must be dialled in advance if **Emergency Call Only with Exchange Line Access Number** is enabled, there is a risk that the exchange line access number is forgotten in an emergency so that no emergency call can be made.

Note: When switched off, an emergency number can be dialled without exchange line access number, but it is not possible to assign internal numbers which begin with the first digits of an emergency number.

Note: If you try to disable **Emergency Call Only with Exchange Line Access Number** after internal numbers beginning with the first digits of an emergency number have been entered, an error message will be displayed. Solve the problem by assigning new phone numbers to the internal subscribers in question.

Emergency Numbers

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > Emergency calls**)

Emergency numbers

With the country settings for the location of the PBX (during first commissioning or on the page **Administration > Dealer and operator data** the country-specific emergency number will be preset. Overall, you can enter up to ten external numbers as emergency call or service phone numbers.

Note: Public emergency numbers, e. g. beginning with 11..., are entered without an area code, which reduces the size of the internal call numbering plan. For example, if the emergency phone numbers are 110 and 112 this means the internal numbers 110 and 112 and also 1100 to 1109 and 1120 to 1129 are no longer available.

Note: All other emergency numbers are entered with area code.

Note: Some countries (e.g. Denmark, Greece, Luxembourg) do not have a fixed number sequence for their area codes. If this also applies to the location at which the PBX is installed, you do not need to enter an area code.

Name

The name is used for display purposes on the system phone. The name along with the phone number also appears on some pages of the Configuration Manager (e.g. Call data list).

+ New

Adds a new emergency or service number.

Important: The exchange line access number is never entered when entering emergency or service numbers.

Delete

Deletes the entries marked in the list.

Service Numbers

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > Emergency calls**)

Service Numbers

Overall, you can enter up to ten external numbers as emergency call or service phone numbers.

Note: Public service phone numbers, e. g. beginning with 11..., are entered without an area code, which reduces the size of the internal call numbering plan. For example, if the service phone number is 115, this means the internal numbers 115 and 1150 to 1159 are no longer available.

Note: All other service phone numbers are entered with area code.

Note: Some countries (e.g. Denmark, Greece, Luxembourg) do not have a fixed number sequence for their area codes. If this also applies to the location at which the PBX is installed, you do not need to enter an area code.

Name

The name is used for display purposes on the system phone. The name along with the phone number also appears on some pages of the Configuration Manager (e.g. Call data list).

+ New

Adds a new emergency or service number.

Important: The exchange line access number is never entered when entering emergency or service numbers.

Delete

Deletes the entries marked in the list.

Analogue

Settings of the Analogue Connection
Expert Configuration of the Analogue Connection

Settings of the Analogue Connection

Analogue Connection
Analogue Phone Number

Analogue Connection

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

Name of the analogue connection

For identification of the port within the PBX.

GSM

The analogue port supports the GSM gateway connected to it. Any phone number can be used for the analogue port. It must be entered in the PBX and GSM gateway configurations.

Note: GSM gateways accept external phone calls automatically. This may cause call charges for the caller, even if no-one picks up the phone at the internal destination.

Note: If the PBX is not connected to the land line, outgoing external calls can only be accepted via a targeted exchange line access using the phone numbers entered on the **Routing > VoIP/GSM Routing**

> Exception numbers page (and the corresponding accesses, also configured on this page) or via a configured preferred exchange line (**exchange line settings** for subscribers or groups).

LCR

Requirements:

- Network provider must support the dial-around service
- Configured prefixes (country prefix and local area code)
- Configured LCR (see LCR)

The analogue connection is used for LCR calls.

Note: LCR must be disabled if the network provider does not support LCR.

Note: LCR cannot be enabled on an external port to which a GSM gateway is connected.

Note: However, you can always perform LCR manually by dialling a provider area code prefix if the network provider supports the dial-around service.

Exchange line has emergency call ability

Requirements:

- The corresponding connection must allow emergency calls

This exchange line can be used for emergency calls.

Note: When switched off, this exchange line cannot be used for emergency calls. You will hear the announcement, "This phone does not allow emergency calls. Please use an alternative."

Note: If you have updated your firmware from Version 6.4 to 6.6, the phone allows emergency calls by default.

Recording on

Requirements:

- Configured voice mailbox for call recordings (page **Devices > Voice mail and fax boxes > Phone numbers**). In order to see the box here, no owner must be assigned to the box.

Selection of the voice mailbox, which will be used for the function **Call recording**.

A voice mailbox for the function **Call recording** can be assigned to only one connection **Analogue / ISDN / VoIP**.

Note: If the voice mailbox for call recordings is removed from the list (Selection of **---**) and becomes a box with the function as answering machine, all recordings are deleted from this box after confirmation.

Note: The function **Call recording** is available only in certain countries.

Expert configuration > Configure

Opens the expert configuration.

Further help under Expert Configuration of the Analogue Connection

Analogue Phone Number

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

Name of the analogue connection

For identification of the port within the PBX.

GSM

The analogue port supports the GSM gateway connected to it. Any phone number can be used for the analogue port. It must be entered in the PBX and GSM gateway configurations.

Note: GSM gateways accept external phone calls automatically. This may cause call charges for the caller, even if no-one picks up the phone at the internal destination.

Note: If the PBX is not connected to the land line, outgoing external calls can only be accepted via a targeted exchange line access using the phone numbers entered on the **Routing > VoIP/GSM Routing**

> Exception numbers page (and the corresponding accesses, also configured on this page) or via a configured preferred exchange line (**exchange line settings** for subscribers or groups).

LCR

Requirements:

- Network provider must support the dial-around service
- Configured prefixes (country prefix and local area code)
- Configured LCR (see LCR)

The analogue connection is used for LCR calls.

Note: LCR must be disabled if the network provider does not support LCR.

Note: LCR cannot be enabled on an external port to which a GSM gateway is connected.

Note: However, you can always perform LCR manually by dialling a provider area code prefix if the network provider supports the dial-around service.

Exchange line has emergency call ability

Requirements:

- The corresponding connection must allow emergency calls

This exchange line can be used for emergency calls.

Note: When switched off, this exchange line cannot be used for emergency calls. You will hear the announcement, "This phone does not allow emergency calls. Please use an alternative."

Note: If you have updated your firmware from Version 6.4 to 6.6, the phone allows emergency calls by default.

Expert configuration > Configure

Opens the expert configuration.

Further help under Expert Configuration of the Analogue Connection

Expert Configuration of the Analogue Connection

Analogue Connection
General Info
Line Connection
Start of Call (Outgoing Calls)
End of the Call
Clip
Feature Codes
Tones
Ring Signal

Analogue Connection

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

Apply country settings for

Accepts the country settings for the selected country.

Note: The analogue port has default settings that are required for analogue ports in the country in which the PBX is operating. The settings are specified by selecting the country when you set up the device for the first time. Changes to the port must be made, for example, when connecting a GSM gateway that was purchased in a different country and therefore requires different default settings.

Note: You do not usually need to change these default values. However, if you do need to, follow the instructions from your network provider.

Copy settings to Execute

Copies the expert configuration settings to the selected port.

General Info

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

Apply country settings for

Accepts the country settings for the selected country.

Note: The analogue port has default settings that are required for analogue ports in the country in which the PBX is operating. The settings are specified by selecting the country when you set up the device for the first time. Changes to the port must be made, for example, when connecting a GSM gateway that was purchased in a different country and therefore requires different default settings.

Note: You do not usually need to change these default values. However, if you do need to, follow the instructions from your network provider.

Copy settings to Execute

Copies the expert configuration settings to the selected port.

Line Connection

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

Waiting for dial tone

Before the PBX dials a phone number it checks whether a dial tone has been sent from the public exchange. It starts dialling as soon as it detects a dial tone for 200 milliseconds

This function prevents the PBX dialling the wrong number by starting dialling too early if the public exchange takes too long to play the external dial tone it generates.

Note: Disable this setting if the dial tone cannot be sent clearly by the public exchange and, as a result, cannot be identified by the PBX.

Waiting time (3-30 seconds)

The waiting time for a line connection setting specifies the time (in seconds) after which the PBX stops establishing the connection if it does not recognise a dial tone, although the wait for dial tone setting is enabled.

If the wait for dial tone setting is disabled, the PBX starts dialling after approximately 4 seconds.

Transmit line access code automatically (0-9, *, #)

The number/character string entered here is automatically added to the selected phone number as a prefix.

This is needed if the public exchange always requires a specific leading number/character string that is not part of the phone number.

Detect line access code and wait (0-9)

A dialling pause of 3 seconds is inserted automatically after the PBX dials the number/character string entered here.

This is needed if the public exchange requires a pause after the specified line access code is dialled.

Loop current detection

Requirements:

- The loop current detection function to detect the network provider's line connection

The PBX checks whether loop current is present on the line on which the call is to be made.

Note: If loop current is detected, the line is used.

Note: If no loop current is detected, the PBX treats the line as if it is either not enabled or not present.

Start of Call (Outgoing Calls)

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

Waiting time after the last dialled digit (1-30 seconds)

Note: This setting is only relevant when the setting **send #** is enabled.

The waiting time serves for post-dialling digits before the PBX recognises that dialling has been finished and sends # to the exchange line. # signalises the exchange line that the number sequence is completed.Time measurement restarts after the entry of each digit. The possibilty of post-dialling digits is not applicable if the PBX has recognized that the number sequence is already completed and automatically sends # to the exchange line.

Note: Select as short a time as possible, because the PBX does not forward the last number to the public exchange until the specified time has passed.

Note: Approximately three seconds is the default value for manual inputs.

Note: You can always press # (the hash key) when you finish dialling the phone number to keep the waiting time between dialling the last number and the connection being set up to a minimum. The PBX recognises this character as the end of the phone number.

Transmit

The PBX transmits the # (hash key) character, appended to the number you are dialling, used to signal that you have finished, to the public exchange.

Busy signal detection

The PBX recognises the busy signal sent by the public exchange (recommended setting).

This is needed to allow the CCBS (callback when busy) function to be enabled using the PBX's internal phones.

Max. hold time until audio connection established (0-30 seconds)

The time to wait until the audio connection is created specifies the maximum time, in seconds, that the PBX will wait for the connection information sent by the public exchange. If the PBX does not receive a busy signal, a continuous tone or a polarity reversal of the line after this time has passed, it creates the connection.

The PBX also creates the connection if it detects a call tone, and then a voice, even before the waiting time has passed.

End of the Call

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

Busy signal detection (incoming calls)

Busy signal detection (outgoing calls)

The PBX recognises the busy signal sent by the public exchange.

It needs this to identify that the call connection has finished and to automatically disconnect an ISDN phone.

Note: Disable this setting if the PBX identifies disruptions on the external line as the busy signal, because this would cause it to interrupt conversations. In this situation, the subscriber must signal to the PBX that the call is ended by disconnecting.

Minimum length of busy signal (3-20 seconds)

The minimum length of the busy signal specifies the time (in seconds) the PBX takes to identify a busy signal sequence. This prevents, for example, the PBX identifying speech as a busy signal.

Continuous tone

Requirements:

- The PBX must recognise the continuous tone sent by the network provider as the end of the call

The PBX recognises the continuous tone sent by the network provider as the end of the call connection.

Loop current detection

Requirements:

- The PBX must recognise the drop in loop current sent by the network provider at the end of the call

The PBX recognises the end of the call when the network provider drops the loop current to zero.

DTMF code 'D'

Requirements:

- The PBX must recognise the DTMF code "D" sent by the network provider at the end of the call.

The PBX recognises the end of the call when the network provider sends DTMF code "D".

Clip

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

CLIP information type

Requirements:

- CLIP feature (transmission and display of the caller's phone number) enabled by network provider

The type of CLIP information specifies whether, and how, the analogue network provider sends the CLIP information.

No

The network provider does not send the CLIP information.

FSK

The network provider uses Frequency Shift Keying to send the CLIP information.

DTMF and DTMF (MaxComm)

The network provider uses Dual Tone Multi Frequency signalling to send the CLIP information.

CLIP sub-version

The CLIP sub-version specifies when the analogue network provider sends the CLIP information.

No evaluation of CLIP information

The data sent by the network provider is not forwarded to the internal subscriber.

Amplification (0 to +25 dB)

CLIP information amplification specifies the number of decibels by which the PBX increases or reduces all the audio signals when CLIP information is being received.

Feature Codes

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

The feature codes specify the character strings the analogue network provider requires to execute a service feature on the analogue connection. If a service feature is requested on one of the PBX's ISDN subscribers (e.g. system phone), the PBX automatically converts it into the character strings required for the analogue network provider.

For the following functions, feature codes can be entered:

Suppress phone number (CLIR)
Initiate callback when busy (CCBS)
Enable CF immediately (CFU)
Disable CF immediately (CFU)
Enable CF when busy (CFB)
Disable CF when busy (CFB)
Enable CF if no response (CFNR)

Disable CF if no response (CFNR)

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters
- **n** as a wildcard for the phone number
- **,** (comma) for a one second pause
- **R** as a wildcard for a flash

Tones

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

The character strings show which tones are sent by the analogue network provider. This means the PBX can then identify the tones and apply them accordingly.

The network provider uses these tones in the following situations:

- **Dial tone**

  When a subscriber picks up the call.

- **Busy signal**

  If the required connection is busy (busy signal 1), or if the network is overloaded (busy signal 2, e.g. directly after dialling the local area code).

- **Ringer**

  During the call phase.

- **Special interest tone**

If the dialled phone number is not present, and the network provider has not recorded a special announcement for this situation.

Note: Please contact the Support team before making any changes to these character strings.

Note: An incorrect character string will cause the PBX to operate incorrectly.

The character strings can be change for the following tones

Dial tone 1 (normal dial tone)
Dial tone 2 (special dial tone)
Dial tone 3 (Centrix tone)
Busy tone 1 (call target busy)
Busy tone 2 (network busy)
Call tone 1
Call tone 2
Special interest tone

Ring Signal

Proceed as follows:

- Separately for analogue ports
- Using the Configuration Manager (page **Public switched tel. networks > Analogue > Phone numbers > Configure**)

Requirements:

- External analogue port (optional, see Technical Data in the Instructions)

ANALOGUE CONNECTION

Select the analogue connection to be displayed/configured.

The feature codes specify the character strings the analogue network provider requires to execute a service feature on the analogue connection. If a service feature is requested on one of the PBX's ISDN subscribers (e.g. system phone), the PBX automatically converts it into the character strings required for the analogue network provider.

For the following functions, feature codes can be entered:

Suppress phone number (CLIR)
Initiate callback when busy (CCBS)
Enable CF immediately (CFU)
Disable CF immediately (CFU)
Enable CF when busy (CFB)
Disable CF when busy (CFB)
Enable CF if no response (CFNR)

Disable CF if no response (CFNR)

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters
- **n** as a wildcard for the phone number
- **,** (comma) for a one second pause
- **R** as a wildcard for a flash

ISDN

Settings of the PTMP Connection
Expert Configuration of the PTMP Connection
Settings PBX Base Number/Bundle
Expert Configuration for the PTP Connection
Settings of the Extension
Settings Additional Numbers ISDN

Settings of the PTMP Connection

Network Provider
Miscellaneous
Multiple Subscriber Numbers

Network Provider

Proceed as follows:

- Separately for external S_0 ports
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection**)

Requirements:

- External S_0 port (optional, see Technical Data in the Instructions)
- Connection type PTMP connection on external S_0 port

PTMP CONNECTION

Select the PTMP connection to be displayed/configured.

Exchange line has emergency call ability

Requirements:

- The corresponding connection must allow emergency calls

This exchange line can be used for emergency calls.

Note: When switched off, this exchange line cannot be used for emergency calls. When you attempt to dial an emergency number, you will hear the announcement, "This phone does not allow emergency calls. Please use an alternative."

Note: If you have updated your firmware from Version 6.4 to 6.6, the phone allows emergency calls by default.

Miscellaneous

Proceed as follows:

- Separately for external S_0 ports
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection**)

Requirements:

- External S_0 port (optional, see Technical Data in the Instructions)
- Connection type PTMP connection on external S_0 port

PTMP CONNECTION

Select the PTMP connection to be displayed/configured.

Name of the PTMP connection

For identification of the connection within the PBX.

Expert configuration > Configure

Opens the expert configuration.

Further help under Expert Configuration of the PTMP Connection

GSM gateway

The S_0 port supports the GSM gateway connected to it. Any phone numbers can be used for the connection. They must be entered in the PBX and GSM gateway configurations.

Note: When you configure the GSM gateway, you must also input the connection type of the ISDN connection.

Note: GSM gateways accept external phone calls automatically. This may cause call charges for the caller, even if no-one picks up the phone at the internal destination.

Note: If the PBX is not connected to the land line, outgoing external calls can only be accepted via a targeted exchange line access using the phone numbers entered on the page **Routing > VoIP/GSM Routing > Exception numbers** (and the corresponding accesses, also configured on this page) or via a configured preferred exchange line (**exchange line settings** for subscribers or groups).

Multiple Subscriber Numbers

Proceed as follows:

- Separately for external S_0 ports
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection**)

Requirements:

- External S_0 port (optional, see Technical Data in the Instructions)
- Connection type PTMP connection on external S_0 port

PTMP CONNECTION

Selection of the PTMP connection to be displayed/configured.

Multiple subscriber number (MSN)

The PTMP connection has up to 10 multiple subscriber numbers. The network provider assigns the phone numbers for the connection. (Select + **New** to enter.)

Phone number for CLIP no screening

Requirements:

- Set up phone number
- CLIP no screening switched on for the ISDN connection

Customer-specific phone number (e.g. a service phone number) that is transferred to the destination by the exchange line without its correctness being checked.

Name

For display purposes on the system phone.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

Recording on

Requirements:

- Configured voice mailbox for call recordings (page **Devices > Voice mail and fax boxes > Phone numbers**). In order to see the box here, no owner must be assigned to the box.

Selection of the voice mailbox, which will be used for the function **Call recording**.

A voice mailbox for the function **Call recording** can be assigned to only one connection **Analogue / ISDN / VoIP**.

Note: If the voice mailbox for call recordings is removed from the list (Selection of **---**) and becomes a box with the function as answering machine, all recordings are deleted from this box after confirmation.

Note: The function **Call recording** is available only in certain countries.

+ New

Adds a new phone number.

Delete

Deletes the entries marked in the list.

Expert Configuration of the PTMP Connection

PTMP Connection
General Info
Routing
Call Forwarding
Number Presentation

PTMP Connection

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection > Expert configuration > Configure**)

Requirements:

- Configured PTMP connection

Country settings for this exchange line

At configuration and relocation of the PBX, the country-specific settings are automatically set.

Country settings have been changed

If you make not country-specific expert settings, you must save them when leaving the page.

Copy settings to | Execute

The settings of this port can be copied an transferred for another port.

Charge information

Requirements:

- Charge information provided by the network provider

The PBX stores the charge information sent by the network provider and uses it to evaluate individual calls.

Important: Disabling this function is only advisable if it causes problems on the external connection (this depends on the network provider/country).

Important: Some telephony providers do not send charge information. If this type of provider is being used, you cannot apply any charges to a particular call.

Important: You cannot check exactly how many PBXs should be charged for each call because, for example, you cannot uniquely assign the charges to one of the two subscribers involved in a query call from an external call.

Important: If you are using the Soft-LCR 4.0 LCR procedure, the duration of the call and your own table of tariffs are used to calculate the charges.

General Info

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection > Expert configuration > Configure**)

Requirements:

- Configured PTMP connection

Country code

International phone prefix (e. g. 0044 for the UK).

Local area code

Phone number prefix (e. g. 0208 for Greater London).

Note: If there is no fixed digit sequence as the area code at the installation location of the PBX, no entry is required here.

Dial level

The selection of the dial level (Location, National, International) is country-specific. Depending on the country and the settings for the local area code above, a dial level must be selected (example: In Belgium, local area codes do not exist. Here, the dial level **Location** must not be selected).

Note: After the port settings once have been made correctly, no further changes should be made.

Source suitable for clock synchronisation

This port is used as source for the clock synchronisation and serves as clock master for the PBX. Thus, interfering noise during calls are avoided.

Note: Select a port a source that is synchronous to the ISDN exchange line.

Layer 1 permanently active

Requirements:

- External S_0 port (optional, see Technical Data in the Operating Constructions)

An activation of the layer 1 is requested from the PBX as soon as the layer is put into the idle state by the network provider. Thus, for example, a delay time is avoided before the external dial tone indicates its action readiness.

Language delay at setup ack with inband info

Interfering noise during dialling phase are avoided.

Always send DTMF digit after inband info

Note: This is a country-specific setting (only relevant in Italy).

All outstanding digits are transferred via DTMF.

No T310 for Info boxes

Note: This is a country-specific setting (only relevant in Italy).

T310 is a timer for ISDN that disconnects a call connection after a certain time.

The call is not disconnected after a certain time.

Routing

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection > Expert configuration > Configure**)

Requirements:

- Configured PTMP connection

Use Soft LCR 4.0 for this connection

Requirements:

- Network provider must support the dial-around service
- Configured prefixes (country prefix and local area code)
- Configured LCR (see LCR)

The ISDN connection is used for LCR calls.

Note: LCR must be disabled if the network provider does not support LCR.

Note: However, you can always perform LCR manually by dialling a provider area code prefix if the network provider supports the dial-around service.

LCR provider digit-recognition for this connection

Requirements:

- Disabled **Adopt settings from PBX location (Germany)**

The digits in the table below are the common LCR number prefixes in Germany. For other countries, more LCR number prefixes (a total of 10) can be entered.

Digit number	Digits begin with
5	010
6	0100

New

A new digit is added to the table.

Delete

Deletes the entries marked in the list.

Call Forwarding

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection > Expert configuration > Configure**)

Requirements:

- Configured PTMP connection

GENERAL SETTINGS FOR THE CALL FORWARDING

GENERAL SETTINGS FOR CALL FORWARDING

Call forwarding method

Call forwardings at the exchange line are configured and performed in different ways. The exchange line, router or the provider determines which of the call forwardings listed below can be used.

CF facility

The call forwarding is controlled in the PTMP exchange line. The standardized message processings are not supported by all providers.

Registro

Special form of the Keypad call forwarding method in Spain.

CR facility

The call forwarding is controlled in the PTP connection. Not every provider supports this method.

Keypad

The call forwarding is controlled in the PTMP exchange line. The digit sequence depends on the provider or router.

CD facility

The call forwarding is controlled in the PTMP connection. Not every provider supports this method.

2. B channel

The call forwarding is performed by the PBX during a call. It is possible for all exchange line types but requires a free voice channel and the number of the caller cannot be displayed unless CLIP no scrrening is used which is not supported by many providers.

Note: The setting **CF facility** has priority to the setting **2. B channel**.

For call forwarding via the PBX, use any available call channel

Requirements:

- Enabled call forwarding method **2. B channel**

When an external phone number is forwarded, the call can be forwarded via a call channel on any of the other ports.

Note: A call can only be redirected via a call channel on the same port. The probability that the call cannot be forwarded because the channels are busy is therefore higher.

Renew call forwarding at PBX reboot

When rebooting the PBX, the saved data of the call forwarding is renewed.

Important: In Switzerland, the configuration of a call forwarding is chargeable. To avoid costs, the check box must be disabled.

Delay time for Call Forwarding on no reply in the PBX

This delay time for external numbers specifies the time (in seconds) after which a call is forwarded if the called subscriber does not respond.

Note: When switched off, the waiting time applies only to call forwarding that was set up using the PBX. If the network provider has configured the call forwarding function, the waiting time depends on their settings and cannot be configured in the PBX.

Routing via exception numbers (VoIP + GSM)

Requirements:

- Configured exception numbers (page **Routing > VoIP and GSM Routing > Exception numbers**)
- Enabled call forwarding method **2. B channel**

The device checks whether the destination number involved in call forwarding for external numbers has been set up as an exception number. If so, the call is sent along the configured path. If the destination number has not been entered in the routing table, the call to be forwarded is sent either via a free call channel that is assigned to the port or via any free call channel (**For call forwarding via the PBX, use any available call channel**).

SETTINGS FOR CALL FORWARDING METHODS CF/CR/CD FACILITY

SETTINGS FOR THE CALL FORWARDING METHODS CF-/CR-/CD-FACILITY

Service Indicator

ISDN voice connections can be performed via different services. The Service Indicator selects the service, The provider defines the available services, however, the usual services are not always supported by all providers.

Service Indicator for the call forwarding query

Select the Service Indicator for the call forwarding query (All services,Speech only, Audio 3kHz, Speech & Audio & Telephony 3 kHz, Automatically, not available)

Service Indicator for the switching

Select the Service Indicator for the switching (All services,Speech only, Audio 3kHz, Speech & Audio & Telephony 3 kHz, Automatically, not available)

Destination number (CDPN) of type

Destination number (CDPN) in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Destination number (CDPN) receives prefix

Enter a prefix (*****, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

Destination number (CDPN) receives postfix

Enter a postfix (*****, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

MSN/DDI fwd. Destination number uses Soft-LCR 4.0 and manual LCR

An LCR prefix is set in front of the phone number.

Own phone number (CGPN) of type

Own phone number (CGPN) of type

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Use CLIP no screening for call forwarding

Requirements

- CLIP no screening switched on for the ISDN connection

If a call is forwarded via the external connection, the caller's phone number is sent to the destination of the forwarded call. If it is an anonymous call, the function behaves as if it was switched off.

Note: When switched off, the PBX's phone number is sent to the destination of the forwarded call.

CALL FORWARDING UNDCONDITIONAL (CFU)

Call forwarding available in exchange line

If the exchange line has a CFU, this function can be enabled.

Note: If the exchange line does not have a CFU, this function must be disabled.

Keypad-/Registro code: Enable CFU

Keypad-/Registro code: Disable CFU

The feature codes specify the character strings the ISDN network provider requires to execute a service feature on the ISDN connection

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters
- **n** as a wildcard for the destination number (CDPN)
- **s** as a wildcard for the own phone number (MSN)

CALL FORWARDING IF BUSY (CFB)

Call forwarding available in exchange line

If the exchange line has a CFB, this function can be enabled.

Note: If the exchange line does not have a CFB, this function must be disabled.

Keypad-/Registro code: Enable CFB

Keypad-/Registro code: Disable CFB

The feature codes specify the character strings the ISDN network provider requires to execute a service feature on the ISDN connection

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters
- **n** as a wildcard for the destination number (CDPN)
- **s** as a wildcard for the own phone number (MSN)

CALL FORWARDING NO REPLY (CFNR)

Call forwarding available in exchange line

If the exchange line has a CFNR, this function can be enabled.

Note: If the exchange line does not have a CFNR, this function must be disabled.

Keypad-/Registro code: Enable CFNR

Keypad-/Registro code: Disable CFNR

The feature codes specify the character strings the ISDN network provider requires to execute a service feature on the ISDN connection

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters
- **n** as a wildcard for the destination number (CDPN)

- **s** as a wildcard for the own phone number (MSN)

Number Presentation

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PTMP connection > Expert configuration > Configure**)

Requirements:

- Configured PTMP connection

OUTGOING CALLS (CDPN)

OUTGOING CALLS (CDPN)

CDPN of type

CDPN in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

CDPN receives prefix

Enter a prefix (*****, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

CDPN receives postfix

Enter a postfix (*****, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

CLIR

The selection of the available options for this function is provider-specific.

Info element

The called party number is sent as info element.

Dial prefix

The called party number is sent as keypad sequence. The dial prefix is provider-specific (e. g. #31).

OUTGOING CALLS (CGPN)

OUTGOING CALLS (CGPN)

Own phone number of type

Own phone number in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Signalising LLC call types as

Audio 3kHz

Selection for fax devices and analogue modems (e. g. alarm systems and emergency devices).

Speech

The exchange line compresses the channel. The transferred amout of data is reduced, whereby the sound quality on the receive side could be slightly changed.

As from subscriber

The LLC call type is adopted from the subscriber.

Automatically

Audio 3kHz is preferably used. At problems with the destination, it is automatically switched to Speech.

CLIP no screening

The called party sees a customer-defined phone number instead of a phone number assgined by the network provider. The customer-defined number is not checked by the exchange line for correctness.

CLIP no screening from type

CLIP no screening in format

Requirements:

- Enabled **CLIP no screening** for incmong calls

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

CONNECTED CALLS (COPN)

CONNECTED CALLS (COPN)

Connected phone number of type

Connected phone number in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Special encoding of the connected phone number

Standard

Must be selected in all countries except for Spain.

Spain

Must be selected in Spain.

Note: Depending on the provider, in Spain the special encoding **Standard** might as well be selected.

FOR INCOMING CALLS

FOR INCOMING CALLS

Waiting for direct dialling (4..8 sec)

Note: This is a country-specific setting (only relevant in Austria).

Enter a waiting time until the direct dialling (between 4 and 8 seconds). This is useful when the phone number is first transferred incompletely.

Settings PBX Base Number/Bundle

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PBX base numbers/trunk bundle**)

Requirements:

- External S_0 port (optional, see Technical Data in the Instructions)
- Connection type PTP connection at external S_0 port

PBX base number

A PTP connection has a PBX base number (base number, header, base phone number) and (Direct Dial In - DDI) extension numbers. The network provider assigns the phone numbers for the connection. The phone number to be dialled is formed from the PBX base number + (DDI) extension number.

Name

For identification of the connection within the PBX.

Extensions block (DDI numbers) | From | To

A PTP connection has a PBX base number (base number, header, base phone number) and (Direct Dial In - DDI) extension numbers. The network provider assigns the phone numbers for the connection. The phone number to be dialled is formed from the PBX base number + (DDI) extension number.

Note: you can enter a zero both as a single number (e.g. 0 to 99, as is usually the case) and as a digit in the set number of places in extension numbers (e.g. 00 to 99).

Note: If linear call distribution for the PTP connection is enabled (**Public switched tel. networks > Call distribution** page), and if the internal numbers match the external extension numbers, the system automatically performs a 1:1 assignment for the extension numbers in the extensions block. If an extension cannot be assigned, the fallback call distribution function is enabled.

Note: If you want an extension number to be different from the ones used in linear call distribution, or if an extension number is needed for a different function (e.g. remote configuration, CLIP no screening), you must also enter it in the phone number list (page **Public switched tel. networks > ISDN > Extension numbers PTP connection**).

Note: For example, for extensions block 0 to 99: enter extension 0 as an extension and assign it in the call distribution.

Note: For example, for extensions block 00 to 99: enter the individual extension numbers 00 to 09 as an extension number and assign them during call distribution. The PBX will then only react when the number of places is reached, for example, if the caller dials 00 as an extension number. In contrast, if the caller only dials 0, the fallback call distribution function is enabled as soon as the time for selecting a suffix dialling digit has passed.

Note: You must enter all the extension numbers for a PTP connection that is not running linear call distribution.

GSM gateway

The S_0 port supports the GSM gateway connected to it. Any phone numbers can be used for the connection. They must be entered in the PBX and GSM gateway configurations.

Note: When you configure the GSM gateway, you must also input the connection type of the ISDN connection.

Note: GSM gateways accept external phone calls automatically. This may cause call charges for the caller, even if no-one picks up the phone at the internal destination.

Note: If the PBX is not connected to the land line, outgoing external calls can only be accepted via a targeted exchange line access using the phone numbers entered on the page **Routing > VoIP/GSM Routing > Exception numbers** (and the corresponding accesses, also configured on this page) or via a configured preferred exchange line (**exchange line settings** for subscribers or groups).

Exchange line has emergency call ability

Requirements:

- The corresponding connection must allow emergency calls

This exchange line can be used for emergency calls.

Note: When switched off, this exchange line cannot be used for emergency calls. When you attempt to dial an emergency number, you will hear the announcement, "This phone does not allow emergency calls. Please use an alternative."

Expert configuration > Configure

Opens the expert configuration.

Further help under Expert Configuration for the PTP Connection

Port x

Assigns the PBX base number entered here to a port.

Note: If the network provider assigns the same PBX base number to a number of PTP connections, these connections can be bundled together. A bundled PTP connection consists of all the ports assigned to the same line that have the same phone number and the same extensions block. Additional settings can also be made for the bundle.

Note: A "primary multiplex interface" is the same as a bundle and cannot be combined with other connections.

+ New

Adds a new phone number.

Delete

Deletes the entries marked in the list.

Expert Configuration for the PTP Connection

PTP Connection
General Info
Routing
Call Forwarding
Number Presentation

PTP Connection

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Separately for PBX base number
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PBX base numbers/trunk bundles > Configure**)

Requirements:

- Configured PBX base number

Country settings for this exchange line

At configuration and relocation of the PBX, the country-specific settings are automatically set.

Country settings have been changed

If you make not country-specific expert settings, you must save them when leaving the page.

Copy settings to | Execute

The settings of this port can be copied an transferred for another port.

Charge information

Requirements:

- Charge information provided by the network provider

The PBX stores the charge information sent by the network provider and uses it to evaluate individual calls.

Important: Disabling this function is only advisable if it causes problems on the external connection (this depends on the network provider/country).

Important: Some telephony providers do not send charge information. If this type of provider is being used, you cannot apply any charges to a particular call.

Important: You cannot check exactly how many PBXs should be charged for each call because, for example, you cannot uniquely assign the charges to one of the two subscribers involved in a query call from an external call.

Important: If you are using the Soft-LCR 4.0 LCR procedure, the duration of the call and your own table of tariffs are used to calculate the charges.

CCBS

Requirements:

- Service feature CCNR (callback on no response) is enabled by the network provider.
- Configured prefixes (country prefix and local area code)

The PBX offers the callback option to an external caller if the connection they dialled is busy.

Important: If this function is enabled, but the network provider does not support it (service feature not unlocked), it might still happen that the caller is given the option of a callback. In this case, the caller's reply is not forwarded to the PBX and therefore the function cannot be performed.

Important: In addition, this function can often only be used if both call partners are using the same network provider.

CCNR

Requirements:

- Service feature CCNR (callback on no response) is enabled by the network provider.
- Configured prefixes (country prefix and local area code)

The PBX offers the callback option to an external caller if their call is not answered.

Important: If this function is enabled, but the network provider does not support it (service feature not unlocked), it might still happen that the caller is given the option of a callback. In this case, the caller's reply is not forwarded to the PBX and therefore the function cannot be performed.

Ringtone for the PTP connection/bundle mentioned above

The ringtone specifies which sequence of sounds is used to signal external calls to the phone numbers of the PTP connection to internal subscribers on the PBX.

Note: Individual ringtones can be assigned to extension numbers that were input in different ways.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

General Info

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Separately for PBX base number
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PBX base numbers/trunk bundles > Configure**)

Requirements:

- Configured PBX base number

Country code

International phone prefix (e. g. 0044 for the UK).

Local area code

Phone number prefix (e. g. 0208 for Greater London).

Note: If there is no fixed digit sequence as the area code at the installation location of the PBX, no entry is required here.

Dial level

The selection of the dial level (Location, National, International) is country-specific. Depending on the country and the settings for the local area code above, a dial level must be selected (example: In Belgium, local area codes do not exist. Here, the dial level **Location** must not be selected).

Note: After the port settings once have been made correctly, no further changes should be made.

Source suitable for clock synchronisation

This port is used as source for the clock synchronisation and serves as clock master for the PBX. Thus, interfering noise during calls are avoided.

Note: Select a port a source that is synchronous to the ISDN exchange line.

Layer 1 permanently active

Requirements:

- External S_0 port (optional, see Technical Data in the Operating Constructions)

An activation of the layer 1 is requested from the PBX as soon as the layer is put into the idle state by the network provider. Thus, for example, a delay time is avoided before the external dial tone indicates its action readiness.

Language delay at setup ack with inband info

Interfering noise during dialling phase are avoided.

Always send DTMF digit after inband info

Note: This is a country-specific setting (only relevant in Italy).

All outstanding digits are transferred via DTMF.

No T310 for Info boxes

Note: This is a country-specific setting (only relevant in Italy).

T310 is a timer for ISDN that disconnects a call connection after a certain time.

The call is not disconnected after a certain time.

Routing

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Separately for PBX base number
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PBX base numbers/trunk bundles > Configure**)

Requirements:

- Configured PBX base number

Use Soft LCR 4.0 for this connection

Requirements:

- Network provider must support the dial-around service
- Configured prefixes (country prefix and local area code)
- Configured LCR (see LCR)

The ISDN connection is used for LCR calls.

Note: LCR must be disabled if the network provider does not support LCR.

Note: However, you can always perform LCR manually by dialling a provider area code prefix if the network provider supports the dial-around service.

LCR provider digit-recognition for this connection

Requirements:

- Disabled **Adopt settings from PBX location (Germany)**

The digits in the table below are the common LCR number prefixes in Germany. For other countries, more LCR number prefixes (a total of 10) can be entered.

Digit number	Digits begin with
5	010
6	0100

New

A new digit is added to the table.

Delete

Deletes the entries marked in the list.

Call Forwarding

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Separately for PBX base number
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PBX base numbers/trunk bundles > Configure**)

Requirements:

- Configured PBX base number

GENERAL SETTINGS FOR CALL FORWARDING

Call forwarding method

Call forwardings at the exchange line are configured and performed in different ways. The exchange line, router or the provider determines which of the call forwardings listed below can be used.

CF facility

The call forwarding is controlled in the PTMP exchange line. The standardized message processings are not supported by all providers.

Registro

Special form of the Keypad call forwarding method in Spain.

CR facility

The call forwarding is controlled in the PTP connection. Not every provider supports this method.

Keypad

The call forwarding is controlled in the PTMP exchange line. The digit sequence depends on the provider or router.

CD facility

The call forwarding is controlled in the PTMP connection. Not every provider supports this method.

2. B channel

The call forwarding is performed by the PBX during a call. It is possible for all exchange line types but requires a free voice channel and the number of the caller cannot be displayed unless CLIP no scrrening is used which is not supported by many providers.

Note: The setting **CF facility** has priority to the setting **2. B channel**.

For call forwarding via the PBX, use any available call channel

Requirements:

- Enabled call forwarding method **2. B channel**

When an external phone number is forwarded, the call can be forwarded via a call channel on any of the other ports.

Note: A call can only be redirected via a call channel on the same port. The probability that the call cannot be forwarded because the channels are busy is therefore higher.

Renew call forwarding at PBX reboot

When rebooting the PBX, the saved data of the call forwarding is renewed.

Important: In Switzerland, the configuration of a call forwarding is chargeable. To avoid costs, the check box must be disabled.

Delay time for Call Forwarding on no reply in the PBX

This delay time for external numbers specifies the time (in seconds) after which a call is forwarded if the called subscriber does not respond.

Note: When switched off, the waiting time applies only to call forwarding that was set up using the PBX. If the network provider has configured the call forwarding function, the waiting time depends on their settings and cannot be configured in the PBX.

Routing via exception numbers (VoIP + GSM)

Requirements:

- Configured exception numbers (page **Routing > VoIP and GSM Routing > Exception numbers**)
- Enabled call forwarding method **2. B channel**

The device checks whether the destination number involved in call forwarding for external numbers has been set up as an exception number. If so, the call is sent along the configured path. If the destination number has not been entered in the routing table, the call to be forwarded is sent either via a free call channel that is assigned to the port or via any free call channel (**For call forwarding via the PBX, use any available call channel**).

SETTINGS FOR THE CALL FORWARDING METHODS CF-/CR-/CD-FACILITY

Service Indicator

ISDN voice connections can be performed via different services. The Service Indicator selects the service, The provider defines the available services, however, the usual services are not always supported by all providers.

Service Indicator for the call forwarding query

Select the Service Indicator for the call forwarding query (All services,Speech only, Audio 3kHz, Speech & Audio & Telephony 3 kHz, Automatically, not available)

Service Indicator for the switching

Select the Service Indicator for the switching (All services,Speech only, Audio 3kHz, Speech & Audio & Telephony 3 kHz, Automatically, not available)

Destination number (CDPN) of type

Destination number (CDPN) in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Destination number (CDPN) receives prefix

Enter a prefix (*****, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

Destination number (CDPN) receives postfix

Enter a postfix (*****, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

MSN/DDI fwd. Destination number uses Soft-LCR 4.0 and manual LCR

An LCR prefix is set in front of the phone number.

Own phone number (CGPN) of type

Own phone number (CGPN) of type

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Use CLIP no screening for call forwarding

Requirements

- CLIP no screening switched on for the ISDN connection

If a call is forwarded via the external connection, the caller's phone number is sent to the destination of the forwarded call. If it is an anonymous call, the function behaves as if it was switched off.

Note: When switched off, the PBX's phone number is sent to the destination of the forwarded call.

CALL FORWARDING UNCONDITIONAL (CFU)

Call forwarding available in exchange line

If the exchange line has a CFU, this function can be enabled.

Note: If the exchange line does not have a CFU, this function must be disabled.

CALL FORWARDING IF BUSY (CFB)

Call forwarding available in exchange line

If the exchange line has a CFB, this function can be enabled.

Note: If the exchange line does not have a CFB, this function must be disabled.

CALL FORWARDING ON NO REPLY (CFNR)

Call forwarding available in exchange line

If the exchange line has a CFNR, this function can be enabled.

Note: If the exchange line does not have a CFNR, this function must be disabled.

Number Presentation

Important: The expert settings for the ISDN connection are settings that should only be changed if e. g. country or provider-specific particularities make it absolutely necessary.

Proceed as follows:

- Separately for PBX base number
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > PBX base numbers/trunk bundles > Configure**)

Requirements:

- Configured PBX base number

OUTGOING CALLS (CDPN)

CDPN of type

CDPN in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

CDPN receives prefix

Enter a prefix (*, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

CDPN receives postfix

Enter a postfix (*****, **#**, **0-9**; max. 4 digits) to send possibly necessary control commands to the provider.

CLIR

The selection of the available options for this function is provider-specific.

Info element

The called party number is sent as info element.

Dial prefix

The called party number is sent as keypad sequence. The dial prefix is provider-specific (e. g. #31).

OUTGOING CALLS (CGPN)

Own phone number of type

Own phone number in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Signalising LLC call types as

Audio 3kHz

Selection for fax devices and analogue modems (e. g. alarm systems and emergency devices).

Speech

The exchange line compresses the channel. The transferred amout of data is reduced, whereby the sound quality on the receive side could be slightly changed.

As from subscriber

The LLC call type is adopted from the subscriber.

Automatically

Audio 3kHz is preferably used. At problems with the destination, it is automatically switched to Speech.

CLIP no screening

The called party sees a customer-defined phone number instead of a phone number assgined by the network provider. The customer-defined number is not checked by the exchange line for correctness.

CLIP no screening from type

CLIP no screening in format

Requirements:

- Enabled **CLIP no screening** for incmong calls

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

CONNECTED CALLS (COPN)

Connected phone number of type

Connected phone number in format

Note: The settings ‚of type' and ‚in format' build a group which means both settings must be made in dependency to each other.

Select the type (**Unknown, Subscriber, National, International**) how the phone number shall be transferred.

Select the phone number format dependent on the phone number type.

Special encoding of the connected phone number

Standard

Must be selected in all countries except for Spain.

Spain

Must be selected in Spain.

Note: Depending on the provider, in Spain the special encoding **Standard** might as well be selected.

FOR INCOMING CALLS

Waiting for direct dialling (4..8 sec)

Note: This is a country-specific setting (only relevant in Austria).

Enter a waiting time until the direct dialling (between 4 and 8 seconds). This is useful when the phone number is first transferred incompletely.

Settings of the Extension

Proceed as follows:

- Separately for PBX base numbers
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > Extension numbers PTP connection**)

Requirements:

- Configured PBX base number

EXTENSION NUMBERS PTP CONNECTION/BUNDLING

Select the extension number/bundle for that the extensions are to be displayed/configured, provided that more then one PBX base number was configured.

Extension (DDI number)

If you want an extension number to be different from the ones used in linear call distribution, or if an extension number is needed for a different function (e.g. remote configuration, CLIP no screening), you must also enter it in the phone number list (click on + **New**). You must enter all the extension numbers for a PTP connection that is not running linear call distribution.

Note: You can input up to 250 external numbers for land line and VoIP connections in the PBX. In other words, the total maximum number of MSNs, DDI numbers and additional numbers is 250.

Phone number for CLIP no screening

Requirements:

- Set up phone number
- CLIP no screening switched on for the ISDN connection

Customer-specific phone number (e.g. a service phone number) that is transferred to the destination by the exchange line without its correctness being checked.

Name

The name is used for display purposes on the system phone.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

Recording on

Requirements:

- Configured voice mailbox for call recordings (page **Devices > Voice mail and fax boxes > Phone numbers**). In order to see the box here, no owner must be assigned to the box.

Selection of the voice mailbox, which will be used for the function **Call recording**.

A voice mailbox for the function **Call recording** can be assigned to only one connection **Analogue / ISDN / VoIP**.

Note: If the voice mailbox for call recordings is removed from the list (Selection of **---**) and becomes a box with the function as answering machine, all recordings are deleted from this box after confirmation.

Note: The function **Call recording** is available only in certain countries.

New

Adds a new phone number.

Delete

Deletes the entries marked in the list.

Settings Additional Numbers ISDN

Proceed as follows:

- Separately for PBX base numbers
- Using the Configuration Manager (page **Public switched tel. networks > ISDN > Additional PTP connection numbers**)

Requirements:

- Configured PBX base number

ADDITIONAL NUMBERS PTP CONNECTION

Select the extension number/bundle for that the additional numbers are to be displayed/configured, provided that more then one PBX base number was configured.

Additional number

Some countries provide additional numbers for a PTP connection, in addition to the DDI numbers. The network provider can assign up to ten phone numbers for the connection.

The additional number specifies which character string a caller must dial from the local phone network to reach the ISDN PTP connection. These numbers are not attached to the PBX base number.

Note: You can input up to 250 external numbers for land line and VoIP connections in the PBX. In other words, the total maximum number of MSNs, DDI numbers and additional numbers is 250.

Phone number for CLIP no screening

Requirements:

- Set up phone number
- CLIP no screening switched on for the ISDN connection
- The exchange line must support CLIP no screening

Customer-specific phone number (e.g. a service phone number) that is transferred to the destination by the exchange line without its correctness being checked.

Name

The name is used for display purposes on the system phone.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

New

Adds a new phone number.

Delete

Deletes the entries marked in the list.

VoIP

Management of the VoIP Provider
Settings of the VoIP Provider
Management VoIP Account
Settings for the VoIP Accounts
Expert Settings of the PTMP Connection
Expert Settings of the PTP Connection
Settings Additional Numbers VoIP

Management of the VoIP Provider

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > VoIP > Provider**)

Configure

Opens the configuration of the VoIP provider.

Further help under Settings of the VoIP Provider

New

Adds a new provider to the list of providers. You must then make the settings for the VoIP provider.

Delete

Deletes the entries marked in the list.

Import

IMPORT PROVIDER

Requirements:

- PC or PBX with existing connection to the Internet

A file with individual provider access data (*.XML) will be downloaded from the update server. Depending on the Internet connection, this file is then either saved to a data storage device (PC) or directly to the PBX.

Note: The file saved to a (PC) data storage device can then be imported.

Note: The access data provided for the download is for information purposes only, and is not mandatory.

Export

The list of providers is saved to the (PC) data storage device in a file called provider.tgz.

Copy

Selects an existing provider from the list of providers and uses that provider as the basis for creating a new VoIP provider.

Settings of the VoIP Provider

SIP
RTP
Preferences
Number Presentation (outgoing)
Evaluation of Incoming SIP Messages

SIP

Proceed as follows:

- Separately for VoIP providers
- Using the Configuration Manager (page **Public switched tel. networks > Voice over IP (VoIP page) > Provider > Configure**)

Note: You can obtain the data for the settings either directly from the VoIP provider or from suitable lists in the Internet.

Requirements:

- For entering a URL: configured DNS server
- For using a STUN server: IP operating mode IPv4

PROVIDER

Select the provider to be displayed/configured.

Domain

The domain (also known as a "realm", "SIP domain" or "SIP realm") is required for the VoIP address. The structure of a subscriber's VoIP address is similar to an e-mail address. It is made up of the VoIP phone number and the domain, separated by the @ character: <subscriber>@domain.

Name resolution

There are both country-specific and provider-specific reasons why name resolution is not always successful via the default gateway. In such cases, another way should be chosen here.

Name resolution via IP configuration

There are several accounts at this domain and the name resolution for another account at this domain has alreday been configured.

Note: It is possible to set the name resolution differently for the accounts of the same domain.

Via used Internet access

The name resolution takes place via the selected IAD under **Used Internet access**. The DNS server provided by the provider is used.

Use own DNS (via IAD)

The DNS servers that are entered below are used via the selected IAD under **Used Internet access**.

First DNS server/Second DNS server

IPv4: 4 blocks with 3 digits separated by a dot (.).

IPv6: 8 blocks with 4 digits or letters separated by a colon (:).

Note: If an IPv6 address is entered as DNS server, then the IAD selected under **Used Internet access** must also have an IPv6 address.

Used Internet access

Requirements:

- If using both network interfaces: Configured IAD for this provider (page **Administration > VoIP Internet gateway**)

Note: A default IAD with the gateway of the first network interface is automatically created. If you only want use the first network interface, you

can select the default IAD. A new entry under **Administration > VoIP Internet gateway** is then optional.

Select the Internet access for this provider.

Registrar | Port

REGISTER requests are sent to the specified registrar (also known as a SIP registrar, registry, SIP server, SIP registry server). Enter this data as an IP address or a URL.

Note: The page **Overviews > Ports** shows an overview of the PBX ports (incoming and outgoing).

Note: You should only switch off the registrar if the VoIP provider requires you to do so. No NAT Keep-Alive is performed if the registrar is disabled. An internal network is usually protected from external manipulation by, for example, a firewall integrated in the router. Without NAT Keep-Alive, the firewall's security functions will defend you against attempted accesses from the Internet. For this reason, you may need to implement port forwarding for incoming SIP packets on the SIP port set up in the VoIP provider's configuration.

Caution: Each time a port is opened on the router, this creates a potential source of danger.

- It is essential you put additional protective measures in place.

Time limit for registration (1 to 60 min.)

The registration time specifies how many minutes the PBX waits before reregistering itself with the VoIP provider.

Note: In the test phase, after configuring the VoIP provider, we recommend you enter a low value (e.g. three minutes). After completing the test phase, you can enter a much higher value, since some VoIP providers also reject calls if too many attempts are made to register.

SIP registration in accordance with RFC 6140

The SIP protocol SIPconnect 1.1 is used by the PBX.

NAT traversal

NAT (Network Address Translation) enables you to convert a particular IP address that is used within a network (e.g. a local network) into a different IP address that can be used by a different network (e.g. the public switched phone network). If NAT traversal is switched on, and a query is sent from a local IP address to the public network, the sending IP address is swapped with the public IP address. This function is performed in the reverse direction for the reply.

Disabled (use local address

NAT traversal is not performed by the PBX. The router which connects the local area network with the Internet should be a properly functioning, SIP-aware router which performs NAT traversal.

Enabled with use of STUN

NAT traversal is performed by the PBX. To achieve this, a STUN server must also be specified for the VoIP provider.

Enabled

NAT traversal is performed by the PBX. To achieve this, the registrar for the VoIP provider must be switched on. The SIP request also contains an "rport" part in which the IP addresses used here (local, public) are transported.

Note: If **Enabled** is selected: If problems occur with single-sided call connections, use **Enabled with use of STUN** instead.

Caution: Each time a port is opened on the NAT router, this creates a potential source of danger.

- It is essential you put additional protective measures in place.

