Manual
be.IP swift

Advanced configuration

Legal Notice

Warranty

This publication is subject to change.

Open source software in this product

Along with other components, this product contains open source software that has been developed by third party suppliers and which is licensed under an open source software license. These open source software files are subject to copyright. For a current list of the open source software programs and the open source software licenses, go to *www.bintec-elmeg.com* .

GEMA

This product uses internal music for calls on hold for which approval from GEMA (German Society for Musical Performance and Mechanical Reproduction Rights) is not required. This has been confirmed by GEMA with the following approval certification. The approval certification can be viewed at the following web address: *www.bintec-elmeg.com* . System hold music: elmeg Song, Hold the line.

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Chapter 1 Purpose of this manual

This manual describes those menus that are accessible in the user interface via the link **Show more** and allow the setup of advanced functions. The setup by means of the **Wizards** is described in the "User Guide" manual. You can find it in the download area of your **be.IP swift**. Both setup approaches are supported by the online help of your device.

Chapter 2 Software update

The functionality of the **be.IP swift** is permanently extended. You can update the software via the **GUI**. This requires an existing internet connection. On the Home Screen there is the following menu:

Search for Updates

Here you can download firmware updates for your box.
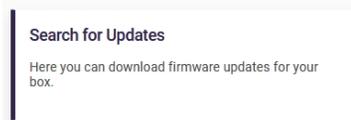
When you click on this card, your device connects to the download server and checks if an updated version of the system software is available. If this is the case, an update of your device will be offered. After installing the new software, you will be prompted to restart the device.

Caution

The update cannot be cancelled after confirming with **Start**. If an error occurs during the update, do not restart the device and contact support.

Chapter 3 Operation via the telephone

The operation and configuration of the system via a connected telephone is described in two separate documents. You will find the document as download under *http://bintec-elmeg.com*

Chapter 4   Assistants

The **Assistants** menu offers step-by-step instructions for basic configuration tasks.

Choose the corresponding task from the navigation bar and follow the instructions and explanations on the separate pages of the Wizard.

Chapter 5 Home

In the **Home** menu, you can manage global system parameters, passwords, access methods, and licenses.

5.1 System administration

5.1.1 System information

The **Information** menu contains general information on the system, the current system time, and the system runtime.

It provides you with a system status overview of the following:

5.1.1.1 System

It provides basic information about your device:

• **System**

  Device name, hardware and software version, bootloader and serial number
• **Current system time**
• **Current system runtime**
• **Language**

5.1.1.2 Internet

It provides you with basic information about your internet connection:

• **DSL Status**
• **Data rate**
• **Internet Connection**

If necessary, you can manually activate or deactivate your internet connection.

5.1.1.3 Telephony

Here you can view basic information about your SIP providers:

• **No.**

- **Description**
- **Registrar**
- **Connection type**
- **Status**

It also shows information about the ISDN interfaces:

- **Interface**
- **Function**
- **Connection information**
- **Status**

5.1.1.4 License

Here you can view license information about the software modules that are used on the device.

5.1.2 Administrative access

5.1.2.1 Configuring remote access

In this screen dialog, you can configure remote access for the **be.IP swift**. If you set the **Allow device configuration via the following IPv4 address** and/or **Allow device configuration via the following IPv6 address** option, you can begin configuring the **be.IP swift** over the internet. You can use **PC address** (IP address), **Port number**, and **Zone** to limit access for computers that are accessing remotely.

⚠️ **Caution**

If you enter *0.0.0.0* (IPv4) or *::0* (IPv6) as the IP address, you can access the **be.IP swift** over the internet from any IP address.

5.1.2.2 SSH

In the menu **SSH** (Secure Shell) you can activate the SSH service of the **be.IP swift** and establish an encrypted connection to the console from the local network, but also from the Internet. To do this, activate the switch **Enable SSH** and make sure that a user has access to the shell.

By default, the user *admin* is included in the list, but must first be enabled for access via SSH. To do this, use the ✎ button to enter the submenu **SSH Client User**. In the user set-

tings, you can also specify whether the user this is equipped with root rights and can thus make extensive changes to the system. With the button **New** you can create additional users with SSH access.

In the menu section **SSH Client IP Addresses** , you can restrict access to the Secure Shell to specific IP addresses or address ranges.

If you want to enable SSH access from the Internet, activate the switch **Enable SSH over WAN** and select the interface through which the connection is coming in for the option **Zone** .

5.2 Maintenance

5.2.1 Firmware and configuration

5.2.1.1 Updating the firmware

Note

The firmware update interrupts any ongoing telephone conversations and internet connections. The STATUS-LED blinks red during loading and during the updating process.

The firmware update takes approx. 3 minutes. Do not turn off the **be.IP swift**, disconnect the power supply, nor interrupt the DSL connection if the update process has not been completed.

If a new firmware version is available, it will be displayed in a pop-up window, and you will have the option to execute the update immediately or at a later time.

Under **Online Update**, you have the option of using the **Check** button to check for a new firmware version directly online and automatically install it.

To manually load new firmware, you must first download a new version of the firmware over the internet. Afterwards, you can select the locally saved firmware file (*.bin) using the **Browse** button. Afterwards, confirm the loading of the new firmware using the **Update** button.

5.2.1.2 Saving the configuration

Here you can save the settings that you have currently made, e.g., to the connected PC. Furthermore, you can reload saved configurations to the **be.IP swift** and thereby overwrite any improperly made configurations with a functional version.

> **Note**
>
> To **Save** and **Load** the configuration file, you can optionally enter a **Password**.

Under **Save configuration**, you can first enter a **Password**. Afterwards, click on **Save**. Depending on the browser, follow the additional steps to locally save the configuration file.

To **Load** the saved configuration, click on **Browse**. In the new dialog window, select the file and enter the **Password** (if it is saved with a password). Afterwards click on **Load**. Afterwards, the **be.IP swift** will be restarted and the **Overview** screen dialog will be displayed once the configuration process has been successful.

> **Note**
>
> Loading a new configuration interrupts all ongoing telephone conversations and internet connections. Please be aware of this before confirming with the **Load** button.

5.2.2 Reset Device

5.2.2.1 Restarting or resetting a device

Using the menu, you can restart the **be.IP swift** or reset it to factory settings.

> **Note**
>
> **Restart** and **Reset** interrupt all ongoing telephone conversations and internet connections.

Restarting device

If you restart the **be.IP swift**, all the settings and configurations are preserved.

To do so, click on the **Restart** button. The **Your device is restarting** message appears while the device is being restarted. Once the restart is complete, the **Overview** screen dialog appears. If you have configured access protection, the **Login** screen dialog appears. Now enter the login data.

Factory settings for be.IP swift

In the event of a misconfiguration, the **be.IP swift** can be reset to factory settings.

⚠️ **Caution**

All the settings that are made are lost. As described in the "Configuration" chapter, you must backup the configuration beforehand.

Note

To protect your personal data, the **be.IP swift** should be reset to factory settings if the device changes owners.

To restore the factory settings, click on the **Reset** button. The **Your device is being reset to factory settings** message appears while the device is being reset. Once the reset is complete, the **Setup Assistant** screen dialog appears. Now you have two options:

- If you would like to reload a previously saved configuration, you can cancel the **Setup assistant**.
- If you wish to reconfigure the **be.IP swift**, you must first carry out a configuration using the **Setup assistant**. Afterwards, you can set up the **be.IP swift** anew and configure it.

If it is no longer possible to access the **be.IP swift**'s user interface or if you have forgotten the password, you can restart or reset the device by pressing the reset button on the **be.IP swift**.

5.2.3 Diagnostics

5.2.3.1 System Messages

The **be.IP swift**'s system notifications are shown on this screen dialog. This information helps you to identify and diagnose irregularities in the **be.IP swift**'s behavior.

In the **Expert syslog file**, you can use the **Save** button to download a detailed system logbook.

All of the system messages and events of the **be.IP swift** are shown in the **Event log**. Service may request the **Event log** during troubleshooting. You have the option of downloading the event log using the **Save** button.

5.2.3.2 Syslog Settings

Here you can configure the functionality of your logbook. You can specify the size of the logbook and the number of log rotations to keep. Furthermore, remote logging can also be carried out. To do so, enter the IP or the host name of the log recipient into the designated column and activate remote logging.

5.2.3.3 Ping Test

This screen dialog lets you verify if a particular network client is reachable on the home network or on the internet and the time that the data packet takes to reach the target host and return.

To do so, enter the network client's IP address and click on the **Execute** button. After a brief period, the results of the ping test are displayed.

5.2.3.4 Snapshot

You can save your router's confidential configuration and status information in a file. You should preferably protect the file from unauthorized access using a password that you set.

5.2.4 Trace

5.2.4.1 Analyzing interface data

Here you can activate recording of packet statistics on any interface of the **be.IP swift** and save it to a file. To do so, select the corresponding interface in the **Connection type** drop-down list and click on **Start**. To stop the trace, click on **Stop**. You will automatically receive the option to choose the storage location as soon as traffic is sent via the selected interface. Afterwards, you can locally view and analyze the data.

5.2.4.2 VOIP/SIP Trace

The **VoIP/SIP trace** menu gives you the option of recording VoIP/SIP messages at various levels and saving them as text files on your computer. This is carried out in two ways:

- As a snapshot: In this case (*Zustand 1* and *Zustand 2*), **START** creates a one-time image that can be downloaded immediately. It is not necessary to stop the action.
- As a continuous trace: In this case (*Ereignisse 1*, *Ereignisse 2* and *SIP*), **START** begins a recording that is afterwards terminated with **STOP** before the trace file can be downloaded.
A description of the information that can be recorded will be displayed depending on your selection

Chapter 6 Telephony

6.1 System Management

The **System Management** menu contains general system information and settings.

You see a system status overview. Global system parameters such as the system name, date/time, passwords and licences are managed and the access and authentication methods are configured.

6.1.1 Global Settings

6.1.1.1 General Settings

Fields in the System Settings menu

Field	Value
Transfer Signalling	Set how the connecting to an internal subscriber shut be managed. Possible values: • *With Ringing Tone* (default value): While being transferred, the caller hears music on hold from the system; after being transferred, the ringing tone. • *With Music On Hold*: The caller hears the system's on-hold music until the targte subscriber accepts the call.
Transfer to busy extension	Set whether a caller may be transferred to a busy subscriber. The function is activated with *Enabled*. The function is disabled by default.
Rerouting to Number	Define where incoming calls should be redirected , e.g., in case of wrong dialling. Possible values: • *None – Busy Tone*: The caller hears the engaged tone by default and cannot be redirected to a destination. • *<Extension number>*: The incoming call is routed to the selected extension.
Interconnect external calls	Select whether, while brokering two external subscribers, these should be connected after you hang up. The function is activated with *Enabled*. The function is disabled by default.

Country settings

Your business is an international company with subsidiaries in several countries. Despite the differences in network structure between countries, you wish to use the same system in each subsidiary. By setting the country option, the system can be adapted to the specificities of the network in the target country.

As system requirements vary from country to country, the functionality of certain performance features must be customised. Basic settings for different country options are saved in the system.

Fields in the Country Settings menu

Field	Value
Country Profile	Select the country in which the system is to be used.
	Note: This will not alter the language used in the system menu of the system telephone.
	Possible values:
	• *Deutschland* (default value)
	• *Nederland*
	• *Great Britain*
	• *België*
	• *Italia*
	• *Danmark*
	• *España*
	• *Sverige*
	• *Norge*
	• *France*
	• *Portugal*
	• *Österreich*
	• *Schweiz*
	• *Česko*
	• *Slovenija*
	• *Polska*
	• *Magyarország*
	• *Ellada*
International Prefix / Country Code	The values for **International Prefix / Country Code** are preset.
	You need this country code if, for example, you wish to automatically generate an international number under **SIP Provider**. You dial the national prefix in the customary way , e.g. 5151 909999, the system then automatically dials +495151 909999. If

Field	Value
	you do not enter the country code, this can result in wrong dial-ling; the system then dials +5151 909999. Without the entry **Generate international phone number** (in the **VoIP**->**Settings**->**SIP Provider** menu) and the **International Prefix / Country Code** the complete number including country code must always be dialled with the SIP providers.
	Note Not all SIP providers support this setting.
National Prefix / City Code	The value for the National Prefix is preset. Enter the area code for the city in which your system is in-stalled. Using **National Prefix / City Code** you can automaticalls gener-ate national numbers (see also **Generate national subscriber number** in the **VoIP**->**Settings**->**SIP Provider** menu. This area code is crucial for a point-to-point connection, as oth-erwise, automatic callback to an external number, for example, is not possible.

The **Show more** menu allows you to make the following settings:

• Telephony services

• Charge settings

• Remote Access to Phone

• Day Mode

• Night Mode

Telephony services

For various telephony functions, different services are required that enable end devices in the network to establish connections to the system via specified ports. The same applies to incoming calls from the Internet. The following services are therefore preconfigured:

• **Telephone Provisioning Service** (TCP 18443) - This service allows transmitting various settings to conncted phones and significantly simplifies their setup.

• **TAPI Service** (TCP 5002) - The TAPI service enables the connection of TAPI applica-tions from the local network. This service is disabled in the delivery state.

- **SIP Service** (UDP/TCP 5060) - This service is the central service for control of incoming and outgoing calls. If it is deactivated, telephony is no longer possible.
- **LDAP Service** (TCP 389) - The LDAP service is required to make the system phone book available.

If you do not need any of these functions, you can disable the corresponding services here. The protocol ports corresponding to the services are then closed and the system no longer accepts connections to them.