Use static public IP address

Requirements:

- Registered fixed IP address (see Preferences)

NAT traversal is performed by the PBX. The VoIP provider requires a connection via a static public IP address.

If the VoIP provider still requires this static IP address in a header, select the appropriate option:

- **Use static public ID address in VIA Header**
- **Use static public IP address in Contact Header**

Interval for NAT Keep-Alive enabled

After the time specified here has passed, NAT Keep-Alive packets are sent to the Firewall to maintain the NAT mapping.

Note: Some VoIP providers block accounts if the **Interval for NAT Keep-Alive** is too short. As a rule, this is reported with error message 503 during SIP registration. If you encounter this problem, we recommend you set a higher value (e.g. 180).

Outbound proxy

The outbound proxy is an intermediate server that processes all VoIP requests and connections going to the provider (except for registration).

automatically

The PBX automatically determines an outbound proxy.

Manual Port

The outbound proxy is a fixed default setting. Enter this data as an IP address or a URL.

Note: The **Overviews > Ports** page shows an overview of the PBX ports (incoming and outgoing).

SIP port (1 to 65535)

The SIP port on the PBX is used both as an incoming port and an outgoing port (i.e. for communications with the VoIP provider).

Note: A different SIP port must be used for each VoIP provider.

Note: The SIP port for a VoIP provider cannot be the same as the **SIP port of the internal VoIP registrar** or the **SIPS port of the internal VoIP registrar**.

Note: The **Overviews > Ports** page shows an overview of the PBX ports (incoming and outgoing).

SIP session timer (5 to 60 min.)

At the specified intervals, the SIP session timer checks whether the connection is still present. (The PBX cannot detect whether a call connection is disrupted due to problems on the provider side.)

Note: When the SIP session timer is switched on, this may result in the call being interrupted more frequently after the specified interval, if a VoIP provider has not implemented session renewal properly. In this case, you should adjust the interval of the SIP session timer or switch it off.

Deactivation of SIP Prack (RFC 3262)

The negotiation of SIP option „100rel“ is disabled. This might be necessary if the provider supports „SIP-Forking“.

Only en-bloc dialling | Delay time after last digit dialled (3 to 15 sec.)

A connection is not set up to transmit the digits dialled here to the provider until the delay time after the last digit is dialled has passed. All the digits are then sent as a block. Time measurement restarts after the entry of each digit. After the selected time has passed, the PBX recognises the most recent digit as the final digit and will not allow any more numbers to be entered.

Note: En-bloc dialling is a good idea, for example, when dialling single digits causes problems in sub-system operation.

Note: Select as short a time as possible, because the PBX does not forward the entire phone number to the public exchange until the specified time has passed.

Note: Approximately three seconds is the default value for manual inputs.

SIPS URL structure

Requirements:

- External VoIP channel that supports SIPS (optional, see Technical Data in the Instructions)
- A certificate for the VoIP provider stored in the PBX

This provider encrypts external connections to prevent listening in. The processes used to set up and shut down a connection, and signalling, are all encrypted by SIPS.

SIPS

Encryption of SIPS via TLS.

SIP/TLS

Encryption of SIP via TLS.

Note: SRTP is used to encrypt call data.

Disable host name check

No verification whether the certificate belongs to the domain/IP.

Manage certificates

Opens the certificates management.

SIP transport

Selection of the transport protocol.

IP protocol

IPv4

IPv4 operating mode is used for communications with the VoIP provider.

IPv6

IPv6 operating mode is used for communications with the VoIP provider.

automatically

The PBX first checks whether the VoIP provider supports IPv6. If so, operating mode IPv6 is used for communications with the VoIP provider. Otherwise, operating mode IPv4 is used.

Note: If the actual IP operating mode is IPv6, STUN cannot be used as a NAT method either for SIP or RTP.

Increased security

The provider provides better levels of security to protect the PBX against external attacks.

RTP

Proceed as follows:

- Separately for VoIP providers
- Using the Configuration Manager (**Public switched tel. networks > Voice over IP (VoIP page) > Provider > Configure**)

Note: You can obtain the data for the settings either directly from the VoIP provider or from suitable lists in the Internet.

Requirements:

- For entering a URL: configured DNS server
- For using a STUN server: IP operating mode IPv4

PROVIDER

Select the provider to be displayed/configured.

Maximum number of VoIP channels provided by the VoIP provider

Number of VoIP channels provided by the VoIP provider. If these channels are occupied additional connections required are made using another provider.

VoIP channels reserved for incoming calls

VoIP channels reserved for outgoing calls

Number of VoIP channels provided by the VoIP provider to be used only for incoming/outgoing calls.

SRTP

Requirements:

- external VoIP channel that supports SRTP (optional, see Technical Data in the Instructions)

Disabled

Forces disabling of call data encryption by SRTP. The connection is not established if the call partner (VoIP provider, other PBX in sub-system operation or external VoIP subscriber) requires call data to be encrypted using SRTP.

preferred

Switches on negotiation for the encryption of call data by SRTP. When a call is made, the call partner will be asked if encryption is possible. If so, the call data will be encrypted and then transmitted, if not, the call data will not be encrypted.

required

Forces enabling of call data encryption by SRTP. The connection is not established if the call partner (VoIP provider, other PBX in sub-system operation or external VoIP subscriber) does not require call data to be encrypted using SRTP.

Note: SIPS should also be enabled at the same time, otherwise the key for SRTP encryption would be readable.

NAT traversal

NAT (Network Address Translation) enables you to convert a particular IP address that is used within a network (e.g. a local network) into a different IP address that can be used by a different network (e.g. the public switched phone network). If NAT traversal is switched on, and a query is

sent from a local IP address to the public network, the sending IP address is swapped with the public IP address. This function is performed in the reverse direction for the reply.

Disabled (use local address): NAT traversal is not performed by the PBX. The router which connects the local area network with the Internet should be a properly functioning, SIP-aware router which performs NAT traversal.

Enabled with use of STUN: NAT traversal is performed by the PBX. To achieve this, a STUN server must also be specified for the VoIP provider.

Caution: Each time a port is opened on the NAT router, this creates a potential source of danger.

- It is essential you put additional protective measures in place.

Use static public IP address

Requirements:

- Registered fixed IP address (see Preferences)

NAT traversal is performed by the PBX. The VoIP provider requires a connection via a static public IP address.

DTMF signalling

Requirements:

- Inband signalling: uncompressed codec (G.711)

Inband

The PBX uses the same channel to transfer the DTMF signals and the voice data (DTMF tones).

Outband (RFC 2833)

The PBX uses different channels to transfer the DTMF signals and the voice data. The DTMF tones are filtered out of the voice data.

Both procedures

The PBX transmits the DTMF signals on two channels (1st contains DTMF tones together with the voice data + 2nd contains the DTMF signal).

Echo Cancellation

This compensates for local echoes and reverberation effects.

Silence Suppression (Voice Activity Detection)

This function allows to add the header SilenceSupp: on/off to the SDP negotiation of the SIP packets.

If **Automatically** was selected (factory settings), the header is not set.

The options **On** or **Off** should only tentatively be used for known problems with silent suppression.

Otherwise always select **Automatically**.

Comfort Noise Support

This function allows to decide whether comfort noise is offered during the SDP media negotiation (**On**) or not (**Off**).

This function is only needed in case of compatibility problems with comfort noise. In this case comfort noise support should be switched off. Otherwise always select **Automatically** (factory settings).

Jitter buffer (40 to 160)

The size of the jitter buffer specifies how many RTP packets can be cached, to buffer disruptions or compensate for them.

maxptime

Specifies the maximum size of a received packet for RTP supported by the PBX (30 ms recommended).

Suppressing

Suppresses the output of the maxptime header.

Automatically

The PBX attempts to determine the required value itself.

Use the exchange line replacement tone

The PBX generates the exchange line replacement tone in the case, for example, of longer dialling phases during which the PBX has not yet sent a call tone. If you find this disruptive or annoying, it can be switched off.

Automatically

The PBX generates the exchange line replacement tone in the case of longer dialling phases during which there is no call tone.

Off

No exchange line replacement tone is generated. No acoustic acknowledgement during longer dialling phases during which there is no call tone.

Codec settings

Requirements:

- external VoIP channel that supports different codecs (optional, see Technical Data in the Instructions)
- VoIP provider who supports the different codecs

The PBX makes various codecs available. The selection of a codec affects the quality of a VoIP call. Different codecs can be configured, depending on the available connection bandwidth, ranging from codecs with the best possible VoIP call quality down to codecs with high compression (low bandwidth). The codec actually used for a call is not fixed until the codec is negotiated with the VoIP provider.

Best available quality

Selects a codec sequence with best possible VoIP call quality (high bandwidth) as highest priority.

Good compromise

Selects a compromise between VoIP call quality (high bandwidth) and compression (lower bandwidth) for the codec sequence.

Best possible compression

Selects a codec sequence with stronger compression (lower bandwidth) as highest priority.

Note: The preset codec sequence can be changed manually if necessary.

Preferences

Proceed as follows:

- Separately for VoIP providers
- Using the Configuration Manager (**Public switched tel. networks > Voice over IP (VoIP page) > Provider > Configure**)

Note: You can obtain the data for the settings either directly from the VoIP provider or from suitable lists in the Internet.

Requirements:

- For entering a URL: configured DNS server
- For using a STUN server: IP operating mode IPv4

PROVIDER

Select the provider to be displayed/configured.

STUN server

IP address or URL | **Port**

Interval for STUN server query

Requirements:

- IP operating mode IPv4
- NAT traversal**Enabled with use of STUN**

A STUN server provides information to subscribers on a private network on request. This information includes the IP address and port outside their private network that is viewed as the source of their data. This information is entered in the requests, instead of the actual private IP address/port.

Note: If you encounter problems with unilateral call connections, enabling the RTP port might help.

Note: The **Overviews > Ports** page shows an overview of the PBX ports (incoming and outgoing).

Caution: Every time port forwarding is performed, there is a security risk.

- For this reason we recommend you use this functionality as little as possible.

Sub-system operation

Like any other PBX for which the VoIP provider has been configured, the PBX can be used as a sub-system.

Note: During sub-system operation, single-digit dialling may cause problems. In this case, en-bloc dialling is usually enabled.

Audio is switched through at call forwarding via 2nd B-channel

The connection between the caller and the PBX is always created when a call is forwarded externally. As a result, the caller will hear all replayed elements (tones, announcements) which are played by the VoIP provider when the connection with the call forwarding destination is established.

Note: If external call forwarding is configured over a VoIP provider, the call tone may not be replayed for the caller (the connection to the call forwarding destination number will therefore be switched through in a "surprising" way).

Audio is switched through when call starts

Requirements:

- The provider supports Early Media

Audio is switched through to the provider. This is useful if, for example, a callback request has to be conformed with "Yes".

EarlyMedia Support

During the call phase audio such as the dialling tone, signal sound, announcement or language can be transmitted (EarlyMedia). The following options define usage of **EarlyMedia** (RFC 3261) in general and the usage of the SIP Header **P-Early-Media** (RFC 5009).

Whether EarlyMedia or the SIP Header P-Early-Media is supported, is at the provider's discretion. It is possible that the provider allows the negotiation but prevents the transmission of audio.

The following options allow the adjustment of behaviour of the PBX to the one of the provider.

Switched off

During the call phase no audio is transmitted. But the status of the call is signalled, the end device of the caller then generates for example the dialling tone or the busy tone.

Announcements are as well suppressed. No tones are generated for the caller on incoming calls.

Outgoing without P-Early-Media support

During the call phase audio is negotiated and transmitted for outgoing calls.

The SIP Header P-Early-Media is not sent or evaluated.

Outgoing with P-Early-Media support (standard)

Audio is negotiated and transmitted for outgoing calls.

The SIP Header P-Early-Media is sent and evaluated.

Outgoing with P-Early-Media and incoming only if P-Early-Media is requested

The SIP Header P-Early-Media is sent and evaluated for outgoing calls.

The SIP Header P-Early-Media is only sent and evaluated for incoming calls, if it has been received. Accordingly, audio is transmitted.

This option is only reasonable for sub-system operation and special accounts.

Outgoing and incoming

There are no limitations regarding the negotiation of EarlyMedia for outgoing and incoming calls, the corresponding headers are considered.

Use URI parameter (user)

Transfers the "User=Phone" parameter to the SIP header.

Note: You may need to switch off this function if there are problems establishing the connection with a particular provider.

Use Mediasec header in acc. with IETF Draft

Transfers the SIP header in accordance with IETF draft "draft-dawes-sipcore-mediasec-parameter".

Note: Some providers require this optional SIP header for establishing and running encrypted VoIP connections.

The provider has emergency call ability

Requirements:

- the associated VoIP provider must allow emergency calls

It is possible to make an emergency call using this provider and the account assigned to it.

Note: When switched off, this provider does not support emergency calls. When you attempt to dial an emergency number, you will hear the announcement, "This phone does not allow emergency calls. Please use an alternative."

Deregister if there are NAT changes

The VoIP provider performs the deregistration and then the reregistration.

Some providers require this to take place every time a new public IP address has been assigned.

DNS query persists SIP session (RFC 3263)

Requirements:

- SIP stack type 2 selected

The SIP stack saves a resource record entry each for PTR, SRV and AAAA with that the preceded destination was reached. As long as the DNS server delivers this entry it will be prioritised.

SIP stack type

Type 1

Software module for the SIP log processing used previously.

Type 2

New extensive software module for the SIP log processing. This software module allows the support of future features of the VoIP providers.

Static public IP address

Some providers provide static public IP addresses and require them for call connection in the VIA header, the Contact header and in the SDP for the audio stream. Enter the provided IP address in the adjacent field and enable the option for the transmission (see previous options).

Support T.38 for providers T.38 negotiation

Requirements:

- external VoIP channel that supports T.38 (optional, see Technical Data in the Instructions)
- VoIP provider support for T.38
- Configured internal fax subscribers

T.38 (procedures for real-time Group 3 facsimile communication over IP networks) enables smooth, extensive fax transmission to take place.

Note: If problems occur when T.38 is switched on, you can change the setting used to negotiate the fax protocol (**T.38 negotiation**).

Compliant to RFC

Two separate media lines for disabling audio and negotiating the fax protocol.

Variant 1

Only one media line for negotiating the fax protocol (the media line for disabling audio is not used).

Number Presentation (outgoing)

Proceed as follows:

- Separately for VoIP providers
- Using the Configuration Manager (page **Public switched tel. networks > Voice over IP (VoIP page) > Provider > Configure**)

Note: You can obtain the data for the settings either directly from the VoIP provider or from suitable lists in the Internet.

Requirements:

- For entering a URL: configured DNS server
- For using a STUN server: IP operating mode IPv4

PROVIDER

Select the provider to be displayed/configured.

Format of called phone number

Format in which the VoIP provider requires the phone number of the destination, to enable the VoIP provider to transfer the call.

Country code with leading zeroes

The destination phone number is transmitted from the PBX to the VoIP provider with the entire country code (e.g. 0044).

Country code with leading + sign

The destination phone number is transmitted from the PBX to the VoIP provider with the entire country code in mobile telephony format (e.g. +44).

Country code without leading zeroes

The destination phone number is transmitted from the PBX to the VoIP provider with a shortened country code, without the international prefix (digit), (e.g. 44).

Without country code

The destination phone number is transmitted from the PBX to the VoIP provider without its own country code but with the local area code (e.g. 05306).

Send as dialled

The PBX transmits the destination phone number to the VoIP provider, as dialled.

Note: The PBX automatically converts the phone numbers dialled into this format for calls made via the VoIP provider.

Note: If prefixes are necessary for conversion, for dialling a number without prefixes, the prefixes set for the account are used (**Public switched tel. networks > Exchange line request**).

Format of own phone number

Format in which the VoIP provider requires the phone number that is to be transmitted.

Country code with leading zeroes

Your own external phone number, that is to be transferred,is sent to the VoIP provider with the entire country code (e.g. 0044).

Country code with leading + sign

Your own external phone number, that is to be transferred, is sent to the VoIP provider with the entire country code in mobile telephony format (e.g. +44).

Country code without leading zeroes

Your own external phone number, that is to be transferred, is sent to the VoIP provider with a shortened country code (without the international prefix (digit, e.g. 44).

Without country code

Your own external phone number, that is to be transferred, is sent to the VoIP provider without the country code, but with the local area code (e.g. 05306).

Only MSN or main number and DDI number

Your own external phone number, that is to be transferred, is sent to the VoIP provider without prefixes (e.g. 920082 or 92008279).

Only DDI number (only possible for PTP connection)

Your own external phone number, that is to be transferred, is sent to the VoIP provider without prefixes or the PBX base number (e.g. 8279).

Area code without leading zero

Your own external phone number, that is to be transferred, is sent to the VoIP provider with the local area code, but without a leading zero (e.g. 5306).

Note: The PBX checks whether any of the specified phone numbers are to be transferred, and if so which ones (**Subscriber (scr.) > Properties > Exchange line settings**).

Note: The transmitted phone number is automatically converted into the selected format. If prefixes are necessary for converting your own phone number, the prefixes set for the account are used (**Public switched tel. networks > VoIP > Accounts > Configure**). For outgoing dialling, the global settings under **Public switched tel. networks > Exchange line**

request are used. For incoming calls, the local area code of the account may be added.

Format of CLIP no screening Phone Number

Requirements:

- **CLIP no screening** for outgoing calls enabled (**VoIP > Accounts > Configure > LOCATION/MISCELLANEOUS > Number Presentation**)

Format in which the VoIP provider needs the telephone number to be transferred.

Method of number presentation suppression

Requirements:

- The VoIP provider supports the number presentation suppression service attribute
- Number presentation suppression is enabled for the subscriber

Area of the From header in which the VoIP provider expects to receive the number presentation suppression.

No display text

If number presentation suppression is switched on, an empty display text is sent in the From header.

Anonymous

The sent display text in the From header is "anonymous" if number presentation suppression is switched on.

User Anonymous

If number presentation suppression is switched on, both the display text and the User name section in the From header contain "anonymous".

Note: Number presentation suppression in accordance with RFC 3325, by using a privacy header, is also used automatically.

Evaluation of Incoming SIP Messages

Proceed as follows:

- Separately for VoIP providers
- Using the Configuration Manager (page **Public switched tel. networks > Voice over IP (VoIP page) > Provider > Configure**)

Note: You can obtain the data for the settings either directly from the VoIP provider or from suitable lists in the Internet.

Requirements:

- For entering a URL: configured DNS server
- For using a STUN server: IP operating mode IPv4

PROVIDER

Select the provider to be displayed/configured.

The phone numbers of incoming VoIP calls listed in the call data list usually need to be converted before they can be used for callbacks.

Note: To check these settings, direct incoming exchange line calls via this VoIP provider, e.g. using a mobile phone, and check the phone number sent along with the call in the call data list (in **User Data > Call data list**).

Note: To identify the user-defined settings, select an incoming call under **Monitoring > Network Data Stream** and view it in the Wireshark programme. Select the options you require in the list fields under **Search**

prioritisation (Wireshark term). The names ("terms") used for the search fields match the names used in Wireshark.

Note: The phone number to be evaluated should be as canonical as possible (+445306.... or 00445306…). Otherwise it is not used for evaluation.

Internationalise unknown phone numbers

If phone numbers that are not in the standard canonical format are received: (e.g. +445306… or 00445306…), they are converted into this format.

Evaluate Connected Party Number (COPN)

Enabling this function prevents that a direct callback fails due to an incorrect depiction of the phone number when the call is answered.

Set prefix for the phone number

Sets a prefix for the country code before the phone number (maximum 10 characters).

Source for called number

Default

Selects an evaluation method for phone numbers which will work for most providers.

Request Line

Forces the phone number to be evaluated using the Request Line.

To Header

Forces the phone number to be evaluated using the To Header.

Analysis type

Default

Selects an evaluation method for phone numbers which will work for most providers.

Default (using the From Header as a Screened Number)

Selects an evaluation method for phone numbers which will work for most providers. The user provided number (unscreened number, corresponds to CLIP no screening) in the "From" header is transferred to the network provided number (screened number). The phone numbers can be used, for example, for Call Through via VoIP.

Caution: As soon as you have enabled this option, your phone and PBX are open to attacks such as the transfer of manipulated phone numbers.

Management VoIP Account

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > Voice over IP (VoIP page) > Accounts**)

Provider name

VoIP provider with whom the account was set up.

Account name

Freely selectable name. This name is used to identify the account within the PBX.

Connection type

Connection type to be used with the account.

PTMP connection

VoIP account with multiple subscriber numbers (MSNs)

PTP connection

VoIP account with a PBX base number and extension numbers (DDI numbers)

Use

The PBX uses the account for external calls.

Configure

Opens the configuration of the VoIP account.

Further help under Settings for the VoIP Accounts

New

Adds a new account to the account list. You must then make the settings for the VoIP account.

Delete

Deletes the entries marked in the list.

Settings for the VoIP Accounts

Network Provider
Location/Access Data
Miscellaneous
Rufnummern am TK-Anlagenanschluss
Rufnummern am Mehrgeraeteanschluss

Network Provider

Proceed as follows:

- Separately for VoIP accounts
- Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > Configure**)

Note: You will receive the settings data from the VoIP provider with whom you have set up the account.

ACCOUNT

Select the account to be displayed/configured.

Account has emergency call ability

Requirements:

- VoIP provider has authorised and switched on emergency call ability.

This account allows emergency calls to be made.

Note: When switched off, this account cannot be used for emergency calls. When you attempt to dial an emergency number, you will hear the announcement, "This phone does not allow emergency calls. Please use an alternative."

Note: The phone allows emergency calls by default. Check you can actually use your account to make emergency calls. Not all network providers allow emergency calls.

Location/Access Data

Proceed as follows:

- Separately for VoIP accounts
- Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > Configure**)

Note: You will receive the settings data from the VoIP provider with whom you have set up the account.

ACCOUNT

Select the account to be displayed/configured.

Exchange line access number (account number)

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This is then used for the specific VoIP access point.

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

User name

User name assigned by the VoIP provider when the account was set up. It is used to identify the user to the VoIP provider.

Note: Note that some providers allocate both login information for the web page and separate login information for using the Voice over IP service (e.g. a dial-in password). Find out what information is needed for the dial-in and enter it accordingly.

Note: The user name is called different things by different providers. For example, the terms Username, Authorization User, SIP code/SIP password and SIP-ID are all used to describe a user name.

Note: If your provider (in Germany) is T-Online, you must also enter the VoIP phone number you received from them under **User name** and **Display name**.

Password

Password assigned by the VoIP provider when the account was set up. It is used to verify the user name.

Note: To change an existing password, click on **Show passwords**.

Note: Note that some providers allocate both login information for the web page and separate login information for using the Voice over IP service (e.g. a dial-in password). Find out what information is needed for the dial-in and enter it accordingly.

Note: The password is called different things by different providers. For example, it can also be called a customer password.

Caution: Visible passwords are a security risk.

- After you enter your password, click on **Hide passwords**.

Authorisation ID

Authorisation ID (authentication ID) assigned by the VoIP provider for the VoIP account.

Note: To change an existing password, click on **Show passwords**.

Note: You can only input an entry in this field if the VoIP provider has assigned an authorisation ID.

Caution: Visible passwords are a security risk.

- After you enter your password, click on **Hide passwords**.

Miscellaneous

Proceed as follows:

- Separately for VoIP accounts
- Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > Configure**)

Note: You will receive the settings data from the VoIP provider with whom you have set up the account.

ACCOUNT

Select the account to be displayed/configured.

Expert configuration Configure

Opens the expert settings page.

Rufnummern am TK-Anlagenanschluss

Proceed as follows:

- Separately for VoIP accounts
- Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > Configure**)

Note: You will receive the settings data from the VoIP provider with whom you have set up the account.

ACCOUNT

Select the account to be displayed/configured.

PBX base number

Extensions block | from | to

Requirements:

- Prefixes (country and local area code) configured for the VoIP account

A PTP connection has a PBX base number (base number, header, base phone number) and (Direct Dial In - DDI) extension numbers. The VoIP provider assigns the phone numbers for the connection. The phone number to be dialled is formed from the PBX base number + DDI number.

Note: You can enter a zero both as a single number (e.g. 0 to 99, as is usually the case) and as a digit in the set number of places in extension numbers (e.g. 00 to 99).

Note: If linear call distribution for the PTP connection is enabled (page **Public switched tel. networks > Call distribution**), and if the internal numbers match the external extension numbers, the system automatically performs a 1:1 assignment for the extension numbers in the extensions block. If an extension cannot be assigned, the fallback call distribution function is enabled.

Note: If you want an extension number to be different from the ones used in linear call distribution, or if an extension number is needed for a different function (e.g. remote configuration, CLIP no screening), you must also enter it in the phone number list (click on + **New**).

Note: For example, for extensions block 0 to 99: enter extension 0 as an extension and assign it in the call distribution.

Note: For example, for extensions block 00 to 99: Enter the individual extension numbers 00 to 09 as an extension number and assign them during call distribution.

Note: You must enter all the extension numbers for a PTP connection that is not running linear call distribution.

Ringtone for the above-mentioned PTP connection

The ringtone specifies which sequence of sounds is used to signal external calls to the phone numbers of the PTP connection to internal subscribers on the PBX.

Note: Individual ringtones can be assigned to extension numbers that were input in different ways.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

CLIP no screening

Requirements:

- CLIP no screening service feature supported and enabled by the network provider
- Type of number presentation configured for the VoIP provider **In accordance with RFC 3325 with P-Asserted-Identity** or **In accordance with RFC 3325 with P-Preferred-Identity**

The exchange line transfers the customer-specific phone number (e.g. a service phone number) entered in **Phone number for CLIP no screening** to the call destination without checking that this number is correct.

Note: In Germany, some providers only offer this service feature for PTP connections. The CLIP no screening function for VoIP connections also depends on which provider is being used.

Note: When an emergency number is dialled, the phone number received from the network provider is always transmitted, not the phone number you specified yourself.

Note: However, the correct phone number is also still transferred to the exchange line, but this cannot be viewed by the person being called.

Note: CLIP no screening must be enabled for every connection to ensure that the phone number for CLIP no screening is transmitted instead of the phone number received from the network provider.

Direct Dial In (DDI)

If you want an extension number to be different from the ones used in linear call distribution, or if an extension number is needed for a different function (e.g. remote configuration, CLIP no screening), you must also enter it in the phone number list. (Select + **New** to enter.) You must enter all the extension numbers for a PTP connection that is not running linear call distribution.

Note: You can input up to 250 external numbers for land line and VoIP connections in the PBX. In other words, the total maximum number of MSNs, DDI numbers and additional numbers is 250.

Phone number for CLIP no screening

Requirements:

- set up phone number
- CLIP no screening switched on for the VoIP account

Customer-specific phone number (e.g. a service phone number) that is transferred to the destination by the exchange line without its correctness being checked.

Note: The phone number must always be configured completely, thus, with the local area code or e. g. as 0800....

Display name

This name is used for display purposes on the system phone.

Note: If no display name has been entered, the account name is displayed.

Note: If your provider (in Germany) is T-Online, you must also enter the VoIP phone number you received from them under **Display name**.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

Recording on

Requirements:

- Configured voice mailbox for call recordings (page **Devices > Voice mail and fax boxes > Phone numbers**). In order to see the box here, no owner must be assigned to the box.

Selection of the voice mailbox, which will be used for the function **Call recording**.

A voice mailbox for the function **Call recording** can be assigned to only one connection **Analogue / ISDN / VoIP**.

Note: If the voice mailbox for call recordings is removed from the list (Selection of **---**) and becomes a box with the function as answering machine, all recordings are deleted from this box after confirmation.

Note: The function **Call recording** is available only in certain countries.

New

Adds a new phone number.

Delete

Deletes the entries marked in the list.

Rufnummern am Mehrgeraeteanschluss

Proceed as follows:

- Separately for VoIP accounts
- Using the Configuration Manager (page **Public switched tel. networks > VoIP > Accounts > Configure**)

Note: You will receive the settings data from the VoIP provider with whom you have set up the account.

ACCOUNT

Select the account to be displayed/configured.

PBX base number

Extensions block | from | to

Requirements:

- Prefixes (country and local area code) configured for the VoIP account

A PTP connection has a PBX base number (base number, header, base phone number) and (Direct Dial In - DDI) extension numbers. The VoIP provider assigns the phone numbers for the connection. The phone number to be dialled is formed from the PBX base number + DDI number.

Note: You can enter a zero both as a single number (e.g. 0 to 99, as is usually the case) and as a digit in the set number of places in extension numbers (e.g. 00 to 99).

Note: If linear call distribution for the PTP connection is enabled (page **Public switched tel. networks > Call distribution**), and if the internal numbers match the external extension numbers, the system automatically performs a 1:1 assignment for the extension numbers in the extensions block. If an extension cannot be assigned, the fallback call distribution function is enabled.

Note: If you want an extension number to be different from the ones used in linear call distribution, or if an extension number is needed for a different function (e.g. remote configuration, CLIP no screening), you must also enter it in the phone number list (click on + **New**).

Note: For example, for extensions block 0 to 99: enter extension 0 as an extension and assign it in the call distribution.

Note: For example, for extensions block 00 to 99: Enter the individual extension numbers 00 to 09 as an extension number and assign them during call distribution.

Note: You must enter all the extension numbers for a PTP connection that is not running linear call distribution.

Ringtone for the above-mentioned PTP connection

The ringtone specifies which sequence of sounds is used to signal external calls to the phone numbers of the PTP connection to internal subscribers on the PBX.

Note: Individual ringtones can be assigned to extension numbers that were input in different ways.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

CLIP no screening

Requirements:

- CLIP no screening service feature supported and enabled by the network provider
- Type of number presentation configured for the VoIP provider **In accordance with RFC 3325 with P-Asserted-Identity** or **In accordance with RFC 3325 with P-Preferred-Identity**

The exchange line transfers the customer-specific phone number (e.g. a service phone number) entered in **Phone number for CLIP no screening** to the call destination without checking that this number is correct.

Note: In Germany, some providers only offer this service feature for PTP connections. The CLIP no screening function for VoIP connections also depends on which provider is being used.

Note: When an emergency number is dialled, the phone number received from the network provider is always transmitted, not the phone number you specified yourself.

Note: However, the correct phone number is also still transferred to the exchange line, but this cannot be viewed by the person being called.

Note: CLIP no screening must be enabled for every connection to ensure that the phone number for CLIP no screening is transmitted instead of the phone number received from the network provider.

Direct Dial In (DDI)

If you want an extension number to be different from the ones used in linear call distribution, or if an extension number is needed for a different function (e.g. remote configuration, CLIP no screening), you must also enter it in the phone number list. (Select + **New** to enter.) You must enter all the extension numbers for a PTP connection that is not running linear call distribution.

Note: You can input up to 250 external numbers for land line and VoIP connections in the PBX. In other words, the total maximum number of MSNs, DDI numbers and additional numbers is 250.

Phone number for CLIP no screening

Requirements:

- set up phone number
- CLIP no screening switched on for the VoIP account

Customer-specific phone number (e.g. a service phone number) that is transferred to the destination by the exchange line without its correctness being checked.

Note: The phone number must always be configured completely, thus, with the local area code or e. g. as 0800....

Display name

This name is used for display purposes on the system phone.

Note: If no display name has been entered, the account name is displayed.

Note: If your provider (in Germany) is T-Online, you must also enter the VoIP phone number you received from them under **Display name**.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

Recording on

Requirements:

- Configured voice mailbox for call recordings (page **Devices > Voice mail and fax boxes > Phone numbers**). In order to see the box here, no owner must be assigned to the box.

Selection of the voice mailbox, which will be used for the function **Call recording**.

A voice mailbox for the function **Call recording** can be assigned to only one connection **Analogue / ISDN / VoIP**.

Note: If the voice mailbox for call recordings is removed from the list (Selection of **---**) and becomes a box with the function as answering machine, all recordings are deleted from this box after confirmation.

Note: The function **Call recording** is available only in certain countries.

New

Adds a new phone number.

Delete

Deletes the entries marked in the list.

Expert Settings of the PTMP Connection

VoIP Account
General Info
Routing
Call Forwarding

VoIP Account

Country settings for this exchane line

At configuration and relocation of the PBX, the country-specific settings are set automatically.

County settings for this exchange line

Requirements:

- Adopt settings from PBX location (Germany) disabled

Select the country for this exchange line.

General Info

Country code

International phone prefix (e. g. 0044 for the UK) for which the account's VoIP provider has provided the phone numbers.

Note: Land line connection prefixes are configured separately (**Public switched tel. networks > Exchange line request**).

Area code

Phone number prefix (e. g. 0208 for Greater London) for which the account's VoIP provider has provided the phone numbers.

Note: If there is no fixed digit sequence as the area code at the installation location of the PBX, no entry is required here.

Dial level

The selection of the dial level (Location, National, International) is country-specific. Depending on the country and the settings for the local area code above, a dial level must be selected (example: In Belgium, local area codes do not exist. Here, the dial level **Location** must not be selected).

Routing

Use Soft LCR 4.0 for this connection

Requirements:

- Network provider must support the dial-around service
- Configured prefixes (country prefix and local area code)
- Configured LCR (see LCR)

The VoIP connection is used for LCR calls.

Note: LCR must be disabled if the network provider does not support LCR.

Note: However, you can always perform LCR manually by dialling a provider area code prefix if the network provider supports the dial-around service.

LCR provider digit-recognition for this connection

Requirements:

- Disabled **Adopt settings from PBX location (Germany)**

The digits in the table below are the common LCR number prefixes in Germany. For other countries, more LCR number prefixes (a total of 10) can be entered.

Digit number	Digits begin with
5	010

6	0100

New

A new digit is added to the table.

Delete

Deletes the entries marked in the list.

Call Forwarding

GENERAL SETTINGS FOR CALL FORWARDING

Call forwarding method

Call forwardings at the exchange line are configured and performed in different ways. The exchange line, router or the provider determines which of the call forwardings listed below can be used.

Keypad

The call forwarding is controlled in the PTMP exchange line. The digit sequence depends on the provider or router.

Call Deflection

Call forwarding shall be executed directly by the exchange line. The PBX sends a corresponding message with new destination number to the exchange line. Accordingly, the call is forwarded to the new destination by the provider/exchange line.

Note: Currently, providers support Call Deflection only at the SIP Trunk.

2. B channel

The call forwarding is performed by the PBX during a call. It is possible for all exchange line types but requires a free voice channel and the number of the caller cannot be displayed unless CLIP no scrrening is used which is not supported by many providers.

For call forwarding via the PBX, use any available call channel

Requirements:

- Enabled call forwarding method **2. B channel**

When an external phone number is forwarded, the call can be forwarded via a call channel on any of the other ports.

Note: A call can only be redirected via a call channel on the same port. The probability that the call cannot be forwarded because the channels are busy is therefore higher.

Renew call forwarding at PBX reboot

When rebooting the PBX, the saved data of the call forwarding is renewed.

Important: In Switzerland, the configuration of a call forwarding is chargeable. To avoid costs, the check box must be disabled.

Delay time for Call Forwarding on no reply in the PBX

This delay time for external numbers specifies the time (in seconds) after which a call is forwarded if the called subscriber does not respond.

Note: The waiting time applies only to call forwarding that was set up using the PBX. If the network provider has configured the call forwarding function, the waiting time depends on their settings and cannot be configured in the PBX.

Routing via exception numbers (VoIP + GSM)

Requirements:

- Configured exception numbers (page **Routing > VoIP and GSM Routing > Exception numbers**)
- Enabled call forwarding method **2. B channel**

The device checks whether the destination number involved in call forwarding for external numbers has been set up as an exception number. If so, the call is sent along the configured path. If the destination number

has not been entered in the routing table, the call to be forwarded is sent either via a free call channel that is assigned to the port or via any free call channel (**For call forwarding via the PBX, use any available call channel**).

Use CLIP no screening for call forwarding

Requirements:

- CLIP no screening enabled for this VoIP account

In the case of call forwarding via the external connection, the phone number of the caller is transmitted to the destination of the call forwarding. In the case of an anonymous call, the function acts as if not enabled.

Note: When switched off, the phone number in the PBX is transmitted to the call forwarding destination.

CALL FORWARDING UNCONDITIONAL (CFU)

Call forwarding available in exchange line

If the exchange line has a CFU, this function can be enabled.

Note: If the exchange line does not have a CFU, this function must be disabled.

Keypad-/Registro code: Enable CFU

Keypad-/Registro code: Disable CFU

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters

- **,** (comma) for a 0,5 second pause
- **n** as a wildcard for the destination number (CDPN)
- **m** as a wildcard for the own phone number (MSN)

CALL FORWARDING IF BUSY (CFB)

Call forwarding available in exchange line

If the exchange line has a CFB, this function can be enabled.

Note: If the exchange line does not have a CFB, this function must be disabled.

Keypad-/Registro code: Enable CFB

Keypad-/Registro code: Disable CFB

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters
- **,** (comma) for a 0,5 second pause
- **n** as a wildcard for the destination number (CDPN)
- **m** as a wildcard for the own phone number (MSN)

CALL FORWARDING NO REPLY (CFNR)

Call forwarding available in exchange line

If the exchange line has a CFNR, this function can be enabled.

Note: If the exchange line does not have a CFNR, this function must be disabled.

Keypad-/Registro code: Enable CFNR

Keypad-/Registro code: Disable CFNR

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection

Possible entries:

- Digits: **0** to **9**
- The ***** and **#** characters
- **,** (comma) for a 0,5 second pause
- **n** as a wildcard for the destination number (CDPN)
- **m** as a wildcard for the own phone number (MSN)

Expert Settings of the PTP Connection

VoIP Account
General Info
Routing
Call Forwarding

VoIP Account

Country settings for this exchane line

At configuration and relocation of the PBX, the country-specific settings are set automatically.

County settings for this exchange line

Requirements:

- Adopt settings from PBX location (Germany) disabled

Select the country for this exchange line.

General Info

Country code

International phone prefix (e. g. 0044 for the UK) for which the account's VoIP provider has provided the phone numbers.

Note: Land line connection prefixes are configured separately (**Public switched tel. networks > Exchange line request**).

Area code

Phone number prefix (e. g. 0208 for Greater London) for which the account's VoIP provider has provided the phone numbers.

Note: If there is no fixed digit sequence as the area code at the installation location of the PBX, no entry is required here.

Dial level

The selection of the dial level (Location, National, International) is country-specific. Depending on the country and the settings for the local area code above, a dial level must be selected (example: In Belgium, local area codes do not exist. Here, the dial level **Location** must not be selected).

Routing

Use Soft LCR 4.0 for this connection

Requirements:

- Network provider must support the dial-around service
- Configured prefixes (country prefix and local area code)
- Configured LCR (see LCR)

The VoIP connection is used for LCR calls.

Note: LCR must be disabled if the network provider does not support LCR.

Note: However, you can always perform LCR manually by dialling a provider area code prefix if the network provider supports the dial-around service.

LCR provider digit-recognition for this connection

Requirements:

- Disabled **Adopt settings from PBX location (Germany)**

The digits in the table below are the common LCR number prefixes in Germany. For other countries, more LCR number prefixes (a total of 10) can be entered.

Digit number	Digits begin with
5	010

6	0100

New

A new digit is added to the table.

Delete

Deletes the entries marked in the list.

Call Forwarding

GENERAL SETTINGS FOR CALL FORWARDING

Call forwarding method

Call forwardings at the exchange line are configured and performed in different ways. The exchange line, router or the provider determines which of the call forwardings listed below can be used.

Keypad

The call forwarding is controlled in the PTMP exchange line. The digit sequence depends on the provider or router.

Call Deflection

Call forwarding shall be executed directly by the exchange line. The PBX sends a corresponding message with new destination number to the exchange line. Accordingly, the call is forwarded to the new destination by the provider/exchange line.

Note: Currently, providers support Call Deflection only at the SIP Trunk.

2. B channel

The call forwarding is performed by the PBX during a call. It is possible for all exchange line types but requires a free voice channel and the number of the caller cannot be displayed unless CLIP no scrrening is used which is not supported by many providers.

For call forwarding via the PBX, use any available call channel

Requirements:

- Enabled call forwarding method **2. B channel**

When an external phone number is forwarded, the call can be forwarded via a call channel on any of the other ports.

Note: A call can only be redirected via a call channel on the same port. The probability that the call cannot be forwarded because the channels are busy is therefore higher.

Renew call forwarding at PBX reboot

When rebooting the PBX, the saved data of the call forwarding is renewed.

Important: In Switzerland, the configuration of a call forwarding is chargeable. To avoid costs, the check box must be disabled.

Delay time for Call Forwarding on no reply in the PBX

This delay time for external numbers specifies the time (in seconds) after which a call is forwarded if the called subscriber does not respond.

Note: The waiting time applies only to call forwarding that was set up using the PBX. If the network provider has configured the call forwarding function, the waiting time depends on their settings and cannot be configured in the PBX.

Routing via exception numbers (VoIP + GSM)

Requirements:

- Configured exception numbers (page **Routing > VoIP and GSM Routing > Exception numbers**)
- Enabled call forwarding method **2. B channel**

The device checks whether the destination number involved in call forwarding for external numbers has been set up as an exception number. If so, the call is sent along the configured path. If the destination number

has not been entered in the routing table, the call to be forwarded is sent either via a free call channel that is assigned to the port or via any free call channel (**For call forwarding via the PBX, use any available call channel**).

Use CLIP no screening for call forwarding

Requirements:

- CLIP no screening enabled for this VoIP account

In the case of call forwarding via the external connection, the phone number of the caller is transmitted to the destination of the call forwarding. In the case of an anonymous call, the function acts as if not enabled.

Note: When switched off, the phone number in the PBX is transmitted to the call forwarding destination.

CALL FORWARDING UNCONDITIONAL (CFU)

Call forwarding available in exchange line

If the exchange line has a CFU, this function can be enabled.

Note: If the exchange line does not have a CFU, this function must be disabled.

CALL FORWARDING IF BUSY (CFB)

Call forwarding available in exchange line

If the exchange line has a CFB, this function can be enabled.

Note: If the exchange line does not have a CFB, this function must be disabled.

CALL FORWARDING NO REPLY (CFNR)

Call forwarding available in exchange line

If the exchange line has a CFNR, this function can be enabled.

Note: If the exchange line does not have a CFNR, this function must be disabled.

Settings Additional Numbers VoIP

Proceed as follows:

- Using the Configuration Manager (page **Public switched tel. networks > VoIP > Additional numbers PTP connection**)

Requirements:

- Configured PTP connection

ADDITIONAL NUMBERS PTP CONNECTION

Select the extension number/bundle for that the additional numbers are to be displayed/configured, provided that more then one PBX base number was configured.

Phone number for CLIP no screening

Requirements:

- Set up phone number
- CLIP no screening switched on for the VoIP connection
- The exchange line must support CLIP no screening

Customer-specific phone number (e.g. a service phone number) that is transferred to the destination by the exchange line without its correctness being checked.

Note: The phone number must always be configured completely, thus, with the local area code or e. g. as 0800....

Name

The name is used for display purposes on the system phone.

Ringtone

The ringtone specifies which sequence of sounds is used to signal external calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

New

Adds a new phone number.

Delete

Deletes the entries marked in the list.

Call Distribution

Management for the Call Distribution
Settings for Call Distribution

Management for the Call Distribution

Call distribution for

Call distribution for

Proceed as follows:

- Separately for configurations
- Separately for external connections
- Using the Configuration Manager (**Public Switched Telephone Networks > Call distribution** page)

Note: External numbers that are required as **Call Through number** (**Functions > Call Through**) are not allowed to be used in call distribution.

Note: In order for a subscriber to signal incoming external calls, it needs at least **Exchange line authorisation** (business) **Only incoming, with emergency call** (**Subscriber (scr.) > Properties > Exchange line settings**).

Note: An internal destination can be assigned to an external number. Internal destinations can be subscribers, groups, Voice mail and fax boxes or the automatic reception functions.

Note: If several subscribers ring simultaneously, they are grouped.

CALL DISTRIBUTION FOR

Select the connections to be displayed/configured.

A separate call distribution is set for each external connection. A single connection as is defined here as:

- A PTMP connection (fixed network or GSM)
- A single PTP connection (fixed network or GSM)

- A bundled PTP connection made up of several S_0 connections with the same PBX base numbers and the same extensions block
- The S_{2M}Connection
- An analogue connection
- A VoIP account

...ISDN PTMP connections

...VoIP PTMP connections

Internal no. | name

Fallback

The caller is transferred to the fallback function.

Internal number: The caller will be transferred to the selected internal number (subscriber, group, voice mail/fax box or automatic reception).

Note: External numbers, e.g., those used as remote switching/ programming numbers or call through numbers, may not be used in the call distribution.

...ISDN PTP connections / ...VoIP PTP connections

Internal no. | name

For a PTP connection without linear call distribution or for extensions differing from linear call distribution, the internal destinations must be selected.

Linear/fallback

The caller will be transferred to the internal number similar to the DDI. If this phone number is not configured in the PBX, the fallback function will be used.

Rejecting a call

The caller will hear the busy signal.

Internal phone number: The caller will be transferred to the selected internal number (subscriber, group, voice mail/fax box or automatic reception).

Note: External numbers, e.g., those used as remote switching/ programming numbers or call through numbers, may not be used in the call distribution.

...VoIP PTP connections / ...VoIP PTMP connections

SIP status on fallback

If an unassigned phone number in the PBX is required on an incoming external call, the caller will be transferred to the fallback function.

In this case, there are the following SIP status codes available for the transfer to the exchange line for a DDI (VoIP PTP connection) or an MSN (VoIP PTMP connection):

SIP status on fallback	Transmitted message	Meaning
Invalid number (404)	Not Found	The destination phone number has not been found or does not exist.
Temporarily not available (480)	Temporarily Unavailable	The called subscriber is temporarily not available.
busy	Busy Here	The called subscriber is busy

Settings for Call Distribution

Call Distribution for
Modem/Fax Call Distribution

Call Distribution for

Proceed as follows:

- Separately for configurations
- Separately for external connections
- Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)

CALL DISTRIBUTION FOR

Select the connections to be displayed/configured.

fallback

The fallback function indicates how the PBX deals with calls which could not be transferred or rejected via call distribution.

Rejecting a call

The caller will hear the busy signal.

If an unassigned phone number in the PBX is required on an incoming external call, the caller will be transferred to the fallback function, the following SIP status codes available for the transfer to the exchange line:

SIP status on fallback	Transmitted message	Meaning
Invalid number (404)	Not Found	The destination phone number has not been found or does not exist.
Temporarily not available (480)	Temporarily Unavailable	The called subscriber is temporarily not available.

SIP status on fallback	Transmitted message	Meaning
busy	Busy Here	The called subscriber is busy

Note: If a configured DDI (VoIP PTP connection) or an MSN (VoIP MPTP connection) is listet below, you can assign an individual SIP status code via **SIP status on fallback** .

Fallback settings: The caller will be transferred according to the settings.

- Only Announcement before Answering: The caller hears the Announcement before Answering and then, the connection is interrupted.
- Fallback phone number and no Announcement before Answering: The call is immediately signaled at the internal number for the fallback.
- Fallback phone number and Announcement before Answering:
  - **immediately**: The callIt is immediately signaled on the internal phone number for fallback and parallel to the announcement before answering.
  - **after Announcement before Answering**The call is signalled after the announcement before answering to the internal number for fallback and, thus, in parallel to the call phase.

Delayed fallback after

Requirements:

- Selected fallback number (see fallback)

All calls which have not been answered will be redirected after the configured time.

Note: The fallback time is the addition of the longest waiting time for the Call Forwarding on no reply and the seconds configured in **after:**. The longest waiting time will be determined by a comparison of the **Delay time for CF on no reply in the PBX** and the **Delay time for CF on no reply**.

Linear call distribution (external = internal)

Requirements:

- Configured connection type PTP connection

A 1:1 assignment for the extensions is made automatically for internal numbers and external extensions that are matching the extensions block.

Modem/Fax Call Distribution

Proceed as follows:

- Separately for configurations
- Separately for external connections
- Using the Configuration Manager (page **Public switched tel. networks > Call distribution**)

CALL DISTRIBUTION FOR

Select the connections to be displayed/configured.

Active

Requirements:

- Configured connection type analogue connections

If the active modem/fax call distribution is enabled, the PBX automatically accepts every external call to the corresponding phone number. Using the dial tones, it attempts to determine whether the incoming call is from a fax machine or from a modem. If the corresponding dial tones are identified within the detection time frame configured, the call is forwarded to the selected internal number (subscriber, group, voice mail/fax box).

No detection

There is no automatic modem/fax call distribution.

Fax

The active fax call distribution is enabled.

Fax and modem

The active modem/fax call distribution is enabled.

Passive

Requirements:

- Configured connection type analogue connections

An external call is transferred to an internal subscriber via the normal call distribution. If the user accepts the call and ends it within the configured detection time frame, the call is automatically transferred to the internal destination selected in the last column (subscriber, group, fax box).

Note: If you have selected the option **Fax** or **Fax and modem** in the **Active** list field, the setting for the **Passive** check box will be ignored and active modem/fax call distribution will be enabled.

Note: The setting in the **Detection duration** list field is not relevant for passive modem/fax call distribution.

Detection duration

Requirements:

- Configured connection type analogue connections

Time interval in seconds that the PBX has for making the identification of fax or modem.

Internal no. | name

Requirements:

- Configured connection type analogue connections

Fallback

The caller is transferred to the fallback function.

Internal number: The caller will be transferred to the selected internal number (subscriber, group, fax box.

Delayed fallback after

Requirements:

- Selected fallback number (see fallback)

All calls which have not been answered will be redirected after the configured time.

Note: The fallback time is the addition of the longest waiting time for the Call Forwarding on no reply and the seconds configured in **after:**. The longest waiting time will be determined by a comparison of the **Delay time for CF on no reply in the PBX** and the **Delay time for CF on no reply**.

Linear call distribution (external = internal)

Requirements:

- Configured connection type PTP connection

A 1:1 assignment for the extensions is made automatically for internal numbers and external extensions that are matching the extensions block.

Call Forwarding

Settings for the Call Forwarding

Settings for the Call Forwarding

Call Forwarding for ...

Call Forwarding for ...

Requirements:

- Configured external numbers

Proceed as follows:

- Separately for configurations
- Separately for external numbers
- Using the Configuration Manager (page **Public switched tel. networks > Call forwarding**)
- Using a character string on the internal phone (see Call Forwarding for External Numbers)
- Using the menu/function key on a system phone (optional, see the phone manual)

CALL FORWARDING FOR

Select the connections to be displayed/configured.

Permanent configuration

The configuration-dependent settings will be overridden by the permanent configuration settings.

Note: On a system phone, you can switch on/off/over the call forwarding of external numbers either for the current configuration or for permanent configuration using the function key (see the manual for the phone).

Note: With a standard phone you can change call forwarding in the permanent configuration and switch on the permanent configuration or the current configuration using a digit sequence.

You can override the configuration-dependent call forwarding by activating the **permanent configuration** option.

CF immediately

External calls to the external number are immediately forwarded to the entered external destination number.

CF on no reply

On no reply, external calls to the external number are forwarded to the entered external destination number.

CF on busy

On busy, external calls to the external number are forwarded to the entered external destination number.

Destination number

Destination number (external number or short-code number) to which external calls to the external number are to be forwarded.

Delete

Deletes the entry fields of the entries marked in the list. The entries themselves remain.

Subscriber (scr.)

Phone numbers
IP provisioning
Profiles
COMfortel WS-500x
Properties / Function overview
Unassigned phones
All subscribers

Phone numbers

Management Subscriber
Add Subscriber

Management Subscriber

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by another internal subscriber.

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Note: The internal number entered here needs to be entered in the ISDN device or in the system phone as the (first) MSN. Also on an ISDN device, make sure that the phone number is transmitted to the PBX.

Name

Freely selectable name. This name is used to identify the subscriber within the PBX and the connected system phones. The name can also be transferred to the VoIP end devices via LDAP server.

Module | Port

Module and port to which the end device (e. g. phone) is connected.

Device type

Device type depending on the end device connected.

Analogue end devices	
Phone	analogue phone, analogue modem
Answering machine	analogue answering machine
Fax machine	analogue fax machine
Door terminal	analogue door terminal, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of a door terminal
Announcement output	analogue announcement output, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of an announcement output

ISDN end devices	
ISDN system phone	System phone 1200/1400/1600/2600
ISDN phone	ISDN phone in compliance with the Euro ISDN Standard (DSS1)

ISDN end devices	
ISDN PC controller	ISDN PC controller in compliance with the Euro ISDN Standard (DSS1)
Fax machine	ISDN fax machine in compliance with the Euro ISDN Standard (DSS1)

VoIP end devices	
COMfortel IP	COMfortel 1400 IP, COMfortel 2600 IP, COMfortel 3200, COMfortel 3500, COMfortel 3600 IP
COMfortel 1200 IP	COMfortel 1200 IP
Standard VoIP phone	Standard VoIP end device (SIP) from other manufacturers, soft phone (SIP)
Handset on COMfortel WS-IP	Handset (e.g., COMfortel M-100/200/210/300/310) on a COMfortel WS-400 IP or COMfortel WS-650 IP
Handset on COMfortel WS-500	Handset COMfortel M-510/520/530 on a COMfortel WS-500S or WS-500M
COMfortel C-400	Conference phone COMfortel C-400
Door terminal	VoIP door terminal, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of a door terminal

Property | Configure

Opens the configuration of the subscriber.

Further help under Settings of the Subscriber Properties

Filter:

The list of subscribers is filtered according to the entered or selected value for **Phone number**, **Name** or **Module**.

+ New

Adds a new subscriber. You must then make the settings for this new subscriber.

Further help under Add Subscriber

Delete

Entries marked in the list are deleted.

Note: If a handset of the device type **COMfortel M-5x0** is deleted as a subscriber, this is also removed in the base station in which it was configured.

Add Subscriber

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > Phone numbers > New**)

Note: PINs and passwords are automatically assigned to the new subscriber by means of a random number generator (change via page **Subscriber (scr.) > Properties**). The following passwords must be adapted:

Note: The device type **Handset on COMfortel WS-IP** requires up to 32 characters (numbers and/or letters - no special characters!).

Add subscriber

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by another internal subscriber.

Several subscribers with a contiguous number range can be created in the fields **from to**.

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Note: The internal number entered here needs to be entered in the ISDN device or in the system phone as the (first) MSN. Also on an ISDN device, make sure that the phone number is transmitted to the PBX.

Name

Freely selectable name. This name is used to identify the subscriber within the PBX and the connected system phones. The name can also be transferred to the VoIP end devices via LDAP server.

Module / Port

Module and port to which the end device (e. g. phone) is to be connected.

Device type

Device type depending on the end device connected.

Analogue end devices	
Phone	analogue phone, analogue modem
Answering machine	analogue answering machine
Fax machine	analogue fax machine
Door terminal	analogue door terminal, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of a door terminal

Analogue end devices	
Announcement output	analogue announcement output, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of an announcement output

ISDN end devices	
ISDN system phone	System phone 1200/1400/1600/2600
ISDN phone	ISDN phone in compliance with the Euro ISDN Standard (DSS1)
ISDN PC controller	ISDN PC controller in compliance with the Euro ISDN Standard (DSS1)
Fax machine	ISDN fax machine in compliance with the Euro ISDN Standard (DSS1)

VoIP end devices	
COMfortel IP	COMfortel 1400 IP, COMfortel 2600 IP, COMfortel 3200, COMfortel 3500, COMfortel 3600 IP
COMfortel 1200 IP	COMfortel 1200 IP
Standard VoIP phone	Standard VoIP end device (SIP) from other manufacturers, soft phone (SIP)
Handset on COMfortel WS-IP	Handset (e.g., COMfortel M-100/200/210/300/310) on a COMfortel WS-400 IP or COMfortel WS-650 IP
Handset on COMfortel WS-500	Handset COMfortel M-510/520/530 on a COMfortel WS-500S or WS-500M
COMfortel C-400	Conference phone COMfortel C-400

VoIP end devices	
Door terminal	VoIP door terminal, selection deletes the subscriber and opens **Devices > Device wizard** for configuration of a door terminal

Assigning the MAC address of the device

The MAC address of a phone is needed to identify a phone that is assigned to a specific subscriber. For all transactions between phone and PBX the PBX must be able to identify the MAC address of the phone so that the phone can be matched to a subscriber and his/her provisioning data.

found devices

Select one of the existing MAC adresses.

assigned devices

MAC adresses already assigned to the subscribers.

manual assignement

Enter manually (only digits) or take over a MAC adress after reading with a bar code scanner.

The MAC address of a device can be found as follows:

- Retrieving from a client certificate sended by the phone
- Reading with a bar code scanner enabled and connected to the computer (e. g. via a USB interface)
- Reading e. g. on the menu oft the phone or on a label

Note: If a MAC address has been entered and assigned to a subscriber, the corresponding phone can already be provided with provisioning data during its first boot process.

Provisioning file

The subscriber receives the required configuration via the provisioning file selected here.

Note: A configuration can also be assigned to this and further ssubscribers via the page **Subscriber (scr.) > IP Provisioning > File management**.

Note: If a required file is not yet available here, a file must first be created on the page **Subscriber (scr.) > IP Provisioning > File management**. There, the file can be assigned to one or more subscribers.

Manufacturer name or product name

Provisioning file already assigned to the manufacturer name/product name.

Other

All other available files.

Files not used by other devices

List of files not yet assigend to a device.

IPUI of the device (only COMfortel M-5x0)

To identify a handset, the IPUI of the handset is used. For all transactions between the handset, base station and PBX, the PBX must be able to determine the IPUI of the handset and use it to assign a handset and its provisioning data to the handset.

already used IPUI

IPUI already assigned to another subscriber.

free IPUI

Slection of an unassigned IPUI.

manual assignment

Manual input (digits or letters) or transfer of an IPUI. This can be determined as follows:

- Read out the barcode label on the package with a barcode reader connected to the computer (e. g. via the USB interface)
- Reading from the handset: Press the control button and enter ***#06#**
- Reading from the package label

Note: If an IPUI is registered and assigned to a subscriber, the corresponding handset can alreday be supplied with provisioning data during the first boot process.

COMfortel WS-500 Base Station (only COMfortel M-5x0)

The subscriber is assigned to a base station here.

Note: This and other subscribers can also be assigned a base station via the page **Subscriber (scr.) > COMfortel WS-500x > COMfortel M-5x0**.

IP provisioning

Settings
File management

Settings

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > IP provisioning > Settings**)

Requirements:

- Automatic provisioning via the provisioning server at the phone(s) switche on

Provisioning server active

IP phones can automactically be configured as provisioning server via the PBX.

Note: When switched off, no automatic configuration of the IP phones is carried out.

Auto Provisioning active

Requirements:

- Activated **SIP Komfortpaket brand plus** (page **Administration > Activation > Open Shop**)

The activation enables the provisioning of SIP end devices from other manufacturers.

Access without client certificate (unsafe operation)

There are IP phones that have a client certificate. Thus, the identity of a subscriber is bound to a unique digital certificate. This way, it can be determined who for example can access the subscriber data.

Note: Only enable this function if you operate phones on the PBX that do not have an own client certificate.

Client certificates are stored in the PBX for the following IP phones:

- COMfortel 1200 IP
- COMfortel 1400 IP/2600 IP/3600 IP
- Snom IP phones
- Yealink IP phones

Provisioning only within local network

Provisioning is only carried out within the local network.

Note: When switched off, provisioning can also be carried out on other possible networks (unsafe operation).

Phone language after provisioning

Phone language that is displayed on the phone after provisioning.

File management

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > IP provisioning > File Management**)

DEVICE TYPES

Select the device to be configured.

If there is no file available for a device type which means that the list for the provisioning files is still empty, a new provisioning file must be created via **New**.

If a template exists for a device type, it is copied to the new file.

Editor presetting

File editor

Via the file editor, settings (e. g. NTP or contacts) are directly made in the provisioning file (*.xml, *.cfg).

Provisioning file (*.xml, *.cfg) > Edit

Variables for the parameters to be provisioned are inserted as palceholders in the configuration files provided by the manufacturer. During provisioning, these are replaced by the corresponding values in the PBX configuration.

Changes in the settings are effective after saving.

COMfortel IP Editor

The COMfortel IP Editor simplifies the provisioning settings (e. g. NTP or function keys) and is only available for the COMfortel xx00 IP devices.The settings are adopted to the provisioning file after saving.

Note: When opening the COMfortel IP Editor, it is automatically checked if a new version is available when the Internet access is available. An update will be carried put if necessary.

Note: If there is no Internet connection, please check the HTTP proxy of your PBX (page **Administration > IP configuration**).

Note: The standard.xml cannot be changed. To edit the file, save it under a different name and assign the devices accordingly.

Provisioning file (*.xml, *.cfg) > Edit

Changes in the setting in the COMfortel IP Editor are transferred to the provisioning file after saving.

New

Creating a new provisioning file and selection of the device type for which this provisioning file is to be used.

Delete

Note: The default provisioning file Standard.xml cannot be deleted.

Import

Importable file types:

- *.xml
- *.cfg
- *.zip

Export

A provisioning file stored in the PBX can be saved on the PC as backup copy.

Note: It is possbile to save more than one provisioning file at the same time. To do this, select the required provisioning files and click **Export**. The provsioning files are generated in a provisioning.zip folder.

Use for device type

Selection of the device type for which this provisioning file is to be used.

Provisioning file > Authorizations

Provisioning files can partly contain sensitive subscriber-specific data. To prevent unauthorised access, subscribers or base stations can be specifically granted or denied access to provisioning files.

Granting access to	Proceed as follows:
individual subscribers / base station	Mark subscriber under **available subscribers** or base station under **available bases** and click > .
all subscribers / base stations	Click >> .
all subscribers of a group	Select group under **Authorisation for all subscribers to the group** .

Header file

Use file as header file.

Note: Does not apply to device type COMfortel WS-500.

Standard.xml > View

An empty provisioning file (Standard.xml) is provided as default file. The Standard.xml file cannot be deleted or edited in the integrated file editor, but it can be viewed.

Provisioning file > Edit

For editing provisioning files (*.xml, *.cfg) the file editor is available for provisioning files from other manufacturers (e.g. Snom or Yealink) and for COMfortel IP phones additionally the COMfortel IP Editor.

Note: The Standard.xml cannot be edited.

Note: We recommend to contact the system administrator or specialised dealer for editing provisioning files.

Note: Under **Account IDs** you find the account IDs for the PBX and the exchange lines.

Further help in the Auerswald DokuWiki.

Profiles

Management Profile
Settings of the Subscriber Profiles
Settings Profile Assignment

Management Profile

Proceed as follows:

- Using the Configuration Manager

Profile name

The available profiles (default profiles, user-defined profiles) are displayed.

View

The selected settings can be viewed (default profiles).

Configure

The user-defined profiles can be adapted to the corresponding requirements (user-defined profiles).

+ New

Creation of a new profile.

Settings of the Subscriber Profiles

Exchange Line Settings
Reachability
Authorisations
Voicemail/ Fax Box
Settings

Exchange Line Settings

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Subscriber (scr.) > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Subscriber (scr.) > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the subscriber setting.

Note: When switched off, the setting under **Subscriber (scr.) > Properties** applies.

Requirements:

- For differently configurable functions for private and business calls: function activated
  - Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)

Exchange line authorisation - business | Expert / Exchange line authorisation - private calls | Expert

Defines which kind of phone numbers may be dialled with a subscriber.

International

All international and domestic telephone numbers can be dialled.

National

All domestic telephone numbers can be dialled.

City

All telephone numbers without a provider area code can be dialled.

Note: If GSM access is used for an outgoing call, the PBX automatically dials the telephone number with the configured area code (national destination code).

Only incoming

Incoming external calls can be accepted.

Only internal

No external calls can be made/answered.

exchange line dependent

The exchange line authorisation is configured separately for each individual external connection via **Expert** .

Preventing emergency calls

Emergency calls cannot be made. The subscriber hears a corresponding announcement. This function is for example useful for remote extensions as the emergency call is issued at the location of the PBX and not at the location of the remote extension.

Call restrictor - (outgoing) business calls

Call restrictor - (outgoing) private calls

Requirements:

- Configured call restrictor
  - Using the Configuration Manager (**User data > Special numbers** page)

The subscriber can not dial the phone numbers entered in the call restrictor irrespective of his exchange line authorisation. So it restricts the exchange line authorisation.

Call deblocker - (outgoing) business calls

Call deblocker - (outgoing) private calls

Requirements:

- Configured call deblocker
  - Using the Configuration Manager (**User data > Special numbers** page)

The subscriber can dial the phone numbers entered in the call deblocker irrespective of his exchange line authorisation. So it extends the exchange line authorisation.

Preferred exchange line - business calls | Expert / Preferred exchange line - private calls | Expert

Defines via which exchange line outgoing external calls will be routed.

Automatic occupation

For outgoing external calls, any available connection is used, whereby first the fixed network connections and then the VoIP connection and at last the GSM connections were taken into account.

Special preferred exchange lines

For outgoing external calls, the connections released under **Expert** are used depending on the selected priority. This makes the definition of a preferred exchange line (priority = 1) with overflow to other connections (priority > 1) possible. Exchange lines with priority **No automatic occupation** are not used for outgoing calls.

Note: If all released connections are busy, it is no longer possible to call. The subscriber can only use a connection with the setting **No automatic occupation** by entering a character string (see).

Note: The same priority may be allocated to different external connections in parallel.

Number presentation - (outgoing) business | Expert / Number presentation - (outgoing) private | Expert

Defines which phone number is presented.

No announcement

The phone number is suppressed.

Default

Both the DDI parallel to the internal number and the MSN on the PTM connection over which the subscriber is called are presented on the PTMP connection. (if the subscriber is not a member of the call distribution for this connection, the first MSN is presented).

exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert** .

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Note: On a connection with **CLIP no screening** enabled, the number presentation defined here is not considered.

Note: Notes for suppressing number presentation

Note: Phone number suppression for VoIP calls is possible but it is not always supported by the providers.

Note: If an emergency number is called while phone number suppression is enabled, phone number suppression is disabled once so that the emergency service can identify the caller.

Note: If the number presentation only needs to be suppressed call by call, the **Number presentation (outgoing) Default** or **exchange line dependent** . For the **exchange line dependent** option, each exchange line can have an individual number presented to the network provider or to the caller separately. The number presentation to the caller is suppressed by the phone before starting a call. The number presentation to the network provider is not suppressed by the phone.

Note: If the number presentation should always be suppressed, the **Number presentation (outgoing) exchange line dependent** . For each exchange line can a separate number be presented to the network

provider. To keep these numbers from being presented to the caller, number presentation can be suppressed per exchange line separately.

Note: If number presentation needs to be suppressed between the PBX and the network provider, the **Number presentation (outgoing) No announcement** .

Note: For the option **No announcement** , the phone service **Calling Line Identification Restriction (CLIR2)** must be released by the network provider.

Phone number presentation - incoming | Expert

Defines which phone number is presented back to the caller.

No announcement

The phone number is suppressed.

Default

Both the DDI parallel to the internal number and the MSN on the PTM connection over which the subscriber is called are presented on the PTMP connection.

exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert** .

Note: For the option **None** , the phone service **Connected Line Identification Restriction (COLR2)** must be released by the network provider.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Reachability

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Subscriber (scr.) > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Subscriber (scr.) > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the subscriber setting.

Note: When switched off, the setting under **Subscriber (scr.) > Properties** applies.

Call deblocker (incoming)

Requirements:

- Configured call deblocker
  - Using the Configuration Manager (page **User data > Special numbers**)

The do-not-disturb function does not apply to callers who use one of the phone numbers entered in the selected call deblocker.

Call restrictor (incoming)

Requirements:

- Configured call restrictor
  - Using the Configuration Manager (page **User data > Special numbers**)

A caller, who transmitted one of the phone numbers entered in the selected call restrictor or no number, is rejected.

Multi-path Call Forwarding / FMC

Multi-path call forwarding behaviour | **Destination number**

- Internal and external calls directed to the subscriber are also signalled at the entered destination number
- CFNR is executed
- Calls directed to a group are not considered

FMC behaviour | **Destination number**

- On a second call, the called subscriber is busy
- The multi-path call is not executed
- CFB is executed
- an answering machine at the 2 nd B-channel (ISDN) ist not called

Call Forwarding | Settings | only external calls

External calls to the subscriber are forwarded. Internal calls are not forwarded.

Call Forwarding | Settings | Cascading

If call forwarding has been configured for the internal destination of an existing call forwarding, it is possible to reroute already forwarded calls (cascading).

Calls also will be rerouted to the new call forwarding destination. During cascading, the PBX memorises the subscribers who have already been called for this call. If the subscriber is the destination of a call forwarding again, the call ends with the previous destination.

The caller hears the dial tone until the call is accepted. If the destination is busy or the maximum call duration has been reached, a busy signal is issued.

Note: When switched off, cascading does not take place. Calls end with the entered destination.

Call Forwarding | Settings | CF for group calls

Calls directed to a group in which the subscriber is logged in as incoming will be forwarded to the CF destination of the subscriber.

Call Forwarding | CF immediately | Destination number:

Internal and external calls to the subscriber are forwarded immediately to the entered destination number.

Call Forwarding | CF on busy | Destination number:

Internal and external calls to the subscriber are forwarded on busy to the entered destination number.

Call Forwarding | CF on no reply | Destination number:

Internal and external calls to the subscriber are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Authorisations

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Subscriber (scr.) > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Subscriber (scr.) > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the subscriber setting.

Note: When switched off, the setting under **Subscriber (scr.) > Properties** applies.

Switching of relays and alarms

The subscriber can switch relays and alarms manually.

Controlling of the PBX via phone

The subscriber may change settings of the PBX (e. g. do-not-disturb) using a menu/function key or a character string.

Open door

No

The subscriber may not open a door.

[Name]

The subscriber can phone with the selected door and open the door by pressing a key.

All doors

The subscriber can phone with all doors and open the doors by pressing a key.

Features to be configured by the network provider (CF, key pad)

The subscriber can

- forward the own external number via phone.
- switch the permanent configuration on/off using a character string.
- send keypad information to the network provider.

Configuration of Scr CF to external numbers

The subscriber can forward the associated external number to an external destination.

Transfer of external calls externally

The subscriber can transfer a call between two external subscribers.

CF (groups) / Follow me

The subscriber can

- switch group properties of groups which the subscriber is a member of.
- switch on Follow-me.

Voicemail/ Fax Box

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Subscriber (scr.) > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Subscriber (scr.) > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the subscriber setting.

Note: When switched off, the setting under **Subscriber (scr.) > Properties** applies.

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on busy

The voice mailbox will be called if the internal number of the subscriber is busy and if no call forwarding on busy has been enabled for the subscriber.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on no reply + on busy

The voice mailbox will be called if the subscriber does not reply or if the internal number of the subscriber is busy.

Delay time for CF on no reply

Call acceptance during the time entered here.

Info call (voice mail function)

The arrival of new voice messages immediately after completion of the recording will be signaled by a call from the voice mailbox to the subscriber. When this is done, the phone number or name of the voice mailbox is shown on the phone display.

Note: COMfortel system phones (ISDN, VoIP) do not receive info calls. New voice message are displayed on these phones via LED signalling (e. g. message button).

Settings

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Subscriber (scr.) > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Subscriber (scr.) > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the subscriber setting.

Note: When switched off, the setting under **Subscriber (scr.) > Properties** applies.

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on busy

The voice mailbox will be called if the internal number of the subscriber is busy and if no call forwarding on busy has been enabled for the subscriber.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on no reply + on busy

The voice mailbox will be called if the subscriber does not reply or if the internal number of the subscriber is busy.

Delay time for CF on no reply

Call acceptance during the time entered here.

Info call (voice mail function)

The arrival of new voice messages immediately after completion of the recording will be signaled by a call from the voice mailbox to the subscriber. When this is done, the phone number or name of the voice mailbox is shown on the phone display.

Note: COMfortel system phones (ISDN, VoIP) do not receive info calls. New voice message are displayed on these phones via LED signalling (e. g. message button).

Settings Profile Assignment

Proceed as follows:

- Using the Configuration Manager

A configuration depending profile can be assigned to each subscriber, group or voice mail/fax box.

Assign profile to all

The same profile is assigned to all subcribers, groups or boxes for one configuration.

COMfortel WS-500x

Management COMfortel WS-500
Settings COMfortel WS-500
COMfortel M-5x0

Management COMfortel WS-500

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > COMfortel WS-500x > COMfortel WS-500**)

Requirements:

- Base station is configured with the MAC address via the PBXBasisstation (use of provisioning). Or the base station is configured with the MAC address via the corresponding Web Configurator (see Intrsuctions of the base station)
- Knowledge of the MAC address of the base station (sticker on the back of the device)
- Base station is configured with the MAC address via the PBX (manually or via provisioning)

Note: At the PBX, only base stations are configured for which the device role "DECT Manager and Integrator" is enabled.Base stations with the role "Base station" are not configured in the PBX but for multi-cell operation at a previously determined base station with the role "DECT Manager and Intergrator".

For more information about role distribution and base station management, see the Instructions of the base station.

MAC address

Required for all transactions between PBX, base station and handset and for the assignment of provisioning data. The address is also required for base station synchronisation in the Integrator/DECT manager.

Name of the base

Freely selectable name. This is used to identify the base station in the PBX and in the Integrator/DECT Manager.

Handsets (IPUI)

Registered handsets at this base station with the following information (left to right):

- Handset IPUI
- Handset phone number
- Handset name

Note: The name can be changed via the page **Subscriber (scr.) > Phone numbers**, the IPUI or the base station can be changed via the page **Subscriber (scr.) > COMfortel WS-500x > COMfortel M-5x0**.

Provisioning file

The provisioning file displayed here gives the base station the required configuration.

Configurator of the base

Link to the Web Configurator of the base station for further configuration.

Properties | Configure

Opens the configuration of the base station.

Further help under Settings COMfortel WS-500

Note: For settings that are required but not offered here, you can open the Web Configurator of the base station via the adjacent link under **Configurator of the base**.

New

A new base station is added. The settings must then be made.

Further help under Settings COMfortel WS-500

Delete

Deletes the entries marked in the list.

Settings COMfortel WS-500

Proceed as follows:

- Using the Configuration Manager (**Subscriber (scr.) > COMfortel WS-500x > COMfortel WS-500 > New**)
- Using the Configuration Manager (**Subscriber (scr.) > COMfortel WS-500x > COMfortel WS-500 > Configure**)

Important: When commissioning a base station for the first time via provisioning, the device may only be connected to the local network after setting up the base station, the handsets and the provisioning file. Provisioning cannot be initiated automatically if the base station is connected before then.

A detailed description of first commissioning (via provisioning or manual) can be found under COMfortel WS-500S / WS-500M.

Requirements:

- See Management COMfortel WS-500

MAC address

Required for all transactions between PBX, base station and handset and for the assignment of provisioning data. The address is also required for base station synchronisation in the Integrator/DECT manager.

Name of the base

Freely selectable name. This is used to identify the base station in the PBX and in the Integrator/DECT Manager.

Password of the base station

Freely selectable password consisting of

- at least 8, maximum 74 characters
- at least one capital letter
- at least one digit
- at least one specialo character

This password is later used for registration in the Web Configurator of the base station.

When configuring the base station via provisioning, the password can only be used in the Web Configurator after provisioning has been completed.

Provisioning file

Selection of a provisioning file already created for this device type.

Note: This and other base stations can also be assigned a configuration via the page **Subscriber (scr.) > IP Provisioning > File management**.

Note: If a required file is not yet available here,a file must first be created on the page **Subscriber (scr.) > IP Provisioning > File management**. There, the file can then be assigned to one or more base stations.

Jitter buffer

Specifies the size of the jitter buffer in milliseconds (duration of the audio signals). In this time period RTP packets are cached (lower values = lower

number, greater values = greater number). Delays in the transmission and loss of packets can be compensated that way.

Echo cancellation

The system compensates for local echoes and reverberation effects during internal VoIP calls.

SIP alert info

Select the options for signalling call tones on end devices. For signalling, the Invite is sent to the end device to transfer the call types. Depending on the device, the call types are either fully converted into call tones, or converted in a simplified version. The following options are available here.

External and internal calls

Device uses this selection of call types to signal call tones.

External, internal, group, door and alarm calls

Device uses all the call types supported by the PBX to signal call tones.

This is the default setting when a subscriber is configured.

If problems occur with signalling on the end device, use one of the options listed above.

RTP encryption

Allows you to set an exception for this base station. The default is **Adopt from Administration ->SIPS/SRTP internal (default)**.

Setting the exception is required if:

- This base station does not support the settings in the PBX. This can lead to problems, e. g. when establishing a call.
- You must bypass the PBX settings for testing or troubleshooting purposes.

Note: The settings for this base station here in the Configuration Manager and the settings in the Web Configurator of the base station must match.

Adopt from Administration ->SIPS/SRTP internal (default)

The setting of the PBX applies for this base station.

Only encrypted

Example: SRTP is disabled in the PBX. The base station requires enabled SRTP.

Only unencrypted

Example: SRTP is enabled in the PBX. The base station does not support SRTP.

Format of the transmitted phone number

The PBX supports several formats to display the the country code of incoming calls on the display of the called handset, e. g. adapted to a small display size.

COMfortel M-5x0

Subscriber
Free IPUI

Subscriber

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr,) > COMfortel WS-500x > COMfortel M-5x0 > Subscriber**)

Requirements:

- At least one subscriber is configured via the page **Subscriber (scr.) > Phone numbers**

Phone number

Phone number configured for this handset.

IPUI

free IPUI

Selection of an unassigned IPUI.

manual assignment

Manual input (digits or letters) or transfer of an IPUI. This can be determined as follows:

- Read out the barcode label on the package with a barcode reader connected to the computer (e. g. via the USB interface)
- Reading from the handset: Press the control button and enter ***#06#**
- Reading from the package label

current IPUI

The currently assigned IPUI for this handset.

WS-500x base

Select a base station to which this handset shall be assigned. With the follwoing information:

- MAC address of the base station
- Name of the base station

Free IPUI

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > COMfortel WS-500x > COMfortel M-5x0 > free IPUI**)

Requirements:

- An IPUI has been assigned to a Handset. This handset is later deleted, making its IPUI available again.
- One or more IPUI were e. g. created by reading with a barcode reader in advance and not yet assigned to a handset.

Phone number | Name

Selection of a subscriber to whom no IPUI has yet been assigned and who should receive the displayed IPUI.

IPUI

Displaying of a still available IPUI.

WS-500x base

Select a base station to which this handset shall be assigned. With the follwoing information:

- MAC address of the base station
- Name of the base station

Properties / Function overview

Settings of the Subscriber Properties

Settings of the Subscriber Properties

Basic Network Settings
Exchange Line Settings
Reachability
Authorisations
Analogue Settings
VoIP settings
Voicemail /Fax Box
Settings

Basic Network Settings

Proceed as follows:

- separately for subscribers
- Using the Configuration Manager
- Subscriber-related (page **Subscriber (scr.) > Properties** page)

  SUBSCRIBER
  Select the subscriber you want to display/configure.
- Function-related (page **Subscriber (scr.) > Function overview** page)

  Configure
  Opens the configuration of the single function for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Subscriber (scr.) > Profiles > ...** page)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function cannot be controlled with a profile.

PIN

Freely selectable PIN that consists of 6 digits. The PIN is used e.g. for access to voice mailbox, fax box and COMfortel Set.

Note: To change current PINs and passwords, click on **Edit PIN and password**.

Note: All of the PINs in the PBX are unique. This means that it is not possible to allocate the same PIN twice.

Note: Do not use dates of birth or dates as PINs. This makes it easy for an attacker to find out the correct PIN. PINs which are easy to guess, such as 111111 or 123456, should also be avoided.

Note: It is also possible not to enter a user PIN.

Caution: Visible PINs and passwords are a security risk.

- After entering the PIN/password, click on **Mask PIN and password**.

Password

Freely selectable password that consists of 8 to 32 characters containing digits, letters (upper and lower case, but no German umlauts and ß) and special characters **- _ . ! ~ * ' ( ) & = + $,**. The password is used for logging-in to the Configuration Manager (associated user name is the internal phone number) and for registering VoIP phones.

Note: To change current PINs and passwords, click on **Edit PIN and password**.

Note: A coloured bar under the entry field shows the security level of the password (red = low level, green = high level).

Note: Passwords with a low security level can not be saved.

Caution: A high level does not necessarily indicate a secure password as this can also be reached by entering names and birth dates in connection with special characters. However, names and birth dates should not be used as passwords.

- To create a secure password you can consult, for example, the guidelines of the German Federal Office for Information Security (BSI, in German).

Caution: Visible PINs and passwords are a security risk.

- After entering the PIN/password, click on **Mask PIN and password**.

Additional passwords Configure

Requirements:

- Enabled **Subscribers may create own passwords for accessing the web server/API** (page **Subscriber (scr.) > All subscribers**)
- Enabled **Subscribers may generate own passwords for accessing Action URLs** (page **Devices > Action URLs**)

Note: If the above-mentioned settings in the Configuration Manager were made as administrator, additional passwords for the Configuration Manager/API and Action URLs can be configured when logging in as subscriber. If only one of the above-mentioned requirements has been enabled, accordingly, the subscriber may only configure additional passwords for this option. The administrator can always configure additional passwords for the subscribers.

Note: The last password for the use of the Configuration Manager/API cannot be deleted.

Note: After the subscriber has assigned another password, he cannot go back to the original password. Only the (sub-)administrator has this authorisation.

Edit passwords

+ New

Adds a new entry to the list.

Description

Enter a name for the entry.

Access to web server / API

The subscriber has access to the Configuration Manager / API.

Access to Action URLs

Note: Additional entries for Action URLs can also be generated system-wide (**Devices > Action URLs**).

Note: The switching or triggering of relays (e. g. door opener) or alarms can also be configured with this setting or excluded in individual cases (**Devices > Alarms > Configure** or e.g. **Devices > Door terminals > Door opener >**).

The subscriber has access to the Action URLs.

Delete

Deletes the entries marked in the list.

Save

By saving a new added entry, the dialogue with the automatically generated password opens.

NEW PASSWORD

Note: The password is only generated if accessing the PBX via a secure HTTPS connection.

The automatically generated subscriber password is displayed only once. Therefore, copy the subscriber password to the clipboard and then close the dialogue.

CTI authorization code

Requirement:

- Subscriber is a phone of the COMfortel D-series

Freely selectable authorization code that consists of 8 to 32 characters containing digits, letters (upper and lower case, but no German umlauts and ß) and special characters **- _ . ! ~ * ' ( ) & = + $,**. The authorization code is used for controlling a phone of the COMfortel D-series via TAPI.

Note: To change an existing authorization code, click on **Edit CTI authorization code**.

Note: A coloured bar under the entry field shows the security level of the authorization code (red = low level, green = high level).

Note: Authorization codes with a low security level cannot be saved.

Caution: A high level does not necessarily indicate a secure authorization code as this can also be reached by entering names and birth dates in connection with special characters. However, names and birth dates should not be used as authorization codes.

- To create a secure authorization code you can consult, for example, the guidelines of the German Federal Office for Information Security (BSI, in German).

Caution: Visible authorization codes are a security risk.

- After you enter your password, click on **Hide CTI authorization code**.

Logged-in to the Configuration Manager Logout

Requirements:

- Administrator

Displays existing logging-in of the user/sub-administrator. By clicking the user/sub-administrator is logged out of the Configuration Manager.

Reset hardware status Execute

Requirements:

- Subscriber is an analogue or standard ISDN phone

The PBX does not automatically detect the removal of an analogue or standard ISDN phone. By clicking, the hardware status of the subscriber is reset on page **Monitoring > Status internal subscriber**.

Exchange Line Settings

Proceed as follows:

- Separately for subscribers
- Using the Configuration Manager
- Subscriber-related (page **Subscriber (scr.) > Properties**)

  SUBSCRIBER
  Select the subscriber you want to display/configure.
- Function-related (page **Subscriber (scr.) > Function overview**)

  Configure
  Opens the configuration of the single function for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Subscriber (scr.) > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Requirements:

- For differently configurable functions for private and business calls: function activated
  - Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)

Exchange line authorisation - business | Expert / Exchange line authorisation - private calls | Expert

Defines which kind of phone numbers may be dialled with a subscriber.

International

All international and domestic telephone numbers can be dialled.

National

All domestic telephone numbers can be dialled.

City

All telephone numbers without a provider area code can be dialled.

Note: If GSM access is used for an outgoing call, the PBX automatically dials the telephone number with the configured area code (national destination code).

Only incoming

Incoming external calls can be accepted.

Only internal

No external calls can be made/answered.

exchange line dependent

The exchange line authorisation is configured separately for each individual external connection via **Expert** .

Preventing emergency calls

Emergency calls cannot be made. The subscriber hears a corresponding announcement. This function is for example useful for remote extensions as the emergency call is issued at the location of the PBX and not at the location of the remote extension.

Call restrictor - (outgoing) business calls

Call restrictor - (outgoing) private calls

Requirements:

- Configured call restrictor
  - Using the Configuration Manager (**User data > Special numbers** page)

The subscriber can not dial the phone numbers entered in the call restrictor irrespective of his exchange line authorisation. So it restricts the exchange line authorisation.

Call deblocker - (outgoing) business calls

Call deblocker - (outgoing) private calls

Requirements:

- Configured call deblocker
  - Using the Configuration Manager (**User data > Special numbers** page)

The subscriber can dial the phone numbers entered in the call deblocker irrespective of his exchange line authorisation. So it extends the exchange line authorisation.

Preferred exchange line - business calls | Expert / Preferred exchange line - private calls | Expert

Defines via which exchange line outgoing external calls will be routed.

Automatic occupation

For outgoing external calls, any available connection is used, whereby first the fixed network connections and then the VoIP connection and at last the GSM connections were taken into account.

Special preferred exchange lines

For outgoing external calls, the connections released under **Expert** are used depending on the selected priority. This makes the definition of a preferred exchange line (priority = 1) with overflow to other connections (priority > 1) possible. Exchange lines with priority **No automatic occupation** are not used for outgoing calls.

Note: If all released connections are busy, it is no longer possible to call. The subscriber can only use a connection with the setting **No automatic occupation** by entering a character string (see).

Note: The same priority may be allocated to different external connections in parallel.

Number presentation - (outgoing) business | Expert / Number presentation - (outgoing) private | Expert

Defines which phone number is presented.

No announcement

The phone number is suppressed.

Default

Both the DDI parallel to the internal number and the MSN on the PTM connection over which the subscriber is called are presented on the PTMP connection. (if the subscriber is not a member of the call distribution for this connection, the first MSN is presented).

exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert** .

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Note: On a connection with **CLIP no screening** enabled, the number presentation defined here is not considered.

Note: Notes for suppressing number presentation

Note: Phone number suppression for VoIP calls is possible but it is not always supported by the providers.

Note: If an emergency number is called while phone number suppression is enabled, phone number suppression is disabled once so that the emergency service can identify the caller.

Note: If the number presentation only needs to be suppressed call by call, the **Number presentation (outgoing) Default** or **exchange line dependent** . For the **exchange line dependent** option, each exchange line can have an individual number presented to the network provider or to the caller separately. The number presentation to the caller is suppressed

by the phone before starting a call. The number presentation to the network provider is not suppressed by the phone.

Note: If the number presentation should always be suppressed, the **Number presentation (outgoing) exchange line dependent** . For each exchange line can a separate number be presented to the network provider. To keep these numbers from being presented to the caller, number presentation can be suppressed per exchange line separately.

Note: If number presentation needs to be suppressed between the PBX and the network provider, the **Number presentation (outgoing) No announcement** .

Note: For the option **No announcement** , the phone service **Calling Line Identification Restriction (CLIR2)** must be released by the network provider.

Phone number presentation - incoming | Expert

Defines which phone number is presented back to the caller.

No announcement

The phone number is suppressed.

Default

Both the DDI parallel to the internal number and the MSN on the PTM connection over which the subscriber is called are presented on the PTMP connection.

exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert** .

Note: For the option **None** , the phone service **Connected Line Identification Restriction (COLR2)** must be released by the network provider.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Reachability

Proceed as follows:

- Separately for subscribers
- Using the Configuration Manager
- Subscriber-related (page **Subscriber (scr.) > Properties**)

  SUBSCRIBER
  Select the subscriber you want to display/configure.
- Function-related (page **Subscriber (scr.) > Function overview**)

  Configure
  Opens the configuration of the single function for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Subscriber (scr.) > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Call waiting

You will be notified of any incoming calls if already on another call.

Do-not-Disturb

Callers are rejected.

Call deblocker (incoming)

Requirements:

- Configured call deblocker
  - Using the Configuration Manager (page **User data > Special numbers**)

The do-not-disturb function does not apply to callers who use one of the phone numbers entered in the selected call deblocker.

Call restrictor (incoming)

Requirements:

- Configured call restrictor
  - Using the Configuration Manager (page **User data > Special numbers**)

A caller, who transmitted one of the phone numbers entered in the selected call restrictor or no number, is rejected.

Follow-me | Destination number:

Internal and external calls to the subscriber are forwarded to the entered destination number.

Multi-path Call Forwarding / FMC

Multi-path call forwarding behaviour | Destination number

- Internal and external calls directed to the subscriber are also signalled at the entered destination number
- CFNR is executed
- Calls directed to a group are not considered

FMC behaviour | Destination number

- On a second call, the called subscriber is busy
- The multi-path call is not executed
- CFB is executed
- an answering machine at the 2 nd B-channel (ISDN) ist not called

Call Forwarding | Settings | only external calls

External calls to the subscriber are forwarded. Internal calls are not forwarded.

Call Forwarding | Settings | Cascading

If call forwarding has been configured for the internal destination of an existing call forwarding, it is possible to reroute already forwarded calls (cascading).

Calls also will be rerouted to the new call forwarding destination. During cascading, the PBX memorises the subscribers who have already been called for this call. If the subscriber is the destination of a call forwarding again, the call ends with the previous destination.

The caller hears the dial tone until the call is accepted. If the destination is busy or the maximum call duration has been reached, a busy signal is issued.

Note: When switched off, cascading does not take place. Calls end with the entered destination.

Call Forwarding | Settings | CF for group calls

Calls directed to a group in which the subscriber is logged in as incoming will be forwarded to the CF destination of the subscriber.

Call Forwarding | CF immediately | Destination number:

Internal and external calls to the subscriber are forwarded immediately to the entered destination number.

Call Forwarding | CF on busy | Destination number:

Internal and external calls to the subscriber are forwarded on busy to the entered destination number.

Call Forwarding | CF on no reply | Destination number:

Internal and external calls to the subscriber are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Authorisations

Proceed as follows:

- Separately for subscribers
- Using the Configuration Manager
- Subscriber-related (page **Subscriber (scr.) > Properties**)

  SUBSCRIBER
  Select the subscriber you want to display/configure.
- Function-related (page **Subscriber (scr.) > Function overview**)

  Configure
  Opens the configuration of the single function for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Subscriber (scr.) > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Sub-admin

The subscriber has the rights of a sub-administrator. For logging-in to the Configuration Manager the password of the subscriber is used (in this case associated user name is **sub-admin**).

You can configure up to four sub-administrators.

Pick up

No

Calls of the subscriber may not be taken by pick-up.

Global

The subscriber may take calls to all other subscribers by pick-up.

Group(s)/voice mailbox(es)

This subscriber may take calls of other subscribers that are members of groups the subscriber is also member of. It does not make any difference if the subscriber is currently logged-in or -out. The subscriber is authorised to take over calls from those voice mailboxes of which he is the owner.

If a voice mailbox has a group as its owner or user, each subscriber who is a member of the group can use this authorisation to take over calls.

Configuration switching

The subscriber can switch configurations manually.

Speaker announcement

No

The subscriber may not perform speaker announcements.

[Name]

The subscriber can perform speaker announcements on the selected announcement output.

All announcement outputs

The subscriber can perform speaker announcements on all announcement outputs.

Switching of relays and alarms

The subscriber can switch relays and alarms manually.

Controlling of the PBX via phone

The subscriber may change settings of the PBX (e. g. do-not-disturb) using a menu/function key or a character string.

Open door

No

The subscriber may not open a door.

[Name]

The subscriber can phone with the selected door and open the door by pressing a key.

All doors

The subscriber can phone with all doors and open the doors by pressing a key.

Features to be configured by the network provider (CF, key pad)

The subscriber can

- forward the own external number via phone.
- switch the permanent configuration on/off using a character string.

- send keypad information to the network provider.

Configuration of Scr CF to external numbers

The subscriber can forward the associated external number to an external destination.

Transfer of external calls externally

The subscriber can transfer a call between two external subscribers.

Delete list of single call records

The subscriber can delete the phone number of the communication partner from the last call made to ensure that this number is not viewed by unauthorised persons.

CF (groups) / Follow me

The subscriber can

- switch group properties of groups which the subscriber is a member of.
- switch on Follow-me.

Call Parking

The subscriber can park a caller in the parking zone or retrieve a waiting caller from the parking zone.

Analogue Settings

Proceed as follows:

- Separately for subscribers
- Using the Configuration Manager
- Subscriber-related (page **Subscriber (scr.) > Properties**)

  SUBSCRIBER
  Select the subscriber you want to display/configure.
- Function-related (page **Subscriber (scr.) > Function overview**)

  Configure
  Opens the configuration of the single function for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Subscriber (scr.) > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Requirements:

- Subscriber is an analogue end device

CLIP info

Requirements:

- Subscriber is a CLIP-capable analogue phone

The phone number information (if transmitted) is displayed on the display of the phone.

Extended CLIP info

Requirements:

- Subscriber is a CLIP-capable analogue phone

In the caller list of the phone date and time of missed calls are recorded. The name can be displayed instead of or in addition to the phone number.

Note: For some end devices, extended CLIP information can cause problems and should therefore be disabled.

Hook flash timing

Hook flash timing enables the PBX to detect a flash signal from the subscriber in question.

For this to work, the hook flash timing must correspond with the time that is configured for the flash signal on the analogue end device (see the phone manual).

Note: DTMF phones that execute various functions using the T-Net function key/menu use a flash of 300 ms to do this. In order to also use these functions on the PBX, the same time as the hook flash timing must be configured on the PBX. Usually, there is an additional flash button

available on this phone that must also be configured with this value (see the phone manual).

Note: Note that the PBX does not detect a receiver being hanged up until after hook flash timing is finished.

Ringtone for internal calls

The ringtone specifies which sequence of sounds is used to signal internal calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

VoIP settings

Proceed as follows:

- Separately for subscribers
- Using the Configuration Manager
- Subscriber-related (**Subscriber (scr.) > Properties**

  SUBSCRIBER)
  Select the subscriber you want to display/configure.
- Function-related (**Subscriber (scr.) > Function overview**

  Configure)
  Opens the configuration of the individual functions for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (**Subscriber (scr.) > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function cannot be controlled with a profile.

Requirement:

- Subscriber is a VoIP end device

Jitter buffer

Specifies the size of the jitter buffer in milliseconds (duration of the audio signals). In this time period RTP packets are cached (lower values = lower number, greater values = greater number). Delays in the transmission and loss of packets can be compensated that way.

Echo cancellation

The system compensates for local echoes and reverberation effects during internal VoIP calls.

SIP alert info

Select the options for signalling call tones on end devices. For signalling, the Invite is sent to the end device to transfer the call types. Depending on the device, the call types are either fully converted into call tones, or converted in a simplified version. The following options are available here.

External and internal calls

Device uses this selection of call types to signal call tones.

External, internal, group, door and alarm calls

Device uses all the call types supported by the PBX to signal call tones.

This is the default setting when a subscriber is configured.

If problems occur with signalling on the end device, use one of the options listed above.

Allowing Update Messages

If the function is enabled and **Online name search** is activated, the name will be updated while the connection is being set up.

When **Blind Transfer** is activated and the function is enabled, the name as well as the telephone number will be updated while the connection is being set up.

Note: In the event of problems while the connection is being set up, this function should be disabled as a possible cause.

RTP encryption

Allows an exception from the default settings of the PBX for the corresponding VoIP subscriber. The default setting is **Adopt from Administration ->SIPS/SRTP internal (default)**.

Applying the exception is required if:

- The corresponding phone does not support the settings of the PBX. This may cause problems e.g., during a call being established.
- You must circumvent the settings of the PBX for test purposes or trouble shooting.

Note: The settings of the PBX for the corresponding subscriber and the settings of the phone must match.

Adopt from Administration ->SIPS/SRTP internal (default)

The PBX setting applies for the subscriber.

Only encrypted

Example: SRTP is not enabled in the PBX. The phone requires an enabled SRTP.

Only unencrypted

Example: SRTP is enabled in the PBX. The phone does not support SRTP.

Format of transferred telephone number

The PBX supports several formats to display the country code of incoming calls on the display of the VoIP phone being called, e. g. adjusted to a small display.

Note: With the phones COMfortel 3200/3500 and COMfortel D-series, the format country code with leading + sign (e. g. + 49 5306) cannot be used.

Properties in the phone | Configure

The Configuration Manager of the phone opens.

Note: For a handset of the **COMfortel M-5x0** device type, the Web Configurator of the base station to which this device is assigned opens.

Assigning the MAC address of the device

The MAC address of a phone is needed to identify a phone that is assigned to a specific subscriber. For all transactions between phone and PBX the PBX must be able to identify the MAC address of the phone so that the phone can be matched to a subscriber and his/her provisioning data.

found devices

Select one of the existing MAC adresses.

assigned devices

MAC adresses already assigned to the subscribers.

manual assignement

Enter manually (only digits) or take over a MAC adress after reading with a bar code scanner.

The MAC address of a device can be found as follows:

- Retrieving from a client certificate sended by the phone
- Reading with a bar code scanner enabled and connected to the computer (e. g. via a USB interface)
- Reading e. g. on the menu oft the phone or on a label

Note: If a MAC address has been entered and assigned to a subscriber, the corresponding phone can already be provided with provisioning data during its first boot process.

IPUI (only COMfortel M-5x0)

current IPUI

The IPUI already assigned to this subscriber.

free IPUI

Selection of an unassigned IPUI for this subscriber.

COMfortel WS-500 (only COMfortel M-5x0)

Assignment of this subscriber to a specific base station.

Header file

Display of the header file which is assigned to the subscriber.

- Using the Configuration Manager (page **Subscriber (scr.) > IP provisioning > File management**)

Provisioning file

Display of the provisioning file which is assigned to the subscriber.

- Using the Configuration Manager (page **Subscriber (scr.) > IP provisioning > File management**)

Format of transferred phone number for LDAP requests

The PBX supports different formats, so you can dial the contact's phone number correctly using an LDAP request.

Note: Country code with leading zeroes (e.g. 00495306...): If exchange line request type Internal phone is set for the subscriber, the exchange line access number is inserted before the phone number. This applies for the following device types: **Standard VoIP telephone**, **COMfortel M-5x0**.

Voicemail /Fax Box

Proceed as follows:

- Separately for subscribers
- Using the Configuration Manager
- Subscriber-related (page **Subscriber (scr.) > Properties**)

  SUBSCRIBER
  Select the subscriber you want to display/configure.
- Function-related (page **Subscriber (scr.) > Function overview**)

  Configure
  Opens the configuration of the single function for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Subscriber (scr.) > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Fax box | Phone number of the box

Requirements:

- Created fax box for the subscriber

Note: If it is not clear whether it is a fax or not, the call will be signaled to the subscriber or it will be accepted by the voice mailbox.

Note: Note. A fax call which is accepted by a voice mailbox will not be forwarded to a fax box.

Voice mailbox | Phone number of the box

Requirements:

- Created voice mailbox for the internal subscriber:

Both the subscriber and the selected voice mailbox will be automatically called for all incoming calls made to the internal number of the subscriber.

Important: The PBX supports only one central answering machine. An answering machine at the COMfortel system phone will be disabled as soon as a voice mailbox is selected for the subscriber. Querying and deleting voice messages as well as replaying own announcements is still possible. The call recording function can be used.

Note: If call forwarding immediately (subscriber/group), the boss/secretary function (system phone) or a waiting field reception (for one waiting field group, one enabled waiting field reception is sufficient) is enabled, the call will not be made to the voice mailbox!

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on busy

The voice mailbox will be called if the internal number of the subscriber is busy and if no call forwarding on busy has been enabled for the subscriber.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on no reply + on busy

The voice mailbox will be called if the subscriber does not reply or if the internal number of the subscriber is busy.

Delay time for CF on no reply

Call acceptance during the time entered here.

Info call (voice mail function)

The arrival of new voice messages immediately after completion of the recording will be signaled by a call from the voice mailbox to the subscriber. When this is done, the phone number or name of the voice mailbox is shown on the phone display.

Note: COMfortel system phones (ISDN, VoIP) do not receive info calls. New voice message are displayed on these phones via LED signalling (e. g. message button).

Settings

Proceed as follows:

- Separately for subscribers
- Using the Configuration Manager
- Subscriber-related (page **Subscriber (scr.) > Properties**)

  SUBSCRIBER
  Select the subscriber you want to display/configure.
- Function-related (page **Subscriber (scr.) > Function overview**)

  Configure
  Opens the configuration of the single function for all subscribers.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place only on page **Subscriber (scr.) > Properties** under **Overridden by profile in configuration**.

Note:[Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Subscriber (scr.) > Profiles > ...**)

Note:not overridden by a profile: The function is currently not overridden by a profile.

Note:(not profile-controlled): The function can not be overridden by a profile.

Baby call (connection without dialling) | Destination number:

Requirements:

- Subscriber is not a VoIP phone or VoIP system phone

After picking up the receiver, the phone number entered is automatically dialled.

Billing factor

Indicates the value used to convert the subscriber's phone costs. This factor is used to multiply the charges transmitted from the network provider or the calculated charges. The phone costs calculated in this way are used in the call data list.

Preliminary pick-up query and monitoring

Requirements:

- VoIP end device which can evaluate the available information by BLF
- VoIP phone from another manufacturer: **SIP convenience package** / **SIP convenience package brand plus** enabled (optional, see activation)

All information about the calls of the subscriber is available for evaluation by BLF.

A preliminary pick-up query allows a subscriber, even before a pick-up, to see which caller he would be picking up.

Note: When switched off, no phone numbers of the subscriber‘s calls will be transmitted by BLF.

Private exchange line access without PIN

Requirements:

- Differently configurable functions for private and business calls activated
  - Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)

The subscriber is able to make private calls without entering a user PIN but only on his own phone.

Note: When authorisation is granted for initiating personal calls without a PIN, there is no protection against misuse by other internal users.

Signalling by tones

Internal dial tone

Specifies which tone the user hears after picking up the receiver or after pressing the flash or query key if a line is available. It is then possible to dial a phone number.