Fields in the Charge Settings menu

Field	Value
Charge Rate Factor	Enter the factor for the connection costs. The default value is *0,00*.
Currency	Enter the name of the currency, e.g., *EUR* (default value) (max 3-digit) here. This entry is only a name not taken up in any calculation of tariff unit factors. Special characters are not allowed.
Charge Information (S0 / Upn Extension)	Select the transmission method for charging information at the internal S0 bus. Possible values: • *Keypad*: Depending on the country and provider, the charging information is transmitted so as to allow direct display by the terminal. • *Functional*: Charging information is transmitted in a binary code and must first be decoded by the terminals (EURO ISDN). • *Both* (default value): Both protocols are recognised.

Fields in the Remote Access to Phone menu

Field	Value
Remote Access (e.g. Follow me, Room Monitoring)	Select whether a remote access shall be permitted on your system. The function is enabled with *Enabled* The function is disabled by default.
PIN2	Only if **Remote Access (e.g. Follow me, Room Monitoring)** is enabled.

Field	Value
	Enter the **PIN2**. The default value is *000000*. With the 6-digit **PIN2** you protect the access from outside (remote access).
	Note Change the default value of **PIN2** to allow external access.

Fields in the Day Mode menu

Field	Value
Global Rerouting	Select the call variant in the day mode that is to apply to the entire system if no particular redirect has been set up. The default value is *Variant 1*.

Night operation

You can switch the system to night operation and thus enable certain call options for team signalling, intercom signalling and redirect functions.

Advanced switching of call options is possible via a code, or the calendar configured for night operation. You configure a calendar for night operation in the **Applications**->**Calendar**->**Calendar**->**New** menu.

Fields in the Night Mode menu

Field	Value
Team Signalling	Select the call option for team signalling in night operation.
Doorcom Signalling	Select the intercom call option for intercom signalling in night operation.
Rerouting of Incoming Distribution	Select the call option for redirect to announcement in night operation.
Extension Rerouting	Select the call option for redirect to direct dial-in in night operation.

Field	Value
Global Rerouting	Select the call option for general redirect to night operation.

6.1.1.2 Timer

In the **Timer** menu, you can configure the times after which specific system features are to be switched on by default.

The menu **System Management**->**Global Settings**->**Timer** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Call Forwarding (CFNR)	Enter the time after which **Call Forwarding (CFNR)** occurs. Possible values are *1* to *99*. The default value is *15*.
Direct Call	Enter the time from when the receiver is picked up at which the configured number is dialled. You wish to configure a telephone for which a call to a specific number is set up even without entry of the number (e.g. emergency telephone). You are not at home. However, there is someone at home who needs to be able to reach you quickly and easily by telephone, if required (e.g. children or grandparents). If you have set up the "Direct Call" function for one or more telephones, the receiver of the corresponding telephone only needs to be lifted. After a period without further entries set in configuration, the system automatically dials the configured direct call number. If you do not dial within the specified period from picking up the receiver, automatic dialling is initiated. Possible values are *1* to *30*. The default value is *5*.
External Door Connections	If an intercom call is queried by an external telephone, you can set the period after which this call is disconnected here. Possible values: *infinite* *60 seconds*

Field	Description
	• *120 seconds* • *180 seconds* (default value) • *240 seconds* • *300 seconds*

Fields in the Timer Settings menu

Field	Value
Explicit Call Transfer	Enter the time after which another call or call waiting is to occur to the initiating subscriber, if the desired subscriber cannot be reached. You have forwarded a caller to another subscriber via transfer or relay. This subscriber could not be reached or is engaged. However, you wish to avoid the subscriber ending the call or being redirected by the system after time. This is achieved by an automatic callback to your phone. For calls forwarded without announcement (special transfer types, UbA), there is a callback or call waiting (if a new call is already in progress) to the initiating subscriber after the time entered here. Possible values are *10* to *179*. The default value is *30*.
Transfer to busy extension	Enter the time after which a subscriber on hold is reconnected to the exchange. The exchange wishes to transfer a call to a specific employee. However, he/she is presently on the phone. In this case, the call can be switched to the subscriber's queue. If the call is not picked up within the period defined here, the switchboard is called again. Possible values are *10* to *600*. The default value is *30*.
System Parked Enquiry	Enter the period after which an open inquiry is terminated, and the subscriber is called again or has a call waiting. You're conducting a call and wish to transfer it to a colleague. Unfortunately, you're not sure of this colleague's current whereabouts. With **System Parked Enquiry**, the subscriber is held in the system's queue. From your telephone, you can now make an announcement advising your colleague of the waiting call. Using a code on the open inquiry, your colleague can take the

Field	Value
	call from any telephone.
	If a call held in the queue is not taken by a subscriber within the period entered here, there is a callback or call waiting to the initiating subscriber.
	Possible values are *10* to *600*. The default value is *30*.

6.1.2 Access Codes

To operate certain performance features in day-to-day operations, you've used codes which you'd like to continue employing with your new system. However, other codes are configured in the basic setting for these performance features. No problem--you can individually extend the codes for individual performance features. Thus, you'll be able to continue using these performance features with your accustomed codes.

6.1.2.1 Alternative Access Codes

In the **Alternative Access Codes** menu, you configure the system's access code plan.

For certain performance features, the access numbers can be individually set in the system configuration. Here, the preset system access number is completed by a call number from the internal system dial plan. For performance features **Open inquiry** and **Bundles**, several access codes can be assigned. Operation of performance features with a modified code occurs as described for the corresponding performance feature You can either use the modified code (internal number) or the code described in the user's guide (except exchange code).

The **System Management**->**Access Codes**->**Alternative Access Codes** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Line Access Digit	Select the exchange code.
	Possible values:
	• *None*
	• *0* (default value)
	• *6*
	• *7*
	• *8*

Field	Description
	• *9*
Pick-up Group	Enter the new code for performance feature **Pick-up Group**.
Pick-up (Extension)	Enter the new code for performance feature **Pick-up (Extension)**.
Assign project codes	Enter the new access code for the **Assign project codes** feature.
Speed Dial	Enter the new access code for the **Speed Dial** feature.
Trunk Group Selection	Create the new access codes for the **Trunk Group Selection** feature. For this, first create a bundle selection by clicking **Add**, select the bundle and enter the desired code for the bundle.
System Parking (Open Enquiry)	Create the new access codes for the **System Parking (Open Enquiry)** feature. For this, first create a queue for the caller to be held by clicking **Add**, then enter the desired code for the queue. You can create up to 10 entries.

6.2 Physical Interfaces

6.2.1 ISDN Ports (PABX)

The ISDN connections of the system are as internal ISDN connections provided for connecting various ISDN terminals (system telephones, ISDN telephones, ...).

6.2.1.1 ISDN Internal

You configure your system's internal ISDN interfaces in the **Physical Interfaces**->**ISDN Ports**->**ISDN Internal** menu.

The internal ISDN interfaces are designed for connecting different kinds of ISDN terminals (system phones, ISDN phones, etc.)

Internal ISDN connections are always point-to-multipoint connections

When connecting terminals to an internal ISDN connection, please note that not all commercially available ISDN terminals can use the performance features offered by the system via your key interface.

The **Physical Interfaces**->**ISDN Ports**->**ISDN Internal** menu consists of the following fields:

Fields in the ISDN Internal menu

Field	Description
Name	Displays the designation of the ISDN interface.
Function	Displays the function of the ISDN interface. Possible value: • *S0*: Interface for ISDN S0 connection.
Default MSN	Shows whether a standard MSN has been assigned for an internal S0 bus. You cannot access configured S0 terminals via a standard MSN. As standard MSN, you can dial internal numbers that have been configured in the **Numbering**->**User Settings**->**Users** menu and which have been assigned to a terminal in the **Terminals** menu.
Status	Displays the status of the interface.

6.2.1.1.1 Edit

Choose the ✎ button to edit an entry.

The menu **Physical Interfaces**->**ISDN Ports**->**ISDN Internal**-> ✎ consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Default MSN	Dial the required extension. You can dial under the numbers that you have configured in the **Numbering**->**User Settings**->**Users**->**Numbers** menu.

Field	Description
	Possible values: • *Not configured* • *<Number>*

6.2.2 Analogue Ports

6.2.2.1 Analogue Internal (FXS)

All of your system's available analogue internal connections are displayed in the **Analogue Internal (FXS)** menu.

The menu **Physical Interfaces**->**Analogue Ports**->**Analogue Internal (FXS)** consists of the following fields:

Fields in the Analogue Internal (FXS) menu

Field	Description
Name	Displays the designation of the analogue interface.
Usage	Displays the function of the analogue interface. Possible values: • *Telephone* • *Doorcom Units* • *Multi Function Device/Telefax* • *Modem* • *Answering Machine* • *Emergency Phone* The analogue terminal's function is configured in the **Terminals**->**Other phones**->**analog** menu.
Status	Displays the status of the interface.

6.3 VoIP

Voice over IP (VoIP) uses the IP protocol for voice and video transmission.

The main difference compared with conventional telephony is that the voice information is not transmitted over a switched connection in a telephone network, but divided into data packets by the Internet protocol and these packets are then passed to the destination over undefined paths in a network. This technology uses the existing network infrastructure for voice transmission and shares this with other communication services.

6.3.1 Settings

In the **VoIP**->**Settings** menu, you set up your VoIP connections.

You can telephone over the internet using all internally connected telephones. The number of connections depends on various parameters:

- The availability of the system's free channels.
- The available bandwidth of the DSL connection.
- The configured, available SIP providers.
- The SIP-out licences that have been entered.

6.3.1.1 SIP Provider

You configure the required SIP provider in the **VoIP**->**Settings**->**SIP Provider** menu.

You modify the status of the SIP provider by pressing the ⌃ button or the ⌄ button in the **Action** column.

After about one minute, registration with the provider has taken place and the status is automatically set to ⊘ (active).

6.3.1.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **VoIP**->**Settings**->**SIP Provider**->**New** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	You can enter a designation for the SIP provider. A 20 digit al-pha-numeric sequence is possible.
Provider Status	Select whether this VoIP provider entry is enabled (*Enabled*, standard value) or not (*disabled*).

Field	Description
Access Configuration	Select which type of VoIP phonenumbers you wish to configure. Possible values: • *Single Number(s)* (default value): Enter the individual DSL phonenumbers. • *Direct Dial-In*: Enter a basic number in conjunction with an extension number block.
Authentication ID	Enter your provider's authentication ID. A 64 digit alpha-numeric sequence is possible.
Password	At this point, you can assign a password. A 64 digit alpha-numeric sequence is possible.
User Name	Enter the user name you received from your VoIP provider. A 64 digit alpha-numeric sequence is possible.
Domain	Enter a new domain name or a new IP address for the SIP proxy server. If you do not make an entry, the entry in the **Registrar** field is used. Note: Enter a name or IP address only if this is explicitly specified by the provider.

Fields in the Outgoing Signalisation Settings menu

Field	Description
Outgoing Signalisation	Select the signalling you want for outgoing calls. Possible values: • *Standard* (default value) • *Global CLIP no Screening Number* • *Individual CLIP no Screening Number* • *Fixed Out DDI* (Only for **Access Type** = *Direct Dial-In*)
Global CLIP no Screening Number	Only for **Outgoing Signalisation** *Global CLIP no Screening Number*

Field	Description
	Enter the number that is to be displayed to the called party for all external connections.

This number is not checked. |
| **Signal remote caller number** | Only for **Outgoing Signalisation** = *Global CLIP no Screening Number* and *Individual CLIP no Screening Number*

You can have the number of an external party shown if it is being signalled.

The function is enabled with *Enabled*.

The function is disabled by default. |
| **Signal fixed out number** | Only for **Outgoing Signalisation** = *Fixed Out DDI*

Enter the number that is to be displayed to the person called with any outward connection. |

Fields in the Registrar menu

Field	Description
Registrar	Enter the DNS name or IP address of the SIP server. A 26 digit alpha-numeric sequence is possible.
Registrar Port	Enter the number of the port to be used for connection to the server. The default value is *5060*. A 5 digit sequence is possible.

If you prefer a DNS SRV request instead of a DNS A record request for this provider, enter port *0* here. For connections offered by Telekom this is a required setting because the SRV entry provides additional server addresses which may provide a better service quality. SIP provider that are created with the Initial Operation menu or with the Telephony assistant automatically use the correct port number. |
| **Transport Protocol** | Select the transport protocol for the connection.

Possible values:

• *UDP* (default value) |

Field	Description
	• *TCP*
	• *TLS*
	• *Automatic* - With this setting, your device supports automatic negotiation of the protocol with your provider's servers. For this setting to work, this negotiation must also be supported by the provider.

Fields in the STUN menu

Field	Description
STUN server	Enter the name or the IP address of the STUN server. STUN = Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs) A STUN server is required to allow VoIP devices access to the internet behind an active NAT. This determines the current public IP address for the connection and uses this for remote addressing. Maximum number of characters: 32.
Port STUN server	Enter the number of the port to be used for the connection to the STUN server. The default value is *3478*. A 5 digit sequence is possible.

Fields in the Timer menu

Field	Description
Registration Timer	Enter the time in seconds within which the SIP client must re-register to prevent the connection from disconnecting automatically. The default value is *600*.

The menu **Show more** consists of the following fields:

Fields in the Proxy menu

Field	Description
Proxy	Enter the DNS name or IP address of the SIP server. A 26 digit alpha-numeric sequence is possible.

Field	Description
Proxy Port	Enter the number of the port to be used for connection to the proxy. The default value is *5060*. A 5 digit sequence is possible.
Transport Protocol	Select the transport protocol for the connection. Possible values: • *UDP* (default value) • *TCP* • *TLS* • *Automatic* - With this setting, your device supports automatic negotiation of the protocol with your provider's servers. For this setting to work, this negotiation must also be supported by the provider.