Note: See Sound for information about sound sequences.

Busy tone at end of call

The PBX generates a busy signal at the end of the call (e. g. the communication partner hangs up).

For some answering machines, this busy signal does not interrupt a recording. These answering machines cannot differentiate between the busy signal and speech. For such cases, the busy signal can be disabled.

Note: See Sound for information about sound sequences.

Special dial tone for CF, Do-not-Disturb, baby call/hotline

The PBX generates a special dial tone to bring attention to a configured function (do-not-disturb, baby call (connection without dialling)).

Note: See Sound for information about sound sequences.

Special dial tone for filled call data memory

the PBX generates a special dial tone to bring attention that call data memory has reached capacity.

Note: See Sound for information about sound sequences.

Special functions

Exchange line request

Requirements:

- System-wide automatic exchange line request switched off and entered exchange line access number
  - Using the Configuration Manager (page **Public switched tel. networks > Exchange line request**)

Internal phone

For external calls, the prefix of the **Exchange line access number** is required.

Direct exchange line phone

When picking up the receiver, an external line is requested instantly. For internal calls, the prefix ** is required.

Direct exchange line request

The PBX automatically distinguishes between internal and external numbers with the analysis of the digit sequence and the adjustment of the dialled phone number and the internal numbering plan.

Note: The selection **Direct exchange line request** is only available for IP DECT phones as well as COMfortel 1200 IP and COMfortel C-400. If you want to use this function for other phones, you must switch on the system-wide automatic exchange line request (page **Administration > Exchange line request**).

Note: The behaviour of the IP system phones when picking up the receiver, depends on the default account selection (see manual of the phone). For some IP system phones, the default account can be determined by the settings for the exchange line request of the PBX.

Call take-over

This is reasonable for the pick-up of calls to answering machines, fax machines or modems.

If call take-over is enabled for an answering machine and this answering machine takes a call earlier than a subscriber, the subscriber may take this call back from the answering machine. The answering machine will receive a busy signal then.

No

Calls of this subscriber may not be taken over.

Global

Calls of this subscriber may be taken over by every other phone.

Group(s)

Calls of this subscriber may be taken over by members of groups the subscriber is also member of. It does not make any difference if the subscriber is currently logged-in or -out.

InterCom (announcement/hands-free calling)

Requirements:

- System phone that supports InterCom (optional, see phone manual)

The subscriber takes automatically an InterCom call directed to him.

Transmission of the display name

Requirements:

- VoIP device with entered display name
- **SIP convenience package** / **SIP convenience package brand plus** enabled (optional, see activation)

The entered display name will be transmitted when the subscriber makes an internal call and displayed on the called phone.

Note: When switched off, the subscriber name entered in the PBX will be displayed on the called phone when the subscriber makes an internal call. If the phone number of the subscriber has been entered in the local phone book of the called phone, the subscriber name is suppressed by the phone book entry.

Alarm device

Requirements:

- System phone that supports signalling for alarm devices (optional, see phone manual)

An internal call of the subscriber is signaled as an alarm call on a system phone.

Unassigned phones

Managing Unassigned Phones

Managing Unassigned Phones

Phones that are still to be used on the PBX can here be assigned to a subscriber.

Phones that are not in use, not connected or not assigned to a subscriber should be removed from the list.

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > Unassigned phones**)

Select the line of the phone to be removed in the register **VoIP phones** or **Other phones** and click **Delete**.

or

Assign a subscriber to a VoIP phone in the list field to the right of the corresponding VoIP phone.

All subscribers

Settings all Subscribers

Settings all Subscribers

Proceed as follows:

- Using the Configuration Manager (page **Subscriber (scr.) > All subscribers**)

Max. duration of switched/forwarded calls: external with external (1..600 min.)

Specifies after how many minutes the PBX end the transferred or forwarded call (external with external), if two call channels of the PBX were used for forwarding.

Ringer frequency

Specifies the ringer voltage frequency the PBX uses to signalise calls to internal analogue subscribers.

ISDN scr may move with phone/change ports

The ISDN devices may be connected to any internal S0/UP0 port. The PBX recognises the subscriber and moves him to another port.

Note: When switched off, an ISDN device may be plugged into another RJ socket of the internal S0/UP0 port, but it has to be the same internal bus.

Delay time for baby / senior call (1..20 sec.)

Specifies after how many seconds the destination number is dialled automatically after picking up the receiver

Differently configurable functions for private and business calls

Two different configurations can be created for each of the functions listed below – one for private calls and one for business calls.

Note: When switched off, only one configuration can be created for the functions listed below. This applies to private calls as well as to business calls.

Note: If differently configurable functions for private and business calls are disabled, the PBX takes over the configuration for business calls as joint configuration.

Note: If differently configurable functions for private and business calls are enabled again, the PBX takes over the settings changed since the last disabling only for business calls. For this reason, you will need to check the configurations for private calls.

Subscribers may create own passwords

Subscribers may generate additional Configuration Manager or API passwords as well as delete existing passwords. If under **Devices > Action URLs**, **Subscribers may generate own passwords for accessing Action URLs** is activated, then the subscriber may also generate or delete passwords for Action URLs.

Groups

Phone numbers
Profiles
Properties

Phone numbers

Management Groups

Management Groups

Proceed as follows:

- Using the Configuration Manager (page **Groups > Phone numbers**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by an internal subscriber. (Select + **New** to enter.)

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the group within the PBX and the connected system phones. The name can also be transferred to the VoIP end devices via LDAP server.

Property Configure

Opens the configuration of the group.

Filter:

The list of groups is filtered by the entered or selected value.

+ New

Adds a new group. You must then make the settings for this new group.

Note: Up to 64 groups can be created.

Delete

Deletes the entries marked in the list.

Profiles

Management Profile
Settings of the Group Profiles
Settings Profile Assignment

Management Profile

Proceed as follows:

- Using the Configuration Manager

Profile name

The available profiles (default profiles, user-defined profiles) are displayed.

View

The selected settings can be viewed (default profiles).

Configure

The user-defined profiles can be adapted to the corresponding requirements (user-defined profiles).

+ New

Creation of a new profile.

Settings of the Group Profiles

Basic Settings
Exchange Line Settings
Reachability
Voicemail-/Fax Box
Settings

Basic Settings

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Groups > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Groups > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the group setting.

Note: When switched off, the setting under **Groups > Properties** applies.

Ringing sequence | after:

All

All subscribers logged-in to the group will ring simultaneously. Selected subscribers will start to ring after the configured delay time (**after:**). If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

All - Split Group

The call is handled almost similar to the ringing sequence **All**. The only difference is the way multiple calls to the group are handled: If the group receives further calls during an active call, the group will be split into several smaller groups ringing for a specific active call. But there will never ring more calls in the group than the number of free available group members. Additional calls will be rejected with a busy then. If a call is accepted the other phones ringing so far for this call are assigned to the other pending calls and will immediately start to ring again. If a group member finishes a call, this subscriber will immediately ring again for the pending calls. The timer for the configured delay time (**after:**) starts after receiving the first call to the group and will not restart for additional incoming calls.

Linear - fixed - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - balancing - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - fixed - rotating - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. If all members of a group are called, the process will be restarted with the first member

of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a subscriber is busy. In the second rotation the busy subscribers will hear knocking (only from external). The order of calls depends on the configured **Priority**.

Note: System phones control the call waiting itself, so the ringing sequence can vary.

Linear - balancing - rotating - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a subscriber is busy. In the second rotation the busy subscribers will hear knocking (only from external). The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call.

Adding - fixed - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If a call was not accepted, although

all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - balancing - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - fixed - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If finally all logged-in members of a group are called, the busy subscribers will hear knocking additionally (only from external).

Adding - balancing - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If finally all logged-in members of a group are called, the busy subscribers will hear knocking additionally (only from external).

Exchange Line Settings

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Groups > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Groups > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the group setting.

Note: When switched off, the setting under **Groups > Properties** applies.

Exchange line authorisation | Expert

Defines which kind of phone numbers may be dialled with a group member logged in as outgoing.

International

All national and international phone numbers can be dialled.

National

All national phone numbers can be dialled.

City

All phone numbers with the same local area code can be dialled.

Exchange line dependent

The exchange line authorisation is configured under **Expert** separately for each connection.

As subscriber

The exchange line authorisation of the subscriber is used.

No

Emergency numbers can be dialled, if the exchange line authorisation of the subscriber permits this.

Call restrictor (outgoing)

None

No call restrictor is used for group members logged in as outgoing.

As subscriber

The call restrictor selected for a subscriber is used.

Call deblocker (outgoing)

None

No call deblocker is used for group members logged in as outgoing.

As subscriber

The call deblocker selected for a subscriber is used.

Preferred exchange line | Expert

Defines via which exchange line outgoing external calls will be routed by a group member logged in as outgoing.

Automatic occupation

For outgoing external calls, any available connection is used, whereby first the fixed network connections and then the VoIP connection and at last the GSM connections were taken into account.

Special preferred exchange lines

For outgoing external calls, the connections released under **Expert** are used depending on the selected priority. This makes the definition of a preferred exchange line (priority = 1) with overflow to other connections (priority > 1) possible. Exchange lines with priority **No automatic occupation** are not used for outgoing calls.

As subscriber

The preferred exchange line of the subscriber is used.

Note: If all released connections are busy, it is no longer possible to call. The subscriber can only use a connection with the setting **No automatic occupation** by entering a character string (see Targeted Exchange Line Access).

Note: The same priority may be allocated to different external connections in parallel.

Number presentation (outgoing) | Expert

Defines which phone number is presented by a group member logged in as outgoing.

No

The phone number is suppressed.

Standard

Both the DDI parallel to the internal number and the MSN on the PTMP connection are displayed on the PTP connection over which the group is called (if the group is not a member of the call distribution for this connection, the first MSN is presented).

Exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert**.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Note: On a connection with **CLIP no screening** enabled, the number presentation defined here is not considered.

Note: For the option **No**, the phone service **Calling Line Identification Restriction (CLIR2)** must be released by the network provider.

Number presentation (incoming) | Expert

Defines which phone number is presented back to the caller by a group member logged in as outgoing.

No

The phone number is suppressed.

Standard

Both the DDI parallel to the internal number and the MSN on the PTM connection over which the group is called are presented on the PTMP connection.

Exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert**.

Note: For the option **No**, the phone service **Connected Line Identification Restriction (COLR2)** must be released by the network provider.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Reachability

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Groups > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Groups > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the group setting.

Note: When switched off, the setting under **Groups > Properties** applies.

Exchange line authorisation | Expert

Defines which kind of phone numbers may be dialled with a group member logged in as outgoing.

International

All national and international phone numbers can be dialled.

National

All national phone numbers can be dialled.

City

All phone numbers with the same local area code can be dialled.

Exchange line dependent

The exchange line authorisation is configured under **Expert** separately for each connection.

As subscriber

The exchange line authorisation of the subscriber is used.

No

Emergency numbers can be dialled, if the exchange line authorisation of the subscriber permits this.

Call restrictor (outgoing)

None

No call restrictor is used for group members logged in as outgoing.

As subscriber

The call restrictor selected for a subscriber is used.

Call deblocker (outgoing)

None

No call deblocker is used for group members logged in as outgoing.

As subscriber

The call deblocker selected for a subscriber is used.

Preferred exchange line | Expert

Defines via which exchange line outgoing external calls will be routed by a group member logged in as outgoing.

Automatic occupation

For outgoing external calls, any available connection is used, whereby first the fixed network connections and then the VoIP connection and at last the GSM connections were taken into account.

Special preferred exchange lines

For outgoing external calls, the connections released under **Expert** are used depending on the selected priority. This makes the definition of a preferred exchange line (priority = 1) with overflow to other connections (priority > 1) possible. Exchange lines with priority **No automatic occupation** are not used for outgoing calls.

As subscriber

The preferred exchange line of the subscriber is used.

Note: If all released connections are busy, it is no longer possible to call. The subscriber can only use a connection with the setting **No automatic occupation** by entering a character string (see Targeted Exchange Line Access).

Note: The same priority may be allocated to different external connections in parallel.

Number presentation (outgoing) | Expert

Defines which phone number is presented by a group member logged in as outgoing.

No

The phone number is suppressed.

Standard

Both the DDI parallel to the internal number and the MSN on the PTMP connection are displayed on the PTP connection over which the group is called (if the group is not a member of the call distribution for this connection, the first MSN is presented).

Exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert**.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Note: On a connection with **CLIP no screening** enabled, the number presentation defined here is not considered.

Note: For the option **No**, the phone service **Calling Line Identification Restriction (CLIR2)** must be released by the network provider.

Number presentation (incoming) | Expert

Defines which phone number is presented back to the caller by a group member logged in as outgoing.

No

The phone number is suppressed.

Standard

Both the DDI parallel to the internal number and the MSN on the PTM connection over which the group is called are presented on the PTMP connection.

Exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert**.

Note: For the option **No**, the phone service **Connected Line Identification Restriction (COLR2)** must be released by the network provider.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Voicemail-/Fax Box

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Groups > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Groups > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the group setting.

Note: When switched off, the setting under **Groups > Properties** applies.

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on busy

The voice mailbox will be called if the internal number of the group is busy and if no call forwarding on busy has been enabled for the group.

on no reply + on busy

The voice mailbox will be called if the group does not reply or if the internal number of the group is busy.

on all logged out

The voice mailbox will be called when all group members are logged out.

Delay time for CF on no reply

Call acceptance during the time entered here.

Settings

Proceed as follows:

- Separately for profiles
- Using the Configuration Manager
- Profile-related (page **Groups > Profiles > Configuration > Configure**)

  PROFILE CONFIGURATION
  Select the profile you want to display/configure.
- Function-related (page **Groups > Profiles > Function overview**)

  Configure
  Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the group setting.

Note: When switched off, the setting under **Groups > Properties** applies.

Number presentation (internally)

Group numbers

All group members logged in as outgoing will present the group number for internal outgoing calls. If the group is internally called the subscriber number is transferred back.

Subscriber number

All group members logged in as outgoing will present their own subscriber number for internal outgoing calls.

Preliminary pick-up query and monitoring

Requirements:

- VoIP end device which can evaluate the available information by BLF
- VoIP phone from another manufacturer: **SIP convenience package** / **SIP convenience package brand plus** enabled (optional, see Technical Data in the Instructions)

All information about the incoming calls of the group is available for evaluation by BLF.

Note: When switched off, no phone numbers of the group‘s calls are transmitted by BLF.

Pick-up of all calls to group members

A member of this group can pick up calls to the phone number of the group as well as direct calls to individual members of the group.

Note: When switched off, group members can only pick up direct calls to the phone number of the group.

Settings Profile Assignment

Proceed as follows:

- Using the Configuration Manager

A configuration depending profile can be assigned to each subscriber, group or voice mail/fax box.

Assign profile to all

The same profile is assigned to all subcribers, groups or boxes for one configuration.

Properties

Settings of the Group Properties

Settings of the Group Properties

Basic Settings
Exchange Line Settings
Reachability
Voicemail/Fax Box
Settings

Basic Settings

Proceed as follows:

- Separately for groups
- Using the Configuration Manager (page **Groups > Properties**)

GROUP

Select the group you want to display/configure.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place on page **Groups > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Groups > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Ringing sequence | after:

All

All subscribers logged-in to the group will ring simultaneously. Selected subscribers will start to ring after the configured delay time (**after:**). If a

call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

All - Split Group

The call is handled almost similar to the ringing sequence **All**. The only difference is the way multiple calls to the group are handled: If the group receives further calls during an active call, the group will be split into several smaller groups ringing for a specific active call. But there will never ring more calls in the group than the number of free available group members. Additional calls will be rejected with a busy then. If a call is accepted the other phones ringing so far for this call are assigned to the other pending calls and will immediately start to ring again. If a group member finishes a call, this subscriber will immediately ring again for the pending calls. The timer for the configured delay time (**after:**) starts after receiving the first call to the group and will not restart for additional incoming calls.

Linear - fixed - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - balancing - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - fixed - rotating - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a subscriber is busy. In the second rotation the busy subscribers will hear knocking (only from external). The order of calls depends on the configured **Priority**.

Note: System phones control the call waiting itself, so the ringing sequence can vary.

Linear - balancing - rotating - Split Group

Only the phone of one group member will ring. After the configured delay time (**after:**) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a subscriber is busy. In the second rotation the busy subscribers will hear knocking (only from external). The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call.

Adding - fixed - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are

already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - balancing - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - fixed - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If finally all logged-in members of a

group are called, the busy subscribers will hear knocking additionally (only from external).

Adding - balancing - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (**after:**) another single group member will be called. If the group receives another call B initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing more members are added to this call until exactly the half is reached. If for the first call A more than half of the group members are ringing, no members are added to this call (frozen). Only the remaining subscribers are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of the calls depends on the configured **Priority**: First of all the member of the group first logged-in gets the position 1 in the order of calls, the second one the position 2, ... If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If finally all logged-in members of a group are called, the busy subscribers will hear knocking additionally (only from external).

Phone number | Name

Group member. (Entry with + **New**)

Priority

Requirements:

- Configured ringing sequence: Not **All** or **All - Split Group**

Specifies the order of calls within the group for the ringing sequence (change with ⬆ | ⬇).

Status of incoming calls

Logged in

Logs the phone into the group.

Logged out

Logs the phone out of the group.

Permanently logged in

Permanently logs the phone of a subgroup into the main group.

Status of outgoing calls

Logged in

Logs the phone into the group.

Logged out

Logs the phone out of the group.

Permanently logged in

Permanently logs the phone of a subgroup into the main group.

Ringing delay

Requirements:

- Configured ringing sequence: **All** or **All - Split Group**

The subscriber rings after the configured delay time (**after:**).

+ New

Selects a new group member from the configured subscribers and groups and adds it to the list.

Note: Up to 40 internal subscribers or groups can be assigned to a group as members.

Delete

Deletes the entries marked in the list.

Exchange Line Settings

Proceed as follows:

- Separately for groups
- Using the Configuration Manager (page **Groups > Properties**)

GROUP

Select the group you want to display/configure.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place on page **Groups > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Groups > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Exchange line authorisation | Expert

Defines which kind of phone numbers may be dialled with a group member logged in as outgoing.

International

All national and international phone numbers can be dialled.

National

All national phone numbers can be dialled.

City

All phone numbers with the same local area code can be dialled.

Exchange line dependent

The exchange line authorisation is configured under **Expert** separately for each connection.

As subscriber

The exchange line authorisation of the subscriber is used.

No

Emergency numbers can be dialled, if the exchange line authorisation of the subscriber permits this.

Call restrictor (outgoing)

None

No call restrictor is used for group members logged in as outgoing.

As subscriber

The call restrictor selected for a subscriber is used.

Call deblocker (outgoing)

None

No call deblocker is used for group members logged in as outgoing.

As subscriber

The call deblocker selected for a subscriber is used.

Preferred exchange line | Expert

Defines via which exchange line outgoing external calls will be routed by a group member logged in as outgoing.

Automatic occupation

For outgoing external calls, any available connection is used, whereby first the fixed network connections and then the VoIP connection and at last the GSM connections were taken into account.

Special preferred exchange lines

For outgoing external calls, the connections released under **Expert** are used depending on the selected priority. This makes the definition of a preferred exchange line (priority = 1) with overflow to other connections (priority > 1) possible. Exchange lines with priority **No automatic occupation** are not used for outgoing calls.

As subscriber

The preferred exchange line of the subscriber is used.

Note: If all released connections are busy, it is no longer possible to call. The subscriber can only use a connection with the setting **No automatic occupation** by entering a character string (see Targeted Exchange Line Access).

Note: The same priority may be allocated to different external connections in parallel.

Number presentation (outgoing) | Expert

Defines which phone number is presented by a group member logged in as outgoing.

No

The phone number is suppressed.

Standard

Both the DDI parallel to the internal number and the MSN on the PTMP connection are displayed on the PTP connection over which the group is called (if the group is not a member of the call distribution for this connection, the first MSN is presented).

Exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert**.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Note: On a connection with **CLIP no screening** enabled, the number presentation defined here is not considered.

Note: For the option **No**, the phone service **Calling Line Identification Restriction (CLIR2)** must be released by the network provider.

Number presentation (incoming) | Expert

Defines which phone number is presented back to the caller by a group member logged in as outgoing.

No

The phone number is suppressed.

Standard

Both the DDI parallel to the internal number and the MSN on the PTM connection over which the group is called are presented on the PTMP connection.

Exchange line dependent

The phone number to be presented is configured separately for each external connection under **Expert**.

Note: For the option **No**, the phone service **Connected Line Identification Restriction (COLR2)** must be released by the network provider.

Note: On a connection with a GSM gateway, the phone number presentation configured in the PBX is not used because the actual phone number presentation is directly configured in the individual GSM gateway.

Reachability

Proceed as follows:

- Separately for groups
- Using the Configuration Manager (page **Groups > Properties**)

GROUP

Select the group you want to display/configure.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place on page **Groups > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Groups > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Reserved call channels | Number

External exchange line: For outgoing external calls of the group, the selected number of call channels of the selected external exchange line

will be reserved. These call channels are never available to the other subscribers or groups.

No reservation

Call channels will not be reserved.

Note: For a VoIP account the possible number depends on the maximum number of VoIP channels, which is configered for the VoIP provider (page **Public switched tel. networks > Voice over IP (VoIP page) > Provider > Configure**).

Note: If a call channel should be reserved for one subscriber this subscriber has to be assigned alone to a group.

Note: A call channel reserved for a group is only available for the members of the group who are logged in the group outgoing.

Busy-on-Busy

Use case

- **Announcement before answering** (**On busy** or **Always**) is enabled
- The number of calls put through to the waiting loop should be limited by **Busy-on-Busy** so that not more callers are in the waiting loop than can be handled.
- Callers who are not put through to the waiting loop due to the limit, hear the busy tone.
- For this, **Busy-on-Busy dynamic** may not be selected.

deactivated

The number of callers to the group is unlimited.

busy from x connections on

The number of callers to the group is limited to a specific number.

dynamic

Note: Selecting **dynamic** can have effects on the function **Do-not-disturb**, **Call waiting** and **Waiting loop** if **Announcement before answering** is enabled.

Registered and logged-in group members are recorded and the number of connections made to the group is automatically corrected downwards if necessary.

Ann. before answering

Off

On busy

When calling a busy group (all group members are busy) the caller hears an announcement before answering for greeting. After the announcement before answering, the caller is put into the waiting loop and hears Music on Hold.

Always

When calling a group the caller always hears the announcement before answering for greeting. If the line is busy, he is put into the waiting loop after the announcement before answering and hears Music on Hold. If the called phones are available, the caller hears Music on Hold also while the phones are ringing.

Ann. before answering | Call signalisation

Immediately

The call is signalled during the announcement before answering.

After announcement before answering

The call is signalled after the announcement before answering when the caller hears Music on Hold.

Ann. before answering | Selection of announcement

Announcement before answering which is played for the caller.

Rework time

Time that a group member cannot be called after an incoming internal or external call (e. g. for Call Center agents). During this time, the do-not-disturb function is enabled for the subscriber.

Note: The rework time can be cancelled at any time by switching off the do-not-disturb function (e. g. using a function key at the system phone).

Note: The following characteristics apply for the devices COMfortel 3200/3500: Enabling this function on the telephone is not displayed in the configuration manager of the PBX. Enabling this function in the configuration manager of the PBX has no influence on the telephone (status is not displayed under the function keys)

Call restrictor (incoming)

None

No call restrictor is used for group members logged in as outgoing.

As subscriber

The call restrictor selected for a subscriber is used.

Call deblocker (incoming)

None

No call deblocker is used for group members logged in as outgoing.

As subscriber

The call deblocker selected for a subscriber is used.

Follow-me | Destination number:

Internal and external calls to the group are forwarded to the entered destination number.

Multi-path Call Forwarding | Destination number:

Internal and external calls to the group are signaled additionally to the entered destination number.

Note: Multi-path call forwarding for a group is executed during the entire calling period independent of the ringing sequence for the group. It is not executed if the group itself is the destination of a forwarded call.

Call Forwarding | Settings | only external calls

External calls to the group are forwarded. Internal calls are not forwarded.

Call Forwarding | Settings | Cascading

If call forwarding has been configured for the internal destination of an existing call forwarding, it is possible to reroute already forwarded calls (cascading).

Calls also will be rerouted to the new call forwarding destination. During cascading, the PBX memorises the subscribers who have already been

called for this call. If the subscriber is the destination of a call forwarding again, the call ends with the previous destination.

The caller hears the dial tone until the call is accepted. If the destination is busy or the maximum call duration has been reached, a busy signal is issued.

Note: When switched off, cascading does not take place. Calls end with the entered destination.

Call Forwarding | Settings | CF on group call

Calls directed to a group of which the group is a member will be forwarded to the CF destination of the group.

Call Forwarding | immediately | Destination number:

Internal and external calls to the group are forwarded immediately to the entered destination number.

Call Forwarding | on busy | Destination number:

Internal and external calls to the group are forwarded on busy to the entered destination number.

Call Forwarding | on no reply | Destination number:

Internal and external calls to the group are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Call Forwarding | on all logged out | Destination number:

Internal and external calls to the group are forwarded to the entered destination number, if all group members are logged out.

Voicemail/Fax Box

Proceed as follows:

- Separately for groups
- Using the Configuration Manager (page **Groups > Properties**)

GROUP

Select the group you want to display/configure.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place on page **Groups > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Groups > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Fax box | Phone number of the box

Requirements:

- Voice mail/fax function (optional, see Technical Data in the Instructions)
- Created fax box for the group

Note: If it is not clear whether it is a fax or not, the call will be signaled to the group or it will be accepted by the voice mailbox.

Note: If a fax call is accepted by the group, it can be forwarded to a fax box by means of blind transfer.

Note: A fax call which is accepted by a voice mailbox will not be forwarded to a fax box.

Voice mailbox | Box number

Requirements:

- Voice mail/fax function (optional, see Technical Data in the Instructions)
- Created voice mailbox for the group

Both the group and the selected voice mailbox will be automatically called for all incoming calls made to the internal number of the group.

Important: The PBX supports only one central answering machine. An answering machine at the COMfortel system phone will be disabled as soon as a voice mailbox is selected for the subscriber. Querying and deleting voice messages as well as replaying own announcements is still possible. The call recording function can be used.

Note: If call forwarding immediately (subscriber/group), the boss/secretary function (system phone) or a waiting field reception (for one waiting field group, one enabled waiting field reception is sufficient) is enabled, the call will not be made to the voice mailbox!

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on busy

The voice mailbox will be called if the internal number of the group is busy and if no call forwarding on busy has been enabled for the group.

on no reply + on busy

The voice mailbox will be called if the group does not reply or if the internal number of the group is busy.

on all logged out

The voice mailbox will be called when all group members are logged out.

Delay time for CF on no reply

Call acceptance during the time entered here.

Settings

Proceed as follows:

- Separately for groups
- Using the Configuration Manager (page **Groups > Properties**)

GROUP

Select the group you want to display/configure.

Important: The settings done are only valid on the condition that they are not overridden by a profile. The simultaneous display takes place on page **Groups > Properties** under **Overridden by profile in configuration**.

Note: [Name configuration] ...: In the displayed configuration, the function is overridden by the displayed setting. Configuration using the Configuration Manager (page **Groups > Profiles > ...**)

Note: not overridden by a profile: The function is currently not overridden by a profile.

Note: (not profile-controlled): The function can not be overridden by a profile.

Number presentation (internally)

Group numbers

All group members logged in as outgoing will present the group number for internal outgoing calls. If the group is internally called the subscriber number is transferred back.

Subscriber number

All group members logged in as outgoing will present their own subscriber number for internal outgoing calls.

Preliminary pick-up query and monitoring

Requirements:

- VoIP end device which can evaluate the available information by BLF
- VoIP phone from another manufacturer: **SIP convenience package** / **SIP convenience package brand plus** enabled (optional, see Technical Data in the Instructions)

All information about the incoming calls of the group is available for evaluation by BLF.

Note: When switched off, no phone numbers of the group‘s calls are transmitted by BLF.

Pick-up of all calls to group members

A member of this group can pick up calls to the phone number of the group as well as direct calls to individual members of the group.

Note: When switched off, group members can only pick up direct calls to the phone number of the group.

Central caller list

Calls saved in the caller list are automatically refreshed for all group members. That means, if a connection is established (incoming or outgoing) from a group member to a caller in the caller list, the phone number is deleted from the caller lists of the other group members, too.

Devices

Voice mail and fax boxes
Device Wizard
Door Terminals
Announcement Outputs
Alarms
Relays (Actuators)
Action URLs

Voice mail and fax boxes

Management Voice mail and fax boxes
Management Profile
Settings Profile Assignment
Settings of the Voice Mail/Fax Function
Settings of the Voice Mailbox
Settings for the Fax Box

Management Voice mail and fax boxes

Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Phone numbers**)

Note: Click the column headings to change the sorting criteria. Click the column headings again to change the sorting order.

Requirements:

- Voice mail and fax function (optional, see activation)

+ New

Depending on the selection, a new voice mailbox or fax box will be added.After that, the settings for the voice mailbox/fax box must be made.

Type

Select a **voice mailbox** or **fax box**.

Note: If under **Public switched tel. networks > Analogue / ISDN / VoIP** in the column **Record on** the voice mailbox was assigned the marking *,the box gets its marking here. Therefore, the box can be used for call recordings.

Note: The function **Call recording** is available only in certain countries.

Phone number or Phone number From to

This is an internal number. You can select any of the numbers in the internal supply of phone numbers.

Single box for one owner

Enter an internal phone number in the entry field **from** or **Phone number**.

Several boxes for one owner

Enter the lowest phone number of the number range between **10** and **9999** in the entry field **from** and enter the highest phone number of the number range in the entry field **to**.

Note: Double allocation of a number is not possible.

Note: Using phone numbers with few digits (two or three-digit numbers) inevitably restricts the possible supply of phone numbers requiring more digits. For example, if the phone number 12 is assigned, the phone numbers 120-129 and 1200-1299 will no longer be available.

Note: The default exchange line access number on the PBX is 0 in the factory settings; however, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the box within the PBX and the connected system phones.

Owner

Select the internal phone number under subscriber or group that is to be assigned to the box.

An owner can query the box. If the administrator has not refused the access authorisation, the owner can perform the following changes/ actions using the Configuration Manager:

- Copying, moving, and deleting messages
- **Properties** of the box not controlled via a profile (except **Users/ authorisations** and the **maximum recording capacity**)
- **Voice mail/fax box properties** of the subscribers and groups not controlled via a profile

For a voice mailbox, the owner can additionally perform the following changes/actions:

- Recording, storing in the PBX, reading from the PBX (downloading), listening to and deleting of announcements
- Properties of the box under **Call acceptance/announcement** that are not controlled via a profile

The following properties of the owner will be applied to the box:

- Exchange line settings of the subscriber or exchange line settings of the group (in a group with the **Exchange line authorisation**: **Use the exchange line authorisation of the subscriber**, the box does not have any exchange line authorisation).
- Only for subscribers: call allowance account

Note: Boxes can be preconfigured by not assigning an owner to the box. Nevertheless, the boxes cannot be used until an owner is assigned.

If voice mailboxes are to be preconfigured for the function **Call recording**, no owner must be assigned to these boxes.

Note: No **owner** can be assigned to a voice mailbox (marked with ***) already set for the function **Call recording** If, at a later point of time, the box is no longer used for call recordings meaning the box is to be used ad answering machine, an owner or several owners can be assigned.

Based on

Select a template from all boxes created so far.

Template:

Voice mailbox

The new voice mailbox obtains all properties of the selected template during creation (except for **Users/authorisations**, **PIN for remote access** and **Announcements**).

Fax box

The new fax box obtains all properties of the selected template during creation (except for **Users/authorisations**).

Standard

The **Basic settings** are used for the box just created.

1:1 assignment

If you want to be able to see from the number to which internal subscriber or internal group the box belongs, you can assign corresponding box numbers:

- You can add a digit (prefix) for the box numbers to an up to 3-digit internal subscriber/group number.

Example: The subscribers 400, 401, 402, 405, 407 are to be assigned the boxes 7400, 7401, 7402, 7405, 7407.

- If the internal subscriber/group numbers have 4 digits, the box numbers can be created by replacing the first digit.

  Example: The subscribers 4400, 4401, 4402, 4405, 4407 are to be assigned the boxes 7400, 7401, 7402, 7405, 7407.

Note: The number range selected for the box numbers needs to be sufficiently large: The number range for the examples (from 7400 to 7407) includes eight numbers, even though only the five corresponding boxes are created.

Note: If the corresponding number range contains an internal number (subscriber, group) which is already owner of a group of the type to be created (voicemail or fax box), no further box will be created for this number.

Note: If the selected number range for the box numbers already contains a corresponding box number, the owner of the box remains the same and a message will be displayed.

Note: If the new owner is not yet an owner of user of another box, the recently created box will automatically be enabled for the subscriber or group.

Requirements:

- Configured internal subscribers and/or groups
- The number range for the numbers of the boxes is still free.

A new box within the box number range **from** ... **to** must completely be configured. Then the box can be assigned by clicking **Save > 1:1 assignment**.

Box numbers which correspond automatically for different owners are created.

Properties Configure

Opens the configuration page for the box.

Filter:

The list of boxes is filtered by the entered or selected value.

Delete

Deletes the entries marked in the list.

Restoring boxes

Due to missing data in on the data storage device (e. g. because of an exchange) or in the database of the PBX, the affected boxes are not ready to use.

The boxes affected by this function are marked with .

Missing data on the data storage device

Restore

Note: The restoration of the boxes does not include the messages, box-specific announcements and language files.

Missing data in the database on the PBX

You can restore these boxes using the following data:

- Internal number and name of the box
- Messages
- Box-specific announcements

Note: Internal numbers must only be assigned once in the PBX. The former phone number of the box to be restored cannot be used if it is still present in the PBX as an internal number (page **Overviews > Phone numbering plan > Total overview**).

Type

Select one of the available options.

Phone number

If the phone number is already present, a new internal number between **10** and **9999** must be entered in the entry field for the box to be restored.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Owner

If necessary, select the internal phone number of the owner.

Restore

Management Profile

Proceed as follows:

- Using the Configuration Manager

Profile name

The available profiles (default profiles, user-defined profiles) are displayed.

View

The selected settings can be viewed (default profiles).

Configure

The user-defined profiles can be adapted to the corresponding requirements (user-defined profiles).

+ New

Creation of a new profile.

Settings Profile Assignment

Proceed as follows:

- Using the Configuration Manager

A configuration depending profile can be assigned to each subscriber, group or voice mail/fax box.

Assign profile to all

The same profile is assigned to all subcribers, groups or boxes for one configuration.

Settings of the Voice Mail/Fax Function

System Information
Automatic clean-up
Memory
Automatic Memory Check
Channels Simultaneously for Outgoing Calls
Language Files

System Information

Proceed as follows:

- Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Basic settings**)
- Jointly for all Voice mail and fax boxes

Requirements:

- Voice mail/fax function (optional, see activation)

Link to the page **Administration > System messages** for specifying how the system reacts to system information.

Generate notification at | … memory utilisation

When the amount of memory used exceeds the specified level, this information is logged and, if specified, sent as an e-mail.

Automatic clean-up

Proceed as follows:

- Using the Configuration Manager (**Devices > Voice mail and fax boxes > Basic Settings** page)
- Jointly for all voice mail and fax boxes

Requirements:

- Voice mail and fax functions (optional, see activation)

Note: If at least one voice mailbox on the PBX is configured for the **Call recording** function, the global setting option for automatically cleaning up the voice mailboxes can no longer be used. However, every voice mailbox can be configured individually, and then operates independently of the device's use as an answering machine or for call recordings. The **Call recording** function is only available in specific countries.

Perform at memory utilisation

When the amount of memory used reaches the specified level, the oldest message that is marked as read is deleted when a new message is received. Archived messages will not be deleted.

Note: If the clean-up fails, this information is logged and, if specified, sent as an e-mail.

Type of clean-up

oldest messages in each box

The oldest messages marked as read in each box will be deleted.

oldest messages in the system

The oldest messages marked as read, of all messages stored in the voice mail/fax system, will be deleted (even if all messages belong to only one box).

Memory

Proceed as follows:

- Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Basic settings**)
- Jointly for all voice mail and fax boxes

Requirements:

- Voice mail/fax function (optional, see activation)

Activate encryption

The whole partition of the data storage device is encrypted using a password.

Important: In order to have the partition encrypted the data storage device needs to be formatted first.

Important: The recorded data is saved unencrypted on the data storage device.

Password

If the encryption is activated for the first time, the admin password of the PBX is set.

Alternatively, you can set a personal password via **Edit password**.

Note: Formatting the data storage device is not possible until you selected save.

Formatting

The inserted data storage device is formatted and the directory tree required for the voice mail and fax function is created.

Important: When the inserted data storage device is formatted, all the data currently present on it is deleted.

Important: After the data storage device has been formatted with the EXT4 file system it can no longer be read on a Windows PC. For this reason, Windows prompts you to format the data storage device. Under no circumstances should you confirm this prompt by selecting "Yes".

Eject data storage device

The inserted data storage device is ejected. The PBX can be powered down and the data storage device securely removed.

Attention: Do not remove the data storage device from the PBX while the system is running.

- Switch off the PBX before removing the data storage device.

Automatic Memory Check

Proceed as follows:

- Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Basic settings**)
- Jointly for all voice mail/fax boxes

Requirements:

- Voice mail/fax function (optional, see activation)

Automatic memory check

The data storage device is automatically checked for errors at the selected interval and at the selected time.

Note: If the memory check fails, this information is logged and, if specified, sent as an e-mail.

Interval

off

The memory check is disabled.

daily

The memory check will be made daily.

weekly

The memory check will be made weekly.

monthly

The memory check will be made on the 1st day of every month.

Time

Time at which the memory check is performed.

Note: The memory check may take a few minutes. The voice mail and fax boxes cannot be accessed during the memory check. For this reason, the memory check should be performed at a time when access to the voice mail and fax boxes is not required (e.g. at night).

Channels Simultaneously for Outgoing Calls

Proceed as follows:

- Using the Configuration Manager (**Devices > Voice mail and fax boxes > Basic settings** page)
- Jointly for all Voice mail and fax boxes

Requirements:

- Voice Mail/Fax function (optional, see activation)

Simultaenously used channels for outgoing calls

Maximum number of call channels that can be used simultaneously for outgoing calls from the Voice Mail/Fax function (message forwarding, fax transfer).

Language Files

Proceed as follows:

- Using the Configuration Manager (**Devices > Voice mail and fax boxes > Basic settings** page)
- Jointly for all Voice mail and fax boxes

Requirements:

- Voice Mail/Fax function (optional, see activation)

The voice mailboxes' default announcements, and also the spoken user guidance for remote access, are stored in "language files" (for English-language announcements, english.fs is used). Up to four language files can be stored and used simultaneously. This means different languages can be used for different voice mailboxes.

Import

The language file (*.fs) stored on a (PC) data storage device is selected and imported.

Default language

Unless a different language has been selected for specific voice mailboxes, the selected language is usually used for all voice mailbox announcements.

Note: The first language imported into the PBX is automatically set as **Default language**.

Settings of the Voice Mailbox

Box settings
Users/Authorisations
Announcements
Call Acceptance/Announcements

Box settings

Using the Configuration Manager (**Devices > Voice mail and fax boxes > Properties**)

VOICE MAIL AND FAX BOXES

Select the voice mailbox for which the settings are to be made.

Note: If a voice mailbox that is already configured for the function **call recording** is selected, certain functions such as **remote access** or **call pick-up** are not available on this page. The function **call recording** is available only in certain countries.

Language of the announcements

If several language files have been imported, the language of the announcements can be configured (**Devices > Voice mail and fax boxes > Basic settings**).

Standard

The language file is used that is set under **Devices > Voice mail and fax boxes > Basic settings**.

e. g. **deutsch - V02.01**

The language of the selected language file is used.

Automatic replay of new messages

Note: This setting does not affect querying the voice mailbox with a system phone.

All new messages will be replayed automatically without the need for further inputs and marked as listened to if the voice mailbox is queried or if message forwarding is used.

Remote access

A voice mailbox can be queried from an external connection. The remote access can be performed by the administrator, sub-administrator, or owner/user after entering the corresponding PIN

PIN for remote access

Freely selectable 6-digit PIN. With the exception of the admin PIN, a PIN must not begin with a zero.

Note: If no entry is made, the PIN of the box owner is automatically used.

Note: All pins in the PBX are unambiguous. This means it is not possible to assign the same PIN twice. As the PINs can also be entered by phone, only numbers are allowed.

Message forwarding

Incoming new messages are forwarded to the configured destination

Destination number

Internal number

External number without exchange line access number

Forwarding after lately recorded messages

Enter the number of messages between **1** and **20**, after which the forwarding can be started (depending on the configured notification time period).

Notification time period from hr until hr

Enter the time period in which the messages are to be forwarded.

daily

The messages are forwarded daily in a determined time period.

Mon - Fri until **Sun**

The messages are forwarded on the selected week days in a determined time period.

Number of repetitions with min. pause

Select the repetitions in the first list field and select the pause length between the forwarding attempts in the second list field respectively in between **0** and **9** repetitions or minutes.

Call take-over

The call take-over function enables a user to take over a call which was accepted by the voice mailbox.

global

Every internal subscriber is permitted to take over calls.

Authorised users

Every owner or user is permitted to take over calls. If a group is owner or user, every group member can take over calls.

Maximum recording capacity

The maximum recording capacity specifies the maximum memory capacity in megabytes (MB) available for the recording of messages of the box. Additionally, the maximum length per recording can be specified.

It indicates the number of minutes after which the voice mailbox ends a recording automatically (in as far as the caller has not already hung up).

Maximum size of the box

Enter a value for the recording capacity between **15** and **1048576** MB. Behind the entry field, it is displayed how much the entered value is in minutes. If no value is entered, the recording capacity is unlimited (maximum memory utilisation).

Maximum length per recording

Enter a value for the recording length between **1** and **99** min. If no value is entered, the recording length is unlimited (maximum memory utilisation).

Note: Under the following circumstances, the entered value is not available to the box:

- There is not enough memory capacity available on the storage media.
- There is not enough memory capacity available on the storage media.

Automatic clean-up

When the amount of memory used reaches the specified level, the memory is cleaned up.

Note: If at least one voice mailbox on the PBX is configured for the **Call recording** function, the global setting option for automatically cleaning up the voice mailboxes can no longer be used. The individual setting option that can be used here for every voice mailbox is used instead, independently of its use as an answering machine or for call recordings. This option can be disabled at any time. The **Call recording** function is only available in specific countries.

Perform at xx % memory utilisation

When the amount of memory used reaches the specified level (in the range **50** to **95** %), the oldest messages that are marked as read are deleted when new messages are received. Archived messages will not be deleted.

Note: If the clean-up fails, this information is logged and, if so configured, sent as an e-mail.

E-mail transfer

Requirements:

- Configured outgoing mail server (under **Administration > E-mail > Outgoing mail server**)

Outgoing mail server

Standard

The e-mails are transferred via the default outgoing mail server.

Name of an outgoing mail server: The e-mails are transferred via the selected outgoing mail server.

Send e-mail in HTML format

Note: When switched off, e-mails are sent in text format.

Send on incoming messages

Attach incoming messages as file

In the case of an incoming message, this information is sent as an e-mail.

Audio file in MP3 format

The audio file is provided in *.MP3 format.

Note: When switched off, the audio file is provided in *.wav format.

Mark sent messages as read

In the message list, a sent message is marked as read.

Send at memory utilisation

Enter a value between **50** and **90** %. When the amount of memory used exceeds the specified level, this information is sent as an e-mail.

Send in the case of failure of automatic clean-up

In the case of failure of automatic clean-up, this information is sent as an e-mail.

(System e-mail)

Link to the page **Administration > System messages** for specifying how the system reacts to system information.

Language file

Select the language file.

Note: If you have created under **Administration > E-mail > Internationalisation > New** a user-defined language file and select it, the e-mails will be sent with the text modules of the selected language file.

Logo

Default logo

The default logo is used.

Name Logo

The selected logo is used.

Message recipients

E-mail transfer @Test

The e-mail addresses of up to 10 message recipients can be entered for e-mail transfer. Subsequently, a test e-mail can be sent to check whether the e-mail transfer to the message recipients entered works without fault.

Requirements:

- Configured DNS server (under **Administration > IP configuration**)
- Configured outgoing mail server (under **Administration > E-mail > Outgoing mail server**)

Note: The PBX enters sent e-mails in a log file (page **Monitoring > System messages**).

Note: E-mails sent by the PBX may be treated as spam mail by the recipient of the message.

Users/Authorisations

Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Properties**)

VOICE MAIL AND FAX BOXES

Select the voice mailbox for which the settings are to be made.

Note: If a voice mailbox that is configured with the function **call recording** is selected, this page **users/authorisations** is not displayed because this type of box must not have an owner. The function **call recording** is available only in certain countries.

User and authorisations

A box can only be used if an owner has been assigned to it (page **Devices > Voice mail and fax boxes > Phone numbers**). In addition to the owner, further users can be assigned to the box.

Requirements:

- Configured internal subscribers and/or groups

Note: An owner is automatically entered as a user with all authorisations.

Note: All group members have the authorisations assigned to the group. If a group member or a subgroup is also entered individually as a user, the authorisations assigned to this user will also apply to this group member/subgroup.

+ New

Opens a new line for selecting another user/group.

User

Select a user/the users. If only one additional user is required, select the user in the first list field. If more than one user is required, they are to be selected via both list fields (from ... to).

Administrate announcements

The authorisations of the announcements are expanded.

- Recording, storing in the PBX, reading from the PBX (downloading), listening to and deleting of announcements.
- Properties of the box under **Call acceptance/Announcement**, which are not controlled via a profile.

Administrate settings

The authorisations of the properties of the box that are not controlled via a profile are expanded (except for **User/authorisations, Maximum recording capacity** and **Fax ID**).

Note: Only a user is permitted to change these settings via phone.

Announcements

Using the Configuration Manager (page **Devices > Voice mail and fax boxes > Properties**)

VOICE MAIL AND FAX BOXES

Select the voice mailbox for which the settings are to be made.

Proceed as follows for saving:

- Using the Configuration Manager

Proceed as follows for recording, listening to and deleting:

- Using the Configuration Manager
- On the internal phone via character string (see Voice Mailbox)

Note: If a voice mailbox that is configured with the function **call recording** is selected, this page **announcements** is not displayed because this type of box must not have an announcement. The function **call recording** is available only in certain countries. .

Upload new recorded accouncement

You can save up to 10 announcements (audio files (*.wav)) lasting a maximum of 3 minutes each for each voice mailbox in the PBX.

The audio files (*.wav) must have the following properties:

- Format: 16 kHz, 16-bit, PCM, mono or 8 kHz, 8-bit, mono, A-law
- File name: up to 32 characters

Note: In the Internet, you can convert the audio files into the audio format required for the PBX (see Auerswald Audio Converter). For

example, use the Audio Converter to convert audio files (*.mp3, *.wav or *.raw) into G.711 (*.wav) audio format.

The audio file (*.wav) stored on a (PC) data storage device is selected and imported.

Read from PBX Read

The announcements for the voice mailboxes saved in the PBX can be saved on the PC as backup copies.

Requirements:

- At least one existing announcement for the voice mailbox

Save or play the file on the PC.

Record announcement Record

Announcements for voice mailboxes can directly be recorded via the receiver of an internal phone.

User

The phone is called.

Sub-administrator/administrator

Select the phone via the list field **Record announcement at the subscriber**. The phone is called.

The call must be accepted and after the tone an announcement can be recorded.

Rename announcement Change

Change the announcement name.

Delete announcement Delete

The announcement will be deleted.

Call Acceptance/Announcements

Using the Configuration Manager (page **Device > Voice mail and fax boxes > Properties**)

VOICE MAIL AND FAX BOXES

Select the voice mailbox for which the settings are to be made.

Note: If a voice mailbox that is already configured for the function **Call recording** is selected, call acceptance and announcements are not available for this type of box. Therefore, this page **Call acceptance/announcements** is not displayed. The function **Call recording** is available only in certain countries.

Readiness

on

Switches the readiness on.

off

Switches readiness off.

configuration-dependent

The setting made in the profile (page **Devices > Voice mail and fax boxes > Profiles > Configuration**) assigned to the currently valid configuration will apply.

Deputy function

Requirements:

- Corresponding announcement saved in the PBX

- Enabled **Readiness** of the voice mailbox

Note: This setting is also possible configuration-dependent via profiles (**Devices > Voice mail and fax boxes > Profiles > Configuration > View**).

A caller can set up a connection to a deputy. For this purpose, the caller dials **1** via DTMF during the announcement or recording phase.

Destination number

Enter an internal or external phone number.

Note: For an external number that has less than five digits, the local area code should additionally be entered.

Announcement with enabled deputy function

Select an announcement that is replayed for callers if the deputy function is enabled.

Call acceptance/announcement

The call acceptance of the voice mailbox can be configured to show different behaviour for different call types. The **Call acceptance/ announcement** defines the standard behaviour for the case that no **Call-specific behaviour** applies.

Requirements:

- **Readiness** enabled
- Disabled **Replacement function**

- If **Box announcement** is selected: the corresponding announcement saved in the PBX

Note: This settings is also possible configuration-dependent via profiles (**Devices > Voice mail and fax boxes > Profiles > Configuration > View**).

No call acceptance

All calls not controlled via **Call-specific behaviour** will not be accepted. The caller hears the call tone.

Reject call

All calls not controlled via **Call-specific behaviour** will not be accepted. The caller hears the busy signal (subscriber/group will not be called additionally) or the call tone (subscriber/group will be called).

Default announcement

All calls not controlled via **Call-specific behaviour** will be accepted. The announcement of the selected language file will be replayed (see **Language of the announcements** under **Devices > Voice mail and fax boxes > Properties > Box settings**).

Box announcement

Select one of the available options in the list field on the right.

Recording

The caller can leave a message and the option **Save short messages** can be enabled.

Save short messages

The phone number of the caller is displayed even when the caller does not leave a message or hangs up during the announcement.

Admin

Page **User data > Messages**

Owner

Page **Messages (Voice mail and fax boxes)**

Note: The voice mailbox only records calls for which a caller leaves a message (of more than two seconds).

Call-specific behaviour (exception table)

The call acceptance of the voice mailbox can be configured to show different behaviour for different call types.

Note: If the call-specific behaviour is permanently set in a configuration by means of a profile, the corresponding table can be viewed. Click the corresponding profile name which is displayed in red font. The corresponding profile configuration is displayed.

Requirements:

- Separately for voice mailboxes
- Separately for configurations
- Using the Configuration Manager

Note: This setting is also possible configuration-dependent via profiles (**Devices > Voice mail and fax boxes > Profiles > Configuration > View**).

Call type

Anonymous callers

Call acceptance for calls without transferred phone number.

Internal calls

Call acceptance for internal calls.

Door terminal calls

Call acceptance for door calls.

External calls

Call acceptance for external calls.

User-defined

Call acceptance for calls from the entered phone number range (**Ext. phone number/range**, e. g. **00** for calls from abroad).

Call acceptance

Standard

For call acceptance see **Call acceptance/announcement** under **Standard behaviour**.

No call acceptance

No reaction on calls, the caller hears the call tone.

Reject call

The caller hears the busy signal (subscriber/group will not be called additionally) or the call tone (subscriber/group will be called).

Default announcement

The announcement of the selected language file will be replayed (see **Language of the announcements** under **Devices > Voice mail and fax boxes > Properties > Box settings**). (No further action is required.)

Box announcement

The announcement of the voice mailbox selected in the **Announcement** list field (Announcement 1 to 10) will be replayed.

Recording

The caller can leave a message. The setting for **Save short messages** also applies to the call-specific behaviour.

Settings for the Fax Box

Box Settings
Users/Authorisations
Call Acceptance

Box Settings

Using the Configuration Manager (page **Device > Voice mail and fax boxes > Properties**)

VOICE MAIL AND FAX BOXES

Select the fax box for which the settings are to be made.

Fax ID

Sender name

Fax ID (phone number)

The fax ID is inserted into the top line of a fax during the fax transfer and, thus, permits an exact identification of the sender.

Note: Recipients of a fax frequently send their responses to the number entered in the header of the fax. Therefore, the number under **Fax ID (phone number)**, should be one of your own extensions/phone numbers with fax acceptance.

Enter a sender name that is to be inserted into the top line of a fax along with the fax ID.

Enter the complete phone number of the own external connection as fax ID. Number in the format +[country prefix][city code][external phone number] (e. g. +495306920094).

Fax transmission

ECM (Error Correction Mode)

Transmission errors occurring during the transmission (e. g. because of faulty lines) will be corrected provided that EMC operation (Error Correction Mode) has also been configured for the opposite end.

Note: If transmission errors occur very often (e. g. overseas lines of VoIP connections with frequent packet losses), the transmission cannot be carried out since the fax machines involved will cancel the transmission due to an excessive number of transmission repetitions. In such cases, it may be sensible to disable EMC error correction. Without EMC error correction, the transmission errors occurred will be displayed as erroneous dots or lines.

A maximum of 9600 baud

The transmission rate of a fax will be adapted in such a way that a faultless transmission quality will be reached, where possible.

Note: The PBX supports a maximum transmission rate of 14400 baud. This transmission rate may be too high for specific lines (e. g. overseas lines or VoIP connections) and may lead to transmission interferences. In such cases, it may be sensible to restrict the transmission rate to 9600 baud

Maximum recording capacity

Maximum size of the box

The maximum recording capacity specifies the maximum memory capacity in megabytes (MB) available for the recording of messages of the box.

Enter a value for the recording capacity between **15** and **999** MB. Behind the entry field, it is displayed how much the entered value is in minutes. If

no value is entered, the recording capacity is unlimited (maximum memory utilisation).

Note: Under the following circumstances, the entered value is not available for the box:

- There is not enough memory capacity available on the storage media.
- The memory capacity on the storage media is utilised by other boxes.

Automatic clean-up

The memory is cleared when a certain fill level is reached.

Perform at xx %… memory utilization

If the maximum number of stored messages or the maximum recording capacity of the box (between **50** up to **95** %) is reached, the oldest messages marked as read will be deleted as soon as new messages arrive. Archived messages will not be deleted.

Note: If the clean-up fails, this information is logged and, if specified, sent as an e-mail.

E-mail transfer

Requirements:

- Configured outgoing mail server (under **Administration > E-mail > Outgoing mail server**)

Outgoing mail server

Standard

The e-mails are transferred via the default outgoing mail server.

Name of an outgoing mail server: The e-mails are transferred via the selected outgoing mail server.

Send e-mail in HTML format

Note: When switched off, e-mails are sent in text format.

Send on incoming messages

Attach incoming messages as file

Note: Faxes are provided as *.pdf files.

Mark sent messages as read

Important: If automatic marking as read is enabled, the fax box will be cleaned up automatically. This option should normally be enabled

Send at memory utilisation

Enter a value between **50** and **90** %. When the amount of memory used exceeds the specified level, this information is sent as an e-mail.

Send in the case of failure of automatic clean-up

If the automatic clean-up fails, an e-mail is sent.

(System-e-mail)

Link to the page **Administration > System messages** for specifying how the system reacts to system information.

Send after fax transfer

off

Disables e-mail transfer after fax transfer.

in the case of errors

Enables e-mail transfer after failed fax transfer.

Always

Enables e-mail transfer after fax transfer.

Attach sent message to file.

Language file

Select the language file.

Note: If you have created under **Administration > E-mail > Internationalisation > New** a user-defined language file and select it, the e-mails will be sent with the text modules of the selected language file.

Logo

Default logo

The default logo is used.

Name logo

The selected logo is used.

Message recipients

E-mail transfer @Test

The e-mail addresses of up to 10 message recipients can be entered for e-mail transfer. Subsequently, a test e-mail can be sent to check whether the e-mail transfer to the message recipients entered works without fault.

Requirements:

- Configured DNS server (under **Administration > IP configuration**)
- Configured outgoing mail server (under **Administration > E-mail > Outgoing mail server**)

Note: The PBX enters sent e-mails in a log file (page **Monitoring > System messages**).

Note: E-mails sent by the PBX may be treated as spam mail by the recipient of the message.

Users/Authorisations

Using the Configuration Manager (page **Device > Voice mail and fax boxes > Properties**)

VOICE MAIL AND FAX BOXES

Select the fax box for which the settings are to be made.

User and authorisations

A box can only be used if an owner has been assigned to it (page **Devices > Voice mail and fax boxes > Phone numbers**). In addition to the owner, further users can be assigned to the box.

Note: An owner is automatically entered as a user with all authorisations.

Note: All group members have the authorisations assigned to the group. If a group member or a subgroup is also entered individually as a user, the authorisations assigned to this user will also apply to this group member/subgroup.

Requirements:

- Configured internal subscribers and/or groups

+ New

Opens a new line for selecting another user/group.

User

Select a user/the users. If only one additional user is required, select the user in the first list field. If more than more user is required, they are to be selected via both list fields (from ... to).

Administrate settings

The authorisations of the properties of the box that are not controlled via a profile are expanded (except for **User/authorisations, Maximum recording capacity** and **Fax ID**).

Call Acceptance

Using the Configuration Manager (page **Device > Voice mail and fax boxes > Properties**)

Voice mail and fax boxes

Select the fax box for which the settings are to be made.

User and authorisations

A box can only be used if an owner has been assigned to it (page **Devices > Voice mail and fax boxes > Phone numbers**). In addition to the owner, further users can be assigned to the box.

Note: An owner is automatically entered as a user with all authorisations.

Note: All group members have the authorisations assigned to the group. If a group member or a subgroup is also entered individually as a user, the authorisations assigned to this user will also apply to this group member/subgroup.

Requirements:

- Configured internal subscribers and/or groups

+ New

Opens a new line for selecting another user/group.

User

Select a user/the users. If only one additional user is required, select the user in the first list field. If more than more user is required, they are to be selected via both list fields (from ... to).

Administrate settings

The authorisations of the properties of the box that are not controlled via a profile are expanded (except for **User/authorisations, Maximum recording capacity** and **Fax ID**).

Device Wizard

Management of the Device Wizard

Management of the Device Wizard

Device Wizard
Door Terminals
Configuring Analogue Door Terminals
Configuring VoIP Door Terminal
Configurig IP Switch Relays
Configuring Loudspeaker Announcements
Online Configuration

Device Wizard

Proceed as follows:

- Using the Configuration Manager (page **Devices > Device wizard > Configure**)

Name

Freely selectable name (maximum of 16 characters).

Device template

Select the device template corresponding to the device.

Module

Select the module on that the device is connected.

Port

Port to which the device is connected.

New

A new device template is created.

Delete

Deletes the entries that are marked in the list.

File management

List of all device templates.

Import a new device template in XML format.

The device template is saved as an *.xml file to the data storage device (PC).

Configure

Opens the configuration of the device template.

Door Terminals

Proceed as follows:

- Using the Configuration Manager (page **Devices > Device wizard > Configure**)

An existing door terminal can be connected to the PBX so that various door functions such as speech connection with the door and actuation of the door opener from internal phones can be carried out. This door terminal must be created in the configuration of the PBX and allocated an internal number where it can be called.

The door terminal and the relays and switch inputs (door bell buttons) available for it can be created with the device wizard. Door calls can be created for the switch inputs of the door terminal by selecting call destinations. To detect their actuation e. g. because a door bell button has been pressed, the PBX needs to know which DTMF sequence or which SIP invite the door terminal sends when a button is pressed (see manual and settings of the device in question).

Note: The DTMF sequence or the SIP invite is the phone number that the device dials when a door bell button is pressed.

Note: After the door terminal has been configured in the PBX, the complete door call distribution will be controlled by the PBX – with regard to the call destinations as well as with regard to the maximum call duration. The maximum call duration can only be limited here, but no longer than the value configured in the door station.

Note: The door call distribution can be set to depend on the configuration – differently for night and day, for example.

Only for VoIP door terminals: To create a connection with a VoIP door terminal, the PBX needs to know the SIP password of the connected device. To control the relays and switch inputs, the PBX also needs to know IP address and port of the connected device as well as authentication data (user name and password) to control the relays (see manual and settings of the device in question).

Various door terminals supported by the PBX are available as device templates. They already include several default values for operation of the door terminals in their factory settings.

If no of the device templates on offer matches your installation, you can select a user-defined device template.

Configuring Analogue Door Terminals

Proceed as follows:

- Using the Configuration Manager (page **Devices > Device wizard > Configure**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number can be called directly by an internal subscriber. When put together with the numbers of the door bell buttons, this creates 3 to 4-digit phone numbers. (Select **Devices > Device wizard >** + **New** to enter).

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Relay X, Switch input X

Name

Enter the names of the relays and switch inputs.

DTMF sequence

Requirements:

- Expert settings

Enter the request that is assigned to the door bell button within the door terminal. Possible entries: Digits and characters (Possible entries: *****, **#**).

Destination number

Select or enter the required call destination (external number, short-code number, internal number of the individual phone, internal number of the group).

Note: The call destination is first entered identically for all configurations. It can be changed later.

Note: If a door call should also be executed via VoIP accounts or GSM gateways, in any case the external number must be entered with the corresponding city code, even if it is the local prefix.

Note: To create further configurations for a door terminal, open the page **Devices > Door terminals** and click **Configure** in the line of the corresponding door terminal.

Configuring VoIP Door Terminal

Proceed as follows:

- Using the Configuration Manager (page **Devices > Device wizard > Configure**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number can be called directly by an internal subscriber. When put together with the numbers of the door bell buttons, this creates 3 to 4-digit phone numbers. (Select **Devices > Device wizard >** + **New** to enter).

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

SIP Password

Enter the SIP password required by the connected device.

IP configuration

IP address

Enter the IP address required by the connected device. Possible entries: IP address with a range of 0 to 255 (example: 192.168.0.240).

Port

Enter the port required by the connected device. Possible entries: 1 to 65535.

User name

Enter the user name (authentication data to control the relays).

Password

Enter the password (authentication data to control the relays).

Relay X, Switch input X

Name

Enter the names of the relays and switch inputs.

Destination number

Select or enter the required call destination (external number, short-code number, internal number of the individual phone, internal number of the group).

Note: The call destination is first entered identically for all configurations. It can be changed later.

Note: If a door call should also be executed via VoIP accounts or GSM gateways, in any case the external number must be entered with the corresponding city code, even if it is the local prefix.

Note: To create further configurations for a door terminal, open the page **Devices > Door terminals** and click **Configure** in the line of the corresponding door terminal.

SIP invite

Enter the SIP invites which the connected door terminal uses as signal. (Possible entries: Digits and the characters * and #).

Note: If default values have already been entered in the device template, the entry fields will not be displayed. If you want to change these values, click **Expert**.

Configurig IP Switch Relays

Proceed as follows:

- Using the Configuration Manager (page **Devices > Device wizard > Configure**)

An existing IP switch relay can be connected to the PBX so that various functions controled by the PBX (see) can be carried out. This IP switch relay must be created in the configuration of the PBX.

The IP switch relay and the relays available for it can be created with the device wizard. To control the relays, the PBX needs to know the IP address, port, user name (optional) and password of the connected device (see manual and settings of the device in question).

Various IP switch relays supported by the PBX are available as device templates. They already include several default values for operation of the IP switch relays in their factory settings.

If no of the device templates on offer matches your installation, you can select a user-defined device template.

In addition, you can select a simplified device template (**IPBox Simple**) which will make it easier for you to create an IP switch relay with the operating mode **Manual relay**.

IP configuration

IP address

Enter the IP address required by the connected device. Possible entries: IP address with a range of 0 to 255 (example: 192.168.0.240).

Port

Enter the port required by the connected device. Possible entries: 1 to 65535.

Note: Normally, the default port is port 80.

User name

Enter the user name required by the connected device.

Password

Enter the password required by the connected device.

Relay X, Name

Enter the names of the relays.

Configuring Loudspeaker Announcements

Proceed as follows:

- Using the Configuration Manager (page **Devices > Device wizard > Configure**)

In order to make loudspeaker announcements, a device that can be used as announcement output (e. g. an a/b-Audiobox at a loudspeaker system) must be connected to the PBX. This announcement output must be created in the configuration of the PBX and assigned an internal number to call it.

The announcement output can be created with the device wizard. Various devices supported by the PBX that can be used as announcement output are available as device templates for this purpose. Their factory settings already include several default values for operation of the announcement outputs.

If no of the device templates on offer matches your installation, you can select a user-defined device template.

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number can be called directly by an internal subscriber. (Entry on page **Devices > Device wizard**)

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Relay X, Name

Enter the names of the relays.

Online Configuration

Proceed as follows:

- Using the Configuration Manager (page **Devices > Device wizard > Configure**)

IMPORTING DEVICE TEMPLATE

A device template is either imported from the Internet **Load configuration** or from a file **Import from file**.

Door Terminals

Management of the Door Terminal
Settings of the Door Terminal
Settings Door Call

Management of the Door Terminal

Door Calls+Relays

Door Calls+Relays

Proceed as follows:

- Using the Configuration Manager (page **Devices > Door terminals**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number can be called directly by an internal subscriber. When put together with the numbers of the door bell buttons, this creates 3 to 4-digit phone numbers. (Select **Devices > Device wizard >** + **New** to enter).

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the door terminal within the PBX and the connected system phones. The name can also be transferred to the VoIP end devices via LDAP server.

Module | Port

Module and port to which the device that can be used as door terminal (e. g. a TFS Universal-Plus) is connected.

Configure

Opens the configuration of the door terminal.

Relais

Requirements:

- Relay/actuator (optional, see Technical Data in the Instructions)
- Created door terminal

A relay with the operating mode **Speaking**, **Door opener** or **Light** is assigned to the door terminal.

A relay used for this purpose can, for example, be part of a connected door terminal or be connected to the same port (a/b Switching Module). Alternatively, it is also possible to use a relay for the door function which is part of another connected device (e. g. an IP switch relay). In this case it must first be created in the configuration of the PBX as part of this other device and then be assigned to the door terminal. It then remains part of this other device in the configuration of the PBX, but takes over the selected door function for the door terminal.

If suitable relays have already been configured on the page **Devices > Relays (Actuators)**, a list field will appear now. Select one of the existing relays and click .

If you do not want to use any of the existing relays, select **New relays** in the list field (✕ cancels the operation). Enter a freely selectable name for the relay and click **Ok**.

Note: Alternatively, you can also create and assign relays on the page **Devices > Relays (Actuators)**.

Note: For a detailed description of the operating modes, refer to the chapter .

Link to the page **Devices > Relays (actuators)**

Deletes the relay or opens the configuration of the relay.

Türrufe

Requirements:

- Switch input (optional, see Technical Data in the Instructions)
- Created door terminal

The door bell buttons of a door terminal can be used to initiate configuration-dependent internal and external door calls. For this purpose, the door bell buttons must be created as switch inputs in the PBX. They must then be allocated call destinations and assigned to the door terminal. To detect a door call, e. g. because a door bell button has been pressed, the PBX needs to know which DTMF sequence or which SIP invite the

connected device sends in this case (see manual and settings of the device in question).

Note: The DTMF sequence or the SIP invite usually is the phone number assigned to the door bell button within the door terminal.
After the door terminal has been configured in the configuration of the PBX, the door call distribution will be controlled completely – including configuration-dependently – by the PBX.

Click the second + under **Door calls + relays** in the line for the corresponding door terminal.

Enter a name for the switch input in the **Switch input** entry field.

Enter a value under **DTMF sequence** (for analogue door terminals) or **SIP invite** (for VoIP door terminals). To detect that the alarm has been triggered, e. g. because a door bell button has been pressed, the PBX needs to know which DTMF sequence or which SIP invite the connected device sends in this case (see manual and settings of the device in question). The DTMF sequence or the SIP invite usually is the phone number assigned to the door bell button within the door terminal.

In the **Destination number** list field, select the required call destination. Or after selecting **Enter destination number**, in the **Destination number** entry field, enter the required call destination. Possible entries:

External number

Short-code number

Internal number of the individual phone

Internal phone of the group

Note: The call destination is first entered identically for all configurations. It can be changed afterwards.

Note: If a door call should also be executed via VoIP accounts or GSM gateways, in any case the external number must be entered with the corresponding city code, even if it is the local prefix.

Click **Ok**.

Note: To create further configurations for a door call or the corresponding switch input, click in the line of the corresponding door call. The configuration is described below and in the chapter Switch Input.

Deletes the switch input or opens the configuration of the door call.

Settings of the Door Terminal

Configuration of the Door Terminal

Configuration of the Door Terminal

Proceed as follows:

- Separately for door terminals
- Using the Configuration Manager (page **Devices > Door terminals > Configure**)

Requirements:

- Created door terminal

Analogue door terminal

Enable connection

Finish connection

Specifies which DTMF sequences the PBX uses to control the analogue door terminal to enable or terminate the connection. For information about which DTMF sequences the connected device requires, see the manual of the device in question.

Note: You can use the comma ',' character as a pause of half a second.

Dead time

Specifies how many milliseconds the PBX will wait before it sends DTMF sequences to the door terminal again. For information about how much time the connected device requires to be ready for reception again after receiving DTMF sequences, see the manual of the device in question.

VoIP Door Terminal

SIP password

To create a connection with a VoIP door terminal, the PBX needs the SIP password of the connected device (see manual and settings of the device in question).

Settings Door Call

Proceed as follows:

- Separately for door calls
- Using the Configuration Manager (page **Devices > Door terminals > Door terminal calls >**)

Requirements:

- Configured door call

CONFIGURATION OF THE DOOR CALL

Select the door call you want to configure.

DTMF sequence

Requirements:

- Created switch input for an analogue device

Specifies which DTMF sequence the PBX detects, for example,as the actuation of a door bell button to then trigger an alarm or a door call. For information about which DTMF sequences the connected device sends, see the manual of the device in question.

Note: The DTMF sequence usually is the phone number assigned to the door bell button within the door terminal.

SIP invite

Requirements:

- Created switch input for a VoIP device

Specifies which SIP invite the PBX detects, for example,as the actuation of a door bell button to then trigger an alarm or a door call. For information about which SIP invite the connected device sends, see the manual of the device in question.

Note: The SIP invite usually is the phone number assigned to the door bell button within the door terminal.

Destination number

Proceed as follows:

- Separately for configurations

Select the destination number you want to assign to a door call.

Ringtone

Proceed as follows:

- Separately for configurations

The ringtone specifies which sequence of sounds is used to signal door calls to internal subscribers on the PBX.

Note: See Ringer tones for information about exceptions to signalling and sound sequences.

Call duration

Proceed as follows:

- Separately for configurations

Select a call duration during which a door call is to be signalled to an internal subscriber.

Note: Set the call duration in the connected door terminal to the maximum call duration. The actual call duration will then be limited by the PBx.

Exchange line dialling/number presentation DDI

Proceed as follows:

- Separately for configurations

The option permits the precise selection of the external connection and even of the phone number to be presented.

All ext. lines or internal

Door calls can be made on internal or on all external channels.

External connection

Door calls will be made on the selected external connection.

In the **DDI** entry field, enter the DDI to be presented from the extensions block under the corresponding slot. If the DDI is displayed in red font subsequently, it either does not belong to the value range or the maximum number of own phone numbers has already been entered.

Second ringer

Proceed as follows:

- Separately for configurations

Requirements:

- Created relay with operating mode **Second ringer**

A second ringer can additionally be installed at locations where ringing of the phone will not always be heard. It will then be triggered by a relay with the operating mode second ringer which is assigned to the door call.

Announcement Outputs

Management Announcement Output
Settings Announcement Output

Management Announcement Output

Proceed as follows:

- Using the Configuration Manager (page **Devices > Announcement outputs**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number can be called directly by an internal subscriber. (Entry on page **Devices > Device wizard**)

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the announcement output within the PBX and the connected system phones. The name can also be transferred to the VoIP end devices via LDAP server.

Module | Port

Module and port to which the device that can be used as announcement output (e. g. an a/b-Audiobox at a loudspeaker system) is connected.

Configure

Opens the configuration of the announcement output.

Relay

Requirements:

- Relay/actuator (optional, see Technical Data in the Instructions)

A relay with the operating mode **Speaking**, which is enabled during the loudspeaker announcement, is assigned to the alarm. This relay can be used to switch over the loudspeaker area of the loudspeaker system, for example, or to switch on an additional lamp which draws attention to the announcement.

If suitable relays have already been configured on the page **Devices > Relays (Actuators)**, a list field will appear now. Select one of the existing relays and click .

If you do not want to use any of the existing relays, select **New relays** in the list field (cancels the operation). Enter a freely selectable name for the relay and click **Ok**.

Note: Alternatively, you can also create and assign relays on the page **Devices > Relays (Actuators)**.

Note: For a detailed description of the operating modes, refer to .

Link to the page **Devices > Relays (actuators)**

Deletes the relay or opens the configuration of the relay.

Settings Announcement Output

Proceed as follows:

- Separately for announcement outputs
- Using the Configuration Manager (page **Devices > Announcement outputs > Configure**)

Requirements:

- Created announcement output

CONFIGURATION OF THE ANNOUNCEMENT OUTPUT

Select the announcement output you want to configure.

Enable connection

Finish connection

Specifies which DTMF sequences the PBX uses to control the analogue announcement outputs to enable or terminate the connection. For information about which DTMF sequences the connected device requires, see the manual of the device in question.

Note: You can use the comma ',' character as a pause of half a second.

Dead time

Specifies how many milliseconds the PBX will wait before it sends DTMF sequences to the announcement output again. For information about how much time the connected device requires to be ready for reception

again after receiving DTMF sequences, see the manual of the device in question.

Alarms

Management of the Alarms
Settings Alarm
Settings of the Switch Inputs

Management of the Alarms

Switch input+Relay

Proceed as follows:

- Using the Configuration Manager (page **Devices > Alarms**)

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number can be called directly by an internal subscriber to initiate an alarm. (Select + **New** to enter.)

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the alarm within the PBX.

Enable alarm

off

The alarm input is disabled. It is not possible to initiate an alarm.

once

The alarm can be initiated only once. Subsequently, the alarm input will be disabled and will have to be enabled once again.

always

After an alarm is triggered off, the alarm input continues to be enabled. A further alarm can be triggered off.

Configure

Opens the configuration of the alarm.

+ New

Adds a new alarm. You must then make the settings for this new alarm.

Delete

Deletes the entries marked in the list.

Switch input+Relay

Relay

Requirements:

- Relay/actuator (optional, see Technical Data in the Instructions)
- Created device (e. g. door terminal, IP switch relay) whose part the relay is

A relay with the operating mode **Alarm**, which is used to enable e. g. a siren when the alarm has been triggered, is assigned to the alarm.

In the list field, select one of the existing relays and click .

If you do not want to use any of the existing relays, select **New relay** in the list field (cancels the operation). Enter a freely selectable name for the relay and select the device as whose part the relay is to be created in the **Add to** list field. Click **Ok**.

Note: The relay remains part of this other device (e. g. door terminal, IP switch relay) in the configuration of the PBX, but takes over the selected alarm function for the alarm.

Note: For a detailed description of the operating modes, refer to .

Link to the page **Devices > Relays (actuators)**

Deletes the relay or opens the configuration of the relay.

Switch input

Requirements:

- Switch input (optional, see Technical Data in the Instructions)
- Created device (e. g. a door terminal) whose part the switch input is (e. g. a still free door bell button)

A switch input that can be used to trigger the alarm, is assigned to the alarm.

In the list field, select one of the existing switch inputs and click .

If you do not want to use any of the existing switch inputs, select **New switch input** in the list field (cancels the operation). Enter a freely selectable name for the switch input and select the device as whose part the switch input is to be created in the **Add to** list field.

Enter a value under **DTMF sequence** (for analogue door terminals) or **SIP invite** (for VoIP door terminals). To detect that the alarm has been triggered, e. g. because a door bell button has been pressed, the PBX needs to know which DTMF sequence or which SIP invite the connected device sends in this case (see manual and settings of the device in question). The DTMF sequence or the SIP invite usually is the phone number assigned to the door bell button within the door terminal.

Click **Ok**.

Note: The switch input remains part of this other device (e. g. door terminal) in the configuration of the PBX, but takes over the selected alarm function for the alarm.

Deletes the switch input or opens the configuration of the switch input.

Settings Alarm

Proceed as follows:

- Separately for alarms
- Using the Configuration Manager (page **Devices > Alarms > Configure**)

Requirements:

- Configured alarm

CONFIGURATION OF THE ALARM

Select the alarm you want to configure.

Enable alarm

off

The alarm input is disabled. It is not possible to initiate an alarm.

once

The alarm can be initiated only once. Subsequently, the alarm input will be disabled and will have to be enabled once again.

always

After an alarm is triggered off, the alarm input continues to be enabled. A further alarm can be triggered off.

Number of alarm loops

Specifies how many alarm loops are to be performed before the alarm is acknowledged.

Alarm announcement

Select the announcement to be played to alarm subscribers.

Alarm subscriber (destination number)

Select up to four internal or external phone numbers to be called when the PBX performs an alarm loop.

Entering the destination number

Use this to enter a phone number manually.

Possible entries:

External number

Short-code dialling number

Internal number of the individual phone

Internal number of the group

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed.

Note: You can set the display text that will be displayed if an alarm call is made to the internal subscribers, which have been selected as alarm subscribers, here (page **Functions > CLIP texts**).

Exchange line dialling / number presentation > DDI

---: Alarm calls can be made on all external channels.

External connection: alarm call is made on the selected external connection. The selected phone number is transmitted.

Note: You must select the DDI number to be sent for a PTP connection from the extensions block. If the DDI number is then displayed in red text, either it is not in the value range, or the maximum number of external phone numbers has already been entered.

Note: Alarm calls should preferably be made via the land line.

Alarm delay time (0..99)

Specifies how many seconds elapse, after the alarm is triggered on the switch input, before the first alarm loop starts.

Alarm waiting time (0..99)

Specifies how many seconds elapse before an alarm is repeated, if it has not been acknowledged.

Alarm can be triggered by Action URL

Provided that the access authorisation to Action URLs is granted system-wide or for a subscriber, the individual alarm can be triggered by Action URLs.

Settings of the Switch Inputs

Switch Input+Relay

Switch Input+Relay

Proceed as follows:

- Separately for switch inputs
- Using the Configuration Manager (page **Devices > Alarms > Switch input >**)

Requirements:

- Created switch input

CONFIGURATION OF THE SWITCH INPUT

Select the switch input to be configured.

DTMF sequence

Requirements:

- Created switch input for an analogue device

Specifies which DTMF sequence the PBX detects, for example,as the actuation of a door bell button to then trigger an alarm or a door call. For information about which DTMF sequences the connected device sends, see the manual of the device in question.

Note: The DTMF sequence usually is the phone number assigned to the door bell button within the door terminal.

SIP invite

Requirements:

- Created switch input for a VoIP device

Specifies which SIP invite the PBX detects, for example,as the actuation of a door bell button to then trigger an alarm or a door call. For information about which SIP invite the connected device sends, see the manual of the device in question.

Note: The SIP invite usually is the phone number assigned to the door bell button within the door terminal.

Relays (Actuators)

Management Relays (Actuators)
Settings of the Relays (Actuators)

Management Relays (Actuators)

Proceed as follows:

- Using the Configuration Manager (page **Devices > Relays (actuators)**)

Name

Freely selectable name (maximum of 16 characters).

Identity number

Identification number of the relay which is automatically assigned when a new relay is created.

Module

Select the module on that the device is connected.

Port

Port to which the device is connected.

Operating mode

Select an operating mode.

Note: Relays can have various functions on the PBX. For a detailed description see here Relays (Actuators).

used by

Assigns a device to the relay.

+ New

A new relay is created and assigned to a device.

Delete

Deletes the entries that are marked in the list.

Configure

Opens the configuration of the device template.

Settings of the Relays (Actuators)

Analogue Door Terminal
VoIP Door Terminal
IP Switch Relay
Loudspeaker Announcements

Analogue Door Terminal

Proceed as follows:

- Using the Configuration Manager (page **Devices > Relays (actuators) > Configure**)

CONFIGURATION OF THE RELAY

Select the relay.

Off control

Requirements:

- Created relay

The PBX will switch off relays with the operating modes Light, Door opener, Alarm and PC relay after the specified switching duration.

Note: When switched off, the hold time of the relay will be used. The PBX does not actively switch off the relay.

Note: As a a pre-requisite, off control must be supported by the relay (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled).

Switching duration

The switching duration of a relay specifies after how many seconds the PBX will switch off the relay. Possible entries: 0 to 6000 (0 only for manual relays).

Note: The switching duration is used for relays with the operating modes Light, Door opener, Alarm, PC relay and Manual relay.

Note: To enable the PBX to switch off the relay after the specified switching duration, off control must be supported by the relay and be switched on (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled). Alternatively, the hold time of the relay can be used.

Hold time

If switching off is not to be controlled by the PBx (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled), the hold time of the relay can be used instead of the switching duration. This will only make sense if the hold time of the relay is sufficiently long, for example for a door opener.

The PBX needs to know how long the relay will be held in order to signalise holding of the relay on the system phones (see the manual of the device in question).

Requirements:

- Created relay
- Off control switched on for the relay

Enter the hold time of the relay in milliseconds. Possible entries: 10 to 10000 ms.

Relay can be triggered by Action URL

Provided that the access authorisation to Action URLs is granted system-wide or for a subscriber, the individual relay can be triggered by Action URLs.

VoIP Door Terminal

Proceed as follows:

- Using the Configuration Manager (page **Devices > Relays (actuators) > Configure**)

CONFIGURATION OF THE RELAY

Select the relay.

PBX is decisive (primary)

The available relays on an IP switch relay can also be toggled without affecting the PBX. In some situations, e.g. if the PBX is switched off at the switching point, the PBX cannot identify the state of the relay.

The PBX restores the relay to the last state it was able to identify after it is rebooted.

Off control

Requirement:

- the relay must have been set up

The PBX will switch off relays with the operating modes Light, Door opener, Alarm and PC relay after the specified switching duration.

Note: When switched off, the hold time of the relay will be used. The PBX does not actively switch off the relay.

Note: As a a pre-requisite, off control must be supported by the relay (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled).

Switching duration

Specifies after how many seconds the PBX will switch off the relay. Possible entries: 0 to 6000 (0 only for manual relays).

Note: The switching duration is used for relays with the operating modes Light, Door opener, Alarm, PC relay and Manual relay.

Note: To enable the PBX to switch off the relay after the specified switching duration, off control must be supported by the relay and be switched on (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled). Alternatively, the hold time of the relay can be used.

Hold time

If switching off is not to be controlled by the PBx (for some relays, e.g. the a/b Switching Modules, switching off cannot be controlled), the hold time of the relay can be used instead of the switching duration. This will only make sense if the hold time of the relay is sufficiently long, for example for a door opener.

The PBX needs to know how long the relay will be held in order to signalise holding of the relay on the system phones (see the manual of the device in question).

Requirement:

- the relay must have been set up

- Off control switched on for the relay

Enter the hold time of the relay in milliseconds. Possible entries: 10 to 10000.

Relay can be triggered by Action URL

Provided that the access authorisation to Action URLs is granted system-wide or for a subscriber, the individual manual relay can be triggered by Action URLs.

IP Switch Relay

Proceed as follows:

- Using the Configuration Manager (page **Devices > Relays (actuators) > Configure**)

CONFIGURATION OF THE RELAY

Select the relay.

PBX is decisive (primary)

The available relays on an IP switch relay can also be toggled without affecting the PBX. In some situations, e.g. if the PBX is switched off at the switching point, the PBX cannot identify the state of the relay.

The PBX restores the relay to the last state it was able to identify after it is rebooted.

Off control

Requirement:

- the relay must have been set up

The PBX will switch off relays with the operating modes Light, Door opener, Alarm and PC relay after the specified switching duration.

Note: When switched off, the hold time of the relay will be used. The PBX does not actively switch off the relay.

Note: As a a pre-requisite, off control must be supported by the relay (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled).

Switching duration

Specifies after how many seconds the PBX will switch off the relay. Possible entries: 0 to 6000 (0 only for manual relays).

Note: The switching duration is used for relays with the operating modes Light, Door opener, Alarm, PC relay and Manual relay.

Note: To enable the PBX to switch off the relay after the specified switching duration, off control must be supported by the relay and be switched on (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled). Alternatively, the hold time of the relay can be used.

Hold time

If switching off is not to be controlled by the PBx (for some relays, e.g. the a/b Switching Modules, switching off cannot be controlled), the hold time of the relay can be used instead of the switching duration. This will only make sense if the hold time of the relay is sufficiently long, for example for a door opener.

The PBX needs to know how long the relay will be held in order to signalise holding of the relay on the system phones (see the manual of the device in question).

Requirement:

- the relay must have been set up

- Off control switched on for the relay

Enter the hold time of the relay in milliseconds. Possible entries: 10 to 10000.

Relay can be triggered by Action URL

Provided that the access authorisation to Action URLs is granted system-wide or for a subscriber, the individual manual relay can be triggered by Action URLs.

Loudspeaker Announcements

Proceed as follows:

- Using the Configuration Manager (page **Devices > Relays (actuators) > Configure**)

CONFIGURATION OF THE RELAY

Select the relay.

Off control

Requirements:

- Created relay

The PBX will switch off relays with the operating modes Light, Door opener, Alarm and PC relay after the specified switching duration.

Note: When switched off, the hold time of the relay will be used. The PBX does not actively switch off the relay.

Note: As a a pre-requisite, off control must be supported by the relay (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled).

Switching duration

The switching duration of a relay specifies after how many seconds the PBX will switch off the relay. Possible entries: 0 to 6000 (0 only for manual relays).

Note: The switching duration is used for relays with the operating modes Light, Door opener, Alarm, PC relay and Manual relay.

Note: To enable the PBX to switch off the relay after the specified switching duration, off control must be supported by the relay and be switched on (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled). Alternatively, the hold time of the relay can be used.

Hold time

If switching off is not to be controlled by the PBx (for some relays, e. g. the a/b Switching Modules, switching off cannot be controlled), the hold time of the relay can be used instead of the switching duration. This will only make sense if the hold time of the relay is sufficiently long, for example for a door opener.

The PBX needs to know how long the relay will be held in order to signalise holding of the relay on the system phones (see the manual of the device in question).

Requirements:

- Created relay
- Off control switched on for the relay

Enter the hold time of the relay in milliseconds. Possible entries: 10 to 10000 ms.

Relay can be triggered by Action URL

Provided that the access authorisation to Action URLs is granted system-wide or for a subscriber, the individual relay can be triggered by Action URLs.

CONFIGURATION OF ANALOGUE RELAYS

Dead time

The PBX uses DTMF sequences to control the analogue relays. For this purpose, the PBX needs to know how much time the connected device requires to be ready for reception again after receiving DTMF sequences (see the manual of the device in question).

Requirements:

- Created analogue relay

Specifies how many milliseconds the pbx will wait before it sends DTMF sequences to the connected device again. Possible entries: 0 to 10000 milliseconds.

Switching

The PBX uses DTMF sequences to control the analogue relays. For this purpose, the PBX needs to know which DTMF sequences the connected device requires to switch the relay on or off (see the manual of the device in question).

Requirements:

- Created analogue relay

Switch on, **Switch off**

Enter DTMF sequences required by the connected device. Possible entries: Digits and the characters * and # and the comma ',' character as a pause of half a second.

Action URLs

Management of the Action URLs (HTTP Request)
Settings Action URL (HTTP Request)

Management of the Action URLs (HTTP Request)

Action URLs (incoming)
Action URLs (outgoing)
Event type

Action URLs (incoming)

Proceed as follows:

- Using the Configuration Manager (**Devices > Action URLs** page)

You can find further information about the structure of an Action URL, the switching of relays, the triggering of an alarm etc. in the Auerswald DokuWiki.

New

Adds a new Action URL. You must then make the settings for this new Action URL.

Delete

Deletes the entries that are marked in the list.

Incoming Action URLs enabled

System-wide access. Requirement for the following settings.

Action URLs only within the local network

Triggered exclusively on the local network.

subscribers may generate own passwords for accessing Action URLs

For the subscriber login, additional passwords can be configured under subscriber properties.

Passwords for accessing Action URLs > Configure

Requirements:

- Enabled Subscribers may generate own passwords for accessing Action URLs

Note: Additional passwords can also be created for individual subscribers (**Subscriber (scr.) > Phone numbers > Configure > Basic settings**).

Page for editing additional passwords.

Access to Action URLs

System-wide access

Save

Displays the automatically created passwords that is only displayed once. Therefore, it should be copied to the clipboard.

Note: When switched off, the password is only created when accessing the PBX via a secure HTTPS connection.

Action URLs (outgoing)

Proceed as follows:

- Using the Configuration Manager (**Devices > Action URLs** page)

You can find further information about the structure of an Action URL, the switching of relays, the triggering of an alarm etc. in the Auerswald DokuWiki.

New

Adds a new Action URL. You must then make the settings for this new Action URL.

Delete

Deletes the entries that are marked in the list.

Name

Freely selectable name to identify the Action URL within the PBX.

Action URL

Protocol, IP/URL and command must be understood by the receiving device (example: http://phoneIP/remote_control?action=HANGUP). You find further information in the device manual.

Configure

Opens the configuration of the Action URL.

Event type

Proceed as follows:

- Using the Configuration Manager (**Devices > Action URLs** page)

You can find further information about the structure of an Action URL, the switching of relays, the triggering of an alarm etc. in the Auerswald DokuWiki.

+ New

Adds a new Action URL. You must then make the settings for this new Action URL.

Delete

Deletes the entries that are marked in the list.

Alarm

+

In the first list feld, select one of the existing alarms, select the alarm type in the second list field and click .

If you do not want to use any of the existing alarms, select **New alarm** in the list field (cancels the operation). Enter the phone number of the alarm and a freely selectable name. In the list field **Alarm type**, select the alarm type. Click **Ok**.

Note: You can successively select all three alarm types for one Action URL.

Phone number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number can be called directly by an internal subscriber to initiate an alarm. (Select + **New** to enter.)

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the alarm within the PBX.

Alarm type

You can select all alarm types for one alarm:

Alarm start

The Action URL will be assigned to the alarm type **Alarm start**. When an alarm is triggered, the Action URL is sent.

Note: If an alarm delay time was entered, the alarm call via the Action URL starts after the alarm delay time.

Alarm acknowledged

The Action URL will be assigned to the alarm type **Alarm acknowledged**. As soon as an alarm subscriber (configured under Devices > Alarms > Configure > Alarm subscriber (destination number)) acknowlegded the alarm, the Action URL is sent

Note: A device from another manufacturer that is not recognized by the PBX, cannot acknowlegde an alarm call by entering the DTMF digit 0.

Alarm timeout

The Action URL will be assigned to the alarm type **Alarm timeout**. If none of the alarm subscribers acknowledges the alarm, the alarm call ends after the set alarm loops and the Action URL is sent.

Link to page **Devices > Alarms**

Delets the alarm or opens the configuration of the alarm.

Settings Action URL (HTTP Request)

Proceed as follows:

- Separately for Action URLs
- Using the Configuration Manager (page **Devices > Action URLs > Configure**)

Requirements:

- Configured Action URL

CONFIGURATION OF THE ACTION URL

Select the alarm you want to configure.

User name

Enter the device‘s username that shall be reached via the Action URL.

Password

Enter the password that belongs to the username.

Response required

If the function is enabled, the PBX waits for a feedback of the receiving device after having sent the Action URL and displays a corresponding message about the correct or incorrect sending (under **Monitoring > System messages**). Therefore, activating this function is generally recommended.

Note: The command to be executed in the Action URL must be correctly understood by the receiving device, also in order to give

according feedback. This should always be controlled when configuring an Action URL.

If the SIP port of the internal VoIP registrar is modified, the internal SIPS port of the internal VoIP registrar is also changed automatically. A message concerning the Action URL that was sent is displayed by the PBX.

Time control

Configurations
Switching Times

Configurations

Management Configuration

Management Configuration

Proceed as follows:

- Using the Configuration Manager

Automatic configuration switching

Requirements:

- Minimum of two configured configurations and configured switching times
- Current PBX time

Time-controlled configurations are toggled according to the PBX internal clock.

Note: If automatic configuration switchover is enabled, the configuration enabled under Time control > Switching times for the current time becomes active. If a different configuration is then switched manually, this remains active until the next switching time.

Controllable with system relay: Aut.Konfig.um. (900)

Requirements:

- Automatic configuration switchover is enabled

Automatic configuration switchover is controllable via phone.

Configuration name

Freely selectable name. This name is used to identify the configuration within the PBX.

Identification number

Automatically issued number. Is used for manual switching of a configuration by a programming sequence.

Note: All previously assigned identification numbers can be viewed sorted by number, name or type under **Overviews > Identification number table**.

Activate

The selected configuration is switched on.

Note: To switch off a configuration, switch on a different configuration.

New

Adds a new configuration to the list. The settings made for the existing configuration are copied. You must then make the individual settings for the configuration.

Delete

Deletes the entries marked in the list.

Copy

Configuration settings are copied from one configuration into other, selected configurations.

Switching Times

Management Switching Times

Management Switching Times

Proceed as follows:

- Using the Configuration Manager

Requirements:

- At least two configured configurations
- Automatic configuration switchover is enabled
  - Using the Configuration Manager (page **Time control > Switching times**)
- For using the holidays: Configured and enabled holidays
  - Using the Configuration Manager (page **Functions > Calendar**)

Time

Time (format hh : mm) when the selected configuration is to be activated. (Select + **New** to enter.)

Note: A configuration applies until another configuration (manual or automatic) is enabled, also beyond the daily change. An exception is the transition from a holiday to a normal weekday. In this case, the configuration activated for the holiday will only be active until 23:59. The normal configuration for a weekday will be activated at 0:00 of the following day (even if a configuration was activated manually on the holiday).

Configuration

Configuration to be activated.

Wake-up announcement

Defines which wake-up announcement the user hears when accepting a wake-up call. Three wake-up announcements are available to indicate the time of day.

New

Adds a new switching time to the list. For this switching time settings must be configured.

Delete

Deletes the entries marked in the list.

Copy

Switching times are copied from one weekday to other selected weekdays.

Routing

Soft-LCR 4.0
VoIP and GSM Routing

Soft-LCR 4.0

Management LCR Subscribers
Management LCR Providers
Management Networks and LCR import
Management Area Codes/Area Code Ranges
Management LCR Tables

Management LCR Subscribers

Proceed als follows:

- Using the Configuration Manager (page **Routing > Soft-LCR 4.0 > LCR subscriber**)

Phone number | Name

Requirements:

- Enabled LCR subscribers

Select the subscribers for whom automatic LCR is to be carried out. (Select + **New** to enter.)

+ New

Adds a new subscriber to the list.

Delete

Deletes the entries marked in the list.

Management LCR Providers

Proceed as follows:

- Using the Configuration Manager (page **Routing > Soft-LCR 4.0 > Provider**)

Note: Note. **LCR for Call Through calls** is switched on/off on the page **Functions > Call Through** under **EXCHANGE LINE SETTINGS**.

Note: Manual LCR by dialling a preceding provider area code is always possible if the network provider supports dial-around service.

Provider area code

Provider area code without an exchange line access number. (Select + **New** to enter.)

Provider name

Freely selectable name. This name is used to identify the provider within the PBX. (Select + **New** to enter.)

+ New

Adds a new provider to the list.

Up to 50 dial-around providers can be created.

Delete

Deletes the entries marked in the list.

Management Networks and LCR import

Proceed as follows:

- Using the Configuration Manager (page **Routing > Soft-LCR 4.0 > Networks and LCR import**)

Create

Requirements:

- Created own local area code

The networks **City**, **Region**, **National** and **International** can automatically be created based on the local area code. The include the data for the **Area code/Area code range**.

Important: When networks are created automatically, the existing LCR data is deleted.

Name

Freely selectable name. This name is used to identify the network within the PBX. (Select + **New** to enter.)

Configure

Opens the configuration of the networks and the corresponding prefixes.

+ New

Adds a new network to the list.

Up to 60 networks can be created manually.

Delete

Deletes the entries marked in the list.

Import

Select and import an LCR configuration file (*.xml, *.txt, also compressed *.gz) that is stored on a (PC) data storage device.

Note: Using special tariff calculators you can calculate an LCR configuration (provider, tariff groups and data) and import this into the PBX (e. g. Telefonsparbuch.de).

Important: When importing configuration data, existing LCR data is deleted.

Export

A completed LCR configuration is saved to a data storage device in a *.gz file.

Management Area Codes/Area Code Ranges

Proceed as follows:

- Using the Configuration Manager (page **Routing > Soft-LCR 4.0 > Netze und LCR-Import > Configure**)

Area codes/area code ranges

Digit sequence for an area code or area code range. (Select + **Neu** to enter.)

A prefix contains the prefix or all of the prefixes from the area that should be collected for the network (e.g., local phone network, regional network, international network).

Name

Freely selectable name. This name is used for identification within the PBX.

+ New

Adds a new area code/area code range to the list.

Up to 1000 entries can be created for all networks in total.

Delete

Deletes the entries marked in the list.

Management LCR Tables

Proceed as follows:

- Using the Configuration Manager (page **Routing > Soft-LCR 4.0 > LCR-Tables > Configure**)

Day

Requirements:

- For use of holidays: holidays created and LCR active
  - Using the Configuration Manager (page **Functions > Calendar**)

Weekdays when dialling is to be carried out using the corresponding provider.

Time

Time point starting on the corresponding **Day** when the dial attempt is to be carried out via the corresponding provider. (Select + **New** to enter.)

Provider

no LCR

The dial attempt is initiated without a provider area code.

generate busy

The dial attempt is ended and a busy signal is issued.

Provider name: The dial attempt is initiated using the area code entered under **Routing > Soft-LCR 4.0 > Providers**.

Note: During the dial attempt, if the selected provider under **Main provider** is busy, the selected provider under **1st fallback provider** is

attempted, etc. If still no connection is established during the dial with the **2nd fallback provider**, a busy signal is issued.

Price per connection

Proceed as follows:

- Separately for main provider, 1st and 2nd fallback provider.

Indicates the fee a provider charges for the establishment of a connection, based on the currency entered in the PBX (page **Functions > Call allowance accounts**).

Price per minute

Proceed as follows:

- Separately for main provider, 1st and 2nd fallback provider

The PBX uses **Price per minute** for internal calculations and display purposes during calls that are initiated by an LCR subscriber and perhaps a Call Through user and routed via the corresponding provider.

The **Price per minute** refers to the currency entered in the PBX (**Functions > Call allowance accounts** under **CHARGES**).

Metering pulse interval 1st pulse

Proceed as follows:

- Separately for main provider, 1st and 2nd fallback provider

Seconds the provider assigns the first meter pulse after the connection has been established (after CONNECT).

Note: This meter pulse is fully billed by the provider even if the call is ended before this time has elapsed.

Second pulse

Proceed as follows:

- Separately for main provider, 1st and 2nd fallback provider

Charge pulse time unit used to bill a call after the first metering pulse.

Note: For second pulses of differing lengths, enter the most unfavourable meter pulse.

Note: The provider bills each started meter pulse in full.

New

Adds a new LCR table to the list.

Delete

Deletes the entries marked in the list.

VoIP and GSM Routing

Settings for VoIP/GSM Routing

Settings for VoIP/GSM Routing

Exception Numbers
Routing Configuration-Dependent

Exception Numbers

Proceed as follows:

- Using the Configuration Manager (page **Routing > VoIP and GSM routing**)

Phone number / range

Exception numbers, for which the selected connection should be determined. Up to 400 exception numbers can be entered. (Select + **New** to enter.)

Note: Do not enter the phone numbers for software updates (system phone) etc. numbers as exception numbers.

Name

Freely selectable name. This name is used to identify the exception number within the PBX.

1st dialling/2nd dialling/3rd dialling

Select the connection to be used for the exception number..

send busy

The dial attempt is ended and a busy signal is issued.

All fixed network lines

The dialling is carried out by one of the external ports without GSM gateway of the configured ISDN PTMP connections or ISDN PTP connections.

PBX base number [extensions block (DDIs)] | Name of connection: The dialling is carried out by the selected external port of the ISDN PTP connection.

Slot x - Port No | Name of connection: The dialling is carried out by the selected external port of the ISDN PTMP connection.

VoIP: all accounts

The dial attempt is made over one of the configured VoIP accounts.

VoIP

All GSM lines

The dialling is carried out by one of the external ports with GSM gateway of the configured ISDN PTMP connections or ISDN PTP connections.

GSM

PBX base number [extensions block (DDIs)] | Name of connection: The dialling is carried out by the GSM gateway at the selected external port of the ISDN PTP connection.

GSM

Slot x - Port No | Name of connection: The dialling is carried out by the GSM gateway at the selected external port of the ISDN PTMP connection.

+ New

Adds a new exception number to the list. You must then make the settings for the exception number.

Delete

Deletes the entries marked in the list.

Export

The columns **Phone number / range** and **Name** are saved to the (PC) data storage device in a file called ausnahmerufnr.csv.

Import

A text file (e.g., *.txt, *.csv, data arranged in rows and separated by semicolons) stored on a (PC) data storage device is selected and imported in the columns **Phone number / range** and **Name**.

Note: The option **All fixed network lines** under **1st dialling** or the option **send busy** for the **2nd dialling** and **3rd dialling** is selected for the imported phone numbers. This must be manually adapted later.

Routing Configuration-Dependent

Proceed as follows:

- Using the Configuration Manager (page **Routing > VoIP and GSM Routing > Routing configuration dependent**)

Configurations [Name]

For all configurations

In case that the selected provider has low prices only for a limited space of time, you can switch on and off single exception numbers by configuration.

Configuration, during which the exception number is used.

Functions

Exchange Line Transfer
Automatic Receptions
Call Through
CLIP Texts
Call Allowance Accounts
Hotel Function
Calendar
Short-Code Macros
LAN TAPI
Call Parking
Online Name Search
Waiting Field
Music on Hold and Announcements
Wake-Up Times

Exchange Line Transfer

Settings for Exchange Line Transfer

Settings for Exchange Line Transfer

Proceed as follows:

- Using the Configuration Manager (page **Functions > Exchange line transfer**)

Exchange line settings for the manually switched exchange line access

Configure special exchange line settings

The exchange line settings (e.g., exchange line authorisation, call restrictor (outgoing), call deblocker (outgoing)) for the manually switched exchange line access will be configured specifically for this purpose.

Use the exchange line settings of the transferring subscriber

The exchange line settings for the manually switched exchange line access are set depending on the transferring subscriber.

Callback option

The requesting subscriber can trigger a callback after initiating an external call.

Charge burden

Transferring subscriber

The charges for a call are billed to the transferring subscriber.