Fields in the Codec Settings menu

Field	Description
Codec Profiles	Select the location of the SIP server. Locations are defined in the**VoIP**->**Settings**->**Codec Profiles** menu. Possible values: • *System Default* (default value): The server is not operated at any defined location. • *<Codec-Profil-Name>*
Video	Select if calls between IP telephones are to support the transmission of video data. Video transmission can only be negotiated between the participants if both support this feature.
SRTP	Select if calls via this SIP provider may be secured with SRTP (Secure Real-Time Transport Protocol).
MediaSec	*MediaSec*: MediaSec regulates the protection of SIP data between the SIP server and your system. For seamless support, automatic negotiation of the transport protocol is mandatory. Fixed transport protocol settings (UDP and TCP) may cause problems during registration. Additionally, the use of SRTP must be allowed. Your VoIP provider must support MediaSec.

Fields in the menu Phone number settings

Feld	Beschreibung
Generate international phone number	If you enable this function and, under **Global Settings**, you have entered the **Country Profile** (*49* for Germany), the 0049 is generated automatically in front of the number when a number with an area code is dialled. The function is activated by selecting *Enabled*. The function is disabled by default.
Generate national sub-scriber number	If you enable this function and, under **Global Settings**, you have entered the **National Prefix / City Code** (e.g. *40* for Hamburg), the number dialed is automatically prefixed with 040. The function is activated by selecting *Enabled*. The function is disabled by default.
Substitution of International Prefix with "+"	Select whether the prefix (e. g. 00) should be replaced by + for international numbers. The function is activated by selecting *Enabled*. The function is disabled by default.
Substitution of Incoming Number Prefix	If the phone number of an incoming call is to be modified for internal forwarding, enter the number sequence to be changed into the first, and the target sequence into the second field.

Fields in the Further Settings menu

Field	Description
From Domain	Enter the SIP provider's "From Domain". It is used after the @ as sender data in the SIP header of the SIP data packages.
Number of allowed simultaneous Calls	Select the maximum number of calls that shall be simultaneously possible Please also note the settings for bandwidth management here. Possible values: • *International* (default value): An unlimited number of simultaneous calls is possible. • *1* • *2*

Field	Description
	• *3* • *4* • *5* • *10*
SIP interface binding	Select the location of the SIP server. Locations are defined in the **VoIP**->**Settings**->**Locations** menu. Possible values: • *Any Location* (default value): The server is not operated at any defined location. • *<Location Name>*
Dial End Monitoring Time	Select the time in seconds (after dialling the last digit of a call number) after which the system begins external dialling. The default value is *5*.
Call Hold inside the PBX system	The network-centric functions call hold, call switching, 3-way conference call and call waiting can be enabled by disabling the option **Call Hold inside the PBX system**. The functions are then no longer provided by the PABX, but by the public network, instead. A corresponding contract with Telekom and customer is required which incluides a bandwidth limitation (number of simultaneously usable voice channels). If a SIP provider - Telekom in particular - offers multiple SIP accounts / numbers via a single connection, this option should be deactivated for each of them. This is required to support the bandwidth limitation for several numbers via a single network connection. If an external call is held, no MoH is played through the PABX, but the public network provides MoH or an announcement for the remotely held party.
Call Forwarding extern (SIP 302)	Select whether calls are to be redirected externally with the SIP provider. The call is forwarded using SIP status code 302. The function is activated by selecting *Enabled*. The function is disabled by default.

Field	Description
SIP-Header for anonymous call signaling	In order to ensure that your system can forward anonymous calls with SIP connections you can specify in which part of the SIP header information the string "anonymous call" is is transferred. The information can be transferred in multiple parts. For most prviders you can simply keep the preconfigured setting *Privacy ID* = *Enabled* . For the service provider 1 & 1 you need to additionally enable *Privacy Header* . Possible values: • *Display* • *Users* • *Domain* • *Privacy Header* • *Privacy User* • *Privacy ID*
SIP Header Field: FROM Display	Not for **Trunk Mode** = *Off* The sender ID is placed in the "Display" field of the SIP header. Possible values: • *None* (default value): The sender ID is not sent. • *Username*: The user-configured user name is displayed. • *Caller Address*: The user-configured number the called party is displayed. • *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.
SIP Header Field: FROM User	Not for **Trunk Mode** = *Off* The sender ID is sent in the "User" field of the SIP header. Possible values: • *Username*(default value): The user-configured user name is displayed. • *Caller Address*: The user-configured number the called party is displayed. • *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.

Field	Description
SIP Header Field: P-Preferred	Not for **Trunk Mode** = *Off* The so-called "p-preferred-identity" field is added to the SIP header and contains the sender ID. Possible values: • *None* (default value): The sender ID is not sent. • *Username*: The user-configured user name is displayed. • *Caller Address*: The user-configured number the called party is displayed. • *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.
SIP Header Field: P-Asserted	Not for **Trunk Mode** = *Off* The so-called "p-asserted-identity" field is added to the SIP header and contains the sender ID. Possible values: • *None* (default value): The sender ID is not sent. • *Username*: The user-configured user name is displayed. • *Caller Address*: The user-configured number the called party is displayed. • *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.
PBX coupling	Select whether another PABX can log into your system. In this way, several PABX systems can be linked. The function is activated by selecting *Enabled*. The function is disabled by default.
SIP Registration behavior after Restart	If after registering with a provider a reset of the system should occur, for example, or a power failure, depending on the provider, another registration may prove impossible. By enabling **Clear Bindings** or **Fetch Bindings**, you can enable re-registration.
Upstreaming Device with NAT	If you enable this function, you can use a gateway with NAT and still make VoIP calls. Without this function, it may not be pos-

Field	Description
	sible to call you with VoIP if you use a gateway with NAT. The function is activated by selecting *Enabled*. The function is disabled by default.
Early media support	Select whether you'll allow exchange of voice and audio data before a receiver accepts a call. The function is enabled with *Enabled*. The function is enabled by default.
Registration type	Specify how registration and authentication at a provider are to be handled, or if they can omitted completely. In the latter case, the relevant data are sent to a particular IP address that is already known to the correspondent. Registration and authentication are not then needed and the Registration function is disabled. An example of this method is Microsoft Exchange SIP. If no registration is required, the device can send **Periodic Options** to maintain the connection. If a registration is required, it can be carried out in either of two ways: • *Single*: With this option, a single MSN is registered with the SIP provider. • *Bulk (BNC)*: With this option, a SIP Trunk (DDI) is registered with the SIP provider, i.e. several numbers are registered under a single address.
T.38 FAX support	Select whether faxes shall be transmitted with T.38. The function is enabled with *Enabled*. The function is enabled by default. If the function is disabled, faxes are transmitted with G.711.
Send SIP UPDATE	This function ensures that after a call transfer the number of the new call partner is displayed for the initial calling party.

Field	Description
	Note Note that this function is not supported by all service providers.
	Enabled activates the functions. The function is not enabled by default.
Called number re-trieved over	In some applications (especially in DDI connections) the target address of a SIP call needs to be extracted from the **Request URI**. By activating this option the address is preferably read from this field of the invite.
Check Source IP	As a response to a DNS SRV request, your SIP provider trans-mits the addresses of valid registration servers. If you activate this option, each SIP invite has its source IP checked against these valid addresses. If it does not originate from one of them, the invite is ignored. The option is not active per default.
TLS certificate check	Only for DDI / SIP trunk connections. If a connection is encryp-ted using TLS (Transport Layer Security) a validity check on the server certificate of the remote station is performed. The option is not active per default.

6.3.1.2 Locations

In the **VoIP**->**Settings**->**Locations** menu, you configure locations of the VoIP subscribers configured on your system, and define bandwidth management for the VoIP traffic.

Individual locations can be created for use of bandwidth management. A location is identi-fied by its fixed IP address, or DynDNS address, as the case applies, or on the basis of the interface to which the device is connected. The available VoIP bandwidth (up- and down-stream) can then be configured for every location.

Fields in the Registration behavior for VoIP subscribers without assigned loca-tion menu

Field	Description
Default Behavior	Specify how the system is to behave when VoIP subscribers for whom no location has been defined are being registered.

Field	Description
	Possible values: • *Registration for Private Networks Only* (default value): The VoIP subscriber is only registered if they are within the private network. • *No Registration*: The VoIP subscriber is never registered. • *Unrestricted Registration*: The VoIP subscriber is always registered.

Fiels in the Registration behavior for VoIP providers without assigned location menu

Feld	Beschreibung
Default Interface	VoIP providers that are not assigned a location require an interface for connection setup. Here you can set one of the created interfaces as default for this case.
Standard interface for VoIP backup	If you have set up a VoIP backup connection, you can specify here which interface should be used to establish the connection.

> **Note**
>
> Note that the VoIP backup setting will not be automatically reset if the corresponding Internet connection is no longer available. You can then switch it to the primary connection here.

6.3.1.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **VoIP**->**Settings**->**Locations**->**New** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter the entry description.
Parent Location	You can cascade the SIP locations at will. Define which previously-defined SIP location shall constitute the higher-level node for the SIP location to be configured here.

Field	Description
Type	Select whether the location shall be defined using IP addresses/ DNS names or interfaces. Possible values: • *Addresses* (default value): The SIP location is defined by an IP address or a DNS name. • *Interfaces*: The SIP location is defined via the available interfaces.
Addresses	Only for **Type** = *Addresses* Enter the IP addresses of the devices at the SIP locations. Click **Add** to configure new addresses. Enter the IP address or the DNS name required under **IP Address/DNS Name**. Also enter the **Netmask** required.
Interfaces	Only for **Type** = *Interfaces* Indicate the interfaces to which the devices of a SIP location are connected. Click **Add** to select the new interface. Under **Interface**, select the desired interface.
Upstream Bandwidth Limitation	Select whether the upstream bandwidth is to be restricted for this location. The bandwidth is reduced with *Enabled*. The function is disabled by default.
Maximum Upstream Bandwidth	Enter the maximum data rate in the send direction in kbits per second.
Downstream Bandwidth Limitation	Select whether the downstream bandwidth is to be restricted for this location. The bandwidth is reduced with *Enabled*.

Field	Description
	The function is disabled by default.
Maximum Downstream Bandwidth	Enter the maximum data rate in the receive direction in kbits per second.

The menu **Advanced Settings** consists of the following fields:

Fields in the DSCP menu

Field	Description
DSCP Settings for rtp Traffic	Select the type of services for RTP data (TOS, type of service). Possible values:
	• *DSCP Binary Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in binary format, 6 bit).
	• *DSCP Decimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in decimal format).
	• *DSCP Hexadecimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in hexadecimal format).
	• *TOS Binary Value*: The TOS value is specified in binary format, e.g. 00111111.
	• *TOS Decimal Value*: The TOS value is specified in decimal format, e.g. 63.
	• *TOS Hexadecimal Value*: The TOS value is specified in hexadecimal format, e.g. 3F.

6.3.1.3 Codec Profiles

In the **VoIP**->**Settings**->**Codec Profiles**, you can define the various codec profiles to control voice quality and set up specific provider-dependent default settings.

When creating codecs, please bear in mind that good voice quality requires corresponding bandwidth, which will limit the number of possible simultaneous calls. Moreover, the remote station must also support the corresponding codec selection.

6.3.1.3.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **VoIP**->**Settings**->**Codec Profiles**->**New** consists of the following fields:

Fields in the Basic Parameters menu

Field	Description
Description	Enter a description for the entry.
Codec Proposal Sequence	Choose the order in which the codecs are offered for use by the system. If the first codec cannot be used, the second is tried and so on. Possible values: • *Default* (default value): the codec in the first position in the menu will be used if possible. • *Quality*: The codecs are sorted by quality. The codec with the best quality is used if possible. • *Low Bandwidth*: The codecs are sorted by required bandwidth. If possible, the codec with the lowest bandwidth requirement is used. • *High Bandwidth*: The codecs are sorted by required bandwidth. If possible, the codec with the highest bandwidth requirement is used.
G.711 uLaw	Not for **Codec Proposal Sequence** = *default* ISDN codec with US characteristic G.711 uLaw passes audio signals in the range of 300-3500 Hz and samples them at the rate of 8,000 samples per second. At 64 kbit/s bit rate the mean opinion score (MOS) is 4,4. This audio codec uses μlaw quantization.
G.711 aLaw	Not for **Codec Proposal Sequence** = *default* ISDN codec with EU characteristic G.711 aLaw passes audio signals in the range of 300-3400 Hz and samples them at the rate of 8,000 samples per second. At 64 kbit/s bit rate the mean opinion score (MOS) is 4,4. This audio codec uses alaw quantization.
G.722	Not for **Codec Proposal Sequence** = *default* G.722 passes audio signals in the range of 50-7000 Hz and

Field	Description
	samples them at the rate of 16,000 samples per second. At 64 kbit/s bit rate the mean opinion score (MOS) is 4,5.
G.729	Not for **Codec Proposal Sequence** = *default* G.729 passes audio signals in the range of 300-2400 Hz and samples them at the rate of 8,000 samples per second. At 8 kbit/s bit rate the mean opinion score (MOS) is 3,9.
G.726 (32 Kbit/s)	Only if **Codec Proposal Sequence** is not *Default* G.726 supports bit rates of 16, 24, 32, and 40 kbit/s. G.726 achieves a Mean Opinion Score (MOS) of about 4.2 for the 40 kbit/s variant and about 3.85 for the 32 kbit/s variant.
G.726 Codec Settings	G.726 can be used with different byte orders according to ITU-T. Select the variant you want to use.
DTMF	Not for **Codec Proposal Sequence** = *default* Select whether the DTMF Outband codec is to be used. First the system attempts to use RFC 2833. If the remote terminal does not use this standard, SIP Info is used. The function is activated by selecting *Enabled*. The function is enabled by default.