Requesting subscriber

The charges for a call are billed to the requesting subscriber.

Time limitation for establishing the external call (1..60 min)

Specifies how many minutes the exchange line access will persist at maximum if a call is not initiated successfully.

Automatic Receptions

Management Automatic Reception
Settings for Automatic Receptions

Management Automatic Reception

- Using the Configuration Manager (page (Seite **Functions > Automatic Receptions**)

Requirements:

- Activated function Automatic Reception (optional, see activation)

Internal number of the reception

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by an internal subscriber.

Note: A number cannot be allocated twice.

Note: Using numbers with few digits (two- or three-digit numbers) inevitably restricts the possible supply of numbers that require more digits. For example, if the phone number 12 is assigned, the phone numbers 120 to 129 and 1200 to 1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the automatic reception within the PBX.

Configure

Opens the configuration of the automatic reception.

New

Adds a new automatic reception. You must then make the settings for this new automatic reception.

Note: A maximum of ten automatic receptions can be created.

Delete

Deletes the entries marked in the list.

Settings for Automatic Receptions

Settings
Destination Number

Settings

Proceed as follows:

- Separately for Automatic Receptions
- Using the Configuration Manager (page **Functions > Automatic Receptions > Configure**)

Requirements:

- A configured automatic reception

Announcement

Select the greeting to be played to callers. Make your selection from the announcements for automatic receptions and the announcements before answering.

Note: Under some circumstances, a greeting with background music may cause errors because the audible touch tones cannot be properly recognized, or parts of the music may be interpreted as touch tones.

Max. number of external calls on hold in parallel

Specifies how many external calls can be kept on hold at the same time. The caller hears the busy signal if this number is exceeded. External calls on hold are calls that have either not yet been transferred or calls that have been transferred back to the automatic reception. This restriction does not apply to internal callers and to callers transferred by a superordinated reception.

Max. number of the repetitions per call

Specifies how often the greeting is played if the caller does not input a valid number. The system react as if the destination was busy/did not reply, after the announcements and the subsequent waiting time has elapsed.

Waiting time between the announcements or until transfer to the default destination

Specifies how long the PBX waits (in seconds) for a valid input between and after the announcements.

Reaction if the destination is busy/does not reply

If the caller enters a valid suffix dialling digit, the destination subscriber is notified that a call is present.

If the call is not answered, the announcement if the destination is busy/ does not reply is played.

If the destination is busy, call waiting is enabled provided that the destination subscriber has enabled this function. If he rejects the call waiting call, the announcement if the destination is busy/does not reply is played.

Note: For internal calls, it is tried to reach the destination number for max. 2 minutes, external calls for max. 8 minutes. This time depends on the settings made. VoIP and ISDN subscribers, for example, can only be called for 3 minutes if the call is not answered.

The system then reacts according to the following setting.

End of the call

Three short busy signals are played and the PBX ends the existing call.

Call default destination

The PBX transfers the existing call to the default destination number entered below. If the call is not answered there either, three short busy signals are played and the PBX ends the existing call.

Note: If the destination number and the default destination number are identical, the PBX cuts off the call without calling again.

Repeat Automatic Reception

The PBX transfers the existing call to the automatic reception that was called previously.

Announcement if destination is busy/does not reply

Select the announcement to be played to a caller if the destination is busy or does not reply. Make your selection from the announcements for automatic receptions and the announcements before answering, or do not select an announcement.

Default destination number

Possible entries:

Internal number of a subscriber

Internal number of a group

Internal number of a voice mailbox

Internal number of another automatic reception

Note: Note that an internal subscriber or a group must have at least **Only incoming, with emergency call** exchange line authorisation to enable them to answer a call transferred by the automatic reception.

Note: An external destination number can be reached by entering an internal destination which is forwarded to an external one (e.g. via call forwarding (subscribers or groups) with the required external phone number as destination number).

Music on Hold during the ringing period

If the Music on hold during the ringing period option has been enabled, a caller transferred by the automatic reception will only hear the music on hold during this process (pressing the R key, dialling the internal number, and ringing period).

Note: When switched off, the caller hears the call tone during the call phase.

Destination Number

Proceed as follows:

- Separately for Automatic Receptions
- Using the Configuration Manager (page **Functions > Automatic Receptions > Configure**)

Requirements:

- A configured automatic reception

DTMF | Destination number

x

Specifies the in-house destination to which the PBX should automatically transfer calls after the caller inputs a DTMF suffix dialling digit 0 to 9.

Possible entries:

Internal number of a subscriber

Internal number of a group

Internal number of a voice mailbox

Internal number of another automatic reception

***** under **DTMF 1- 9**: All internal numbers for subscribers, groups, voice mailboxes and automatic receptions that start with a number marked with * can be selected directly by redialling.

Note: If the caller enters ***** e.g. for DTMF 4: All internal subscribers, groups, voice mailboxes and automatic receptions whose phone numbers

start with 4 (4xxx) can be reached by directly redialling their phone number.

Note: If the caller enters *: The subscriber is given a maximum of 10 seconds in which to enter each digit. If this time period is exceeded, or if they dial an invalid phone number (subscriber, group, automatic reception), the announcement if the destination is busy/does not reply is played, and the caller will be automatically connected with the automatic reception they called previously.

Note: Note that an internal subscriber or a group must have at least **Only incoming, with emergency call** exchange line authorisation to enable them to answer a call transferred by the automatic reception.

Note: An external destination number can be reached by entering an internal destination which is forwarded to an external one (e.g. via call forwarding (subscribers or groups) with the required external phone number as destination number).

Call Through

Settings for Call Through

Settings for Call Through

Exchange Line Settings
Call Assignment
Call Through User

Exchange Line Settings

Proceed as follows:

- Using the Configuration Manager (page **Functions > Call Through**)

LCR for Call Through calls

Requirements:

- Configured LCR
  - Using the Configuration Manager (page **Routing > Soft-LCR 4.0**

Even for Call Through calls the most cost-effective phone service provider is determined.

PIN for Call Through calls

In the case of a call, the PIN of the caller is transmitted via DTMF. The PIN is then compared with the Call Through PIN.

Note: The transmission of the phone number is required. Therefore, anonymous callers do not get a Call Through authorisation.

Note: If the PIN is locked, the access is only locked for the Call Through subscriber who made the misentry.

Max. duration of an external Call Through call (1..99 min.)

Specifies after how many minutes an external Call Through call is ended by the PBX, due to security reasons.

Special exchange line settings Configure

You can make the following exchange line settings for internal subscribers, but also for Call Through calls:

- Exchange line authorisation
- Call restrictor (outgoing)
- Call deblocker (outgoing)
- Preferred exchange line
- Number presentation (outgoing)
- Number presentation (incoming)

Call Assignment

Proceed as follows:

- Using the Configuration Manager (page **Functions > Call Through**)

Call Through number

Requirements:

- To use VoIP numbers: **Standard (use of From header as screened number)** configured for the VoIP provider as evaluation type of incoming SIP messages
  - Using the Configuration Manager (page **Public switched tel. networks > VoIP > Provider > Configure**

Select the external phone number, that the Call Through user must dial to call the PBX for Call Through.

Note: Call Through numbers are not included in call distribution and cannot be diverted.

Max. number

Specifies how many how many simultaneous Call Through calls are permitted for the exchange line.

---: No Call Through calls are permitted.

Call Through User

Proceed as follows:

- Using the Configuration Manager (page **Functions > Call Through**)

Short-code dialling number

Requirements:

- Contact created for the Call Through user (e.g., mobile phone).
  - Using the Configuration Manager (page **User data > Address books > Contacts and assignment**)

Select the Call Through user according to its display name. (Select + **New** to enter.)

Note: Call Through is automatically configured for each of the four phone numbers that can be entered for each contact, but only for a total of 400 contacts.

Call restrictor/deblocker

If call restrictor and deblocker for Call Through are configured, the phone number dialled by the Call Through user is checked.

+ New

Adds a new Call Through user to the list. You must then make the settings for the Call Through user.

Delete

Deletes the entries marked in the list.

CLIP Texts

Settings CLIP Texts

Settings CLIP Texts

CLIP texts can be configured for the following calls and functions:

- **CCBS callbacks**

  The configured CLIP text is displayed on system phones during a callback on busy.

- **CCNR callbacks**

  The configured CLIP text is displayed on system phones during a callback on no reply.

- **Alarm calls**

  The configured CLIP text is displayed on system phones during an alarm call.

- **Wake-up calls**

  The configured CLIP text is displayed on analogue phones during a wake-up call.

- **Exchange line reservation**

  The configured CLIP text is displayed on analogue phones during a call to signal that a line that was previously busy has become available.

- **TAPI**

  The configured CLIP text is displayed on analogue phones during a call that signals the LAN-TAPI function.

- **Callbacks**

  The configured CLIP text is displayed on analogue phones during a callback (on busy or on no reply).

Proceed as follows:

- Using the Configuration Manager (page **Functions > CLIP texts**)

Requirements:

- System phone or analogue phone with alphanumerical CLIP information
- **Phone number display (CLIP)**: **CLIP info** and **Extended CLIP info** enabled for the analogue subscriber

Call Allowance Accounts

Settings of the Call Allowance Account

Settings of the Call Allowance Account

Charges
Account Debit
Call Allowance Accounts

Charges

Proceed as follows:

- Using the Configuration Manager (page **Functions > Call allowance accounts**)

Requirements:

- ISDN connection
- The AOCD service attribute released by the network provider (charge transmission during the connection)
- Configured LCR
  - Using the Configuration Manager (page **Routing > Soft-LCR 4.0**)

Currency name

Specifies which abbreviation is used for the currency in the Configuration Manager (e.g., "EUR" for Euros).

Price per unit

Is used for internal calculations and the display. The value refers to the currency entered (e.g., EUR 0,0600 for 6 cents).

Note: For Least Cost Routing (LCR), the tariff information of dial-around providers is saved in the PBX. This data (prices + charge pulse interval) replace the cost per call unit entered here for all LCR calls.

Maximum credit for an account (max. 500 EUR)

Specifies the highest amount that a call allowance account may not exceed.

Call charge recording changes with scr

After transferring a call to another internal end device, the account of the other subscriber is debited.

Account Debit

Proceed as follows:

- Using the Configuration Manager (page **Functions > Call allowance accounts**)

Requirements:

- ISDN connection
- The AOCD service attribute released by the network provider (charge transmission during the connection)
- Configured LCR
  - Using the Configuration Manager (page **Routing > Soft-LCR 4.0**)

Charge account for business calls

Charge account for private calls

Charge account for group calls

The configured call allowance accounts are debited in the case of business calls, private calls and/or group calls.

Call Allowance Accounts

Proceed as follows:

- Using the Configuration Manager (page **Functions > Call allowance accounts**)

Requirements:

- ISDN connection
- The AOCD service attribute released by the network provider (charge transmission during the connection)
- Configured LCR
  - Using the Configuration Manager (page **Routing > Soft-LCR 4.0**)

Important: You cannot enter/set/recharge a call allowance balance if **Unlimited allowance** under **Charge according the following pattern** is selected.

Call allowance balance (EUR)

Specifies the current call allowance balance of the call allowance account.

Charge according the following pattern

Unlimited allowance

The call allowance account has no effect.

Manual

The call allowance account is set only manually.

Increase ... by

The call allowance account is increased at intervals by the configured allowance charge amount. Charging manually is possible.

Set ... to

The call allowance account is set at intervals to the configured allowance charge amount. Charging manually is possible.

Allowance charge amount (EUR)

Specifies the value with which the call allowance account is charged manually or at the configured intervals.

Recharge call allowance

Several call allowance accounts are increased simultaneously by the configured allowance charge amount. Select the check boxes of the subscribers in question and click **Save**.

Set call allowance

Several call allowance accounts are set simultaneously to the same value. Select the check boxes of the subscribers in question, enter the call allowance balance under **on** and click **Save**.

Set allowance amount

For several call allowance accounts the allowance charge amounts are set simultaneously to the same value. Select the check boxes of the subscribers in question, enter the allowance charge amounts under **on** and click **Save**.

Hotel Function

Settings of the Hotel Function
Management of the Hotel Function

Settings of the Hotel Function

Printer Settings
Print Options

Printer Settings

Proceed as follows:

- Using the Configuration Manager (page **Functions > Hotel function > Create print form**)

Active

The telephone bill is issued directly to a printer.

USB printer

Printers that are compatible with PCL4 printer language (HP Laserjet) are supported. PCL6 or PCL5c emulation is downward compatible and includes PCL4.

Network printer

IP address and port must be entered. Printers with an interface compatible with HP JetDirect and the print format PCL3 (and later versions) are supported.

Form feed

Each telephone bill is printed out on an individual page.

Note: When switched off, the telephone bills (original document + copies) are printed out in sequence without blank lines.

Number of printouts

Number (max. 4) of invoices to be printed out (original document + copies).

Note: When printing out an interim invoice, this setting is ignored and only the original is printed.

Print quality

Draft

Fast print in draft quality.

Letter

Invoice in letter quality.

Print Options

Proceed as follows:

- Using the Configuration Manager (page **Functions > Hotel function > Create print form**)

Header text | Bold | Header X

7 lines (max. 80 characters per line) are available. If an empty line is required, you must also leave it out here.

Subject | Bold | Invoice headline

Max. 80 characters.

Invoice data

Headlines of the invoice data **Invoice no**, **Check-in** and **Check-out** are freely selectable (recommended: preset texts).

Column headlines

The column headlines of the itemised bills **Date**, **Time**, **Duration**, **Phone number** and **Amount** are freely selectable (recommended: preset texts).

Final amount

Headlines of the final amount **Sum**, **(incl. VAT)**, **Tax** and **VAT** are freely selectable (recommended: preset texts).

Print tax

The included VAT is shown additionally.

Currency name

The currency appears after the invoice amount.

Missing call data

If the call data storage capacity is exceeded, a note appears on the telephone bill that is entered here (e. g. **Calls before the following date are not included in the overview:**)

Footer text | Bold | Footer X

3 lines (max. 80 characters per line) are available.

Separator | Date | Time

For example: 01.01.2000 - 12:00

Preadjustment

Existing entries are discarded and automatic text suggestions are entered.

Print test page

To check the layout.

Management of the Hotel Function

Hotel Reception Subscribers
Room Subscribers

Hotel Reception Subscribers

Proceed as follows:

- Using the Configuration Manager (page **Functions > Hotel function > Hotel reception subscribers**)

+ New

Adds new hotel reception phone from the configured ISDN system phone subscribers.

Delete

Deletes the entries marked in the list.

Room Subscribers

Proceed as follows:

- Using the Configuration Manager (page **Functions > Hotel function > Room subscribers**)

Prevent Room2Room

Calls between hotel room phones are not possible. Calls to other internal subscribers (e. g. reception, room service, etc.) are still possible.

+ New

Adds a new hotel room phone from the configured subscribers.

Delete

Deletes the entries marked in the list.

Calendar

Management Calendars

Management Calendars

Proceed as follows:

- Using the Configuration Manager (page **Functions > Calendar**)

+ New

For individual data (e. g. company vacations). General holidays are added with **Add holiday list** or alternatively by double-clicking a day.

Note: Editing holidays: Tapping the holiday and clicking .

Fill out **Name** and **Date** and select an **Option**:

- **active**: Day is enabled as holiday.
- **once**: Day is enabled as holiday only for the current year.
- **LCR active**: Day is enabled as LCR.

Deleting calendar

All holiday entered into the calendar will be deleted.

Import

Select and import a holiday file (*.ft) previously exported from another PBX of the same design and saved on a data storage device (PC)

Export

The entered holidays are saved to the (PC) data storage device in an *.ft holiday file so they can be imported afterwards to a different PBX of the same design.

Add holiday list

The holidays valid for the country/federal state are added for an indefinite period. Holidays with data that changes yearly are calculated automatically by the PBX.

Note: Holidays inserted in this manner can be edited or deleted individually as needed. Each of these changes is carried out automatically for all calendar years.

Short-Code Macros

Management Macro Short-Code Number

Management Macro Short-Code Number

Proceed as follows:

- Using the Configuration Manager (page **Functions > Macro short-code number**)

Important: For short-code macros, the automatic exchange line request is always enabled. Therefore, short-code macros with postdialling could have problems with the length check so that the area code must always be dialled.

Note: The replacement of an internal number by another internal number is not supported.

Note: It is not possible to integrate a macro into another macro.

Macro types:

Simple digit sequence that is replaced by the macro short-code.

- **0-9**
- *****
- **#**

Character sequence that by dialling the macro short-code is transferred to the macro dialler and then started.

- **Z** (0-9, *, #)
- **W** (1-5) waiting time/break time of 1-5 s
- **H** Hang up with hook key

- **C** Waiting for connection

Note: If the macro does not end with an **H** and the macro dialler is after the execution of a macro in an active connection, the call will be connected to the subscriber who triggered it.

Macro short-code number

Note: A short-code number must not begin with **0** or the exchange line access number.

For example: the internal number 60 for the pick-up sequence ##06.

Macro name

For example: Pick-up

Macro sequence

Digit sequence (e. g. ##06) or sequence of macro commands that are selected or triggered by the short-code.

New

Delete

Print

Prints a list of all configured short-code macros.

LAN TAPI

Settings for LAN TAPI

Settings for LAN TAPI

Settings
Subscribers

Settings

Proceed as follows:

- Using the Configuration Manager (page **Functions > LAN TAPI**)

Network port of the TAPI server (1024-65535)

Port on the PBX which is used for communications with the TAPI server.

Note: The new configuration will only become effective after the PBX or the TAPI server has been restarted.

Password for the network connection of the TAPI server

Freely selectable password that consists of 8 to 32 characters containing digits, letters (upper and lower case, but no German umlauts and ß) and special characters **- _ . ! ~ * ' ( ) & = + $,**. The password is used for communications with the TAPI server.

Note: To change current passwords, click on **Edit passwords**.

Note: A coloured bar under the entry field shows the security level of the password (red = low level, green = high level).

Note: Passwords with a low security level can not be saved.

Caution: A high level does not necessarily indicate a secure password as this can also be reached by entering names and birth dates in connection with special characters. However, names and birth dates should not be used as passwords.

- To create a secure password you can consult, for example, the guidelines of the German Federal Office for Information Security (BSI, in German).

Caution: Visible passwords are a security risk.

- After entering the PIN/password, click on **Hide passwords**.

Restart TAPI server | Restart

Restarts the TAPI server.

Use PBX Call Assist

Requirements:

- PBX Call Assist function enabled (optional, see activation)

PBX Call Assist is a CTI software available from Auerswald. Its use must be switched on in the PBX.

All available TAPI lines are administrated by the corresponding server software UCServer. Other TAPI applications cannot be used in parallel.

Further steps:

- Install the server and client software. You will find the installation files on the Internet (see Auerswald Service Portal).

Subscribers

Proceed as follows:

- Using the Configuration Manager (page **Functions > LAN TAPI**)

+ New

Selects a LAN TAPI subscriber from the configured phones and adds it to the list. All subscribers that should be controlled over a PC must be configured for LAN-TAPI.

Note: If PBX Call Assist is switched on, all available TAPI lines will instead be administrated by the corresponding UCServer server software.

Delete

Deletes the entries marked in the list.

Call Parking

Settings Call Parking

Settings Call Parking

Proceed as follows:

- Using the Configuration Manager (page **Functions > Call parking**)

Internal base number

This is an internal number. You can select any of the numbers in the internal supply of phone numbers. When put together with the ten possible park positions 0-9, this creates ten 3 to 4-digit phone numbers. These numbers can then be used to select the parking zone for call parking.

Note: Double allocation of a number is not possible.

Note: Using phone numbers with few digits (two or three-digit numbers) inevitably restricts the possible supply of phone numbers requiring more digits. For example, if the phone number 12 is assigned, the phone numbers 120-129 and 1200-1299 are no longer available.

Note: The default exchange line access number on the PBX is 0 in the factory settings; however, this can be changed. Note that changing the exchange line access number restricts the phone numbering plan.

Name

Freely selectable name. This name is used to identify the internal phone number within the PBX.

Hold time (1..20 min.)

Specifies how many minutes a call is kept in the internal parking zone at most. After the hold time has elapsed, the subscriber that put the call in the parking zone is again signaled that the call is still there. If this subscriber is busy, the call is put back on hold in the parking zone. If the hold time elapses a second time without the call being transferred, the connection is disconnected and the caller hears the busy signal.

Online Name Search

Management Provider Online Name Search
Management Online Name Search Group Filters
Settings Online Name Search Group Filters
Settings Online Name Search Text Converter

Management Provider Online Name Search

Proceed as follows:

- Using the Configuration Manager (page **Functions > Online name search > Basic settings**)

Note: The service provider list accommodates up to five service providers.

Online name search

Requirements:

- A configured DNS server and, if necessary, an HTTP proxy server
  - Using the Configuration Manager (page **Administration > IP configuration**)
- At least one online service provider must have been entered for the online name search

The PBX attempts to obtain the missing name from the selected service provider.

Active

Select the service provider to be used for the online name search. (Select **New** to enter.)

Configure

Opens the configuration of the provider and the corresponding group filters.

Online service provider

Requirements:

- PBX with an existing Internet connection

A file with individual provider access data (*.n2n) will be downloaded from the update server. This file is then saved directly to the PBX.

Note: The access data provided for the download is for information purposes only, and is not mandatory.

New

Adds a new provider to the list of providers. You must then make the settings for the VoIP provider.

Delete

Deletes the entries marked in the list.

Import

Select and import a provider file (*.n2n), that is stored on a (PC) data storage device or download a configuration from the Internet.

Note: The data of the import file is added to the existing list of providers, without verification whether a provider is already contained in the list (double entries).

Export

The selected provider is saved to the (PC) data storage device in a file called nr2name_config.n2n.

Copy

Selects an existing provider from the list of providers and uses that provider as the basis for creating a new VoIP provider.

Standard

Resets the data of the service providers to default values.

Note: The data of the service providers is added to the existing list of providers, without verification whether a provider is already contained in the list (double entries).

Note: If resetting the data would cause the number of providers to exceed the maximum number of five entries, the data is not imported and an error message is displayed.

Test

Requirements:

- A configured DNS server, a configured gateway and, if necessary, an HTTP proxy server
  - Using the Configuration Manager (page **Administration > IP configuration**)
- Entries in the provider list

Tests the search properties. In the **Phone number** entry field, enter the phone number to be tested, in the **Filter (search properties)** list field, select how many of the configured search properties are to be processed, and click ✓ **Execute**.

Management Online Name Search Group Filters

Proceed as follows:

- Using the Configuration Manager (page **Functions > Online name search > Basic settings > Configure**)

Important: To create group filters, you need know-how in HTML, HTTP, and regular expressions.

Note: The filter list accommodates up to five group filters.

Group filter

↑ | ↓

Freely selectable name for the group filter. (Select + **New** to enter.)

Note: If several group filters have been created, the sequence of application of these filters can be determined (priority). A group filter will only be processed if the application of group filters with a higher priority has not been successful.

Configure

Opens the configuration of the group filter.

+ New

Adds a new group filter to the list of filters. You must then make the settings for the group filter.

Delete

Deletes the entries marked in the list.

Copy

Selects an existing group filter from the list of filters and uses that group filter as the basis for creating a new group filter.

Settings Online Name Search Group Filters

The group filters specify according to which search commands or character strings the response of the service provider (HTML code) will be searched.

If one of these search properties is found in the response of the service provider, the response will be modified according to the configured properties. The further search properties will be processed accordingly and the search result will be output.

Proceed as follows:

- Using the Configuration Manager (page **Functions > Online name search > Basic settings > Configure > Configure**)

Important: To create group filters, you need know-how in HTML, HTTP, and regular expressions.

Note: The filter list accommodates up to five group filters.

Search property

| |

Search command (regular expression) or the character string you want to search for. (Select **New** to enter.)

Note: If several search properties have been created, the sequence of application of these search properties can be determined (priority). A search property will only be processed if the application of search properties with a higher priority has not been successful.

Mode

Use Match

The response of the server (HTML code) will be filtered according to the search property and only the search result will be further processed.

Delete Match

The response of the server (HTML code) will be filtered according to the search property and the search result will be deleted. Only the rest will be further processed.

Replace Match

The response of the server (HTML code) will be filtered according to the search property and the search result will be replaced by the substitute text. The modified response will then be further processed completely.

Line Mode

The search is made line-by-line.

New

Adds a new search property to the list. You must then make the settings for the search property.

Delete

Deletes the entries marked in the list.

Copy

Selects an existing search property from the list and uses that search property as the basis for creating a new search property.

Settings Online Name Search Text Converter

The server transmits the search result of an online name search in HTML code. To view the result of the online name search, some characters in the determined name may have to be converted. The characters will be converted automatically if the characters, character strings, or texts to be substituted are entered into the text converter. During conversion, the character string will be substituted by the corresponding text.

Note: Up to 64 character strings can be entered into the text converter.

Proceed as follows:

- Using the Configuration Manager (page **Functions > Online name search > Text converter**)

Received string

Characters, character strings, or texts to be substituted. (Select **New** to enter.)

Text substitute

Characters, character strings, or texts, with which the received strings should be substituted.

New

Adds a new character string to the list. You must then make the settings for the character string.

Delete

Deletes the entries marked in the list.

Standard

Resets the character strings to default values.

Waiting Field

Settings Waiting Field

Settings Waiting Field

Proceed as follows:

- Using the Configuration Manager (page **Functions > Waiting field**)

Automatic call take over

No automatic

Incoming calls must be put manually into the waiting field.

On busy

Incoming calls are automatically put into the waiting field if all waiting field reception phones are busy. (In this case, “busy” means that there is an ongoing call in the active waiting field receptions or an outgoing or incoming call is ringing. If a caller is waiting in the waiting field, this does not count as “busy”.)

After 0..120 sec.

Incoming calls are indicated for a duration of x seconds in the waiting field reception and then automatically put into the waiting field if not already accepted manually.

Subscriber/group number for the Waiting Field

Select the group that is to be used as waiting field (alternatively single waiting field reception).

Music on Hold and Announcements

Management of the Music On Hold and Announcements
Settings for Music On Hold and Announcements

Management of the Music On Hold and Announcements

Own Files
System Files

Own Files

Proceed as follows:

- Using the Configuration Manager (Page **Functions > Music on Hold and Announcements > File management**)

MEMORY UTILISATION

The combined total size of all audio files (*.wav) stored in the PBX must be less than 20 MB. The amount of memory already used, and the remaining capacity, are displayed (in MB and as a percentage).

This list is where you manage the audio files you have created for the different PBX functions. You can use an external programme to create the audio files and then import them. Alternatively, you can record the announcements one of the PBX's internal phones (see Short reference). You can then assign the following file names to the audio files you create for the different functions:

- AudioIn_MoH.wav (music on hold)
- Phone_Alarm_x.wav (announcement for alarm)
- Phone_Auto_x.wav (announcement for automatic reception)
- Phone_Hold_Line.wav (announcement for music on hold)
- Phone_TvM_x.wav (announcement for announcement before answering)
- Phone_Wakeup_x.wav (announcement for wake-up function)
- emergency_call_not_possible_de.wav (announcement for emergency call)

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

Delete

Deletes the entries marked in the list.

Import

The audio file (*.wav) stored on a (PC) data storage device is selected and imported.

Note: The audio files (*.wav) must have the following properties: Format: 8-bit, A-law, 8 kHz, mono and file name containing 32 characters.

Note: In the Internet, you can convert the audio files into the audio format required for the PBX (see Auerswald Audio Converter). For example, use the Audio Converter to convert audio files (*.mp3, *.wav or *.raw) into G.711 (*.wav) audio format.

System Files

Proceed as follows:

- Using the Configuration Manager (Page **Functions > Music on Hold and Announcements > File management**)

MEMORY UTILISATION

The combined total size of all audio files (*.wav) stored in the PBX must be less than 20 MB. The amount of memory already used, and the remaining capacity, are displayed (in MB and as a percentage).

The audio files already stored in the PBX for various different PBX functions are displayed in this list. The **Standard assignment** column shows which audio file is usually used for each function.

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

Settings for Music On Hold and Announcements

Source for music on hold
Music on hold with announcement
music on Hold during the ringing period
Music on hold
Alarm
announcement before answering
Wake-Up Announcements
Automatic receptions
Emergency call

Source for music on hold

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Source of the Music on Hold

No

No music on hold is played.

Internal (PBX)

The music on hold is played from the PBX.

Music on hold with announcement

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Volume of music on hold during the announcement

Specifies how loudly the music on hold should be played during the announcement.

Mix announcement into music on hold

An announcement is played together with music on hold.

Note: When switched off, music on hold is played without an additional announcement.

Announcement delay time (0..60 sec.)

Specifies how many seconds delay there will be between the individual repetitions of the announcement.

music on Hold during the ringing period

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Blind Transferring

A caller connected by a blind transfer will exclusively hear music on hold during this process (pressing the R key, dialling of internal number, and ringing period).

Note: When switched off, the caller hears the ringtone during the ringing phase.

Ann. before answering

A caller hears the music on hold after the playback of the announcement before answering.

Note: When switched off, the caller hears the ringtone during the ringing phase.

Music on hold

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Audio file (*.wav)

Select the audio files that are to be used for the music on hold and function.

Standard announcement

For standard case, the PBX uses an audio file shown under **Functions** > **Music on Hold/announcements** > **File management** > **System Files**.

Volume on phone (in %)

Specifies how loudly the audio file is to be played on the phone.

Note: The volume at the phone for emergency calls cannot be set under 50 %.

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

Alarm

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Audio file (*.wav)

Select the audio files you want to use as alarm announcements. Specify which alarm announcement is to be used for an alarm ounder **Boxes** > **Alarms** > **Configure**.

Standard announcement

For standard case, the PBX uses an audio file shown under **Functions** > **Music on Hold/announcements** > **File management** > **System Files**.

Volume on phone (in %)

Specifies how loudly the audio file is to be played on the phone.

Note: The volume at the phone for emergency calls cannot be set under 50 %.

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

announcement before answering

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Audio file (*.wav)

Select the audio files that are to be used as announcement before answering files. Specify which announcement before answering is to be used for the Ann. before Answering function in the **Groups > Properties > Reachability** and **Public switched tel. networks > Call distribution** pages.

Default ann.

For the standard announcement, the PBX uses the audio file specified as the standard file in **Functions > Music on Hold and Announcements > File management > System Files**.

Volume on phone (in %)

Specifies how loudly the audio file is to be played on the phone.

Note: The volume at the phone for emergency calls cannot be set under 50 %.

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

Wake-Up Announcements

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Audio file (*.wav)

Select the audio files you want to use as wake-up announcements. Specify which wake-up announcements are to be used for the wake-up function on the **Time control > Switching times** page.

Default ann.

For the standard announcement, the PBX uses the audio file specified as the standard file in **Functions > Music on Hold and Announcements > File management > System Files**.

Volume on phone (in %)

Specifies how loudly the audio file is to be played on the phone.

Note: The volume at the phone for emergency calls cannot be set under 50 %.

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

Automatic receptions

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Audio file (*.wav)

Select the audio files you want to use as welcome announcements. Specify which welcome announcements are to be used for an automatic reception in **Functions > Automatic Receptions > Configure**.

Default ann.

For the standard announcement, the PBX uses the audio file specified as the standard file in **Functions > Music on Hold and Announcements > File management > System Files**.

Volume on phone (in %)

Specifies how loudly the audio file is to be played on the phone.

Note: The volume at the phone for emergency calls cannot be set under 50 %.

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

Emergency call

Proceed as follows:

- Using the Configuration Manager (**Functions** > **Music on Hold/ announcements** > **File assignment** page)

Audio file (*.wav)

Select the audio file that is to be used for the emergency call function.

Standard announcement

For standard case, the PBX uses an audio file shown under **Functions** > **Music on Hold/announcements** > **File management** > **System Files**.

Volume on phone (in %)

Specifies how loudly the audio file is to be played on the phone.

Note: The volume at the phone for emergency calls cannot be set under 50 %.

Listen

A suitable programme is used to open the audio file. The file is then either played or saved to a (PC) data storage device.

Wake-Up Times

Settings for Wake-Up Functions

Settings for Wake-Up Functions

Settings
Wake-up Times for Subscribers

Settings

Proceed as follows:

- Using the Configuration Manager (Page **Functions > Wake-up times**)

Maximum number of wake-up call attempts per wake-up time

Specifies how often a wake-up call is made during each wake-up time. When a wake-up call is acknowledged, the wake-up process for each wake-up time is interrupted.

Call duration of a wake-up call (30..120 sec.)

Specifies how long an individual wake-up call is played on an end device.

Pause between the first and second wake-up call

This specifies how many minutes pause there is between the first and second wake-up call.

Pause between all following wake-up calls

Specifies how many minutes pause there is between subsequent wake-up calls.

Record wake-up calls in the call database

Wake-up calls are recorded in the call database.

Use wake-up times of Sundays for holidays

Requirements:

- holidays must have been configured and enabled

The wake-up calls set up for a Sunday are played on enabled holidays.

Note: When switched off, the wake-up calls of the corresponding weekdays are played on holidays.

Wake-up Times for Subscribers

Proceed as follows:

- Using the Configuration Manager (Page **Functions > Wake-up times**)

WAKE-UP TIMES FOR SUBSCRIBERS

Select the subscriber whose wake-up times are to be displayed/ configured.

Wake-up times

Time (format: hh:mm) at which the wake-up call is to be made. (Select + **New** to enter.)

Configured via

wake-up time was configured using the Configuration Manager.

wake-up time was configured using a character string on the internal phone.

Day

Weekday/s on which the wake-up call is made.

Always

Multiple wake-up calls are enabled. The wake-up call is made for the duration of the specified time, until the user switches it off.

Note: Single wake-up call is enabled. A wake-up call is made once, then switched off automatically.

activated

Wake-up call is switched on and will be made on the next possible date.

+ New

Adds a new wake-up time.

Note: A maximum of 20 wake-up times can be configured per subscriber.

Delete

Deletes the entries marked in the list.

User data

Call Data
Address Books
Special Numbers
Messages

Call Data

Management Call Data
Call Data Column Settings
Settings Filter
Settings Call Data

Management Call Data

Proceed as follows:

- Using the Configuration Manager (page **User data > Call data > Call data list**)

Date and time

Display of the date/time when the call was started.

Duration

Display of the call duration.

LCRNo

Display of the dialled number of the LCR provider, if an external number was dialled via a provider (manual entry or automatic LCR).

External partner

Displays call partner's external phone number.

- **Dialled calls**: Displays the phone number that was dialled.
- **Incoming calls**: Displays the phone number transferred by the caller (CLIP information; not for unknown phone numbers).
- **Wake-up calls**: Displays the phone number of the person receiving the call.

Important: The phone numbers in an incoming VoIP conversation can only be used for callbacks if **Evaluation of incoming SIP messages** is active.

Note: In the case of private calls, the last three digits are not recorded. Only an anonymised phone number is displayed (the last three digits are represented by ***).

	The phone number is entered in a call restrictor.
	The call partner's name and phone number are transferred to the address books.
	Online name search: This is where you can add the name to the call data record and also check and update it. Requirement: Online name search is configured and enabled.
	Soft Call: Dials the phone number directly. Requirement: User is logged on to the configuration manager. Requirements for D-series telephones: Action URL function activated in the telephone. Identical token in the PBX and telephone.

External Name

Displays the name of the external call partner if stored under **User data > Address books** (Users: **Address books**) or transmitted via CLIP. For VoIP calls, the **Display name** (**Public switched tel. networks > VoIP > Provider > Configure**) is displayed if there is no entry in the address book for this VoIP number.

Note: If there is no **External name** available, the data record can be updated via the **Online name search** function.

Scr No.Invoice

Displays the internal subscriber or group number for:

- **Outgoing calls** (business): Subscriber number of the end device. If the internal subscriber is logged-in to a group, the group number is displayed.
- **Private calls**:

  Initiation with PIN: Internal number
  Initiation without PIN: Internal number of the executing end device
- **Incoming calls**: Internal subscriber or group number dialled by the caller
- Alarm calls: Phone number of this alarm
- **Call Through call**: External number of the Call Through user
- **Door-to-external calls**: Phone number of the door terminal
- **Wake-up calls**: Internal number of the person to be woken up

Scr Name Invoice

Displays the subscriber or group name for:

- **Internal subscriber numbers**
- **Internal group numbers**
- **Alarm calls**
- **Call Through users**
- **Door-to-external calls**

Scr no.real

Displays the internal subscriber or group or group number for:

- **Outgoing calls** (business or private): Internal number is displayed.

- **Incoming calls**: Internal number of the accepting end device or external number of the accepting external call forwarding destination (unsuccessful calls are not displayed).
- **Alarm calls** to an internal subscriber: Phone number of the internal end device (alarm calls to external subscribers are not displayed).
- **Door-to-external calls**: Internal subscriber number who forwarded the call to an external connection (direct calls to external subscribers are not displayed).

Scr Name real

Displays the subscriber or group name for:

- **Internal subscriber numbers**
- **Internal group numbers**
- **Alarm calls**
- **Call Through users or external numbers**
- **Door-to-external calls**

Connection no.

Displays the own phone number.

- **Outgoing call**: Phone number used for forwarding the call to the public exchange
- **Incoming call**: Phone number dialled by the caller
- **Call made via ISDN PTMP connection**: MSN
- **Call made via ISDN PTP connection**: PBX base number + DDI or additional number
- **Call made via analogue connection**: Analogue number

- **VoIP call**: VoIP number

Charges

Amount charged for the call

- LCR data
- Data transmitted by the network provider (units are calculated into charges with the price per unit entered under **Functions > Call allowance accounts**.

Note: If the charges for the calls are to be recorded, transmission of the call charges must be enabled on the external S_0 ports.

Note: If using an analogue exchange line or a VoIP account for external calls, no charges will be displayed unless charges have been entered via Soft LCR tables and the subscriber uses uses the LCR function.

Direction

Outgoing or incoming call.

Billing Type

business	Business call
private	Private call: External call with private exchange line access
successful	Successful call
unsuccessful	Unsuccessful call

Call Type

The text entered in the print options or the default text for this call type is displayed.

Project (Proj.)

Call charges or times can be billed according to individual projects. Business calls are then initiated with project numbers that are recorded and displayed if the call was successfully transferred.

Hotel invoice number (HotId)

The check-in via the hotel reception phone automatically generates a 5-digit billing number which is displayed for calls via the hotel function.

Standard sorting

The call data is sorted by date and time.

Refresh

The call data list is refreshed.

Delete

Deletes the entries marked in the list.

Delete all

All marked entries in the list will be deleted.

Export

The current data in the table (gesp_dat.csv.gz) can be exported (gespr_dat.csv) for further processing.

Important: Note the following legal notes before saving the call data.

The call data include, amongst others, the following legally especially protected components:

- The time of a call
- The phone numbers of the internal and external participants
- The duration of a call

Note: Before the call data are being saved, all subscribers to the PBX must be informed that these contents will be forwarded. They must also be informed about who will use the data received.

Note: It is absolutely necessary to store the data safely and in accordance with the legal requirements.

Note: If you want to make the call data available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

Print

All entries are printed.

Call Data Column Settings

Proceed as follows:

- Using the Configuration Manager (page **User data > Call data > Print options**)

Date

Display of the date at the beginning of the call.

Time

Displays the time at the beginning of the call.

Duration

Display of the call duration.

LCRNo

Display of the dialled number of the LCR provider, if an external number was dialled via a provider (manual entry or automatic LCR).

External partner

Displays call partner's external phone number.

- **Dialled calls**: Displays the phone number that was dialled.
- **Incoming calls**: Displays the phone number transferred by the caller (CLIP information; not for unknown phone numbers).
- **Wake-up calls**: Displays the phone number of the person receiving the call.

Important: The phone numbers in an incoming VoIP conversation can only be used for callbacks if **Evaluation of incoming SIP messages** is active.

Note: In the case of private calls, the last three digits are not recorded. Only an anonymised phone number is displayed (the last three digits are represented by ***).

	The phone number is entered in a call restrictor.
	The call partner's name and phone number are transferred to the address books.
	Online name search: This is where you can add the name to the call data record and also check and update it. Requirement: Online name search is configured and enabled.
	Soft Call: Dials the phone number directly. Requirement: User is logged on to the configuration manager. Requirements for D-series telephones: Action URL function activated in the telephone. Identical token in the PBX and telephone.

External Name

Displays the name of the external call partner if stored under **User data > Address books** (Users: **Address books**) or transmitted via CLIP. For VoIP calls, the **Display name** (**Public switched tel. networks > VoIP > Provider > Configure**) is displayed if there is no entry in the address book for this VoIP number.

Note: If there is no **External name** available, the data record can be updated via the **Online name search** function.

Scr No.Invoice

Displays the internal subscriber or group number for:

- **Outgoing calls** (business): Subscriber number of the end device. If the internal subscriber is logged-in to a group, the group number is displayed.
- **Private calls**:

  Initiation with PIN: Internal number
  Initiation without PIN: Internal number of the executing end device
- **Incoming calls**: Internal subscriber or group number dialled by the caller
- Alarm calls: Phone number of this alarm
- **Call Through call**: External number of the Call Through user
- **Door-to-external calls**: Phone number of the door terminal
- **Wake-up calls**: Internal number of the person to be woken up

Scr Name Invoice

Displays the subscriber or group name for:

- **Internal subscriber numbers**
- **Internal group numbers**
- **Alarm calls**
- **Call Through users**

- **Door-to-external calls**

Scr no.real

Displays the internal subscriber or group or group number for:

- **Outgoing calls** (business or private): Internal number is displayed.
- **Incoming calls**: Internal number of the accepting end device or external number of the accepting external call forwarding destination (unsuccessful calls are not displayed).
- **Alarm calls** to an internal subscriber: Phone number of the internal end device (alarm calls to external subscribers are not displayed).
- **Door-to-external calls**: Internal subscriber number who forwarded the call to an external connection (direct calls to external subscribers are not displayed).

Scr Name real

Displays the subscriber or group name for:

- **Internal subscriber numbers**
- **Internal group numbers**
- **Alarm calls**
- **Call Through users or external numbers**
- **Door-to-external calls**

Connection no.

Displays the own phone number.

- **Outgoing call**: Phone number used for forwarding the call to the public exchange

- **Incoming call**: Phone number dialled by the caller
- **Call made via ISDN PTMP connection**: MSN
- **Call made via ISDN PTP connection**: PBX base number + DDI or additional number
- **Call made via analogue connection**: Analogue number
- **VoIP call**: VoIP number

Charges

Amount charged for the call

- LCR data
- Data transmitted by the network provider (units are calculated into charges with the price per unit entered under **Functions > Call allowance accounts**.

Note: If the charges for the calls are to be recorded, transmission of the call charges must be enabled on the external S_0 ports.

Note: If using an analogue exchange line or a VoIP account for external calls, no charges will be displayed unless charges have been entered via Soft LCR tables and the subscriber uses uses the LCR function.

Billing factor

Dispalay of the cobversion factor for the costs. The value can be changed depending on the subscriber. The change then applies only to calls after the change.

Note: For logged in users, the **Billing factor (Fact.)** is not visible. Instead, the calculated product of transmitted charges and **Billing factor (Fact.)** is displayed under **Charges**.

Direction

Outgoing or incoming call.

Billing Type

business	Business call
private	Private call: External call with private exchange line access
successful	Successful call
unsuccessful	Unsuccessful call

Call Type

The text entered in the print options or the default text for this call type is displayed.

Project (Proj.)

Call charges or times can be billed according to individual projects. Business calls are then initiated with project numbers that are recorded and displayed if the call was successfully transferred.

Hotel invoice number (HotId)

The check-in via the hotel reception phone automatically generates a 5-digit billing number which is displayed for calls via the hotel function.

Call data list

The table column is added to the call data list.

Description in call data list

Displays the description of the colum in the call data list.

Standard

All settings will be reset to the standard values.

Settings Filter

Filter Criteria
Configure

Filter Criteria

Proceed as follows:

- Using the Configuration Manager (page **User data > Call data > Filter**)

Filter name

Name of the used filter (standard or own filter).

Time period

Filter criteria for the time period.

Configure

Proceed as follows:

- Using the Configuration Manager (page **User data > Call data > Filter**)

PERIOD

Default period

For example **today**

Configuring the following parameters:

- **Period** (only for the selection **user-defined**): Fixed period
- **Daytime**: Fixed daytime
- **Call duration**: Displays calls of a certain length
- **Charges**: Displays calls of a certain call charge

DIRECTION

Incoming calls, **Outgoing calls**

Calls that are displayed after being filtered.

BILLING TYPE

Private Calls, **Business Calls**, **Successful Calls**, **Unsuccessful Calls**

Calls that are displayed after being filtered.

HOTEL FUNCTION

Hotel invoice number (1..65535)

Entering the hotel invoice number.

CALL TYPE

Characteristic a call may have

Call type if only these calls are to be displayed after being filtered.

SPECIFIC FILTERING

Possible entries for the following settings: Phone number, name and replacement *

Subscriber (Scr no. invoice + scr no.real)	Calls made by selected internal subscribers.
Connection number	Calls initiated via the own external number.
External partner	Calls made with selected external subscribers.
Project number/s	Calls assigned to certain projects.

Back

Back to previous page.

Settings Call Data

Call data management
SFTP transfer

Call data management

Proceed as follows:

- Using the Configuration Manager (**User data > Call data > Settings**)

Selection of the calls to be recorded in order to avoid filling the call data memory too fast.

Dialled calls

Note: The transmission of the charge information can be disabled for ISDN (**Public switched tel. networks > ISDN > PTMP connection** and **Public switched tel. networks > ISDN > PBX base numbers/trunk bundles > Configure**).

The following calls can be recorded:

- **Chargeable calls**

  Chargeable calls and unsuccessful outgoing calls (e. g. transfer charges).

- **Free of charge/calls without charge information**

  Calls that are free of charge or for which no charge information is transmitted.

  - **Successful only**

    Calls that are free of charge or for which no charge information is transmitted (unsuccessful calls are not displayed)

- **Private calls with partial telephone number privacy**

  For reasons of data protection, the traceability of external numbers for outgoing private calls can be dispensed with. The last three digits of

the phone number are displayed with ***. The remaining call data is recorded.

Incoming Calls

- **Successful**
  - **Only with charges**

    Calls that cause charges (e.g. collect calls).
- **Unsuccessful**

  Unsuccessful calls.
- **Charges independent from calls**

  Calls with additional charges (e.g. the transfer charge for collect calls).

General settings

- **Record Wake-up calls in the call database**

  Wake-up calls to internal subscribers (see **Functions > Wake-up times**)

SFTP transfer

Proceed as follows:

- Using the Configuration Manager (**User data > Call data > Settings**)

A call data file is provided in configurable intervals through the SFTP server of the PBX for pickup.

General Settings

Determines in which form the call data is provided.

- **Data separator**
- **Generate header**

call data file

Determines when and how often a call data file is provided.

- **File creation**
- **Cleanup interval**
- **Automatic deletion if (1..200 MB) is used**
- **Folder of call data files** /calldata/

Hotel function

Settings for providing subscriber-related call data in hotel operation.

- **Creation of a subscriber-relating file for each check-out**
- **Cleanup interval**
- **Automatic deletion if (1..200 MB) is used**
- **Folder of call data files** /checkoutdata/

Address Books

Management Contacts and assignment
Settings of the Contacts
Settings of the Contact Groups
Settings of the Contact Group Authorisations
Settings Standard Contact Groups
Settings of the Address Books
Information about Transfer via LDAP Server

Management Contacts and assignment

Contacts
Contact Groups and Assignment
Google Contact Groups
CSV Import Group

Contacts

Proceed as follows:

- As administrator: page **User data > Address books > Contacts and assignment**
- As a subscriber: page **Contacts and assignment**

Requirements:

- For subscribers: read access to at least one contact group

All contacts

All contacts are displayed.

Note: Searching for contacts can be restricted in the following way:

- via contact groups by clicking a contact group.
- using the filter by entering at least one character or digit in one or more of the entry fields.
- using the tabs by clicking **All**, **Contacts without contact groups**, or an initial letter.

Buttons for each contact for:

	View
	Editing
	Displaying the QR code
	Exporting a vCard (Format vCard 2.1)

Soft Call

Requirements:

- Logged in as subscriber

The stored phone number is automatically dialled.

New

Delete

In the **All contacts** view, you can delete either individual contacts or all contacts from a standard or DECT contact group. To delete a contact from a **Google contact group**, you first need to delete the contact in Gmail. The contact will then be deleted from the Google contact group linked to Gmail after the next synchronisation.

Important: Deleting a contact cannot be undone.

Proceed as follows for subscribers:

- Write access to at least one contact group
- Only contacts that are assigned to the contact groups for which the subscriber has write access

Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

Note: The **Delete** button is only available in the **All contacts** view.

Delete all

If you have administrator or sub-administrator authorisation, you can delete all the contacts simultaneously in the **All contacts** view.

Important: If you delete all the contacts you cannot undo this step.

Note: The **Delete all** button is only available in **All contacts** view.

vCard v2.1 import

Import of a vCard in version 2.1.

csv import

If you have exported an address book in *.csv format, e.g. from a COMmander 6000 that is using an older firmware version or from a different PBX, you can import the contact data in that file to **All contacts** view and then assign the contacts to one or several contact groups. Contacts whose fields are identical to existing contacts will be overwritten. In all other cases a new contact will be created.

Requirements:

- For users: read access to at least one contact group
- A text file (*.csv file) in address book format, saved on a data medium (PC)

Note: The **csv import** button is only available in **All contacts** view.

Note: If contacts with identical contact data are already present they will be overwritten.

Note: If contacts with different contact data are already present, new contacts are created.

Note: The contacts are automatically assigned to the predefined standard contact groups.

CSV FILE

A csv file (* .csv) is loaded into the PBX via the Configuration Manager.

IMPORT

Select an import template.

PREVIEW

Display the entries in the CSV file.

csv export

Requirements:

- For subscriber: read access to at least one contact group

You can export contact groups, contacts with the same initial letter or **All contacts** in a single text file (*.csv).

Important: Note the following legal notes. The contacts may contain the following legally protected components:

- Private and work addresses
- Private and work phone and fax numbers
- Birthdays
- Photos of the person (in connection with the data backup)
- Electronic contact addresses (private and work), e. g. the e-mail addresses
- Position and department within a company

Note: All subscribers must be informed prior to the storage of the contacts that a transfer of this content takes place and by whom the data obtained are processed.

Note: A secure storage of the data within the framework of the legal regulations is necessary.

Note: If you want to make the contacts available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

Contact Groups and Assignment

Proceed as follows:

- Using the Configuration Manager (page **User data > Address books > Contacts and assignment**)

Note: Only one type can be selected per contact group. It is not possible to change the contact group type afterwards.

Note: If you have not selected a type, the **Standard contact group** type is automatically assigned to the new contact group.

Delete contacts in contact group

The selected contact will be removed from all contact groups.

The selected contact will be removed permanently.

Important: Deleting a contact cannot be undone.

Proceed as follows for subscribers:

- Write access to at least one contact group
- Only contacts that are assigned to the contact groups for which the user has write access

Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

csv export

Requirements:

- For subscriber: read access to at least one contact group

You can export contact groups, contacts with the same initial letter or **All contacts** in a single text file (*.csv).