6.3.1.4 Options

General VoIP settings are located in the **VoIP**->**Settings**->**Options** menu.

The **VoIP**->**Settings**->**Options** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
RTP Port	Enter the port via which the RTP data is to be transmitted. The default value is *10000*.
Client Registration Timer	Enter a default value for the period in seconds within which the SIP clients must re-register to prevent automatic disconnection.

Field	Description
	The default value is *60*.

The menu **Advanced Settings** consists of the following fields:

Fields in the Advanced Parameter menu

Field	Description
DSCP Settings for SIP Traffic	Select the Type of Service (TOS) for SIP data. Possible values: • *DSCP Binary Value* (default value): Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in binary format, 6 bit). The default value is *110000*. • *DSCP Decimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in decimal format). • *DSCP Hexadecimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in hexadecimal format). • *TOS Binary Value*: The TOS value is specified in binary format, e.g. 00111111. • *TOS Decimal Value*: The TOS value is specified in decimal format, e.g. 63. • *TOS Hexadecimal Value*: The TOS value is specified in hexadecimal format, e.g. 3F.
SIP Port	Specify the port SIP data are to be transferred through. The default value is *5060*. **Note** If you change the port during operation, the change only becomes effective after the next reboot of your device.
Client Subscription Timer	Enter a value for the amount of time in seconds after which a SIP client must have re-registered all its configured busy lamp field keys in order for the status information not to get lost.

Field	Description
	The default value is *300*.
	You can usually keep the default value, but in case you have many keys configured, it may be a good idea to increase it.

Fields in the SIP dual Stack (IPv4/IPv6) menu

Field	Description
SIP dual Stack (IPv4/IPv6)	Enable this option if you want to support IPv6 for VoIP. Both, IPv4 as well as IPv6 are supported, and if a VoIP provider supports IPv6, IPv6 is preferred. If a VoIP provides does not support IPv6, IPv4 is used.
	With selection of *Activated* the option is anabled.
	The function is not enabled by default. This means that only IPv4 is used until you enable dual stack SIP.
SIP Service	This service is the central service for controlling incoming and outgoing calls. If it is deactivated, telephony is no longer possible. If you do not need the functions, you can disable the service here . The corresponding protocol ports (UDP/TCP 5060) will then be closed and the system will no longer accept connections to them.

6.4 Numbering

6.4.1 Trunk Settings

Your system is a telecommunication installation for external connection to the Internet.

6.4.1.1 Trunks

In the menu **Numbering**->**Trunk Settings**->**Trunks** you can see the configured external connections of your system. External connections are configured in the menu **VoIP**->**Settings**->**SIP Provider** or through the respective configuration assistant.

6.4.1.2 Trunk Numbers

In the menu **Numbering**->**Trunk Settings**->**Trunk Numbers** menu, you assign the external numbers and the name indicated in a system telephone display to the external connections you've defined.

External numbers at the point-to-point connection

For a point-to-point connection, you receive a PBX number together with a 1-, 2-, 3- or 4-character extension number range. This extension number range comprises the direct dial-in numbers for the PBX connection. If you've requested several point-to-point connections, the number of extensions can be expanded, or you receive another PBX number with your own extension number range.

With a point-to-point connection, external calls are signalled to the subscriber whose assigned internal number corresponds to the dialled extension number. You configure the internal numbers to be accessed directly via the extension number range as **Internal numbers** in the **Numbering**->**User Settings**->**User**->**Add**->**Trunk Numbers**->**Internal Numbers** menu.

Example: You have a point-to-point connection with the PBX number *1234* and extension numbers from *0* to *30*. A call under *1234-22* is normally signalled at the internal subscriber with call number *22*. However, if you enter extension number *22* in this list, you can define that calls under *1234-22* are signalled at the internal subscriber by call number *321*.

External subscriber numbers at point-to-multipoint connection

For a point-to-multipoint connection, you can request up to 10 numbers (MSN, multiple subscriber number) per ISDN connection. These MSNs are the external subscriber numbers for your ISDN connections. The internal numbers are specified under **Numbering**->**User Settings**->**User**->**Add**->**Trunk Numbers**.

6.4.1.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Choose the **New** button to create new numbers.

The **Numbering**->**Trunk Settings**->**Trunk Numbers**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Trunk	Select the connection defined in **Assistants**->**PBX**->**Trunks** for

Field	Description
	which you want to configure the numbers.
Type of Number	Select the call number type to be defined according to connection type.
	Possible values:
	• *Single Number (MSN)*: Only for point-to-multipoint connections.
	• *P-P Base Number*: Only for point-to-point connections.
	• *P-P DDI Exception*: Only for point-to-point connections.
	• *P-P Additional MSN*: Only for point-to-point connections.
Displayed Name	In general, you enter the name to be displayed for this number in the called system telephone's display.
	For **Type of Number** = *P-P Base Number* this field displays the name of the connection.
Single Number (MSN)	Here, enter the MSN for a point-to-multipoint connection.
P-P Base Number	Here, enter the number for the point-to-point connection (without direct dial number).
P-P DDI Exception	Here, enter the direct dial exception for a point-to-point connection.
	Note: Only enter the extension according to your extension number range that should be routed to differing internal subscriber numbers. Direct dial at the point-to-point connection always proceeds to the subscriber whose number was dialled along as extension. E.g. the internal subscriber has the number *16*. If this subscriber is called from outside on *1234567-16*, the call is signalled at his telephone. However, if with direct dial *16* a subscriber with the number *888* is to be called, enter *888* as the exception number. In **Incoming Distribution** you then assign the exception number to the subscriber with the number *16*. You can subsequently make additional settings in **Incoming Distribution**.
P-P Additional MSN	Here, enter an additional MSN for a point-to-point connection.
	With some providers, it's possible to also transmit a point-

Field	Description
	to-multipoint number on a point-to-point connection in parallel to the direct dial number; e.g. a fax number pre-existing setup of a point-to-point connection, or the old point-to-multipoint number.

6.4.1.3 Trunk Groups

In the **Numbering**->**Trunk Settings**->**Trunk Groups** menu, you can group the various external connections and individually provide these to the users.

You wish to assign specific external connections to internal subscribers for outgoing connections. You can join these external connections together to create bundles and supply these to extensions for outgoing calls. In this way, all extensions start external dialling with the same dialling code, but can only establish a connection using the bundle released for the extension in question.

The external connections of your system can be grouped into bundles. You can configure up to 99 bundles (01 - 99). The code number for bundle assignment can be modified (menu **Alternative Access Codes**).

When initiating an external call through the bundle code number, the bundle cleared for the subscriber is used in connection setup.

Only for compact systems: A predefined entry with the parameters **Description** = *ISDN External* and **Sequence in Trunk Group** = *ISDN External* is displayed.

6.4.1.3.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create a new bundle.

The **Numbering**->**Trunk Settings**->**Trunk Group**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Sequence of Trunk Lines in Group	Select the desired external connections for a bundle. The order when dialling to the outside matches the sequence of external connections in this list. You wish to assign specific external connections for outgoing connections to the internal subscribers of your system. You can group external connections into bundles and provide these to

Field	Description
	subscribers for the outgoing dialling. In this way, all subscribers initiate the external dialling with same bundle access code, but can only set up a connection over the bundles for which they have been cleared.

6.4.2 User Settings

In this menu, you configure and administer your system's users. The users are organised into authorisation classes to which the desired external lines are assigned, and which may use performance features according to request. The user assigned to an authorisation class receives an internal number and specific authorisations. A default authorisation class (Default CoS) is preset ex-works, to which new users are automatically assigned.

After it's been defined in User Settings which functions and authorisations a user, or several users, have access to, authorisation of user settings is assigned to a terminal in **Terminals**. In this way, its possible to create settings for several terminals via an authorisation class, e.g. a user setting *Boss*, a user setting *Department Head* and a user setting *Clerk*. Now, all that's left to do is assign the corresponding terminals to one of these **Class of Service**.

6.4.2.1 Users

In the **Numbering**->**User Settings**->**Users** you configure the users of your system, their class, and assign them internal and external numbers.

You see an overview of the users that have been created. The entries in the **Name** column are sorted alphabetically. Click the column title of any other column to sort entries in ascending or descending order

Only for compact systems: The following users are predefined:

- *User 1 to User 4 analog Tel*
- *User 5 and User 6 Sys Tel*
- *User 7 DECT*
- *User 8 and User 9 ISDN*

Choose the ✏ icon to edit existing entries. Select the **New** button in order to create new users.

6.4.2.1.1 Basic Settings

In the **Numbering**->**User Settings**->**Users**->**Basic Settings** you enter basic information about the user.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Name	Enter the name of the user. This name is displayed in the phone book if you have entered a number and cleared it for the phone book under **Mobile Number Home Number**. The name is displayed with the codes (M) for mobile communication, and (H) for home number in the system telephone display.
Description	Enter additional user information.

Fields in the External Numbers menu

Field	Description
Mobile Number	Enter a number under which the user can be reached via mobile phone. Also select whether this number is to be shown in the system telephone display so that it can be dialled on the system telephone from the system phone book (**Access from system phone** option).
Home Number	Enter a number under which the user can be reached via home phone. Also select whether this number is to be shown in the system telephone display so that it can be dialled on the system telephone from the system phone book (**Access from system phone** option).
E-mail Address	Enter the e-mail address for the user.

Fields in the Class of Service menu

Field	Description
Standard	Select the classes of service = CoS. The authorisation class and the creating of new authorisation classes is done under **Numbering**->**User Settings**->**Class of Services**. Only the selecting is done in this setting.

Field	Description
	Possible values: • *Uneingeschr. AutoAmt* (default value): All connections can be established and taken, automatic outside line is enabled. • *Uneingeschränkt*: All connections can be established and taken, automatic outside line is not enabled. • *Not allowed*: No class of service • *<Authorisation class>*
Optional	Select an optional authorisation class. This CoS is required for the calendar settings. The authorisation class and the creating of new authorisation classes is done under **Numbering**->**User Settings**->**Class of Services**. Only the selecting is done in this setting. Possible values: • *Uneingeschr. AutoAmt* (default value): All connections can be established and taken, automatic outside line is enabled. • *Uneingeschränkt*: All connections can be established and taken, automatic outside line is not enabled. • *Not allowed*: No class of service • *<Authorisation class>*
Night	Select the authorisation class for night operation. This CoS is required for the calendar settings. The authorisation class and the creating of new authorisation classes is done under **Numbering**->**User Settings**->**Class of Services**. Only the selecting is done in this setting. Possible values: • *Uneingeschr. AutoAmt* (default value): All connections can be established and taken, automatic outside line is enabled. • *Uneingeschränkt*: All connections can be established and taken, automatic outside line is not enabled. • *Not allowed*: No class of service • *<Authorisation class>*

Fields in the Further Options menu

Field	Description
Busy on busy	Select whether the performance feature Busy on Busy shall be enabled for this user.
	If a user for whom more than one phone number has been set up is currently engaged, you can decide whether additional calls for this user should be signalled. If "Busy on Busy" is set for this user, other callers get the **Engaged** signal if the user is calling on one of their numbers.
	The function is activated by selecting *Enabled*.
	The function is disabled by default.

6.4.2.1.2 Numbers

The internal numbers which are later assigned to the terminals can be entered in the **Numbering**->**User Settings**->**Users**->**Numbers** menu. Depending on the type, one or more numbers can be assigned per terminal.

The **Numbering**->**User Settings**->**Users**->**Numbers** menu consists of the following fields:

Fields in the Internal Numbers menu

Field	Description
Internal Numbers	Enter the internal numbers for the user and the description to be shown in the system telephone display (**Displayed Description**). In addition, select whether this internal number shall be displayed in the **System Phonebook**, and whether the LED next to the corresponding function key (**Busy Lamp Field**) should light up.
	The functions are activated by default.
	Add new **Internal Numbers** with **Add**.
	Only for compact systems: Users with the internal numbers *10*, *11*, *12*, *13*, *20*, *21*, *22*, *30* and *35* are predefined.

6.4.2.1.3 Outgoing Signalisation

In the **Numbering**->**User Settings**->**Users**->**Outgoing Signalisation** menu, select the outgoing numbers for the user.

For an outgoing call, if the remote subscriber should not see the number assigned to your own connection, one of the existing numbers can be selected here for display. If no number is defined, the system transmits no number to the provider.

Fields in the list Outgoing Signalisation

Field	Description
Internal Number	Displays the internal numbers configured for the user.
Displayed Description	Displays, for each internal number, the description configured for the system telephone display.
Outgoing Signalisation	Select the signal you want for outgoing calls. Possible values: • *Default, own DDI Signalling*: The user's own extension is used as the **Outgoing Signalisation**. This option is available when there is a point-to-point configuration or a SIP provider with direct dialling. • *Standard*: No **Outgoing Signalisation** is sent. In this case, the switchboard uses the port's main number. • *<Fixed phone number>*: For a FXO port, the phone number configured is already assigned as the **Outgoing Signalisation** and is displayed. • *<Phone number>* : When more than one number has been configured, you can select a number that you wish to use as the **Outgoing Signalisation**.

Select the ✎ symbol to define, for every internal number (indicated in the table by **Internal Number** and **Displayed Description**), the number that should be displayed for outgoing calls. Here, for each configured external connection, select one of the numbers configured for this purpose.