Important: Note the following legal notes. The contacts may contain the following legally protected components:

- Private and work addresses
- Private and work phone and fax numbers
- Birthdays
- Photos of the person (in connection with the data backup)
- Electronic contact addresses (private and work), e. g. the e-mail addresses
- Position and department within a company

Note: All subscribers must be informed prior to the storage of the contacts that a transfer of this content takes place and by whom the data obtained are processed.

Note: A secure storage of the data within the framework of the legal regulations is necessary.

Note: If you want to make the contacts available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

Google Contact Groups

Proceed as follows:

- Using the Configuration Manager (page **User data > Address books > Contacts and assignment**)

Max. 5 Google groups are displayed. If you have a Gmail account with contact data, they can be transferred into the address books of the PBX and dragged into the desired contact group (exception: Google groups).

On **Google groups** you have only read access. Contacts from Google groups can only be copied into another contact group. Changes must be made directly in the Gmail account. They are adopted with the next synchronization. If you delete a Gmail contact from a standard or DECT contact group, it still remains in the Google group.

Delete contacts in contact group

The selected contact will be removed from all contact groups.

The selected contact will be removed permanently.

Important: Deleting a contact cannot be undone.

Proceed as follows for subscribers:

- Write access to at least one contact group
- Only contacts that are assigned to the contact groups for which the user has write access

Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

csv export

Requirements:

- For subscriber: read access to at least one contact group

You can export contact groups, contacts with the same initial letter or **All contacts** in a single text file (*.csv).

Important: Note the following legal notes. The contacts may contain the following legally protected components:

- Private and work addresses
- Private and work phone and fax numbers
- Birthdays
- Photos of the person (in connection with the data backup)
- Electronic contact addresses (private and work), e. g. the e-mail addresses
- Position and department within a company

Note: All subscribers must be informed prior to the storage of the contacts that a transfer of this content takes place and by whom the data obtained are processed.

Note: A secure storage of the data within the framework of the legal regulations is necessary.

Note: If you want to make the contacts available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

CSV Import Group

Proceed as follows:

- Using the Configuration Manager (page **User data > Adress books > Contacts and assignment**)

Display of imported CSV contacts that are stored temporarily in the PBX. You can select the contact data and drag it into the view **All contacts** or directly into a contact group (except from Google contact groups).

Delete contacts in the CSV contact group

The selected contact is deleted from the CSV import group.

Important: Deleting a contact cannot be undone.

Requirements for subscribers:

- Write access to at least one contact group
- Only contacts that are assigned to the contact groups for which the subscriber has write access

Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

Delete all CSV contact groups

All contacts are deleted from the CSV import group.

Important: Deleting a contact cannot be undone.

Requirements for subscribers:

- Write access to at least one contact group
- Only contacts that are assigned to the contact groups for which the user has write access

Adopting contacts from CSV import group

Select all or a selected CSV contact you want to adopt from the CSV import group into the pbx.

Important: Deleting a contact cannot be undone.

CONTACT GROUPS

Select all or a selected contact group.

Move

The CSV contacts are moved into all or a selected contact group.

Requirements for subscribers:

- Write access to at least one contact group
- Only contacts that are assigned to the contact groups for which the subscriber has write access

Settings of the Contacts

Proceed as follows:

- Using the Configuration Manager (page **User data > Address books > Contacts and assignment >** + ,)

A contact must at least consist of a phone number and a corresponding entry in the entry fields **Surname** and **Company**. Additionally, you can add further phone numbers (up to 4), names, e-mail and Internet addresses, postal addresses and a contact image. Furthermore, a contact is assigned to one or more contact group(s).

For the displaying in the call data list and for various connected phones the **display name** is used. If you have not entered anything in the entry field **display name**, the display name will automatically be generated from the entries in the entry fields **First name** and **Surname**. If there are no entries at all in the entry fields **First name** and **Surname**, the display name will then be generated from the entry in the entry field **Company**.

Note the following differences when using the contact data in the connected phones:

COMfortel 1400 IP, 2600 IP, 3200, 3500 und 3600 IP

In the contact list of the phone almost all contact data will be displayed except for display name, department name and birthday. Exception: If no first name or surname was entered at all for a contact, the display name will be displayed.

If there already is a contact entry with the same phone numbers in the local address book, the contacts will be merged in the phone. The local address book name entry is preserved.

COMfortel 1400, 2600

In the phone‘s contact list the display name will be displayed.

Additionally to the display name, four phone numbers per contact will be displayed.

COMfortel 1200, 1600

In the contact list of the phone only the first 16 characters of the display name will be displayed (perhaps more, depending on space and character width).

Additionally to the display name four phone numbers per contact will be displayed.

+ New

Settings of the Contact Groups

Proceed as follows:

- Using the Configuration Manager (**User data > Address books > Contact groups**)

Contact groups

Name of the contact group.

Number of contacts in contact group

Specifies how many contacts are existing in a contact group.

Contact group type

To organise large address books, up to 20 contact groups can be created and assigned for **All contacts**.

Administrators and sub-administrators can create own contact groups and assign types to them. These types set the possible size of the contact group and specify the phones to which the contacts in this contact group are to be transmitted.

Contact group type	
Standard	• A maximum of 2000 contacts • For the system phones - COMfortel 1400 IP/2600 IP/3600 IP - COMfortel 3200/3500 - COMfortel 1200/1400/1600/2600

Contact group type	
Google	• For synchronisation with Gmail

Note: Only one type can be selected per contact group. It is not possible to change the contact group type afterwards.

Note: If you have not selected a type, the **Standard contact group** type is automatically assigned to the new contact group.

Last Google update

Displays the date and time on which the Google contact group was last synchronised with the Gmail account's contact data, either automatically or manually.

Configure

Use the **PBX Google contact group** contact group type to transfer contact data from a Gmail account to the PBX's address books. Once you have created a Google contact group, you can synchronise it with one or all the contact groups managed under Gmail. You can either do this manually or on a time-specific basis.

Note: From firmware version 8.0 onwards, synchronisation is performed via a different interface. Reauthentication is required for existing Google contact groups. A note is displayed next to the button to this effect. After authentication, you can continue configuring the contact group.

ACCESS DATA

Release code

This code is generated and transferred automatically by Google.

Copy

Copies the release code to the clipboard so it can be used for authentication.

Authentication

You are transferred directly to Google, so you can authenticate your Google account. Input the required data in the authentication dialogue.

CONFIGURATION

Google synchronisation active

The Google contact group is synchronised with the contact data in the Gmail account.

Last update

Displays the date and time on which the Google contact group was last synchronised with the Gmail account's contact data, either automatically or manually.

Update interval

Enter the time interval for automatic synchronisation.

Google Contact Groups

Select the Google contact groups that are to be synchronised automatically.

Manual synchronisation

Manual synchronisation of the Gmail managed contact group can be initiated at any time, regardless of automatic synchronisation. The status can be queried on the **User data > Address books > Contacts and assignment** page.

	There is no synchronisation yet
	Synchronisation successful
	Synchronisation is running
	Synchronisation not successful

Note: A maximum of 2000 contacts can be synchronised.

New

A new contact group is added.

Save

The changes are saved.

Delete

The selected contact group is deleted.

Requirements for subscribers:

- Write access to at least one contact group
- Only contacts that are assigned to the contact groups for which the subscriber has write access

Note: You can select and delete multiple contact groups at once by holding down the Shift or Ctrl key.

Settings of the Contact Group Authorisations

Authorisations
Contact Group Filter

Authorisations

The administrator or sub-administrator assigns read and/or write authorisations to the individual subscribers for the contact groups.

A subscriber with read access to at least one contact group may via the **Contacts and assignment** page:

- search for and display a contact
- dial from within the contacts (Soft Call)
- read the QR code
- export a contact data record (vCard)

A subscriber with write an read acces may via the **Contacts and assignment** and **Standard contact groups** pages additionally:

- create contacts
- import contact images
- edit contacts
- import contact data sets (vCard)
- adopt contacts from the call data list
- enter short-code numbers
- delete contacts
- import address books
- export address books
- switch the use as standard contact group on/off
- assign contacts to a contact group
- delete contacts from a contact group

Contact Group Filter

Proceed as follows:

- Using the Configuration Manager (page **User data > Address books > Contact group authorisations**)

Requirements:

- At least one configured contact group

CONTACT GROUP FILTER

Select a contact group or all contact groups for editing the authorisations for individual subscribers.

Filter:	Enter filter criteria for the search of contacts into the entry fields **Phone number** and **Name**.
All:	The subscriber receives read and write access for all contact groups.
Read access:	The subscriber receives read access for this contact group.
Write access:	The subscriber receives write access for this contact group.

Settings Standard Contact Groups

Information
Logged-in Subscriber
Subscriber with Write Authorisation

Information

New contacts are automatically assigned to the specified standard contact groups either as soon as they are created (when you click the + **New** button), or they are imported as a vCard or from the call data list. However, contacts that have been created by importing an address book file are not automatically assigned to a standard contact group.

You can define one or more contact group(s) as your standard contact group. If you are logged on as a user, you must define at least one contact group as your standard contact group before you can create a contact. This is because users cannot create a contact without a contact group.

A Google contact group cannot be used as standard contact group.

Logged-in Subscriber

Proceed as follows:

- As administrator: page **User data > Address books > Standard contact groups**
- As a subscriber: page **Contacts and assignment**

Requirements:

- For subscribers: read access to at least one contact group

Name

Name of the logged-in subscriber.

Standard contact groups

The contact group is defined as standard contact group.

Subscriber with Write Authorisation

Proceed as follows:

- As administrator: page **User data > Address books > Standard contact groups**
- As a subscriber: page **Contacts and assignment**

Requirements:

- For subscribers: write access to at least one contact group

Phone number

Phone number of the phone.

Name

Name of the phone.

Save

The changes are saved.

Settings of the Address Books

LDAP server
Conversion

LDAP server

Proceed as follows:

- Using the Configuration Manager (**User data > Address books > Settings**)

Enable server

The PBX's LDAP server is one of the databases based on this protocol. It stores the contact data for the PBX's central address books. As the LDAP server displays the contact data stored in the central address books, it can also hold up to 2000 contacts like the address books.

The LDAP server is needed to transfer the PBX's contact data to those IP end devices which support this transfer method (e.g. IP DECT servers or SIP phones with LDAP support). Any contact data that is input directly on a phone is not transferred to the central address books.

Auerswald/FONtevo provides LDAP support for the following IP end devices:

- IP phone: COMfortel 1200 IP
- IP phones: COMfortel 1400 IP/2600 IP/3600 IP (firmware version 2.4A, or later)
- IP-DECT server: COMfortel WS-400 IP, COMfortel WS-650 IP (only Auerswald)
- IP-DECT handsets: COMfortel M-100/200/210/300/310 (only Auerswald)

Contact data can be transferred to the IP end devices.

Note: When switched off, the security and performance of the PBX are improved. No contact data can be transferred to the IP end devices.

Note: The LDAP server's IP address matches the PBX's IP address and is predefined.

Simplified user name

Freely selectable part of the user name.

The user name along with the associated password is needed for authentification of IP end devices on the LDAP server so that they have (only read) access to all the contact data that is currently present on the LDAP server.

Exported contact groups

Requirements:

- At least one contact group that is configured in the address books

Select the contact groups (up to 5), which are transferred to the LDAP server and therefore also to the IP end devices.

The user name resulting from the name of the contact group along with the associated password is needed for authentification of IP end devices on the LDAP server so that they have (only read) access to the contact data for this contact group.

Password

Freely selectable password that consists of 8 to 32 characters containing digits, letters (upper and lower case, but no German umlauts and ß) and special characters **- _ . ! ~ * ' ( ) & = + $,**.

Note: To change current passwords, click on **Edit passwords**.

Note: A coloured bar under the entry field shows the security level of the password (red = low level, green = high level).

Note: Passwords with a low security level can not be saved.

Caution: A high level does not necessarily indicate a secure password as this can also be reached by entering names and birth dates in connection with special characters. However, names and birth dates should not be used as passwords.

- To create a secure password you can consult, for example, the guidelines of the German Federal Office for Information Security (BSI, in German).

Caution: Visible passwords are a security risk.

- After entering the PIN/password, click on **Hide passwords**.

Format of transferred phone number for LDAP requests

The PBX supports different formats, so you can dial the contact's phone number correctly using an LDAP request.

Conversion

Proceed as follows:

- Using the Configuration Manager (page **User data > Address books > Settings**)

Display name conversion

If you did not enter any data in the **Display name** field when you created a contact, the display name is generated automatically from the entries in the **First name** and **Surname** entry fields, in accordance with the predefined conversion process, when you save the entry.

Surname,**First name**: Display names are formed from the surname and the first name (separated by a comma and a blank space).

First name Surname: Display names are formed from the first name, a blank space and the surname.

Note: If no data is present either in the **First name** entry field or the **Surname** entry field, the display name is generated from the entry in the **Company** entry field.

Information about Transfer via LDAP Server

For the transfer of contact data from the LDAP server of the PBX to the IP end device you have to enter the following values into the end device.

For the transfer of all selected contact groups (e. g. in the factory settings):

- User name **cn=auerswaldschandelah,dc=auerswald**
- BaseDN **dc=auerswald**
- Password **auerswaldsecret**

For the transfer of single contact groups (e. g. for contact group **DECT1**):

- User name **cn=DECT1,dc=auerswald**
- BaseDN **ou=DECT1,dc=auerswald**
- Password is configurable

For the transfer of contacts of a single subscriber (e. g. for the internal number **31**):

- User name **telephoneNumber=31,dc=auerswald**
- BaseDN **dc=auerswald**
- In this case the password is the user password of the subscriber.

Note: You must enter the complete character string

Note: Ensure you type the capital and lower-case letters correctly, and do not type any blank spaces before or after the **,** and **=** characters.

Note: In order to avoid input errors, on page **User data > Address books > Settings**, in the lines below the entry and list fields you can mark and copy the user name and BaseDN.

Note: In the Configuration Manager of the IP DECT server COMfortel WS-400 IP or COMfortel WS-650 IP you have to enter the user name and the BaseDN on page **Administration > Phonebook** in the following fields: the user name (e. g. **cn=auerswaldschandelah,dc=auerswald**) in the entry field **Bind user** and the BaseDN (e. g. **dc=auerswald**) in the entry field **Base**.

Special Numbers

Management
Configuration

Management

Proceed as follows:

- Using the Configuration Manager (page **User data > Special numbers**)

Name

Name of the call deblocker/call restrictor.

Type

This column displays the direction (incoming/outgoing) of the call deblocker/call restrictor.

Reject anonymous calls

Anonymous incoming calls are rejected by the call restrictor.

Reject calls with special characters

The following characters are valid for phone numbers: 0-9, space, (), - , +. Phone numbers containing other characters will be rejected.

Ext.number/range / Name

Display of the phone number/range and name of the corresponding number.

Configuration

Proceed as follows:

- Using the Configuration Manager (page **User data > Special numbers > Configure**)

Phone number / range

Phone number and range of the corresponding call deblocker/call restrictor.

Name

Name of the corresponding phone number.

New

A new call deblocker/call restrictor is added.

Save

The changes are saved.

Delete

The selected entry will be deleted.

Messages

Management

Management

Requirements:

- At least one received voice message or fax
- For the replay of the voice message: installed audio player
- To read the fax: installed Acrobat Reader

Proceed as follows:

- Using the Configuration Manager
  - For admin: **User data > Address books > Messages**
  - For user: **Messages (Voice mail-/fax boxes)**

MESSAGES

Selcet the voice mailbox in question according to its name or number.

Filter

all

Shows all messages/faxes including those in the archive.

new

Shows only new messages/faxes.

today to **last month**

Shows all messages/faxes received in the specified time which have not yet been archived.

Archive

Shows all messages/faxes moved to the archive.

Direction

all

Shows all messages/faxes including those in the archive.

received

Shows only the received messages/faxes.

sent

Shows only the sent messages/faxes.

Search (phone number or name) Search

Phone number

Only messages/faxes are displayed which contain the searched phone number in the **Caller**, **Call destination**, or **Exchange line** columns.

Phone number ranges

Only messages/faxes are displayed which contain the searched phone number range in the **Caller**, **Call destination**, or **Exchange line** columns (e. g. **00** for international calls).

Name

Only messages/faxes are displayed which contain the searched name in the **Caller** or **Exchange line** columns.

Name fragment

Only messages/faxes are displayed which contain the text with the searched character string in the **Caller** or **Exchange line** columns.

Note: Filters configured under **Filter** will not be considered. All messages corresponding to the search term will be shown in any case.

Note: During the fax transfer the status of the transfer when calling the pages is displayed under **Options**. It is not updated automatically. Click **Refresh** to reload the page and update the data displayed.

Note: If the fax could not be transferred, is displayed under **Options**. Moreover, a status message will be output.

	Opens or saves the message/fax.
	Sends the message/fax via e-mail. If the configurations for e-mail transfer have not yet been completed (e. g.missing message recipient), will be displayed and the message/fax will not be sent via e-mail.
Delete	Deletes the entries marked in the list.
Read	Entries marked in the list are set to read.
Unread	Entries marked in the list are set to unread.
Copy/Move	Moves, copies or archives the messages/faxes.

Administration

Dealer and Operator Data
Activation
Access Authorisations
Data Backup
Firmware update and restart
Date and time of the PBX
IP Configuration
VoIP Configuration
Remote Configuration
SIPS/SRTP Internal (VoIP)
VoIP Internet Gateway
IP Block- and Allowlist
Certificates
E-Mail
System Messages
Logging
Service Data

Dealer and Operator Data

Management of the PBX, Dealer and Operator Data

Management of the PBX, Dealer and Operator Data

Administrator Access
PBX Data
Dealer Data
Operator and Location of the PBX

Administrator Access

Proceed as follows:

- Using the Configuration Manager (page **Administration > Dealer and operator data**)

Administrator access > Change

Requirements:

- Knowledge of the current admin password

The user data for the administrator access are used for administrator registration on the PBX.

Note: Changes at the administrator access are only possible with an encrypted HTTPS connection.

Note: After a change, a new regisration is required.

Admin user name

Consists of up to 32 characters containing digits, letters (upper and lower case, but no umlauts and ß).

Change admin PIN

Enables the input option (6 characters (digits)) for **Admin PIN** and **Repeat admin PIN** (e. g. for access to COMfortel Set).

Note: All PINs in the PBX are unambiguous. Therefore, you cannot assign the same PIN twice.

Note: Avoid using PINs that can easily be found out (e. g. birthdays, specific dates, 123456, etc.)

Change admin password

Enables the input option **Admin password** and **Repeat admin password** (8-32 characters containing digits, letters (upper and lower case, but no umlauts and ß), special characters **- _ . ! ~ * ' ( ) & = + $,**).

Note: A coloured bar under the entry field shows the security level of the password (red = low level, green = high level).

Note: Passwords with a low security level cannot be saved.

Caution: A high level does not necessarily indicate a secure password as this can also be reached by entering names and birth dates in connection with special characters. However, names and birth dates should not be used as passwords.

- To create a secure password you can consult, for example, the guidelines of the German Federal Office for Information Security (BSI, in German).

Please confirm your changes with the current admin password

Protection against unauthorised changes.

PBX Data

Proceed as follows:

- Using the Configuration Manager (page **Administration > Dealer and operator data**)

PBX name

For identification of the PBX. It is displayed in the status line and the PBX data overview (click on the logo).

PBX description

This is used to enter any additional PBX information that may be required.

Dealer Data

Proceed as follows:

- Using the Configuration Manager (page **Administration > Dealer and operator data**)

Company ... Homepage

Data of the specialised dealer or qualified electrician who commissioned the PBX and is servicing it. They appear in the overview when you click the logo.

Operator and Location of the PBX

Proceed as follows:

- Using the Configuration Manager (page **Administration > Dealer and operator data**)

Company ... Coverage

Location data of the PBX. Appears on the overview when you click on the logo.

Activation

Management Activation

Management Activation

If you log in to the Online Shop, the upgrading status of the PBX is recognised on basis of the serial number, you will be presented with the possible upgrade options.

Note: You can carry out the expansion activation in the online shop or have your dealer do it for you.

Requirements:

- A user account in the Online Shop

NUMBER OF ENABLED FUNCTIONS

Overview of the enabled functions.

ACTIVATION CODE FOR EXTENDED FUNCTIONS

Enter activation/voucher code

Important: When you change a voucher code into an activation code, the right of withdrawal expires before expiry of the withdrawal period.

Enter and confirm an activation or voucher code.

Open Voucher Center

Opens the Voucher Center, to change a voucher by entering the voucher code and the serial number.

Purchase voucher code | Open Shop

Opens the online shop.

Refresh

The overview of the enabled functions is refreshed.

Access Authorisations

Configuring Access Authorisation

Configuring Access Authorisation

Proceed as follows:

- Using the Configuration Manager (page **Administration > Access authorisations**)

1. Click next to the required page to list all subpages.
2. Select or clear all check boxes for the pages to which sub-admin and user shall have access.

Data Backup

Management of the Configuration Data

Management of the Configuration Data

Store Configuration Data in the PBX
Read Configuration Data from the PBX
Reset Configuration
Regenerate Configuration
Restore old Configuration Data
Insert Restoration Point

Store Configuration Data in the PBX

Proceed as follows:

- Using the Configuration Manager (page **Administration > Data backup**)
- For automatic upload: Using the PBX

Store in the PBX

Important: Saving an old configuration overwrites the current configuration data.

Note: A configuration file that was created with an older firmware version, may possibly not contain all of the configuration data (for example, data for new functions that were added by a firmware update).

Select and import a configuration file (*.cfg) that is stored on a (PC) data storage device. The following data packages can be selected for import:

- **Apply configuration data** (cannot be edited)
- **Save network settings**
- **Apply PBX data**
- **Transfer music on hold and announcements**
- **Contact images**
- **COMfortel IP Editor**

Read Configuration Data from the PBX

Proceed as follows:

- Using the Configuration Manager (page **Administration > Data backup**)
- For automatic upload: Using the PBX

Read from PBX

Important: Note the following legal notes before saving the data backup.

The data backup includes, amongst others, the following legally especially protected components:

- Configuration of the PBX
- Names of internal subscribers
- Passwords of PBX, subscribers, VoIP accounts and VoIP subscribers
- Address books of the PBX
- Connection data sets with destination, source, and time information of a telecommunication connection

Note: Before the data backup is being saved, all subscribers to the PBX must be informed that these contents will be forwarded. They must also be informed about who will use the data received.

Note: It is absolutely necessary to store the data safely and in accordance with the legal requirements.

Note: If you want to make the data backup available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

The configuration is saved to the (PC) data storage device as configuration file (*.cfg) The following data packages can be selected for export:

- **Read configuration data** (cannot be edited)
- **Read network settings**
- **Read PBX data**
- **Read music on hold and announcements**
- **Contact images**
- **COMfortel IP Editor**

Reset Configuration

Proceed as follows:

- Using the Configuration Manager (page **Administration > Data backup**)
- For automatic upload: Using the PBX

Reset configuration

When resetting the configuration, the following parameters are maintained in contrast to a reset to the ex-factory state via the buttons:

- Admin user name, admin PIN and admin password
- IP address settings
- Web server port setting
- Time
- NTP server settings
- Dealer and operator data
- Saved hold music and announcements
- Call data
- System messages
- Restoration points

Regenerate Configuration

Proceed as follows:

- Using the Configuration Manager (page **Administration > Data backup**)
- For automatic upload: Using the PBX

Regenerate

When regenerating the configuration, the configuration data is read out of the PBX. The database is reset, and the configuration data is restored to the PBX once more. This eliminates internal errors the configuration data may contain. The configuration of the PBX remains unchangend.

Restore old Configuration Data

Proceed as follows:

- Using the Configuration Manager (page **Administration > Data backup**)
- For automatic upload: Using the PBX

Configuration file

Requirements:

- Set restoration point

Select the configuration file to be restored.

Restore network settings

Enables the restoration of the network settings.

Restore

Important: Restoring an old configuration overwrites the current configuration data.

Note: A configuration file that was created with an older firmware version, may possibly not contain all of the configuration data (for example, data for new functions that were added by a firmware update).

Requirements:

- Selected configuration file

The configuration data is restored at the restoration point. The PBX restarts.

Insert Restoration Point

Proceed as follows:

- Using the Configuration Manager (page **Administration > Data backup**)
- For automatic upload: Using the PBX

Insert restoration point

If a restoration point for the configuration data is set, the current configuration data is saved in a configuration file in the PBX. The configuration files are archived with information about the time and date they were created. They are available in the **Configuration file** list field under **RESTORE OLD CONFIGURATION DATA**.

A maximum of ten restoration points are set. If an eleventh restoration point is set, the oldest of the current restoration points is overwritten.

Note: A restoration point is automatically inserted at every firmware update.

Firmware update and restart

Management of the Firmware Updates

Management of the Firmware Updates

Store the Firmware in the PBX (Manual Firmware Update)
Firmware Update via Update Server
Firmware Update for VoIP System Phones
Restore Firmware (Reboot)
Reset of the PBX (Restart)
Power down of the PBX

Store the Firmware in the PBX (Manual Firmware Update)

Proceed as follows:

- Using the Configuration Manager (page **Administration > Firmwareupdate/restart**)
- To restore firmware, to restart and (only COMmander 6000/R/RX) to power down the PBX: By clicking a button on the PBX

Start firmware update

Requirements:

- A firmware file (*.rom) saved on a data storage device (PC)

Note: You can find a new firmware file on the Internet (see Auerswald: Auerswald Service Portal, FONtevo: FONtevo Support). Alternatively, you can get a new firmware file from your specialised dealer.

A firmware file (*.rom) is saved in the PBX via the Configuration Manager and the firmware is enabled after a reboot.

Automatic restart

Automatically restarts the PBX after a click on **Import** and enables the firmware. The step **Initiate reboot** will be skipped.

Initiate reboot

Restarts the PBX and enables the firmware.

Select and import a firmware file (*.rom) that is stored on a (PC) data storage device.

Firmware Update via Update Server

Proceed as follows:

- Using the Configuration Manager (page **Administration > Firmwareupdate/restart**)
- To restore firmware, to restart and (only COMmander 6000/R/RX) to power down the PBX: By clicking a button on the PBX

Address of the update server

Address of the update server that consists of four blocks with three digits for IPv4 addresses or eight blocks with four digits or characters for IPv6 addresses.

Note: The factory settings are already configured with the address of the update server (www.auerswald-update.de).

Update

Requirements:

- PBX connected to broadband Internet access
- Configured address of the update server
- Configured DNS server and, if necessary, an HTTP proxy server (page **Administration > IP configuration**)
- Configured gateway (page **Administration > IP configuration**)

When an instant firmware update is enabled via the update server, the firmware update is initiated automatically. A connection to the update server is set up and then the server sends the latest firmware to the PBX.

You can perform immediate firmware updates both for the PBX and for the COMfortel system phones connected to it.

PBX firmware update

ISDN COMfortel firmware update

Enables the immediate firmware update of the PBX and/or all connected ISDN COMfortel system phones.

Interval Period hr

Requirements:

- PBX connected to broadband Internet access
- Configured address of the update server
- Configured DNS server and, if necessary, an HTTP proxy server (page **Administration > IP configuration**)

If the regular firmware update option is enabled, you can specify the frequency and time period in which the system is to look for new firmware. The firmware update is then triggered automatically once a connection to the update server has been set up and new firmware has been found.

ISDN COMfortel firmware update

Enables the regular firmware update of all connected ISDN COMfortel system phones.

Update status

If an update of the PBX has been initiated from the update server, you will find the following information :

- Update status: information about the update that is currently being performed. An error message appears if the update encounters a problem
- Date of the last update attempt
- Date of the last successful update

Only the update status is displayed (during the update) for system phone updates.

Note: When you open the page it first shows the status, which is automatically updated every 30 seconds.

Search online for firmware updates

If searching for available firmware updates is switched on, the PBX searches the update server for available firmware updates. Their availability is signalled by a system message when you log on to the Configuration Manager (as administrator) or when you open the page **Administration > Firmware update and restart** (as administrator or sub-administrator).

Requirements to carry out a search:

- PBX connected to broadband Internet access
- Configured address of the update server
- Configured gateway

Firmware Update for VoIP System Phones

Proceed as follows:

- Using the Configuration Manager (page **Administration > Firmwareupdate/restart**)
- To restore firmware, to restart and (only COMmander 6000/R/RX) to power down the PBX: By clicking a button on the PBX

If COMfortel 1400 IP/2600 IP and/oder COMfortel 3600 IP system phones have been configured at the phones itself as firmware-distributing phones, you will find the following information here:

- IPv4 address of the distributing phone
- Version number of the current firmware stored in this phone

Restore Firmware (Reboot)

Proceed as follows:

- Using the Configuration Manager (page **Administration > Firmwareupdate/restart**)
- To restore firmware, to restart and (only COMmander 6000/R/RX) to power down the PBX: By clicking a button on the PBX

Restore firmware

Requirements:

- Firmware update carried out

The PBX retains not only the current firmware but also the firmware previously used in the flash memory. If necessary, the old firmware can be restored.

Caution: The PBX reboots automatically after you restore the previous firmware. This interrupts all the connections with the web server, all the calls and the data services. Unsaved data gets lost!

Note: The previous firmware will only be retained in the flash memory for four weeks. After four weeks, the previous firmware will be overwritten by the current firmware to prevent the restoration of obsolete data.

Reset of the PBX (Restart)

Proceed as follows:

- Using the Configuration Manager (page **Administration > Firmwareupdate/restart**)
- To restore firmware, to restart and (only COMmander 6000/R/RX) to power down the PBX: By clicking a button on the PBX

Restart

Requirements:

- Configured timeframe until restart

The **Power** LED remains lit in green until all of the calls are terminated or until the end of the waiting time. If all calls, all connections to/from the web server and all data services are terminated, the PBX will perform a restart (even before the defined time has elapsed). If a call (a connection) is active after the time has elapsed, this call will be interrupted to perform the restart.

The **Power** LED lights up/flashes in red and orange for up to two minutes.

If the **Power** LED lights up in green again, the restart is complete and the PBX is ready for operation.

Note: If the **Power** LED remains continuously lit in red, an error has occurred. Please contact your dealer or the manufacturer directly.

minutes

Select a timeframe until restart or power down.

Power down of the PBX

Proceed as follows:

- Using the Configuration Manager (**Administration > Firmwareupdate/restart** page)

Shutdown

As soon as the **Power** LED is red, you have 90 seconds to pull the power plug of the PBX and accessory equipment – if required – out of the socket.

If the power plug is not pulled out of the socket after this time has run out, the PBX restarts (Restart).

Date and time of the PBX

Settings of Date and time of the PBX

Settings of Date and time of the PBX

One-Time Configuration of Date and time
Automatic Synchronisation of the PBX Time/Date
Automatic Distribution of the PBX Time/Date

One-Time Configuration of Date and time

Proceed as follows:

- Using the Configuration Manager (page **Administration > Date and time of the PBX**)
- Using a character string on the internal phone (see Time and Date (of the PBX))

Manual

The time can be entered manually.

Time

Enter in format: hh: mm: ss

Date

Enter in format: JJJJ. MM. TT

Time zone

UTC

The UTC (Coordinated Universal Time - world time) is a combination of the International Atomic Time (TAI) and Universal Time (UT). It is used instead to Greenwich Mean Time (GMT).

Location name

E. g. Berlin

Set now

Take from PC

The time is took from the PC.

Time zone

UTC

The UTC (Coordinated Universal Time - world time) is a combination of the International Atomic Time (TAI) and Universal Time (UT). It is used instead to Greenwich Mean Time (GMT).

City name

E. g. Berlin

Automatic Synchronisation of the PBX Time/Date

Proceed as follows:

- Using the Configuration Manager (page **Administration > Date and time of the PBX**)
- Using a character string on the internal phone (see Time and Date (of the PBX))

From the ISDN network

Note: The network provider usually transmits a time precisely to the minute.

Note: The time is updated at most once per hour.

Requirements:

- External S_0 port (optional, see Technical Data in the Instructions)
- Disabled automatic updating of the PBX time via the network

After each ISDN call

The time transmitted by the network provider during an outgoing external call is used for setting the PBX time.

oder

Automatic ISDN call

(**no entry = first MSN/DDI entered**)

The time transmitted by the network provider during an outgoing external call is used for setting the PBX time.

From the Internet from an NTP server

Requirements:

- PBX connection to the server network
- Disabled automatic updating of the PBX time after each ISDN call

Update interval

Select one of the availbale options.

Note: The automatic updating of the PBX time via the network is performed at the selected time interval (e.g., every 5 hours).

Public time server NTP server

The PBX time will be accepted from an NTP server at the selected time interval. Select the country in the upper list field and select the NTP server in the lower entry field.

or

Manual server Port

Enter the address of the local NTP server or name of an external NTP server in the first entry field and the corresponding port in the second entry field.

Note: You should receive the address from the system administrator of the network.

Automatic Distribution of the PBX Time/Date

Proceed as follows:

- Using the Configuration Manager (page **Administration > Date and time of the PBX**)
- Using a character string on the internal phone (see Time and Date (of the PBX))

NTP server service

Requirements:

- PBX connection to the network

The PBX can be used as an NTP server by other devices in the network.

IP Configuration

Settings of the IP Configuration

Settings of the IP Configuration

Server Configuration
Ethernet Configuration
NET Module
DHCP Server Settings for NET module
DNS Configuration
IP Routing
HTTP Proxy Configuration
SFTP Server

Server Configuration

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

Port of the webserver

Port 80 is used for accessing the web server by default. The address can thus be entered as follows: http://192.168.55.13, for example. Only when accessing via Ethernet the value (e. g. 8081 instead of 80) has to be entered when modifying the port (e. g. http://192.168.55.13:8081). A change of the port should be made for security reasons especially if the PBX has to be accessible from the Internet (via a router and a fixed IP address).

You can view the ports already configured on the **Overview > Ports** page.

Caution: A port change will be made out of security reasons especially if the PBX has to be reachable from the Internet (via router and fixed IP address).

HTTPS port of the webserver

Port 443 is used for accessing the web server by default. The address can thus be entered as follows: http://192.168.55.13, for example. Only when accessing via Ethernet the value (e. g. 4432 instead of 443) has to be entered when modifying the port (e. g. https://192.168.55.13:8081). A change of the port should be made for security reasons especially if the PBX has to be accessible from the Internet (via a router and a fixed IP address).

You can view the ports already configured on the **Overview > Ports** page.

Caution: A port change will be made out of security reasons especially if the PBX has to be reachable from the Internet (via router and fixed IP address).

Force HTTPS connection

The connection is carried out via HTTPS to secure the access to the Configuration Manager. An access via HTTP will be forwarded to HTTPS.

Note: When switched off, the connection is carried out via HTTP.

Webserver access only within local network

For the increased security, the webserver access is limited to the local network.

API access only within local network

For the increased security, the API access is limited to the local network.

Increased security/Content Security Policy

The CSP header is activated. Cross-Site-Scripting attacks by foreign sources should be prevented by this.

Note: Enabling this function can lead to problems in connection with some add-ons.

Configuration port for system phones

Enter the configuration port for system phones with the range **1** to **65535**. It indicates which port on the PBX is used by the configuration software on the ISDN system phone during the transmission.

Language

Select the language.

Forced logout if no action happens after Minutes (5..60)

The time lapse with the range **5** to **60** for a forced logout defines after how many minutes a user is automatically logged out of the Configuration Manager if no action has been taken in the Configuration Manager within a configured time period. If an action happens after the automatic logout, the user will see the page for a new login.

UPnP

If the UPnP service is installed and enabled on the PC being used, the PBX is identified and selected as a device with its name in the network environment.

Ethernet Configuration

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

Receive the IP address automatically (DHCP client)

The PBX automatically receives its IP address from the DNS server.

Assigning IP address manually

Requirements:

- Disabled DHCP client

IP address

IPv4 address in the range 0-255 (example: 192.168.0.240).

Apply immediately

The IPv4 address is immediately applied and the PBX is routed to the new IPv4 address. Renewed registration required.

Apply at restart

The IPv4 address is not applied until the next time the PBX is restarted.

Subnet mask

IPv4 address of the subnet mask in the range 0-255 (example: 255.255.255.0).

Gateway

IPv4 address of the gateway in the range 0-255 (example: 192.168.0.2).

Permanent IP address from the APIPA address range

The PBX can be reached via the IP address 169.254.1.240, if

- it is directly connected to an individual computer which automatically receives its IP address.
- it is connected to a router that automatically assigns IP addresses.

IPv6 Configuration

Opens the IPv6 configuration.

IPv6 mode

If IPv6 is switched on, an IP address complying with version 6 of the Internet protocol can be assigned to the PBX. Thanks to the greater length of IPv6 addresses (eight blocks with four digits each, example: 3001:00FF:ABC0:0EAC:0001:0000:0000:000F), IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

disabled

Switches the use of IPv6 addresses off.

SLAAC

Automatically assigns an IPv6 address to the PBX.

static

Assigns a user-defined IPv6 address to the PBX.

IPv6 settings

Requirements:

- IPv6 status **IPv6 static** selected

IPv6 address

Enter the IPv6 address of the PBX.

Prefix length

Enter the prefix length in the range **1** to **128**. The prefix specifies to which subnet the IPv6 address belongs.

IPv6 Gateway

Enter the IPv6 gateway. Possible entries: Digits, characters and **:**.

NET Module

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

Requirements:

- Slot assigned with NET module (page **Hardware > Selection of modules**)

Receive the IP address automatically (DHCP client)

Note: In the factory settings, DHCP is enabled.

The PBX receives its IP address automatically from the DHCP server.

Assigning IP address manually

Requirements:

- Disabled DHCP client

IP address

Enter the IPv4 address in the range 0 to 255 (example: 192.168.0.240).

Apply immediately

The IPv4 address is immediately applied and the PBX is routed to the new IPv4 address. After the rerouting, you need to log in again.

Apply at restart

The IPv4 address is not applied until the next time the PBX is restarted.

Subnet mask

Enter the IPv4 address of the subnet mask in the range 0 to 255 (example: 255.255.255.0).

IPv6 Configuration

Opens the IPv6 configuration.

IPv6 mode

If IPv6 is switched on, an IP address complying with version 6 of the Internet protocol can be assigned to the PBX. Thanks to the greater length of IPv6 addresses (eight blocks with four digits each, example: 3001:00FF:ABC0:0EAC:0001:0000:0000:000F), IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

disabled

Switches the use of IPv6 addresses off.

SLAAC

Automatically assigns an IPv6 address to the PBX.

static

Assigns a user-defined IPv6 address to the PBX.

IPv6 settings

Requirements:

- IPv6 status **IPv6 static** selected

IPv6 address

Enter the IPv6 address of the PBX.

Prefix length

Enter the prefix length in the range **1** to **128**. The prefix specifies to which subnet the IPv6 address belongs.

IPv6 Gateway

Enter the IPv6 gateway. Possible entries: Digits, characters and **:**.

DHCP Server Settings for NET module

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

When connecting SIP devices to the 2nd network interface, it is advisable to enabled the DHCP server in order to supply the SIP devices with IP addresses and other information (NTP server address, gateway address etc.).

Use DHCP server

Requirements:

- If internal and external VoIP telephony are to be connected at the same network interface: Disabled **Receive the IP address automatically (DHCP client)** (page **Administration > IP Configuration** under NET MODULE)

Begin of DHCP address range

End of DHCP address range

Enter begin and end of the IP address range from that the IP addresses are to be assigned. Only IP addresses within the changeable range of the corresponding network class can be entered.

DHCP Configuration

Opens the DHCP configuration for importing and editing the configuration.

Import

A configuration saved on the data storage device (PC) can be selected and is saved onto the PBX. The configuration can be edited in the DHCP config editor.

Standard

The settings are reset.

DNS Configuration

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

Note: If you do not enter a DNS server, the gateway is automatically used instead.

Use DNS for NET module

Requirements:

- DHCP enabled

First DNS server Second DNS server

IPv4 address

Four blocks separated by a dot (.).

IPv6 address

Eight blocks separated by a colon (:).

Note: Click **Test connection**, to check functioning of the Internet connection. The PBX will then send a Ping command to test whether www.google.de can be reached. www.google.de has been chosen because it can usually be reached

A DNS server is necessary for Internet access. The data is specified by the Internet service provider or system administrator of the network.

Requirements:

- DHCP disabled or DHCP enabled and **Use own DNS server** enabled

Use own DNS server

Requirements:

- DHCP enabled

Despite the enabled DHCP client, the DNS server set here is used.

Note: When switched off, the PBX receives the DNS server address automatically from a DHCP server.

IP Routing

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

IP Routing Configure

IP routing is used to define routes that send IP packets to remote networks through a gateway other than the default gateway.

Requirements:

- At least one configured IAD under Administration > VoIP Internet gateway

Network address

Network or host address to send to.

Prefix length

To define the address range to be used. This applies to both IPv4 and IPv6.

Gateway

HTTP Proxy Configuration

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

HTTP-Proxy

An HTTP proxy is a proxy for the HTTP protocol, over which Internet sites are called or files downloaded.

The HTTP-Proxy is enabled and additional settings must be made:

IP address or URL | **Port**

Enter the IP address or URL:

IPv4 address

Four blocks separated by a dot (.).

IPv6 address

Eight blocks separated by a dot (.).

URL

Characters (example: www.proxyserver.de)

Enter the related port number (port) in the range **1** to **65535**.

Note: If you do not enter anything in this entry field, port 3128 will be entered automatically.

Proxy authorisation

The proxy authorisation is enabled and additional settings must be made:

User name

Enter a username for the authorisation to the HTTP proxy.

Password

Enter the related password.

SFTP Server

Proceed as follows:

- Using the Configuration Manager (page **Administration > IP Configuration**)

SFTP server

The SFTP network protocol is used for the secure data transfer between the PBX and external programmes. For this purpose, the PBX uses an SFTP server from which call data files can be downloaded.

Enable the SFTP server if you want to forward this file to external programmes (e.g. evaluation programmes).

A password is required.

8 to 32 digits:

- Digits: **0**- **9**
- Characters: **a**- **z**, **A**- **Z**, no German umlauts and ß
- Special characters **- _ . ! ~ * ' ( ) & = + $,**

Note: To change current passwords, click on **Edit passwords**.

Note: For registering at the SFTP server of the PBX an external programme requires the user name and password. The user name is predefined **netstorage** and displayed under **User name**.

Note: A coloured bar under the entry field shows the security level of the password (red = low level, green = high level).

Note: Passwords with a low security level can not be saved.

Caution: A high level does not necessarily indicate a secure password as this can also be reached by entering names and birth dates in connection with special characters. However, names and birth dates should not be used as passwords.

Caution: Visible passwords are a security risk.

- After entering, click **Hide passwords**.

VoIP Configuration

Settings for VoIP

Settings for VoIP

Presetting of the VoIP Channels
Distribution of the VoIP Channels
SIP
RTP
Quality of Service
STUN Server for the Connection of External VoIP Subscribers

Presetting of the VoIP Channels

Proceed as follows:

- Using the Configuration Manager (page **Administration > VoIP Configuration**)

Requirements:

- VoIP channels (optional, see Technical Data in the Instructions)

VoIP operating mode

Classic

This operating mode is basically selected after commissioning or after a firmware update and corresponds to the possibilities before updating to firmware 7.4A.

Mixed

Provides a maximisation of the provided VoIP channels with full availability of classic communication technologies.

Broadband (only COMpact 5500)

Priority is given the use of broadband codecs for the best possible VoIP call quality.

VoIP oriented (only COMpact 5500)

Priority is given to a higher number of available VoIP channels. Provides minimal support for classic communication technologies, for example for integrating fax machines or door terminals.

VoIP maximum

The operating mode with the highest possible number of available VoIP channels. No classic telephony is possible (analogue/ISDN).

Note: The actual available **VoIP channels**, **codecs internal**, **codecs external** and **module** are listed under the respective VoIP operating mode.

Note: The amount of codecs varies depending on the operating mode, is individually limited to provide the highest possible number of VoIP channels.

Overview of the available channels, codecs and modules

COMpact 5200/R

VoIP operating mode		Classic	VoIP-oriented	VoIP maximum
VoIP channels (max. possible)		16	28	32
Codecs internal	G.711	+	+	+
	G.729	+	–	–
	G.722	+	–	–
	G.726	+	–	–
	iLBC	+	–	–
Codecs external	G.711	+	+	+
	G.729	+	+	–
	G.722	+	–	–
	G.726	+	–	–
	iLBC	+	–	–

VoIP operating mode		Classic	VoIP-oriented	VoIP maximum
Number modules (without NET module)	1	+	+	–
	2	+	–	–
	3	+	–	–
	4	+	–	–
	5	+	–	–
NET module		+	+	+

COMpact 5500/R

VoIP operating mode		Classic	Mixed	Wideband	VoIP-oriented	VoIP maximum
VoIP channels (max. possible)		32	64	64	72	80
Codecs internal	G.711	+	+	+	+	+
	G.729	+	–	–	–	–
	G.722	+	–	+	–	–
	G.726	+	–	–	–	–
	iLBC	+	–	–	–	–
Codecs external	G.711	+	+	+	+	+
	G.729	+	+	–	+	–
	G.722	+	–	+	–	–

VoIP operating mode		Classic	Mixed	Wideband	VoIP-oriented	VoIP maximum
	G.726	+	–	–	–	–
	iLBC	+	–	–	–	–
Number modules (without NET module)	1	+	+	+	+	–
	2	+	+	–	–	–
	3	+	+	–	–	–
	4	+	+	–	–	–
	5	+	+	–	–	–
NET module		+	+	+	+	+

Missing availability of a VoIP operating mode

If an operating mode cannot be selected it is because too many modules are installed. The module is dispalyed in red or grey in the module overview (page **Hardware > Selection of modules**). A port assignment to this module is not possible.

Note: Furthermore, the relevant element in the VoIP module overview is highlighted in colour for the cause correction. For example, are due to a large number of detected extension modules certain operating modes not selectable, the cause of the error is indicated by a red displaying of the number under **Max. number modules**.

Adjustments for the achievement of a VoIP operating mode

An operating mode must always be changed manually. This does not happen automatically.

It could be necessary to remove one or more modules. In this case, first change the VoIP operating mode before changing the hardware configuration.

If a module has been added so that the current VoIP operating mode is not possible. no devices can be configured on this module.

Display of available channels

When displaying the possible channels, the configures VoIP operating mode is taken into account.

Example: 80 VoIP channels were unlocked but the operating mode **Classic** is enabled. Only the 32 VoIP channels that are possible in this operating mode are displayed.

Distribution of the VoIP Channels

Proceed as follows:

- Using the Configuration Manager (page **Administration > VoIP Configuration**)

Requirements:

- VoIP channels (optional, see Technical Data in the Instructions)

Channel reserved for voicemail/fax

Number of channels to be used for voice mail/fax and not for VoIP.

Available VoIP channels

External VoIP channels

Number of VoIP channels to be used only as external VoIP channels.

Internal VoIP channels

Number of VoIP channels to be used only as internal VoIP channels.

Available VoIP channels for dynamic assignment

Number of VoIP channels to be used as internal or external VoIP channels depending on the need.

SIP

Proceed as follows:

- Using the Configuration Manager (page **Administration > VoIP Configuration**)

Requirements:

- VoIP channels (optional, see Technical Data in the Instructions)

SIP port of the internal VoIP registrar (1 to 65535)

You will need to change this if one of the VoIP providers you use uses the default value 5060 set here.

Important: If the **SIP port of the internal VoIP registrar** is modified while it is running, all the calls will be interrupted.

Note: If the SIP port of the internal VoIP registrar is modified, the internal SIPS port of the internal VoIP registrar is also changed automatically. The internal SIPS port is calculated automatically from the **SIP port of the internal VoIP registrar** + 1).

Note: The page **Overviews > Ports** shows an overview of the PBX ports (incoming and outgoing).

Note: You must change the **SIP port of the internal VoIP registrar** in every VoIP phone that is connected to the PBX.

Network interface of the VoIP registrar

The network interface for the VoIP registrar determines what options are available for the use of two network interfaces (see 2nd Network Interface).

LAN 1

- If only one network interface is to be used.

  or:

- The external VoIP telephony is to be realised via the network interface LAN 2 (section VoIP provider at the LAN 2 connection).

LAN 2

- The external VoIP telephony is to be realised via the network interface LAN 1 (section VoIP internal at the LAN 2 connection).

  or:

- Internal and external telephony are to be realised via the network interface LAN 2 (section VoIP completely at the LAN 2 connection).

RTP

Proceed as follows:

- Using the Configuration Manager (page **Administration > VoIP Configuration**)

Requirements:

- VoIP channels (optional, see Technical Data in the Instructions)

CPU

The RTP port is a port on the local system that is used as the outgoing port for RTP transfers.

Note: The page **Overviews > Ports** shows an overview of the PBX ports (incoming and outgoing).

Quality of Service

Proceed as follows:

- Using the Configuration Manager (page **Administration > VoIP Configuration**)

Requirements:

- VoIP channels (optional, see Technical Data in the Instructions)

DiffServ

Requirements:

- Existing and enabled DiffServ support on all active network components

DiffServ is used to prioritise the handling of language packages (VoIP) and to achieve better voice quality for VoIP calls. DiffServ can be evaluated in networks that include active network components, such as routers or switches, to enable packets to be forwarded in accordance with their priority.

As proposed in RFC 4594, the DSCP values are set for VoIP calls (signalling (SIP): CS5, voice (RTP): EF).

Note: DiffServ is not usually supported by VoIP providers. As a result, this setting may not be taken into consideration under certain circumstances on the Internet.

Note: To adopt the change, reboot the PBX.

STUN Server for the Connection of External VoIP Subscribers

Proceed as follows:

- Using the Configuration Manager (page **Administration > VoIP Configuration**)

Requirements:

- VoIP channels (optional, see Technical Data in the Instructions)

STUN server

IP address or URL

Port

Requirements:

- support for public DNS mapping by the router on the PBX network
- call forwarding from the SIP port of the internal VoIP registrar to the PBX's IP address in the router
- configured DNS server

The STUN server entered here can be used to connect to external private branch exchanges.

Note: If all external private branch exchanges are connected to the PBX network via a VPN tunnel, no STUN server needs to be entered.

Note: If IP protocol IPv6 is selected, STUN cannot be used as a NAT method either for SIP or RTP.

Note: The page **Overviews > Ports** shows an overview of the PBX ports (incoming and outgoing).

Remote Configuration

Settings for the Remote Configuration

Settings for the Remote Configuration

Remote Programming
Authorisation for Remote Configuration
Remote Configuration for System Phones
Remote Configuration via Remote Access Server in the Internet

Remote Programming

Proceed as follows:

- Using the Configuration Manager (page **Administration > Remote configuration**)

ARA PIN

The ARA PIN is required for the following purposes:

- As a PIN for remote programming from an external phone (e.g. for enabling remote configuration or switching the alarm on/off)

Note: To change an existing PIN, click on **Edit PIN**.

Note: All the PINs in the PBX are unique. For this reason, you cannot assign the same PIN twice.

Note: Do not use dates of birth or dates as PINs. This makes it easy for an attacker to find out the correct PIN. Do not use PINs which are easy to guess, such as 111111 or 123456.

Caution: Visible PINs are a security risk.

- After you enter your PIN, click on **Hide PIN**.

Remote switching number Tel. no.:

Requirements:

- External numbers have been entered

---: no remote switching number.

The remote switching number is required for the following purposes:

- As a PBX phone number for remote programming from an external phone (e.g. for enabling remote configuration or switching the alarm on/off)

Note: If no remote switching number has been entered, you must dial a character string on an internal phone to unlock the PBX for remote configuration. After this, the PBX can be accessed via every associated MSN/DDI number.