If more than one external connection has been configured, you can specify the procedure for outgoing calls. When an external line is engaged, the order of the entries determines the sequence in which the other lines assigned will be used to dial.

The configured **Outgoing Signalisation** can be hidden for each outgoing line; to do so, put a tick under **Hide Number** in the relevant row.

If you wish to move an entry in the list displayed, select the ↑↓ icon in the relevant row. A new window opens.

The selected entry is displayed under **External Connection**, here e. g. *ISDN_1*.

Proceed as follows to move the selected entry:

(1) Under **Move**, select in the list the entry relative to which you wish to move the selected entry, here e. g. *1.SIP-Provider_1*.

(2) Select whether you want to insert the entry *above* or *below* the selected entry in the list, here e. g. *above*.

(3) Select **Apply**.
     The entries display in the changed order.

(4) If the list contains more than two entries, move other entries if you wish.

(5) Close the window by clicking **OK**.

The sequence configured here overwrites the setting that is assigned by the permission class. However, the assigned permission class continues to determine whether a user has access to a particular external connection.

6.4.2.1.4 Optional Rerouting

In the **Numbering**->**User Settings**->**Users**->**Optional Rerouting** menu you can assign a **rerouting application** and a **Active Variant (Day)** to each of the internal numbers displayed for an extension.

Here, for example, you can control which colleague calls should be rerouted to if you are involved in a conference, and whether the switchboard is responsible for calls during the lunch break.

Fields in the Optional Rerouting menu

Field	Description
Internal Number	Displays the internal numbers configured for the user.
Displayed Description	Displays, for each internal telephone number, the description that is configured for the display in the system telephones' display.
Rerouting Application	From the dropdown list, select the rerouting application that you want to assign to the internal number. You may select from the rerouting applications that you have configured in the **Applications**->**Rerouting**->**Rerouting Applications**->**New** menu with **Type of Rerouting Application** = *Internal Subscriber* . Possible values: • *None* (default value)

Field	Description
	• *<Type of Rerouting Application>*
Active Variant (Day)	Select the rerouting application variant that should be currently enabled. If a switch is set up for the variants via the calendar, this setting will be switched back again in a timely manner. Possible values: • *Variant 1* • *Variant 2* • *Variant 3* • *Variant 4*

6.4.2.1.5 Authorizations

In the **Numbering**->**User Settings**->**Users**->**Authorizations** menu you can allow this user to perform certain configurations himself via HTML configuration. For this, a user name and password must be entered in the user HTML configuration, and personal access author-ised. Once logged out, you can view and modify the corresponding settings after entering this user name and password.

The **Numbering**->**User Settings**->**Users**->**Authorizations** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Password for IP Phone Registration	Enter the password with which a user IP telephone must log into the system. The password can remain free if IP telephones log in but need not authenticate themselves.
PIN for Phone Access	Here you can create the PIN for access to protected functions. These functions are: • Access to the voice mailbox from a phone not assigned to the user • Access to the configuration of the system via the phone via dial codes . In the default configuration no PIN is created.

Fields in the User HTML Configuration menu

Field	Description
Personal Access	Select whether this user shall receive access authorisation to a personalised user interface where he can perform his own entries and settings. The function is activated by selecting *Enabled*. The function is disabled by default.
Login Name	Only for **Personal Access** = enabled. Enter a user name for this user. This is required for login on the user interface.
Password	Only for **Personal Access** = enabled. Enter a password for this user. This is required for login on the user interface.

Call Through

Call Through consists in dialin to the system via an external connection and the call put through from the system via another external connection.

> **Note**
>
> In the connection data records, one data record is created for the incoming connection and one for the outgoing connection.

Fields in the Further Options menu

Field	Description
Call Through	Select whether Call Through should be authorised for this user. The function is activated by selecting *Enabled*. The function is disabled by default. If you enable the function you must select, under **Use routing and signalisation from number**, the internal number from which the authorised external lines and call options for Call Through shall be used.

6.4.2.2 Class of Services

In the **Numbering**->**User Settings**->**Class of Services** (CoS) the functions and perform-ance features for the user settings are defined. These authorisation classes can then be assigned to individual users (user groups) in the user settings.

Choose the ✎ icon to edit existing entries. Choose the **New** button to create additional au-thorisation classes. The authorisation class *CoS Default* is configured by default.

6.4.2.2.1 Basic Settings

The basic settings and the name for the new CoS is specified in the **Numbering**->**User Settings**->**Class of Services**->**Basic Settings** menu. The authorisation class can be loc-ated via the name.

The **Numbering**->**User Settings**->**Class of Services**->**Basic Settings** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.

Fields in the Line Access Authorization menu

Field	Description
Line Access Authoriza-tion	Select line access authorisation for the authorisation class.
	Line access authorisation determines which calls (internal, ex-ternal,...) are allowed. The system distinguishes several author-isation levels.
	Possible values:
	• *Unlimited*: The telephones have unrestricted dialling au-thorisations and can initiate all connections.
	• *National*: The telephones can initiate all calls except inter-national calls. If a number starts with the code for international dialling, the number cannot be dialled.
	• *Incoming*: The telephones can receive incoming external calls, but cannot initiate any external calls. Internal calls are possible.
	• *Region*: The telephones cannot make any national or inter-national calls. For this dial permission, 10 exception numbers

Field	Description
	allowing national or international dialling can be configured. An exception number can consist of complete call numbers or sections thereof (e.g. the first numerals).
	• *Local*: The telephones can make local calls. National and international calls are not possible.
	• *Internal*: The telephones do not have authorisation for incoming or outgoing external calls. Only internal telephone calls are possible.
Automatic Outside Line	This setting defines whether automatic outside line is set up for this authorisation class. With automatic outside line, users of this authorisation class hear the external dialling tone after picking up the receiver and can immediately dial outside. To make internal calls, press the star key after picking up the receiver.
	Note
	If you are conducting an external call when automatic outside line access is active and then want to start a second external call, you must select a leading *0* for the second call so that it can be set up.
	If you have set up an automatic outside line for an internal subscriber, the keypad functions cannot be directly used. First disable the **Automatic Outside Line** or dial the star key, then the code for manual outside line (e. g. 0) followed by keypad dialling, beginning with the star or hash key.
Trunk Line Selection with Line Access Number	Select the connections over which outgoing calls from these telephones shall be externally routed. The order of entries determines in which sequence, in case of an engaged external line, dialling shall occur over the other assigned lines.
Allow manual trunk group selection	Besides general exchange access, a telephone can also selectively use a bundle. Here, an external connection with the corresponding code number is initiated for selective use of the bundle, and not by dialling the exchange code.
	To be able to perform a selective bundle assignment, the authorisation class must possess the appropriate authorisation. The authorisation can also include bundles that the authorisation class can otherwise not assign. If a telephone does not pos-

Field	Description
	sess the authorisation for selective bundle assignment, or if the selected bundle is in use, the busy tone is heard after dialling the code. If **Automatic outside line** is set up for an authorisation class, users of this authorisation class must press the star key before selective bundle assignment, then initiate external dialling with the code for bundle assignment.
	The function is activated by selecting *Enabled*.
	The function is disabled by default.
	Then select the bundles for which manual bundle assignment is to be allowed. You configure bundles in the **Numbering**->**Trunk Settings**->**Trunk Groups** menu.

Number display

If you call a subscriber, your number is displayed to him. The person you're calling thus sees that you are calling even before picking up the receiver. If you don't want the person you're calling to see your number before picking up the receiver, you can prevent display of your number to your called party.

If your called party has set up call forwarding, you won't know at which telephone you've reached him. In this case, you can display the number to which your called party has forwarded the call. However, the person you're calling also has the option of preventing display of this number.

Call number display allows display of the caller's number already at call signaling, even on analogue telephones. Thus, you know who wishes to speak to you even before you've accepted the call.

Note

Transmission of analogue CLIP data can be set up separately for every analogue connection. Please refer to the users' guides for your analogue terminals to determine whether these support the CLIP and CLIP off Hook performance features.

Not all described performance features are included in the ISDN standard connection. Please inquire of your network operator the extent to which individual performance features must be separately ordered for your ISDN connection.

The menu **Advanced Settings** consists of the following fields:

Fields in the Further Settings menu

Field	Description
Dial Control	Select whether the numbers entered in the **Call Routing**->**Out-going Services**->**Dial Control** menu shall be allowed or denied also for this CoS. The function is activated by selecting *Enabled*. The function is disabled by default.
Automatic Route Se-lection (ARS)	Select whether the routing rules entered in the **Call Routing**->**Automatic Route Selection** menu shall also be applied to this authorisation class. The function is activated by selecting *Enabled*. The function is disabled by default.
Show Outgoing Number (CLIP)	Select whether the caller number shall be displayed to the called party. The function is activated by selecting *Enabled*. The function is enabled by default.
Show Connected Number (COLP)	Select whether the called party number shall be displayed to the caller. If, for example, the called party has set up call forwarding to a third subscriber, the caller can display the number of the call forwarding destination using this performance feature. The function is activated by selecting *Enabled*. The function is enabled by default.
Additional Info for Ex-tern Call	Select what should be displayed for an exchange call. Possible values: • *Trunk and Number Name*: The display shows the exchange connection and the assigned name alternatively. • *Trunk Name Only*: Only the name assigned to the exchange connection is displayed. • *Number Name Only* (default value): The display shows the

Field	Description
	name assigned to the external number only.
	• *None*: Display is blank.

6.4.2.2.2 Features

Additional functions are set up in the **Numbering**->**User Settings**->**Class of Services**->**Features** menu.

Call pickup

A call is signalled to a co-worker who is presently absent from his work station. You now have two options to respond to the caller. You could walk over to your colleague's telephone, or transfer your colleague's call to your phone. Assignment is done by the option **Pick-up Group** in the menu **Features**; the group is then assigned to a user. If the values are identical, a call pickup is possible. Call pickup is not possible for open inquiry.

System telephones can pick up calls via programmed function keys. You can set up line keys, connection keys and team keys on system telephones.

- Line key: An ISDN connection or a VoIP provider is set up under a connection key. The LED assigned to the line key indicates the connection status. The LED lights up if both B channels of a connection are in use, or when the maximum number of simultaneous connections over a VoIP provider is reached. If an external call is signalled on another internal telephone, you can pick it up by pressing this line key.

- Line key: A system user is set up under a connection key. The LED assigned to the connection key indicates the subscriber status (call, connection,...). If a call is signalled for this internal subscriber, you can pick it up by pressing this connection key.

- Team key: A team key is a normal line key to which the internal number of a team is assigned. The LED assigned to the team key indicates the team status (call, connection,...). If a call is signalled for this team, you can pick it up by pressing the team key.

Call waiting

As far as possible, you want to accept calls from every customer, even while you're already on the phone. If another call is signalled to your phone by a call-waiting tone or display notification, you can decide with which of two customers you wish to speak.

If a currently engaged subscriber is called, she gets automatic call-waiting. Call-waiting is possible for internal and external calls. The call-waiting connection is signalled to the called party visually and/or acoustically, depending on the terminal.

The called party can:

- Decline the call-waiting connection and proceed with the current call. The caller is then

signalled engaged.

- Accept the call-waiting connection and hold the current connection.
- Accept the call-waiting connection after the current connection is ended.
- Ignore the call-waiting connection. Call-waiting automatically ends after 30 seconds and the caller hears a busy signal.

Analogue terminals

The call-waiting option can be individually configured for every subscriber. Allowing call waiting or not can be set via configuration or via a code number in operations.

Analogue terminals get the system call waiting tone. The number of the call-waiting party can be shown in the analogue telephone display if it features the corresponding perform-ance feature (CLIP off Hook). CLIP off Hook is disabled for analogue terminals in the basic setting, but may be enabled via configuration.

Call waiting can only occur simultaneously in the system for a limited number of analogue connections. If call waiting is already operating with this maximum number of call-waiting tones on analogue connections, additional call-waiting callers will get the busy tone.

If you hear the call-waiting tone during a call, you can take that call and transfer the ongo-ing call An operating procedure allows transfer of the ongoing call and acceptance of the call waiting. The following conditions apply here:

- Every dialled number is accepted by the system.
- After the operation procedure, the subscriber and the call-waiting subscriber are immedi-ately connected to each other (no acknowledge tones).
- Transfer to one's own number is possible, then call waiting.
- Internal, external target subscribers as well as teams can be dialled.
- A return call occurs in case of invalid or engaged target number.
- If the subscriber is free, a return call is made according to the target subscriber's defined period.
- With transfer to a team number, there is no return call in case of an engaged or unreach-able team
- With transfer to a team number only return call after time is supported.

ISDN terminals

Configuration and operation of call waiting occurs as described in the users' guides of the corresponding terminals. ISDN terminals use their own tones to signal call waiting.

Note

Call waiting is not possible:

- for conference calls
- for do not disturb (analogue terminals)
- for announcements
- for room monitoring
- for terminals in which the data protection performance feature is enabled (e.g., fax, modem)
- in analogue subscriber's dialling status (the receiver has been picked up, but there is no connection yet)
- for current call-waiting protection
- for dialling a team number. Then there is no call waiting for analogue team subscribers.

ISDN telephones can also transfer a call waiting to another subscriber via the Call Deflection performance feature. An active connection is ended by replacing the receiver, for example. The call waiting connection is then signalled and can be accepted, e.g. by picking up the receiver.

The **Numbering**->**User Settings**->**Class of Services**->**Features** menu consists of the following fields:

Fields in the Feature Authorization menu

Field	Description
Pick-up Group	Enter the number of the group in which calls may be picked up.
Call Waiting	Select whether call waiting shall be allowed for this authorisation class. The function is activated by selecting *Enabled*. The function is enabled by default.
Use global rerouting	Select whether a global redirect shall be allowed for this authorisation class. The function is activated by selecting *Enabled*. The function is disabled by default.

Field	Description
	Note The redirect target must be in a class of service that does not allow global redirect.
Switch signalling variants manually	Select whether manual switching of call options shall be allowed for this authorisation class. The function is activated by selecting *Enabled*. The function is enabled by default.
Call Through	Select whether Call Through shall be allowed for this authorisation class. The function is activated by selecting *Enabled*. The function is disabled by default.

Simplex operation

The simplex operation function allows you to set up a connection from a system telephone to another system telephone without this connection having to be actively accepted by the called system telephone (pick up receiver, switch on loudspeaker/hands-free). As soon as the system telephone has accepted the simplex operation connection, the connection is set up. The caller and the called system telephone hear an attention tone at the beginning of the simplex operation. Duration of the simplex operation is limited to two minutes. If the receiver of a concerned telephone is picked up during this period, the call is translated into a normal connection.

System telephones can initiate a simplex operation call via the system telephone menu or a programmed function key. If the simplex operation is initiated via a function key, notifications appear in the system telephone display as with a normal connection and the simplex operation key LED is switched on. The simplex operation can be ended by renewed pressing of the function key or by pressing the loudspeaker key. The LED switches off again at conclusion of the simplex operation.

If a telephone or a system telephone is the destination of a simplex operation call, the caller's number is indicated in the display. The simplex operation call is signalled over the loudspeaker with an attention tone. Simplex operation can be terminated with the ESC key.

A function key can also be configured on a system telephone to deny or allow simplex operation calls.

Note

Simplex operation calls are automatically accepted by the called telephone by enabling the hands-free function, if:

- the telephone is not in use
- simplex operation is allowed and
- the "Do not disturb" function (Call Protection) is disabled.

If a simplex operation connection is not ended by both subscribers, the connection is automatically ended by the system after ca. 2 minutes.

Message

Do you wish to call your co-workers to a meeting or to a meal? You could call each of them individually, or simply use the announcement function. With just one call, you reach all the announcement-enabled telephones without subscribers having to pick up the receiver.

Caution

Although you can be heard with the announcement, you cannot hear any comments your colleagues or family members make.

The announcement function allows you to set up a connection to another telephone without this connection having to be actively accepted by the latter (pick up receiver or switch on loudspeaker/hands-free). As soon as a telephone has accepted the announcement, the connection is active. The announcer and the called subscriber initially hear a positive acknowledge tone. Announcement duration is unlimited.

Announcements are possible to ISDN and analogue telephones if these support the announcement performance feature. Please refer to the user's guide for your telephones to determine whether the performance feature is supported.

Announcements can be allowed or denied to telephones via a code number.

System telephones

Announcement to and from system telephones is possible. System telephones can initiate an announcement via the system telephone menu or using a programmed function key. If the announcement is initiated via a function key, notifications appear in your telephone display as with a normal connection and the announcement key LED is switched on. The announcement can be ended by renewed pressing of the function key or by pressing the

loudspeaker key. The LED switches off again at conclusion of the announcement.

If a system telephone is the destination for an announcement, the number of the announcer appears on the display. The announcement is signalled with a positive acknowledge tone over the loudspeaker. The announcement can be terminated with the ESC key.

A function key with associated LED can also be set up on a system telephone to deny or allow announcements.

Individual announcement

You can initiate the announcement in a selective manner by dialling an internal number. The announcement can be allowed or denied by the destination subscriber via an operating procedure. The announcement is signalled to the destination subscriber and the announcer with a positive acknowledge tone.

Team announcement

An announcement can also be made to a team by dialling a team number. The team subscribers hear the announcement simultaneously. The announcement is signalled to the destination subscribers and the announcers with a positive acknowledge tone. The announcement to a team is also possible from an inquiry. With a team announcement, it can take up to four seconds before the connection to the individual team subscribers is established. The announcement then proceeds to the team subscribers who have accepted the announcement within this period.

Note

Announcements are automatically accepted by the called telephone by enabling the loudspeaker function, if:

* the telephone is not in use
* the announcement is set up and
* the "Do not disturb" function is not active.

MWI (Message Waiting Indication)

You've got new messages in your mailbox, or new e-mails waiting at your Internet service provider. as you have no prior knowledge, you must constantly check whether you do actually have new messages. With the MWI performance feature, your system receives the information about new messages from the corresponding service provider. Now you merely need query your mailbox or e-mail POB if new messages really are present. You can also send a MWI from a voicebox connected to the system, or from a system telephone set up

as a reception telephone.

This information can be displayed or signalled on terminals (analogue terminals, ISDN terminals and system telephones) that support this performance feature. MMW information from outside is conveyed transparently by the system. When an MMI is present, the bintec elmeg telephone displays an envelope symbol and a text generated in the telephone, along with the caller's phone number.

Analogue terminals

- Switching on the MMI can only occur with receiver replaced.
- If there's a message from a voicemail system, there's a short call. Depending on the terminal, a symbol, a text generated in the telephone as well as the caller's telephone number can be displayed. If MWI information is deleted, there is no signalling.
- For the terminal, CLIP must be set up and enabled in the configuration.
- Callback to the voice mail system or reception telephone is possible; the MMI information is deleted in the process.

ISDN terminals

- Switching on the MWI is possible at all times (also during the call).
- If there's a message from a voicemail system, there's a short call. Depending on the terminal, a symbol, a text generated in the telephone as well as the caller's telephone number can be displayed. If MWI information is deleted, there is no signalling.
- Callback to the voice mail system or reception telephone is possible; the MMI information is deleted in the process.

System telephones

- Switching on the MWI is possible at all times (also during the call). The caller's number is entered in the caller list. Depending on the type of system telephone, e. g. external voicemail, Netbox Heute, the name and number of the caller are entered. In addition, the **Caller list** LED flashes.
- Callback to the voice mail system or reception telephone is possible; the MMI information is deleted in the process.

Hotel room telephone

- If a message from a voicemail system is present, a special dialling tone is heard after the receiver is picked up.

Reception telephone

- MWI information can be switched on and off from a reception telephone to a room tele-

phone via a telephone procedure. If MWI information is switched to a room telephone, the reception telephone number is entered into the caller list and the special dialling tone is enabled.

Disabling the MWI announcement

• Manual disabling via reception telephone procedure.

• Call from reception telephone to room telephone. The MWI information is automatically deleted in call status.

• Callback from room telephone to reception telephone deletes the MWI information.

Note

This performance feature must be requested for your ISDN connection from the network operator. There, you will also be informed of available services. The information can only be displayed on the internal ISDN terminal if an external MSN has been assigned to the terminal in the configuration.

All MWI data are deleted after a system reset.

Net Direct (keypad)

Some time ago, you purchased the most advanced telephone of the time. Since then, however, a number of new performance features have appeared on the public network, which cannot be used by simply pressing a key. You can use the keypad function to employ your network operator's current ISDN functions by entering a key sequence from your ISDN or analogue telephone.

The keypad function allows control of service or performance features in your operator's network by entering character and numerical sequences.

Note

You can only use the keypad performance feature if it is supported by your network operator and has been requested for your ISDN connection. If you have set up an automatic outside line for an internal subscriber, the keypad functions cannot be directly used. First disable the **Automatic Outside Line** or dial the star key, then the code for manual outside line (e. g. 0) followed by keypad dialling, beginning with the star or hash key.

Keypad functions can only operate from terminals that have been assigned an external multiple subscriber number (MSN) in configuration and possess a keypad authorisation.

> Your network operator's performance features are always set up for the number (MSN) sent by your terminal.

The menu **Advanced Settings** consists of the following fields:

Fields in the Advanced Settings menu

Field	Description
Receive System Intercom Call	Select whether simplex operation calls to the system telephone shall be allowed for this authorisation class. The function is activated by selecting *Enabled*. The function is enabled by default.
Receive Announcement Calls	Select whether this authorisation class may receive announcements. The function is activated by selecting *Enabled*. The function is enabled by default.
Receive MWI Information	Select whether this authorisation class may receive information about existing messages (MWI = Message Waiting Indication). The function is activated by selecting *Enabled*. The function is enabled by default.
Net Direct (Keypad)	Select whether you wish to use your network operator's current ISDN functions also from older ISDN or analogue telephones by entering a key sequence. The function is activated by selecting *Enabled*. The function is enabled by default.

6.4.2.2.3 Applications

Additional applications are set up in the **Numbering**->**User Settings**->**Class of Services**->**Applications** menu.

The menu consists of the following fields:

Fields in the Application Authorization menu

Field	Description
System Phonebook Authorization	Select whether this authorisation class may use entries in the system phone book and, if so, to what extent. Possible values: • *Yes, according to line access authorization* (default value): System phone book entries may be used unless located beyond the configured line access authorisation. • *Yes, without restrictions*: System phone book entries may be used in unrestricted access. • *No*: System phone book entries may not be used.
Music on Hold	Select whether and which MoH (Music on Hold) shall be used. Possible values: • *Off* (default value): A caller on hold shall hear no music-on-hold. • *<MoH-Wave file>*: A caller on hold should hear the selected Wave file as music-on-hold. • *MOH Intern 1* (default value for compact systems) • *MOH Intern 2* • *MoH Wave 1 to 8*
Doorcom Access	Select whether this authorisation class may connect to the door intercom. The function is activated by selecting *Allowed*. The function is enabled by default.
TAPI	Select whether this authorisation class may use the system's TAPI functionalities. The function is activated by selecting *Allowed*. The function is enabled by default.
Save call data records	Define whether the connection data of this authorisation class shall be saved. The function is activated by selecting *Enabled*.

Field	Description
	The function is enabled by default.
Transmit charge information	Select whether the transferred charge information shall be transmitted to terminals of this authorisation class. The function is activated by selecting *Allowed*. The function is enabled by default.
Relay Contact(s) Access	Within an authorisation category, you can enable or prohibit the permission to configure a relay for each contact individually. The function is activated by selecting *Allowed*. The function is disabled by default.

6.4.2.3 Parallel Ringing

In the **Numbering**->**User Settings**->**Parallel Ringing** menu, you configure whether there should be signalling to another external number in the case of incoming calls to an internal number.

6.4.2.3.1 Edit or New

Choose the ✎ icon to edit existing entries. Choose the **New** button to create other entries.

The **Numbering**->**User Settings**->**Parallel Ringing**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Internal Number	Select the internal number for which the parallel call performance feature is to be set up.
External Number	Under **New Number**, enter the external telephone number to which a call should be signalled in parallel. If a mobile number and a home number have been set up under **Users**->**Basic Settings**->**External Numbers**, they will be displayed and can be selected under **Configured Home Number** or **Configured Mobile Number**.
Parallel Ringing	Select whether this parallel call entry is to be enabled.

Field	Description
	The function is activated by selecting *Enabled*.
	The function is disabled by default.

6.4.3 Groups &Teams

In this menu, you configure your system's teams.

6.4.3.1 Teams

In the **Numbering**->**Groups &Teams**->**Teams** menu, you configure you system's teams.

Teams are groups of people working together to realise an objective. In practice, this means that all people within a team can be reached under the same subscriber number for external and internal calls. In the PABX, each team of telephones/terminals can thus be assigned a specific subscriber number to guarantee accessibility to internal and external calls. Individual structures of companies can be mapped by teams. Thus departments such as Service, Sales or Development can be called from inside or outside in a selective manner via team numbers. Within a team, the call can, for example, be signalled simultaneously to all, or first to one telephone, then also to a second, etc. In one team, answering machines or voice systems can also be used.

Four team call options are assigned to each team. Switching between call options can occur manually or via one of the calendars.

Only for compact systems: The *Team global* is configured by default.

Choose the ✎ icon to edit existing entries. Select the **New** button to create a new team.

6.4.3.1.1 General

In the **Numbering**->**Groups &Teams**->**Teams**->**General** menu, the basic conditions in the team are configured. Among these are the team name and the internal team number.

For internal team calls, a team number and team name can be assigned to the team in the configuration. If a team number is dialled, the caller sees the team name until a team subscriber accepts the call. The name of the team subscriber is then displayed.

The **Numbering**->**Groups &Teams**->**Teams**->**General** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the team.
Internal Number	Enter the internal number of the team.

Fields in the Further Settings menu

Field	Description
Switch call signalling	Define whether the call option configured for the team shall be enabled manually over the telephone, or via the calendar. For this, calendar and switching times must first have been configured. You can create up to four call variants for each team in the menu **Numbering**->**Groups &Teams**->**Teams**->**New**->**Variant1-4** . Possible values: • *No calendar,only manually* (default value): Manual switch is enabled. • *<Calendar>*: Select one of the configured calendars.
Active Variant (Day)	Select one of the call options to be currently enabled. If a switch is set up via the calendar, this setting will be switched back again in a timely manner. The default value is *Signalling Variant 1*.
Permit Call Forwarding	Define whether call forwarding may occur for the team. The function is activated by selecting *Enabled*. The function is disabled by default.
Call Forwarding to External Numbers	Define whether there shall be call forwarding within the system itself (**Through PABX**, default value) or via an exchange (provider, **Through Exchange Office**). Please note that for call forwarding within the system two external connections are used.

The menu **Advanced Settings** consists of the following fields:

Fields in the Timer menu

Field	Description
Team Speed Timer	Here, enter the **Team Speed Timer** after which call forwarding

Field	Description
	after time shall be done in the team. The default value is *15* seconds.
Simultaneous after time	With linear and rotating team calls, there is the option for all team subscribers to be simultaneously called after a defined period. The default value is *60* seconds.
Wrap-up Timer	This setting is only enabled when **Signalling** *Even Distribution (Longest Free)*. For every subscriber who has ended a call, a **Post processing time** is configured, during which he receives no more calls. Calls received by the subscriber on his number rather than via the team and self-initiated calls are not included in the time calculation. The default value is *0* seconds; the range *0... 999* seconds.