Note: You cannot use call forwarding to divert this phone number.

Note: You must select the DDI number to be sent for a PTP connection from the extensions block. If the DDI number is then displayed in red text, either it is not in the value range, or the maximum number of external phone numbers has already been entered.

Authorisation for Remote Configuration

Proceed as follows:

- Using the Configuration Manager (page **Administration > Remote configuration**)

Dealer access number X

Specifies the connection from which the PBX can be configured remotely without manual release. To do this, the PC connected to the PBX via an external ISDN connection must send the dealer access number that is entered here.

Note: Enter the phone number along with its local area code.

Note: If you do not enter a dealer access number, or if the transmitted number does not match the number you enter, you must dial a character string on an internal phone to unlock the PBX for remote configuration.

Remote Configuration for System Phones

Proceed as follows:

- Using the Configuration Manager (page **Administration > Remote configuration**)

Remote switching number Tel. no.:

Requirements:

- External numbers have been entered

The system phone remote switching number is used to set up an external ISDN connection using the COMfortel Set configuration software.

Note: You cannot use call forwarding to divert this phone number.

Note: You must select the DDI number to be sent for a PTP connection from the extensions block. If the DDI number is then displayed in red text, either it is not in the value range, or the maximum number of external phone numbers has already been entered.

Remote Configuration via Remote Access Server in the Internet

Proceed as follows:

- Using the Configuration Manager (page **Administration > Remote configuration**)

Enable dealer access via Internet

Requirements:

- Configured network connection with DNS server and Gateway

The PBX can be configured remotely via the Internet with the (Auerswald) Remote Access Server (see Configuring the PBX remotely via the Internet with (Auerswald) Remote Access Server)

SIPS/SRTP Internal (VoIP)

Settings SIPS/SRTP

Settings SIPS/SRTP

SIPS/SRTP
Root Certificate
PBX Certificate

Proceed as follows:

- Using the Configuration Manager (page **Administration > Internal SIPS/SRTP (VoIP)**)

SIPS/SRTP

Proceed as follows:

- Using the Configuration Manager (page **Administration > Internal SIPS/SRTP (VoIP)**)

SIPS/SRTP

Requirements:

- internal VoIP channels that support SIPS and SRTP (optional, see Technical Data in the Instructions)
- suitable keys and certificates stored on the PBX (valid PBX certificate, private key of the PBX certificate, valid root certificate)

Internal VoIP call connections are encrypted to prevent listening in. The connection setup and termination, and also call signalling, are encrypted with SIPS. The call data is encrypted with SRTP.

Note: The certificates/keys this process requires can either be generated in the PBX or externally, and then imported from a (PC) data storage device.

Note: SIPS/SRTP must also be enabled in the VoIP phones. In addition, the certificate must be verified. For this, you may need the **Fingerprint**.

Important: SIPS/SRTP must be switched off before you can change a certificate. When switching it on again, it may happen that calls are interrupted.

Root Certificate

Proceed as follows:

- Using the Configuration Manager (page **Administration > Internal SIPS/SRTP (VoIP)**)

All properties of the certificate are displayed:

- the period of validity
- by whom and for whom the certificate has been issued
- more information (e.g. serial number and version)

Publisher

Publisher of the root certificate.

Import

Select a certificate that is stored on a data storage device and store it in the PBX.

Export

The certificate stored in the PBX is downloaded from the device and saved to a (PC) data storage device.

Renew

The certificate is renewed. This affects, among other things, the period of validity and the fingerprint of the certificate.

Delete

Deletes a certificate currently saved in the PBX after user confirmation at the prompt.

Fingerprint

Fingerprint of the root certificate. The fingerprint clearly identifies the root certificate with a numerical code.

Note: If the root certificate is renewed, a new fingerprint is generated automatically.

PBX Certificate

Proceed as follows:

- Using the Configuration Manager (page **Administration > Internal SIPS/SRTP (VoIP)**)

All properties of the certificate are displayed:

- the period of validity
- by whom and for whom the certificate has been issued
- more information (e.g. serial number and version)

Note: A pbx certificate can not be saved when a change is made, it must be renewed. This affects, among other things, the period of validity and the fingerprint of the certificate.

Owner

Owner of the pbx certificate.

Common name (CN)

Entry of an IP address or domain.

SIPS URI

URI entries.

DNS

DNS entries.

Import

Select a certificate that is stored on a data storage device and store it in the PBX.

Export

The certificate stored in the PBX is downloaded from the device and saved to a (PC) data storage device.

Renew

The certificate is renewed. This affects, among other things, the period of validity and the fingerprint of the certificate.

Delete

Deletes a certificate currently saved in the PBX after user confirmation at the prompt.

Private key

Private key is available

A private key is available in the pbx certificate.

No private key available

A private key is not available in the pbx certificate.

Note: If the pbx certificate is renewed, a new private key is generated automatically.

Import

Select a certificate that is stored on a data storage device and store it in the PBX. An existing certificate will be overwritten.

Delete

Deletes the private key currently saved in the PBX after user confirmation at the prompt.

VoIP Internet Gateway

Management VoIP Internet Gateway

Management VoIP Internet Gateway

Proceed as follows:

- Using the Configuration Manager (page **Administration > VoIP Internet Gateway**)

Requirements:

- To select LAN2: Slot assigned with NET module (page **Harwdare > Selection of modules**)

+ New

A new IAD is added to the list.

Name

Freely selectable name (max. 16 characters).

IP address

Enter the gateway address of the IAD.

Interface

Select the network interface.

Note: A default IAD with the data of the first network interface is automatically created.

IP Block- and Allowlist

Settings of the IP Block- and Allowlist

Settings of the IP Block- and Allowlist

IP Block- and Allowlist

IP Block- and Allowlist

Proceed as follows:

- Using the Configuration Manager (page (Seite **Administration > IP block- and allowlist**)

Enable IP blocklist

If traffic from a particular IP address is evaluated as being too high - and therefore probably malicious- this IP address is blocked. This means that accesses from this IP address to the PBX are initially prevented for one minute (single block by block time). The administrator is informed about the block (**Monitoring > Status report**). During the block time the PBX continues to monitor the traffic from this IP address. If the traffic from this IP address to the PBX continues to be too high, the single block time is extended (block time restarts).

If the number of attacks from IP addresses over the Internet exceeds a threshold, all traffic outside of the local network is blocked (global block). The local network is the network configured for the CPU (**Administration > IP configuration** under **ETHERNET CONFIGURATION**).

The administrator must then put in place appropriate measures to protect the PBX within its infrastructure (e.g. using routers or bridges).

Note: The global block deletes all further entries (single blocks) from the blocklist.

Note: When you open the page it first shows the status, which is automatically updated every 30 seconds.

Important: All the entries in the blocklists are deleted when you reboot the PBX.

Network-based access attempts, such as the number of data packets per second or SIP authentication failures are monitored and evaluated.

+ New

IP address

Enter a new IP address in the entry fields.

IPv4 address

Four blocks separated by a dot (.).

IPv6 address

Eight blocks separated by a colon (:).

Subnet mask

Subnet mask

Four blocks separated by a dot (.).

Prefix

Digits **1** to **128**.

Note: The subnet mask is entered automatically for a network computer when you click **Network calculator**. You can then also set the **CIDR suffix** and so the **CIDR notation**.

The values for **Host IP** appear automatically in the list.

Transfer to allowlist

Configured locks can be transferred into the allowlist.

Select the blocklist entry **Transfer**

Delete

The administrator can view automatically configured restrictions, and if necessary cancel them, in the IP blocklists.

The marked entry will be deleted.

Certificates

Management of the Certificates

Management of the Certificates

Information
Not trustworthy
Trustworthy
Certificate Authorities
Webserver
SIPS/SRTP internal (VoIP)
View

Information

When an encrypted connection (e. g. SIPS) is being set up, it is important to verify the authenticity of the server and, consequently, the identity of a web page. During setup of a connection the certificate provided by the server is checked against a root certificate stored in the PBX.

Not trustworthy

Proceed as follows:

- Using the Configuration Manager (page **Administration > Certificates**)

If checking of the server certificate against the root certificate (or stack of certificates) stored in the PBX fails, the connection will be interrupted and the certificate will be regarded as not trustworthy.

Issuer

The column displays the issuer of the certificate.

Issued on

The column displays when the certificate has been issued.

Valid until

The column displays the validity period of the certificate.

Trust certificate

The marked certificate is considered trustworthy.

Trustworthy

Proceed as follows:

- Using the Configuration Manager (page **Administration > Certificates**)

If checking of the server certificate against the root certificate (or stack of certificates) stored in the PBX does not fail, the connection will be set up and the certificate will be regarded as trustworthy.

Issuer

The column displays the issuer of the certificate.

Issued on

The column displays when the certificate has been issued.

Valid until

The column displays the validity period of the certificate.

Import

Select a certificate that is stored on a data storage device and store it in the PBX.

Certificate Authorities

Proceed as follows:

- Using the Configuration Manager (page **Administration > Certificates**)

A certificate authority is an organisation that issues digital certificates. A digital certificate serves to assign a specific public key to a person or organisation. The certificate authority certifies this assignment by adding its own digital signature.

Issuer

The column displays the issuer of the certificate.

Issued on

The column displays when the certificate has been issued.

Valid until

The column displays the validity period of the certificate.

+ New

Individual certificates are added again or all certificates are restored.

Save

The changes are saved.

Delete

Deletes a certificate currently saved in the PBX after user confirmation at the prompt.

Webserver

Proceed as follows:

- Using the Configuration Manager (page **Administration > Certificates**)

Issuer

The column displays the issuer of the certificate.

Issued on

The column displays when the certificate has been issued.

Valid until

The column displays the validity period of the certificate.

SIPS/SRTP internal (VoIP)

Proceed as follows:

- Using the Configuration Manager (page **Administration > Certificates**)

Requirements:

- internal VoIP channels that support SIPS and SRTP
- suitable keys and certificates stored on the PBX (valid PBX certificate, private key of the PBX certificate, valid root certificate)

Note: The certificates/keys this process requires can either be generated in the PBX or externally, and then imported from a (PC) data storage device.

Note: SIPS/SRTP must also be enabled in the VoIP phones. In addition, the certificate must be verified. For this, you may need the **Fingerprint**.

Important: SIPS/SRTP must be switched off before you can change a certificate. When switching it on again, it may happen that calls are interrupted.

Issuer

The column displays the issuer of the certificate.

Issued on

The column displays when the certificate has been issued.

Valid until

The column displays the validity period of the certificate.

Delete

Deletes a certificate currently saved in the PBX after user confirmation at the prompt.

View

Proceed as follows:

- Using the Configuration Manager (page **Administration > Certificates**)

All properties of the certificate are displayed:

- the period of validity
- by whom and for whom the certificate has been issued
- more information (e.g. serial number and version)

Publisher

Publisher of the root certificate.

Import

Select a certificate that is stored on a data storage device and store it in the PBX.

Export

The certificate stored in the PBX is downloaded from the device and saved to a (PC) data storage device.

Renew

The certificate is renewed. This affects, among other things, the period of validity and the fingerprint of the certificate.

Delete

Deletes a certificate currently saved in the PBX after user confirmation at the prompt.

E-Mail

Settings Outgoing Mail Server
Management Logo
Management Internationalisation
Settings Internationalisation

Settings Outgoing Mail Server

Proceed as follows:

- Using the Configuration Manager (page **Administration > E-mail > Outgoing mail server**)
- Separately fpr outgoing mail servers

Requirements:

- Knowledge of the access data for the outgoing mail server

OUTGOING MAIL SERVER (SMTP)

Select the outgoing mail server for which the settings are to be made.

+ New

A new outgoing mail server is added.

Name

Freely selectable name. This name is used to identify the outgoing mail server within the PBX and from connected system phones.

Address

Enter the address of the outgoing mail server:

IPv4 address

Four blocks separated by a dot (.).

IPv6 address

Eight blocks separated by a colon (:).

Port

Note: In the factory settings, the port is set to 25.

Requirements:

- Knowledge of the access data for the outgoing mail server

Enter the port number of the outgoing mail server between **1** and **65535**. You can obtain the port number from your e-mail provider or read it up in the Internet.

Authentication type

No authentication

Disables authentication.

User name / Password (AUTH PLAIN)

Enables authentication via password.

User name / Password (AUTH LOGIN)

Enables authentication via password.

MD5 challenge-response (AUTH CRAM-MD5)

Enables authentication via MD5 challenge-response (MD5 = Message-Digest Algorithm 5).

automatically

The authentication type is set automatically.

User name

Requirements:

- Enables authentication

Enter the username. In the most cases, this is your e-mail address.

Password

Requirements:

- Enabled authentication

Enter the password that belongs to the username.

Encryption (SSL)

If encryption is enabled, the connection of the outgoing mail server is secured using the SSL (Secure Socket Layer) protocol. The data is transferred via https.

No encryption

Switches encryption off.

Encryption (SSL) to STARTTLS

Switches encryption on after STARTTLS.

Encryption (SSL) immediately

Switches encryption immediately on.

Default outgoing mail server

The outgoing mail server first configured in the PBX is automatically switched on as **Default outgoing mail server**. To switch it off, use as **Default outgoing mail server** must be switched on for another outgoing mail server.

The selected outgoing mail server is used as default.

Note: The setting **Default outgoing mail server** is shown in gray to indicate that the outgoing mail server is now being used as default. To switch it off, use as **Default outgoing mail server** must be switched on for another outgoing mail server.

Sender

Enter the address of the e-mail sender. It is used to display it to the recipient of the e-mails.

Delete

The outgoing mail server will be deleted.

Management Logo

The PBX supports any number of logos (limited by the total memory). These can be freely assigned to individual e-mail functions of the PBX (system e-mails, forwarding of voice mail and fax messages).

Proceed as follows for configuration:

- Using the Configuration Manager (page **Administration > E-mail > Logo management**)

Import

Note: The logo must have the file format *.jpg and should have a size of 160 x 60 pixels (width x height).

Note: The file may have a size of 50 kB max. The memory for all logos is limited to 500 kB.

Note: If do not import your own logo, the Auerswald/FONtevo logo included in the scope of supply will be used.

Select and import an image file (*.jpg) that is stored on a data storage device (PC).

Delete

Deletes the entries marked in the list.

Management Internationalisation

Proceed as follows:

- Using the Configuration Manager (page **Administration > E-mail > Internationalisation**)

New

A new language file is added.

Import

Note: The name of the file must be unambiguous. It is displayed in the Configuration Manager and cannot be changed.

Note: The file structure of the imported language file must be identical to the file structure of the default language file. To this end, export a language file to your PC and edit the text fields of the language file without changing the structure of the *.xml file.

Select and import a language file (*.xml) that is stored on a data storage device (PC).

Export

The list of providers is saved to the data storage device (PC) in an *.xml file.

View

Opens the page to view the different text elements.

Configure

Opens the configuration of the user-defined language file.

Delete

Deletes the entries marked in the list.

Settings Internationalisation

Proceed as follows:

- Using the Configuration Manager (page **Administration > E-mail > Internationalisation > View**)

LANGUAGE FILE NAME - MAIL TYPE

Select a text element category

Test e-mail

Text modules to generate a test e-mail.

Mailbox: New message

Text modules to generate an e-mail with information about a new message.

Mailbox: Automatic clean-up failed

Text modules to generate an e-mail regarding failure of automatic clean-up.

Mailbox: Storage space warning

Text modules to generate an e-mail when a specified storage capacity has been exceeded.

Mailbox: Fax transmission successful

Text modules to generate an e-mail regarding successful fax transmission.

Mailbox: Fax transmission failed

Text modules to generate an e-mail regarding failure of fax transmission.

System log

Text modules to generate an e-mail with system information, malfunctions and safety warnings.

Common

General text modules.

Date and time

Text modules with variables for date and time.

- Date:

  Day: **DD**
  Month: **MM**
  Year: **YYYY**

- Time:

  Hour (24-hour format): **hh24**
  Hour (12-hour format): **hh12**
  Minute: **mm**
  Second: **ss**
  am/pm (only English language template): **ampm**

System Messages

Settings of the System Message

Settings of the System Message

Transfer of System E-Mails
Logs to be Sent as E-Mails
Events for System Logs
Status Messages and Notes

Transfer of System E-Mails

Proceed as follows:

- Using the Configuration Manager (page **Administration > System messages**)

Requirements:

- configured outgoing mail server
  - Using the Configuration Manager (page **Administration > E-mail > Outgoing Mail Server**)
- configured DNS server
  - Using the Configuration Manager (page **Administration > IP Configuration**)

Outgoing mail server

Select the outgoing mail server that will be used to send system e-mails.

Default

System e-mails are sent by the default outgoing mail server.

Send e-mail in HTML format

System e-mails are sent in HTML format.

Note: When switched off, system e-mails are sent in text format.

Language file

Select the language file whose text modules are to be used for system e-mails.

Note: If you have created under **Administration > E-mail > Internationalisation > New** a user-defined language file and select it, the e-mails will be sent with the text modules of the selected language file.

URL format for hyperlinks

Select the format for the link in system e-mails to the system message overview on the **Monitoring > System messages** page.

with IPv4 address

Link starts with https: and the PBX's IPv4 address.

with IPv6 address

Link starts with https: and the PBX's IPv6 address (if present).

User-defined

A URL prefix defined by the user is set.

URL prefix

Requirements:

- user-defined URL format

network-specific part of the URL which comes before the device-specific part of the URL. This enables you to call hyperlinks from a phone that is not on the PBX's internal network.

Example: Input **https://www.auerswald.de** for this link **https://www.auerswald.de/statics/html/index.htm#page_systemlogfiles.html**

Logo

Select the logo to be used on system e-mails.

Message recipients E-mail address 1

E-mail address to which system e-mails are sent. Up to 10 e-mail addresses can be entered here.

@ Testen

Send a test e-mail to the specified message recipients.

Logs to be Sent as E-Mails

Proceed as follows:

- Using the Configuration Manager (page **Administration > System messages**)

Requirements:

- configured outgoing mail server
  - Using the Configuration Manager (page **Administration > E-mail > Outgoing Mail Server**)
- configured DNS server
  - Using the Configuration Manager (page **Administration > IP Configuration**)

System information

System information is sent as e-mails (except for system information about the sending of system e-mails)

Note: When switched off, system information is only logged.

Simple malfunctions

Simple malfunctions are sent as e-mails.

Note: When switched off, simple malfunctions are only logged.

Serious malfunctions

Serious malfunctions are sent as e-mails.

Security warnings

Security warnings are sent as e-mails.

x Waiting time until e-mail is sent

Specifies how long (in minutes) the PBX waits after the system event has occurred until it sends an e-mail. System events that occur during the waiting time are also included in this e-mail.

Events for System Logs

Proceed as follows:

- Using the Configuration Manager (page **Administration > System messages**)

Requirements:

- configured outgoing mail server
  - Using the Configuration Manager (page **Administration > E-mail > Outgoing Mail Server**)
- configured DNS server
  - Using the Configuration Manager (page **Administration > IP Configuration**)

Temperature monitoring CPU alarm temperature

A serious malfunction is logged when the CPU temperature rises above this threshold.

Temperature monitoring Power supply alarm temperature

A serious malfunction is logged when the power supply temperature rises above this threshold.

Relay monitoring

A system information log is created when a relay is switched on or off.

Status Messages and Notes

Proceed as follows:

- Using the Configuration Manager (page **Administration > System messages**)

Requirements:

- configured outgoing mail server
  - Using the Configuration Manager (page **Administration > E-mail > Outgoing Mail Server**)
- configured DNS server
  - Using the Configuration Manager (page **Administration > IP Configuration**)

Reset hidden pop-up elements for users Execute

Click on **Do not display this notice again** to hide warnings and notes that are displayed as pop-up elements during configuration or after you register in the Configuration Manager. You can also restore the hidden pop-up elements again if required.

Logging

Settings of the Logging

Settings of the Logging

Logging

Logging

Proceed as follows:

- Using the Configuration Manager (page **Administration > Logging**)

Logging activated

Note the following legal notes before permitting D-channel output via IP.

After permission, every computer within the LAN network can record a D-channel protocol of the PBX without any further password protection.

It might be that the recorded protocol includes the following components:

- Connection data sets with destination, source, and time information of a telecommunication connection
- Configuration data of the PBX and connected phones, as well as the corresponding PINs and passwords
- Address books of the PBX

Before any recording, all participants in the telecommunication connection must be informed that these contents will be recorded and must also be informed who will use the recorded data.

If you want to make the recorded data available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

Connection data can be recorded with the D-channel decoder PC programme (D-channel output via IP) or on a syslog server. Enter **ok** in

the entry field, to accept the legal notes and then click **D-channel output via IP**.

Logging system

Connection data can be logged using the D-channel decoder PC programme or on a syslog server configured for this purpose.

Important: If **Logging enabled** is enabled and **D-channel** has been selected, anyone within the network can record a D-channel protocol over IP. A D-channel protocol also displays sensitive data, such as call connection data. That is why you should only enable the **Logging enabled** function if you need to create logs for error analysis purposes.

D-channel decoder

Install the D-channel decoder PC programme and initiate logging (see Recording the D-Channel Protocol).

Syslog server

Enter the IP address and the port of the syslog server.

Configuring the syslog server

A protocol can be recorded to log and display control and management information about the connections to the PBX e.g. to display errors. You will thus be able to see the error type, and maybe even its cause, from the list of transferred information.

Requirements:

- Syslog server configured by the system administrator
- Knowledge of the IPv4 or IPv6 address and UDP port number of the syslog server

Syslog server IP

Enter the IPv4 or IPv6 address of the syslog server.

Syslog server port

Enter the UDP port number of the syslog server.

Logging of the connection data will start as soon as you have saved your settings.

Service Data

Exporting Service Data

Exporting Service Data

Proceed as follows:

- Using the Configuration Manager (page **Administration > Service data**)

If the PBX exhibits a malfunction which is not caused by incorrect operation or external influences, the service data can be used to diagnose the malfunction.

Note the following legal notes before saving the service data.

The service data include, amongst others, the following legally especially protected components:

- Configuration of the PBX
- Names of internal subscribers
- Passwords of PBX, subscribers, VoIP accounts and VoIP subscribers
- Address books of the PBX
- Connection data sets with destination, source, and time information of a telecommunication connection

All PBX subscribers must be informed prior to storing the service data that transmission of this content takes place and by whom the data obtained is processed.

If you want to provide the service data to your specialised dealer or to the manufacturer for error analysis, you must ensure that the legal requirements are met.

Proceed as follows:

1. Accept the legal notes by entering **ok** in the entry field.
2. Click **Download service data**.

3. Save the file.

Import FTP directory

The service data is imported together with the FTP directory.

Note: If the FTP directory is too full, importing the service data may fail.

Export of Announcement Directory

The service data is exported together with the directory of announcements and music on hold.

Note: Too large amounts of data due to a larger number of WAV-files in this directory may result in a longer download. Do not choose this option, if this is not wanted.

Appendix Hardware

Status Displays on the LEDs
Buttons
Printers and Data Storage Devices at the USB Host

Status Displays on the LEDs

Status LED:	User prompt:
Flashes red (50/50 ms)	Pressing the **Software default** button is detected.
Flashes red (0,5/0,5 s)	The **Software default** button was pressed long enough.
Flashes orange (50/50 ms)	Pressing the **Software altern.** button is detected.
Flashes orange (0,5/0,5 s)	The **Software altern.** button was pressed long enough.

Status LED:	Power LED:	PBX status (after restart):
Lit in red	Lit in red	The kernel is being loaded from the partition.
Lit in orange	Lit in red	The kernel is being initialised and the partition is checked.
Lit in green	Lit in red	The system is being initialised.
Lit in green	Lit in orange	The kernel modules are being loaded.
Lit in green	Flashes orange	The data base is being initialised/converted.
Lit in green	Lit in green	The PBX is ready for operation.

Status LED:	Power LED:	PBX status (in the case of an error):
Flashes red (1/1 s)	Lit in red	No valid boot partition was found. System is paused.

Status LED:	Power LED:	PBX status (in the case of an error):
Flashes orange (1/1 s)	Lit in red	In both boot partitions no constant kernel data were found. System is paused.

Note: If the **Power** LED remains constantly lit in red even after a restart, an error has occured. Please contact your dealer or the manufacturer directly.

Status LED:	Power LED:	PBX status (during powering down):
Lit in green	Flashes green	The delayed powering down has been enabled and the PBX waits for the termination of the last call.
Lit in green	Lit in red	The PBX is powered down completely.

LAN LED (COMpact 5200):	Status of the Ethernet Interface:
Off	No Ethernet connection to a computer/network exists.
Lit in orange	An Ethernet connection to a computer/network exists (link).
Flashes orange	An Ethernet connection to a computer/network exists (link) and packets are being transmitted.

LAN-Link LED (COMpact 5020R/5500R):	Status of the Ethernet Interface:
Off	No Ethernet connection to a computer/network exists.

LAN-Link LED (COMpact 5020R/5500R):	Status of the Ethernet Interface:
Lit in green	An Ethernet connection to a computer/network exists (link).

LAN-Activity LED (COMpact 5200R/5500R):	Transfer Activities on the Ethernet Interface:
Off	No transfer activity on the Ethernet interface.
Flashes orange	Packets are being transmitted.

Buttons

Information
Performing a Restart Immediately by Pressing a Button
Restoring Firmware at the Touch of a Button
Restore Factory Settings at the Touch of a Button

Information

On the CPU module of the PBX, the various buttons are located to e. g. reboot the PBX, to restore the previous firmware or to set the PBX to factory settings.

Performing a Restart Immediately by Pressing a Button

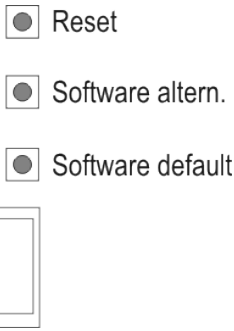
Warning: Touching voltage-carrying conductors or phone connections can cause life-threatening electric shocks.

- For some installation and maintenance work, it is necessary to open the PBX while it is in operation (qualified electrician only). Make sure that the PBX is never left unattended while working with an open casing. Close the casing after completing this work.
- Only a qualified electrician may perform installation work within an open casing or service work using the buttons inside the casing. If necessary, commission a specialised dealer to perform this work.

Warning: Power surges, which can occur during electrical storms, can cause life-threatening electric shocks and damage or destroy the PBX.

- Do not open the PBX during an electrical storm.

1. Press the **Reset** button on the CPU module.



The **Power** LED lights up/flashes in red and orange for up to two minutes.

When the **Power** LED lights up in green again, the restart is complete and the PBX is ready for operation.

Note: If the **Power** LED remains continuously lit in red, an error has occurred. Please contact your dealer or the manufacturer directly.

Restoring Firmware at the Touch of a Button

Requirements:

- Firmware update, carried out

Warning: Touching voltage-carrying conductors or phone connections can cause life-threatening electric shocks.

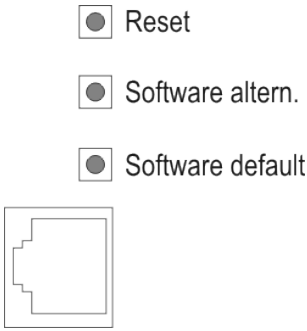
- For some installation and maintenance work, it is necessary to open the PBX while it is in operation (qualified electrician only). Make sure that the PBX is never left unattended while working with an open casing.
- Only a qualified electrician may perform installation work within an open casing or service work using the buttons inside the casing.

Warning: Power surges, which can occur during electrical storms, can cause life-threatening electric shocks and damage or destroy the PBX.

- Do not open the PBX during an electrical storm.

Note: The previous firmware will only be retained in the flash memory for four weeks. After four weeks, the previous firmware will be overwritten by the current firmware to prevent the restoration of obsolete data.

1. Press the **Reset** button on the CPU module.



2. Within the space of two seconds, push the **Software altern.** button on the CPU module and hold it.

   The **Status** LED (LED at the centre) starts to quickly blink in orange.

   After a few seconds, the blinking of the **Status** LED will slow down.

3. Let go of the **Software altern.** button.

   The Firmware is restored.

Restore Factory Settings at the Touch of a Button

If the factory settings on the PBX are restored, all of the settings are deleted or reset to the original settings before the device was first configured.

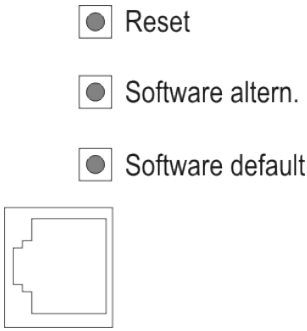
Warning: Touching voltage-carrying conductors or phone connections can cause life-threatening electric shocks.

- For some installation and maintenance work, it is necessary to open the PBX while it is in operation (qualified electrician only). Make sure that the PBX is never left unattended while working with an open casing. Close the casing after completing this work.
- Only a qualified electrician may perform installation work within an open casing or service work using the buttons inside the casing. If necessary, commission an authorised dealer to perform this work.

Warning: Power surges, which can occur during electrical storms, can cause life-threatening electric shocks and damage or destroy the PBX.

- Do not open the PBX during an electrical storm.

1. Press the **Reset** button on the CPU module.



2. Press the **Software default** button within 2 seconds and hold it for 10-15 seconds.

   The **Power** LED starts to quickly blink in red.

   After 5 seconds the LED **Status** starts to blink slowly.

3. Release the **Software default** button.

Note: If the **Power** LED remains continuously lit in red, an error has occurred. Please contact your dealer or the manufacturer directly.

Note: If you release the **Software default** button too late (the **Power** LED is no longer flashing), restart the PBX without making any changes.

Printers and Data Storage Devices at the USB Host

Connection possibilities:

- USB memory stick
- USB printer

Important: The data storage devices used require an EXT4 formatting. For read access to USB memory sticks, for example for automatic transfer of a configuration, FAT or FAT32 formatting is also supported.

Note: The connection of a USB hub is not supported.

Note: You can use any printer with at least PCL4 emulation and USB port. PCL6 or PCL5c emulation is downward compatible and includes PCL4.

Glossary

Technical Terms and Functions
Abbreviations

Technical Terms and Functions

A
B
C
D
E
F
G
H
I
J
L
M
N
O
P
Q
R
S
T
U
V
W
Z

A

Account
Exchange Line
APIPA
(Auerswald) Remote Access Server
Remote Extension

Account

Authorisation to access a VoIP provider or a VoIP PBX. Users must identify themselves by logging in with a username and a password.

Exchange Line

General term for a connection that links the PBX to the public switched phone network. The exchange line joins the provider‘s local exchange and the phone connection.

APIPA

Automatic Private IP Addressing - allows network interfaces to be configured automatically. A device that supports APIPA allocates itself a random IP address from the range 169.254.1.0 to 169.254.254.255 or has a permanent IP address from the APIPA address range.

(Auerswald) Remote Access Server

(Auerswald) Remote Access Server - for remote configuration of the PBX via user interface and Internet access (Auerswald) Remote Access.

Remote Extension

Internal subscriber (e.g., home office, outlying parts of the building, branch offices) connected to a central phone system via a public network (e.g., the Internet). The distance between the phone system and the outlying extension is not relevant.

B

B-Channel

B-Channel

Bearer channel) - ISDN channel for the transportation of user data (e. g. speech) for direct communication between subscribers.

C

Dial-around
CD
Client
CLIP No Screening
Codec

Dial-around

Service that allows a subscriber to make calls via another provider than the provider of the phone connection. In this way, the subscriber can use other - often less expensive - providers.

CD

Call Deflection - refers to call forwarding during the ringing period. If Call Deflection has been activated, the called subscriber can deflect the call on a case-to-case basis while the phone is still ringing.

Client

Piece of computer hardware or software that accesses a service made available by a server.

CLIP No Screening

Phone service for outgoing calls that allows the party called to see a customer-defined phone number instead of the phone number assigned by the network provider.

In this context, no screening means that the customer-defined number is not checked by the public exchange against the assigned numbers. It may contain any phone number you define (e.g., a service phone number).

Codec

Method to encode (digitise) analogue voice data for transmission and decode it again to convert it into back into voice. There are various codecs that feature different voice data compression rates thereby require different band widths for data transmission. The equality of VoIP calls is dependent on the codec used.

D

DDI
DECT
DHCP
DiffServ
D-Channel
DNS
Domain
DSL
DynDNS

DDI

Direct Dial In - extension number for dialling directly to a specific subscriber on a phone system. Example: In the number 0 53 06 / 92 00 - 700, 700 is the Direct Dial In number (DDI) for the Auerswald info-line.

DECT

Digital Enhanced Cordless Telecommunications (previously Digital European Cordless Telephony) - a standard for digital cordless phones and data transmission (also see GAP). Auerswald DECT systems provide encrypted transmission between mobile handset and base station.

DHCP

The IP address is temporarily allocated for a certain amount of time. If the address is no longer required by the client, the server has access to it again and can allocate it to another client.

DiffServ

Differentiated Services Flag - flag evaluated by active network components such as routers or switches in networks in order to forward packets according to their priority. This is necessary, for example, to give voice packets (VoIP) priority and to achieve better voice quality.

D-Channel

Delta channel - ISDN channel for the transportation of signalling information between end device and public exchange.

DNS

Domain Name Service - needed to translate Internet addresses. The name of a computer on the Internet (for example, Auerswald) is assigned to the corresponding IP address. This service is provided by DNS servers at the various Internet service providers or by upper domain servers.

Domain

Globally unique name of a website, consisting of third-level domain (e. g. the service name “www“), second-level domain (e. b. “auerswald“) and top-level domain (e. g. the country code “uk“). The domain is part of the URL.

DSL

Digital Subscriber Line - Internet access technology using a digital transmission standard that provides wide bandwidths for the transmission of data via telecommunication networks.

DynDNS

DynDNS services make it possible to access an Internet connection despite a dynamic IP address. To do this, the service provider assigns the changing IP address of an Internet connection to a permanent domain name (e.g., YourName.DynDNS.org).

E

Echo Cancellation
End Device
Ethernet

Echo Cancellation

Local echos and reverberation effects are compensated using echo cancellation. For example, an echo is generated on the receiver of a phone; the voice of the caller emanates from the loud speakers and reaches the microphone where it is then transmitted back.

End Device

Device that can be operated on a communication network or on a PBX, e.g. phone, fax machine, answering machine etc.

Ethernet

Network system with a speed of 10/100/1000 Mbit/s that was developed by the companies INTEL, DEC and Xerox.

F

Firewall
Firmware
Flash

Firewall

Network security component that uses a set of security rules to protect a computer network or an individual computer against unauthorised access to or from the network.

Firmware

Operating software on the phone or PBX that is stored in the device itself. The device cannot function without firmware.

Flash

Storage device (e. g. a USB memory stick) for the non-volatile storage of data within a minimum of space without permanent power supply.

G

GAP
Prefix
Gateway

GAP

Generic Access Profile - transmission protocol for cordless phones. Describes the basic functions for the communciation of DECT devices from different manufacturers among themselves.

Prefix

Number range containing the prefix or all of the prefixes from the area that should be collected for the network (e.g., local phone network, regional network, international network).

Gateway

PC or router used as an intermediary between two networks. The Internet service provider is the gateway for direct Internet dial-up connections. If you use a router, this is the gateway in a local network.

H

Host
HTTP Proxy
HTTPS
Hyperlink

Host

Component of a data processing system that manages larger application programs and data volumes and makes them available to lower-order servers and clients.

HTTP Proxy

Proxy for the HTTP protocol, over which Internet sites are called or files downloaded.

HTTPS

Hypertext Transfer Protocol Secure - communication protocol for tap-proof data transfer with encryption within the World Wide Web.

Hyperlink

Electronic cross reference, e. g. inside an Internet document, to a defined location within the same or a different document.

I

IAD
IP Address
IPv4 Address
IPv6 Address
ISDN
PD

IAD

The IAD (Integrated Access Device) is a device for realising an Internet connection. Usually, this device is provided by the Internet provider.

IP Address

Unique numerical address within a TCP/IP network that is assigned to one device and ensures that data packages reach the correct recipient.

IPv4 Address

Certain ranges are reserved for the operation of local networks:

- Class A: 10.x.x.x (for networks with up to 16.5 million PCs)
- Class B: 172.16.x.x to 172.31.x.x (for networks with up to 65534 PCs)
- Class C: 192.168.0.x to 192.168.255.x (for networks with up to 254 PCs)

These addresses have no validity on the Internet; this means that data packets with this sender type or recipient address cannot be transported on the Internet. However, they can be used in local networks without restriction. The advantage of this is that if data from a local network configured in this way comes in contact with the Internet, none of the data on the computers in the local network can leak out or be accessed externally.

IPv6 Address

IPv4 addresses comprise eight blocks with four digits or characters each that are separated by a colon, for example, 3001:00FF:ABC0:0EAC:0001:0000:0000:000F, or, abbreviated, 3001:FF:ABC0:EAC:1::F.

Thanks to the greater length of IPv6 addresses, IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

ISDN

Integrated Services Digital Network) - digital network that combines the different communication services, e. g. telephony, telefax groups 2 and 3, data transmission, etc. Unlike analogue networks, ISDN uses digital transmission.

PD

Pulse dialling - an older dialling method that transmits the dialled number to the exchange by interrupting the current in the A-wire of the phone connection.

J

Jitter Buffer

Jitter Buffer

Buffers RTP packages, to compensate irregular transfer rates of the Internet.

A high jitter buffer reduces short interruptions during the call. In return the delay of the RTP packages increases.

L

LAN
LDAP
LED

LAN

Local Area Network - connects computers over company or home networks.

LDAP

Lightweight Directory Access Protocol) - enables you to query the data in a directory service (a hierarchical database) over the network.

LED

Light emitting diode - electronic device similar to a lamp and used for displays (also multi-coloured).

M

MAC Address
PTMP
DTMF
MSN

MAC Address

Media Access Control address - specific, unique hardware address specified for each individual single network card used in the whole world. This means that the network card is unique and can be controlled and detected detected by the network without any problems.The IP addresses on the TCP/IP network are always mapped to the respective MAC address.

MAC addresses are represented in a byte-by-byte format, with the individual bytes being separated by colons or hyphens, e. g. 00:09:52:04:4B:13.

PTMP

Point-to-Multipoint - ISDN connection with 3 to 10 independent phone numbers (MSNs). They can be assigned to one or more devices, including PBXs.

DTMF

Dual Tone Multi Frequency dialling - dialling method that transmits the dialled number to the exchange as two superimposed sinusoidal signals.

MSN

Multiple Subscriber Numbers - multiple phone numbers where an ISDN PTMP connection can be reached. The MSN is the part of the phone number without a local prefix.

N

NAT
Extension
Network
NTP

NAT

Network Address Translation - permits the mapping of a specific IP address used within a network to another IP address used by another network. This function is e. g. performed by a router which connects a local area network to the Internet.

Extension

Internal connections of a PBX together with the devices, e. g. phones and telefaxes, connected to them.

Network

System of more than one computer and other communication devices that enable multiple users to access common resources such as files, printers, etc.

NTP

If a local NTP server is available in a local network, it can be selected, or a public NTP server from a public list on the Internet can be used.

O

Outbound Proxy

Outbound Proxy

Intermediate server that processes all VoIP requests and connections going to the provider (except registration).

P

PIN
PoE
Port
Proxy Server

PIN

Personal Identification Number - multi-digit secret code that has to be entered to get access to services or authorisations.

PoE

PoE is described in the IEEE 802.3af standard.

Port

A single PC can simultaneously establish multiple connections and provide multiple services for other PCs. Ports are used to differentiate these connections from parallel connections. Due to a common agreement, PCs usually provide their services on ports 1 to 1023. Outgoing connections are usually established starting at port 1024. Most services that use the standard port numbers (for example, web browsers use port 80).

Proxy Server

Acts as the interface to the Internet or between two networks. A proxy server receives queries from a computer in an Intranet, for example. It then connects to the Internet (on behalf of the computer) to forward the queries and return the (if necessary, filtered) responses. To achieve this, the HTTP proxy uses the Internet's HTTP protocol.

Q

QoS
QR Code

QoS

Quality of service - service of the router or switch that provides constantly high speech quality by prioritising speech data (RTP data packages).

QR Code

Quick Reponse Code - two-dimensional code for binary representation of data in a matrix of black and white squares: .

R

Registrar
Router
Routing

Registrar

A registrar (domain name registar) is an organisation or a company that registers Internet domains.

Router

Routers connect two separate networks. This means, for example, that you can connect the local network to another LAN or a WAN. When a PC wants to send a data packet, this packet must travel over a router. The router uses the IP address to detect the network it must send the data to. In addition to connecting to networks, routers can also execute certain control functions, such as maintaining a simple firewall. In a home network, routers are usually used to connect the local network to the Internet and to enable simultaneous Internet access for multiple users.

Routing

Identification of paths for the flow of data within computer networks. Routing determines the complete path of a set of data through the network.

S

S0 Basic Access
S0 Port
S2M Primary Rate Multiplex Access
SD/SDHC/SD-XC Card
Server
SIP
SIP-DDI (SIP Trunking)
SIP Port
SMTP
Soft Call
Structured Cabling
STUN
Subnet mask
Switch
Syslog Server
System Phone

S_0 Basic Access

ISDN basic access with a maximum of two speech channels and one data channel (D-channel). A difference is made between external and internal S_0. An external S_0 port can be used as PTP or PTMP connection. COMfortel system phones, other ISDN devices and ISDN PC cards are connected to the internal S_0 port. The internal bus acts as PTMP connection.

S_0 Port

ISDN port with two payload channels and a data channel. Two simultaneous connections are possible on an S_0 port.

S_{2M} Primary Rate Multiplex Access

A S_{2M} primary rate multiplex access provides the PBX with up to 30 speech channels (B-channels) and one data channel (D-channel). It is designed as a PTP connection and can be operated in combination with ISDN basic access connections (S_0 ports).

SD/SDHC/SD-XC Card

Secure, digital memory cards that use flash-type storage (non-volatile storage). Their special shape protects them from wrong insertion into the card slot. A lateral device for write protection (sliding switch) protects them from accidental deletion of data. SD cards can store up to 2 GB (SD 1.1 standard) whereas SDHC cards can store up to 32 GB (SD 2.0 standard). SC-XC cards can store from 32 GB to - theoretically - 2.048 GB (SD 3.0 standard). Information on recommended memory cards can be found in the Internet (see Auerswald: Auerswald Service Portal, FONtevo: FONtevo).

Server

Computer or a software that fulfils different tasks as part of the network. It provides, for example, other users (clients) in this network with certain information, data and services.

SIP

Session Initiation Protocol - network protocol which provides establishing a communication session between two or more subscribers. Via SIP only communication conditions are arranged. Other protocols such as RTP are responsible for actual data transmission.

SIP-DDI (SIP Trunking)

Some VoIP providers also support a VoIP PTP connection with a Direct Dial In option. The provider allocates a PBX base number with additional Direct Dial In numbers (DDI numbers), e.g., 9200–0.

SIP Port

Port of the local system used as the communication port for the SIP transfer.

SMTP

Simple Mail Transfer Protocol - network protocol used for the secure transfer of e-mails between the PBX and external mail servers. The PBX needs an external SMTP server (outgoing mail server) to send system e-mails or forward voice mail messages and fax messages.

Under **Administration > E-mail > Outgoing mail server** you can enter different outgoing mail servers. If no outgoing mail server has been entered, no e-mails can be sent.

Soft Call

With the soft call function, a user can use the Configuration Manager of the PBX to dial an external number directly from the phone book or the call data list.

Structured Cabling

Structured cabling provides a uniform cable infrastructure for different services such as speech (analogue, ISDN) and data (IP-based networks). Structured cabling is subdivided into primary, secondary and tertiary structured cabling.

STUN

Simple Transversal of UDP over NATs - used to determine the public IP address of an Internet connection. To do this, a STUN request is sent to a STUN server, which then shares its own IP address allocated by the VoIP provider with the Internet connection.

Subnet mask

The value of the subnet mask, as with IP addresses, is often given in decimal form (for example, 255.255.255.0 for IPv4 addresses).

Switch

Active network distributor that distributes data packages to a destination defined by an IP address among the different segments of a network.

Syslog Server

Records protocols with control and management information about the PBX. This information helps to identify errors and, where possible, their causes.

System Phone

Thanks to their integration into the system, system telefphones simplify the operation of the entire system and additionally offer numerous possibilities to use PBX functions and signalise their status visually and, in part, accustically.

T

TAPI
TCP
Template
PTP
T-Net Functions
Driver

TAPI

Phone Application Programming Interface - name of a standardised software interface for computer-assisted telephony (CTI). TAPI ensures that software and hardware from various manufacturers are compatible with each other.

TCP

Transmission Control Protocol - transport protocol that segments data into packets up to a specified size and reliably sends these individual data packets in the correct sequence to the recipient address. In this process, every data packet sent must be resent until it has been confirmed as arrived. In order to make sure this happens, a large amount of information is sent along with the actual payload data. Most Internet services are implemented with TCP, e.g., HTTP (WWW), SMTP/POPS (e-mail), etc.

Template

File (here: provisioning file) with default provisioning data that can be replaced by subscriberspecific provisioning data.

PTP

Point-to-Point - ISDN connection with Direct Dial In option. A provider allocates a PBX base number with additional Direct Dial In numbers (DDIs), e.g., 9200–0. PBXs are usually operated on a PTP connection.

T-Net Functions

Functions available on the analogue T-Net in Germany. There are analogue comfort phones with pre-defined function keys or menus for T-Net functions such as callback on busy, call forwarding, toggling between conversations or conference calling.

Driver

Program or software module that controls the interaction between an operating system or program and the interfaces to connected devices (keyboard, printer, monitor, etc.) or virtual devices.

U

UDP
UP0
UPnP
URL

UDP

User Datagram Protocol) - protocoll used to send data packets over non-secure communication lines without a connection. This means that successful transmission is dependent on the application and is therefore not always guaranteed. UDP itself does not verify whether data has been transmitted successfully. When a UDP packet is sent, the sender cannot assume that the packet will indeed arrive at the recipient. This particular protocol needs only a small amount of additional information, and results in a better data throughput rate in a well-functioning network, e. g., on a LAN. UDP is used by DNS servers, for example.

U_{P0}

The U_{P0} interface is used as PTMP connection to link up digital end devices via a single twin wire. System phones of the COMfortel series are directly connected to the internal U_{P0} port; other ISDN devices can be connected to the U_{P0} port the via a U_{P0}/S_0 adapter.

UPnP

UPnP (Universal Plug and Play) is a standard for networking various devices on home networks. It ensures that computers, televisions, multimedia equipment and household appliances can be automatically detected by each other without the user having to spend a lot of time configuring settings.

URL

Uniform Source Locator - complete address of a resource (e. g. a website), consisting of a scheme (e. g. „http://“) and a scheme-specific part (e. g. domain „www.auerswald.de/“ and path „en/products/pbx/home-office.html...“).

V

Public Exchange
VoIP Account
VoIP Provider
Voucher
VPN

Public Exchange

Node point on the public telecommunication network.

VoIP Account

Account configured with a VoIP provider who provides the necessary access data required for VoIP calls. For this purpose, you need to register your name and address with a provider via their web site. Finally, one or more phone numbers accessible from the public switched network and the Internet as well as an account with a user name and password are allocated. Most of the time, it only takes a few minutes to configure the registered connection which can then be used almost immediately.

VoIP Provider

Internet service provider offering internet telephony (VoIP, Voice over Internet Protocol).

Voucher

Voucher for a function release for a specific device type (e.g. automatic receptions for COMpact 4000). This is not tied to an individual serial number.

VPN

Virtual Private Network - connects remote or neighbouring networks (LAN) to the internal network using a VPN tunnel. The tunnel functions basically as a channel in which embedded data is transported from one network to the other.

W

WAN

WAN

Wide Area Network - a public network, e. g. the Internet.

Z

Certificate

Certificate

A (digital) certificate is similar to a digital passport that serves to identify the owner of the certificate. Its authenticity can be verified by cryptographic procedures. The certificate thus confirms the identity of a person or a company by means of an asymmetric key, thus ensuring that the person or company really is the person/company it claims to be. Certificates are issued by an officicial institution, the Certification Authority (CA).

Abbreviations

AOCD	Advice Of Charge During Call
AOCE	Advice Of Charge End of Call
APIPA	Automatic Private IP Addressing
ARA	Auerswald Remote Access
BLF	Busy Lamp Field
CAPI	Common ISDN Application Programming Interface
CCBS	Completion of Calls to Busy Subscriber
CCNR	Completion of Calls on No Reply
CD	Call Deflection
CD (PR)	Call Deflection (Partial Rerouting)
CFB	Call Forwarding on Busy
CFNR	Call Forwarding on No Reply
CFU	Call Forwarding Unconditional
CIDR	Classless Inter-Domain Routing
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CNG	Comfort Noise Generation
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction

COLR	Connected Line Identification Restriction
CoNDIS	Connection-Oriented Network Driver Interface Specification
CSV	Comma-Separated Values
CTI	Computer Telephony Integration
DDI	Direct Dial In
DECT	Digital Enhanced Cordless Telecommunications
DHCP	Dynamic Host Configuration Protocol
DiffServ	Differentiated Services Flag
DIP	Dual Inline Package
DNS	Domain Name Service
DSCP	Differentiated Services Code Point
DSL	Digital Subscriber Line
DSP	Digital Signal Processor
DTMF	Dual Tone Multi Frequency
DynDNS	Dynamic DNS
ECM	Error Correction Mode
EMC	Electro-Magnetic Compatibility
GAP	Generic Access Profile
GSM	Global System for Mobile communication
HLC	High Layer Capability
HTML	Hypertext Markup Language

HTTP	Hypertext Transfer Protocol
HTTPS	Hypertext Transfer Protocol Secure
IP	Internet Protocol
IPUI	International Portable User Identity
ISDN	Integrated Services Digital Network
LAN	Local Area Network
LDAP	Lightweight Directory Access Protocol
LCR	Least Cost Routing
LED	Light Emitting Diode
MAC	Media Access Control
MSN	Multiple Subscriber Number
MWI	Message Waiting Indication
NAT	Network Address Translation
NTBA	Network Termination for ISDN Basic rate Access
NTP	Network Time Protocol
NTPM	Network Termination Primary rate Multiplex acces
PBX	Private Branch Exchange
PCL	Printer Command Language
PD	Pulse Dialling
PIN	Personal Identification Number
PMP	Protected Monitoring Point

PoE	Power over Ethernet
PPP	Point-to-Point Protocol
PTMP	PTMP
PTP	PTP
QoS	Quality of Service
RJ	Registed Jack
RSS	Really Simple Syndication
RTP	Real-Time Transport Protocol
SC	Short Code (Dialling Number)
Scr.	Subscriber
SD	Secure Digital
SDHC	Secure Digital High Capacity
SFTP	Secure File Transfer Protocol
SIP	Session Initiation Protocol
SIPS	Session Initiation Protocol Secure
SMTP	Simple Mail Transfer Protocol
SRTP	Secure Real-Time Transport Protocol
STUN	Simple Transversal of UDP over NATs
SW	Software
TAPI	Phone Application Programming Interface
TCP	Transmission Control Protocol

UDP	User Datagram Protocol
UPS	Uninterrupted Power Supply
URL	Uniform Resource Locator
USB	Universal Serial Bus
UTC	Coordinated Universal Time
VMF	Voice Mail and Fax
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
WAN	Wide Area Network
XML	Extensible Markup Language
XSS	Cross-Site Scripting