6.4.3.1.2 Variant 1 - 4

In the **Numbering**->**Groups &Teams**->**Teams**->**Variant 1-4** you configure a team's four call variants. You can create up to four different call options for each team. For this, assign either an internal or external number to the call option, and define how an incoming call should be signalled within the team.

Under **Internal Assignment**, select the internal subscribers who are to belong to this team. If you wish to temporarily exclude one of the team subscribers from call signalling (e.g. a team subscriber is on holiday), you can **Logout** this subscriber. Team calls are not signalled to logged out subscribers. Every team subscriber can also control login and logout himself via a system code.

For internal team calls, a team number and team name can be assigned to the team in the configuration. If a team number is dialled, the caller sees the team name until a team subscriber accepts the call. The name of the team subscriber is then displayed. A call to a team can be simultaneous, linear, rotating, setting up or parallel after time. With linear and rotating team calls, there is the option for all team subscribers to be simultaneously called after a defined period (1...99).

The **Numbering**->**Groups &Teams**->**Teams**->**Variant** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Assignment	You can assign several internal numbers to each team, or an external number to each. Define whether calls for a team shall be signalled to internal or external subscribers. Possible values: • *External*: The entered external number is called. • *Internal* (default value): The subscribers assigned to the selected number are called according to the defined signalling.
Internal Assignment	Only if **Assignment** = *Internal* Select the internal team subscribers. With **Add**, you add more internal numbers. Only for compact systems: The numbers *10*, *20*, *21*, *22* are assigned to the *Team global*.
External Assignment	Only if **Assignment** = *External* Enter the number of the external subscriber.
Route and Charge Assignment	Only if **Assignment** = *External* Charges for the call and assignment of an external connection occur via the selected internal subscriber.

Automatic call acceptance in the team

You want a caller to be accepted already at call signalling and not to hear the ringing tone. That's no problem if you're using automatic call acceptance for team calls. In this case, the caller is automatically accepted by the system and hears an announcement or system music-on-hold. During this time, the call is signalled to the entered team subscribers. If a subscriber takes the call, the connection to the caller is established.

If a team is called, it can be defined in configuration that the call is automatically accepted, and that the caller hears an announcement or music. The target subscriber(s) are called during this time. After the receiver is picked up, the announcement or music is turned off and the subscribers are connected to each other.

Possible settings for automatic call acceptance:

• *Simultaneous*: All assigned terminals are called simultaneously. If a terminal is busy,

call waiting can be used.

- *Linear*: All assigned terminals are called in the sequence of their entry in the configuration. If a terminal is engaged, the next free terminal is called. The call is signalled ca. 15 seconds per subscriber. This period can be set between 1 and 99 seconds (per team) in the configuration. If subscribers are on the phone or logged out, there is not forwarding time for these.

- *Rotating*: This call is a special case of the linear call. After all terminals are called, call signalling begins again with the first entered terminal. The call is signaled until the caller replaces the receiver or the call is ended by the exchange (after ca. 2 minutes).

- *Adding*: The terminals are called in the sequence of their entry in the subscriber list. Every terminal that has already been called is called again, until all entered terminals are called.

- *Linear,Simultaneous after time* or *Rotating,Simultaneous after time*: Rotating or linear is set for the team call. After defined times have run out, all team subscribers can be called in parallel (simultaneously). Example: A precondition is that the sum of forwarding times is larger than the time **parallel after time**. There are 4 subscribers to a team. The forwarding time for each subscriber is 10 seconds, 40 seconds in total. The time **parallel after time** is set to 38 seconds. Every subscriber will be called. If a subscriber logs out of the team or is engaged, forwarding time is only 30 seconds, after which the **parallel after time** call is no longer made.

- *Even Distribution (Longest Free)*: Even distribution equates to **SignallingRotating** and ensures that all team subscribers receive the same number of calls. For every subscriber who has ended a call, a **Wrap-up Time** (0...999 seconds) is set up for the team/subscriber, during which he receives no more calls. Calls received by the subscriber on his number rather than via the team and self-initiated calls are not included in the even distribution calculation. Even distribution begins with the subscriber who hasn't received calls for the longest time, on restart with the first subscriber entered in the subscriber list. A subscriber who has logged out of the team (code number or function key) is no longer taken into account for the even distribution. After a system power failure, the existing **Even distribution** calculation is deleted and the process begins again. If all team subscribers are in **Post processing time**, external calls are routed to the preset redirect destination; internal calls hear the busy tone. If the same time since the last call is calculated for several team subscribers, the sequence of entries in **Internal Assignement** applies.

Fields in the Options menu

Field	Description
Signalling	You can call team subscribers with a broadcast call. Possible values: • *Simultaneous* (default value)

Field	Description
	• *Linear* • *Rotating* • *Adding* • *Linear,Simultaneous after time* • *Rotating,Simultaneous after time* • *Even Distribution (Longest Free)*
Busy on busy	Select whether the performance feature Busy on Busy shall be enabled for this call option. If a team subscriber is currently engaged, you can decide whether additional calls for this team should be signalled. If "Busy on Busy" is set for a team, other callers are signalled as "Engaged". The function is activated by selecting *Enabled*. The function is disabled by default.
Automatic Call Pick-up with	Select whether an incoming call should be automatically accepted, and the caller hear the desired music-on-hold or announcement. Signalling of the call to the team proceeds. The caller bears the costs for the existing connection. The function is activated by selecting *Enabled*. The function is disabled by default. Also select the desired music-on-hold or announcement. Possible values: • *<File_x>* • *MOH Internal 1* • *MOH Internal 2* • *MoH Wave 1 to 8*

The menu **Advanced Settings** consists of the following fields:

Fields in the Rerouting Functions menu

Field	Description
Rerouting on no re-sponse	Select whether and, if so, to which team an incoming call should be redirected on no reply. Possible values: • *None* • *<Team>* Also enter the time after which the call should be redirected.
Further Rerouting	Select whether and, if so, to which redirect option an incoming call shall be switched. Possible values: • *Off*: No other redirect options are used. • *Immediately*: The incoming call is immediately rerouted to the redirect function selected in **Immediately**. • *On Busy*: The incoming call is rerouted to the redirect function selected in **On Busy**.
Immediately	Only if **Further Rerouting** = *Immediately* Select the redirect function for immediate redirect. You configure the redirect functions in **Applications**->**Rerouting**->**Rerouting Functions**.
On Busy	Only if **Further Rerouting** = *On Busy* Select the redirect function for redirect on engaged. You configure the redirect functions in **Applications**->**Rerouting**->**Rerouting Functions**.
Busy starting with	Only if **Further Rerouting** = *On Busy* Select from which number of subscribers the team is considered engaged.

6.4.3.1.3 Log on / Log off

In the **Numbering**->**Groups &Teams**->**Teams**->**Log on / Log off** menu, the individual team members are logged in/out.

The **Numbering**->**Groups &Teams**->**Teams**->**Log on / Log off** menu consists of the fol-

lowing fields:

Fields in the Basic Settings menu

Field	Description
Numbers	Indicates the internal number of assigned team members.
Status	Select whether the team member is logged into the team. The team member is logged in by selecting *Logged on*. Only for compact systems: All team members are *Logged on* by default.

6.4.4 Call Distribution

In this menu, you configure internal forwarding of all incoming calls.

6.4.4.1 Incoming Distribution

In the **Numbering**->**Call Distribution**->**Incoming Distribution** menu you configure the assignment of incoming calls to the desired internal numbers.

Under call assignment, you assign numbers entered under **External Numbers** to e. g. the teams or an internal number.

6.4.4.1.1 Edit

Choose the ✎ icon to edit existing entries.

The **Numbering**->**Call Distribution**->**Incoming Distribution**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
<Name of Number Entry>	Displays the number configured.
Trunk	Displays the external connection for which call assignment is configured.
Assignment	Select the internal number or the desired function to which incoming calls shall be assigned via the line selected in **Trunk**.

Field	Description
	Possible values: • *Internal Number* (default value): The internal team number is selected for assignment to a team. • *Call Through* • *Rerouting Application* • *Phone Remote Access* • *Mini Call Center*

Fields in the Internal Number and Rerouting Settings menu

Field	Description
Internal Number	Only for **Assignment** = *Internal Number* Select the internal number to which incoming calls shall be assigned via the line selected in **Trunk**.
Rerouting Application	Only for **Assignment** = *Rerouting Application* Select the rerouting application that you wish to be assigned to the number. You configure rerouting applications in the **Applications**->**Rerouting**->**Rerouting Applications** menu.
Active Variant (Day)	Only for Rerouting Application= *<configured rerouting application>* Select the rerouting application variant that should be currently enabled. If a switch is set up for the variants via the calendar, this setting will be switched back again in a timely manner. Possible values: • *Variant* • *Variant 2* • *Variant 3* • *Variant 4*

Fields in the Call Through Settings menu

Field	Description
Authorization	Only for **Assignment** = *Call Through*

Field	Description
	Define the authorisation for which the Call Through function shall be released. Possible values: • *Number screening*: Dialling release occurs after matching the entered number with the entry in the system phone book or with the user's call number entries (**Mobile Number** and **Home Number**). • *Number screening and PIN*: Dialling release occurs after matching the entered number with the entry in the system phone book or with the user's call number entries (**Mobile Number** and **Home Number**) AND entering the PIN. • *PIN*: Dialling release occurs after PIN entry. • *Number screening or PIN*: Dialling release occurs after matching the entered number with the entry in the system phone book or with the user's call number entries (**Mobile Number** and **Home Number**) OR entering the PIN.
PIN (6 Digit Numeric)	Only for **Authorization** = *Number screening and PIN*, *PIN*, *Number screening or PIN* The system checks the caller's authorisation for Call Through, then activates a simulated external dialling tone for the call. Authorisation is granted if the caller has entered the correct 6-digit PIN.
Internal Number and Rerouting Settings	Select the internal subscriber via which Call Through is to occur. One of the system's telephone numbers is defined in the configuration for Call Through. An external caller using this telephone number first hears the system's attention tone.

6.4.4.2 Misdial Routing

In the menu **Numbering**->**Call Distribution**->**Misdial Routing** for every external connection, you define the subscriber or the team to which the call shall go in any of the following cases:

- an incoming call has a wrong or truncated number / extension
- all members of the called team or call center are logged off.
- all members of the called c all center are in post-processing.

6.4.4.2.1 Edit

Choose the ✎ icon to edit existing entries.

The **Numbering**->**Call Distribution**->**Misdial Routing**-> **Edit** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Trunk	Displays the external connection for which redirect for wrong dialling is configured.
Rerouting to Number	Select the type of rerouting: • *None*: No redirect here, the caller gets a busy tone. • *Global Settings*: Redirect occurs as entered in **System Management**->**Global Settings**->**System**->**Rerouting to Number**. • *<Internal number of a user or team>*: The call is redirected to this user or team.

6.4.4.3 Caller number distribution

This menu allows you to specify to which internal number an incoming call is distributed to in dependence from the caller's number. This function can also be used as a blacklist when incoming calls from specific numbers are distributed neither to an internal number nor to an announcement. These calls are rejected.

6.4.4.3.1 Edit or New

Select the ✎ button in order to edit exiting entries, or select the **New** button to add further caller numbers.

The menu consists of the following fields:

Fields in the menu Basic Settings

Field	Description
Caller Number	Specify the caller's number whose calls are to be distributed to a specific internal number. Possible applications are: • complete numbers (0911987654) • area codes (0911) • country codes (001) • prefixes of special numbers (0137) . Numbers from your own public network have to be specified with their area code, the local country code is ignored. **Note** Any incoming number is matched against the specified number starting with the first digit and without considering any possible groups of digits. A single *0*, therefore, matches **all** calls coming in with a leading *0*. This means that a digit sequence matches the more calls the shorter it is. If you select the option *anonymous* instead of specifying a number, all calls are filtered that come in without transmitting a caller number.
Description	Enter a description for the number settings you have just made, e.g., *Family* or *Advertising*.
Assignment	here, you specify how your device is to respond to an incoming call. Possible choices: • *None*: The incoming call is not distributed, at all, and is refused. • *Internal Number*: The call is distributed to an internal number. If you choose this option, another card **(Assignment)** is displayed that allows you to choose from the

Field	Description
	available internal numbers.
	• *Announcement*: The caller is played an announcement. You can choose from the available announcements in the card **Assignment**.

Note

If you want to assign more than one internal number to an incoming number, create multiple entries for the same incoming number.

6.5 Terminals

6.5.1 elmeg System Phones

In this menu, you perform the assignment of configured internal numbers to the terminals and manage additional functions depending on the type of terminal.

The system telephone end devices (or DECT bases, respectively) are listed alphabetically in the **Description** column. Click the column title of any other column to sort entries in ascending or descending order.

Connected telephones or DECT bases are automatically recognized and listed in the respective summary; they can, however, also be manually configured before being connected to the system.

6.5.1.1 elmeg IP

The **Terminals**->**elmeg System Phones**->**elmeg IP** menu displays a list of IP telephones. The upper part of the overview displays the manually configured, the lower part displays the automatically detected devices. For an automatic discovery we recommend the use of DHCP (Activate the option *Use this device as DHCPv4 server* in the menu **Assistants**->**First steps**)

If you want to assign a static IP address, you must enter your PABX system as provisioning server (*http://<IP address of the provisioning server>/ eg_prov*). As soon as a **Description** for the telephone is entered and saved with **OK**, the entry for that device is moved to the upper part of the overview.

Note

Key Extension Modules are not discovered automatically, but must be manually configured using the GUI.

If a configured Key Extension Module is deleted the corresponding function keys are likewise deleted.

Choose the ✎ icon to edit existing entries.

After a short time, the icons 🖳 and ↻ are displayed for this device.

After clicking the **Apply** button it takes several seconds until the changes have been transmitted to the respective IP telephone.

Choose the ≡₊ synbol in order to copy an existing entry. This can be useful if you intend to create an entry that differs only in a few parameters from the already existing entry. In this case, copy the entry and change only the desired paramaters.

Select the 🖳 button to go to the **IP1x0** telephone user interface administrator page. This is described in the telephone user guide.

Select the **New** button to manually set up a new IP end device.

Employ automatic provisioning in order to have your PABX system transmit basic telephony parameters to the IP telephones. When using the assistent **First steps** activate the option *elmeg IP1x/DECT* for the field **Transmit Provisioning Server for** in the **Assistants**->**First steps**->**Advanced Settings**->**Add** section. Alternatively, you can create a new entry in the menu **Local Services**->**DHCP Server**->**DHCP Configuration**->**New**->**Advanced Settings**->**DHCP Options** and set the fields **Option** = *URL (Provisioning Server)* and **Value** = *http://<IP address of the provisioning server>/eg_prov* .

To register the handsets you first set the base station to login mode. Then you perform the registering of the handsets on the handests themselves. To configure the base station in any more detail, you need to use the DECT system's web configurator.

Select the button ↻ to trigger an update of the device's provisioning. If the update is successful, the updated value displays in the **Last seen** column within 10 seconds.

Note

If you wish to test whether your base station is correctly configured and accessible, select the button ⟳ and check whether an updated value is displayed within 10 seconds in the **Last seen** column.

6.5.1.1.1 General

You can change the IP telephone's basic settings in the **Terminals**->**elmeg System Phones**->**elmeg IP**->**General** menu.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	To clearly identify the telephone in the system, enter a description for the telephone.
Phone Type	Displays the type of your IP telephone. Possible values: • *elmeg IP620* • *elmeg IP630* • *elmeg IP640*
Location	Select the location of the telephone. You define location in the **VoIP**->**Settings**->**Locations** menu. Depending on the setting in this menu, default behaviour for registration of VoIP subscribers for which no location should be defined is displayed for selection. Possible values: • *Not defined (Unrestricted Registration)*: No location is defined. According to set default behaviour, the subscriber is nevertheless registered. • *Not defined (No Registration)*: No location is defined. According to set default behaviour, the subscriber is not registered. • *Not defined (Registration for Private Networks Only)*: No location is defined. According to set de-

Field	Description
	fault behaviour, the subscriber is only registered if located in a private network. • *<Location>*: A defined location is selected. The subscriber is only registered if at this location.
MAC Address	Shows the MAC address of the telephone.
IP/MAC Binding	Displays the IP address automatically assigned by DHCP. Here you have the option of permanently assigning the displayed IP address to the device with the displayed MAC address. This option should be enabled for quick login after a functional disruption.

Key Extensions

Fields n the menu Extensions

Fiekd	Description
Ext. Module No 1 - 3 (depends on **Phone Type**)	Displays if you are operatnig the IP telephone with an key extension module. Only the number of modules supported by the respective phone type is offered for configuration. Possible values: • **Not available** • **Available**

The menu **Advanced Settings** consists of the following fields:

Fields in the Further Settings menu

Field	Description
No Hold and Retrieve	The performance features hold a call and retrieve a held call are not available on certain telephones. The function is activated by selecting *Enabled*. The function is disabled by default.

Fields in the Codec Settings menu

Field	Description
Codec Profile	Select the Codec profile terminal to be used. Codec profiles are configured in the **VoIP**->**Settings**->**Codec Profiles** menu.

6.5.1.1.2 Numbers

In the menu **Terminals**->**elmeg System Phones**->**elmeg IP**->**Numbers** you assign an IP telephone up to twelve internal phone numbers using **Add**.

The available internal phone numbers are created under **Numbering**->**User Settings**->**Users**->**New**.

You can delete assigned numbers from the list with 🗑 .

Values in the list Number Settings

Field	Description
Connections Nr.	Shows the serial number of the connection.
Internal Number	Displays the assigned internal number.
Displayed Description	Displays the description that will be displayed on the IP telephone's display.
User	Displays the user's name.

6.5.1.1.3 Keys / T100

The menu **Terminals**->**elmeg System Phones**->**elmeg IP**->**Keys** displays the configuration of your system telephone's keys.

> **Note**
>
> You can configure the key assigment either through your PABX system or on the telephone itself. We recommend using your PABX system for this, since it overwrites the telephone configuration.
>
> You can avoid the overwriting for individual keys that have already been configured on the telephone by choosing *Not configured* on the PABX system.

Your telephone is equipped with several function keys that allow the assignment of different functions. The functions available for programming are different across different types of telephones.

Values in the list Keys

Field	Description
Key	Displays the name of the key.
Label Description	Displays the configured key name. This appears on the labelling page (label strips).
Key Type	Displays the key type.
Settings	Displays the additional settings with a summary

Print allows you to print out a label sheet for the description field of your system phone or key extension.

Edit

Choose the ✎ icon to edit existing entries. In the pop-up menu, you configure the functions of your system telephone keys.

You can use the following functions with system telephones:

- *Dial Key (Standard)*: You can store a number on each function key. External numbers have to be prefixed by the exchange code *0* if *no automatic outside line* has been configured for your **Class of Service** on the telephone.
- *Dial Key (DTMF)*: You can store a DTMF sequence on every function key.
- *Extension Key (User)*: You can set up dialling to an internal extension using a line key. After pressing the corresponding key, hands free is switched on and the internal extension entered is selected. If a call is signalled on the internal extension you have entered, you can pick this up by pressing the line key.
- *MSN Selection Key*: Assigns a specific connection (i.e. a specific SIP account) to the function key. You can use this key to initiate a call via this connection, or you can accept a call coming in via this connection. The key flashes if a call is received, it is lit if the connection is busy. Select the desired connection. All configured connections are available. Configure SIP accounts exclusively on your PABX system.
- *System Parking (Open Enquiry)*: The called extension enters an enquiry and dials a code. The telephone is now open for additional operations like e.g. an announcement. A second subscriber can accept the call by picking up the receiver and dialing the code corresponding to the call. The codes are determined by the PABX, but can also be assigned to the functions keys of one or more system phones. If a call is put into open enquiry by pressing a function key, this is indicated by the flashing of the respective function key LED on all system phones with a corresponding configuration. Pressing the function key accepts the call. This function is only available if a call has been parked.

- *Login / Log Out, Team*: You can log into out out of one of the configured teams.

- *Call Forwarding (enable)*: Assigns activating or deactivating a call forwarding that has been configured on the telephone. You can only store a single call forwarding on the device; it is applied to all calls.

- *Parallel Ringing*: Incoming calls to an internal number are signaled at another external number in parallel.

- *Night Mode*: You can set the phone into night mode oprtation.

- *Call Variant, Team*: With this function key the option **Active variant (Day)** of a team can be switched between variant 1 to 4. A separate key must be programmed for each signaling variant (2, 3, 4). The LED signals the respective active variant, variant 1 is indicated by switched off LEDs.

  If the variant is switched by the calendar, the key LEDs are updated accordingly, no manual change can be made by the keys (the LEDs flash three times when the key is pressed).

  The corresponding user must be allowed by their class of service to change the variants.

- *Reroute Variant, Extension*: With this function key the **Active Variant (Day)** of an extension can be switched between variants 1 to 4. This key can only be used on the telephone to which the extension belongs. The function is the same as for the function key **Call Variant, Team** .

- *Reroute Variant, Trunk Number*: With this function key the **Active Variant (Day)** of a rerouting application assigned to an extension number can be switched between 1 to 4. This key can be used system-wide, but requires the authorization to switch variants via the corresponding class of service.

- *Voice Box Mode, Extension*: This function key can be used to switch the status of the owner of a voice box between *In the office* and *Out of office*. Pressing the button switches the mode to *Out of office* and the LED lights up. Pressing the key again switches back to the mode *In the office* and the LED goes out.

  This key can only be used on the phone to which the voice mail box belongs.

- *Voice Box Mode, Team*: This function key can be used to switch the status of a Team Voice Mail box between *In the office* and *Out of office*. This key can be used system-wide, but for switching the mode the extension must be a member of the team.

- *Not configured*: The function key is managed by the telephone itself and not by the PABX system.This options locks the key for the provisioning by your PABX system.

The menu **Terminals**->**elmeg System Phones**->**elmeg IP**->**Keys**->**Edit** consists of the following fields:

Fields in the menu Keys

Field	Description
Key name	Enter a name for the key to be used as text for the corresponding key when the ID labels are printed.
Key Type	Depending on the model, telephones have seven or 14 keys that can have functions assigned to them. Optional key extension modules extend the number of available functions keys. Possible values: • *Dial Key (Standard)* • *Dial Key (DTMF)* • *Extension Key (User)* • *MSN Selection Key* • *System Parking* • Login / Log out Team • *Call Forwarding (enable)* • *Parallel Ringing* • Night Mode • *Call Variant, Team* • *Reroute Variant, Extension* • *Reroute Variant, Trunk Number* • *Voice Box Mode, Extension* • *Voice Box Mode, Team* • *Not configured* Depending on which function is assigned to a key, the required options such as **Internal MSN**, **External number**, **Teams** and **Call variant**are displayed in a selection menu or an input field. In a selection menu, only those values are available that have already been created (e.g. internal phone numbers or teams).

Transfer key

Select the ↑↓ icon to move configured function keys.

Fields in the menu Key

Field	Description
Key name	Displays the name of the key.

Field	Description
Key Type	Displays the key type.
Settings	Displays the additional settings with a summary

Fields in the menu Move to

Field	Description
Phone	Select one of the connected telephones.
Module	Select *Telephone* or a key extension.
Key	Select the key to which you wish to transfer the configured function.

6.5.1.1.4 Settings

You can reset the phone's administrator password in the **Terminals**->**elmeg System Phones**->**elmeg IP**->**Settings** menu.

The menu consists of the following fields:

Fields in the System Phone menu

Field	Description
Admin Password	Select whether the administrator password should be reset. The function is activated by selecting *Enabled*. The function is disabled by default. As soon as you select the **OK** button, the password is reset to the default setting.
Display Language	Select the display language for your telephone. Possible values: • *Deutsch* • *Dutch* • *English* • *Italian* • *Spanish*

Field	Description
	• *French*
	• *Portugues*
	• *Česko*
	• *Greek*
	• *Polish*
	• *Romanian*
	• *Slovak*

6.5.1.2 elmeg DECT

The menu **Terminals**->**elmeg System Phones**->**elmeg DECT** displays the base stations of the connected DECT single-cell and multi-cell systems.

All base stations that are connected are automatically detected and listed in the lower part of the overview. For an automatic discovery we recommend the use of DHCP. (Activate the option *Use this device as DHCPv4 server* in the menu **Assistants**->**First steps**.)

If you want to assign a static IP address, you must enter your PABX system as provisioning server (*http://<IP address of the provisioning server>/ eg_prov*).

As soon as a **Description** for the base station is entered and saved with **OK**, the entry for that device is moved to the upper part of the overview.

After a short time, the icons ▐▀ and ↻ are displayed for this device.

Choose the ✎ icon to edit existing entries.

After clicking the **Apply** button it takes several seconds until the changes have been transmitted to the respective device.

Select the **New** button to manually set up a new base station.

Select the button ▐▀ to go to the base station's Web configurator. This is described in the user guide for the relevant DECT system.

To register the handsets you first set the base station to login mode. Then you perform the registering of the handsets on the handests themselves. To configure the base station in any more detail, you need to use the DECT system's web configurator.

Select the button ↻ to trigger an update of the device's provisioning. If the update is successful, the updated value displays in the **Last seen** column within 10 seconds.

Note

If you wish to test whether your base station is correctly configured and accessible, se-lect the button ↻ and check whether an updated value is displayed within 10 seconds in the **Last seen** column.

Note

If you wish to change the language currently used with a DECT single-cell system, the system has to be connected to the provisioning server of the PABX.

6.5.1.2.1 General

In the menu **Terminals**->**elmeg System Phones**->**elmeg DECT**->**General** you make the basic settings for base stations.

The menu consists of the following fields:

Fields in the menu Basic Settings

Field	Description
Description	To clearly identify the base station in the system, enter a de-scription for the telephone.
Phone Type	Displays the type of base station. Possible values: • *elmeg DECT150* ⚠ **Important** Please note that the DECT150 can only be operated with system software version 2.59 or higher on a **be.IP swift**. In-formation on updating can be found in the download area of the product if necessary. • *elmeg DECT160/210*
Location	Select the location of the base station. You define locations in the **VoIP**->**Settings**->**Locations** menu. Depending on the set-ting in this menu, default behaviour for registration of VoIP sub-

Field	Description
	scribers for which no location should be defined is displayed for selection.
	Possible values:
	• *Not defined (Unrestricted Registration)*: No location is defined. According to set default behaviour, the subcriber is nevertheless registered.
	• *Not defined (No Registration)*: No location is defined. According to set default behaviour, the subscriber is not registered.
	• *Not defined (Registration for Private Networks Only)*: No location is defined. According to set default behaviour, the subscriber is only registered if located in a private network.
	• *Location*: A defined location is selected. The subscriber is only registered if at this location.
MAC Address	Shows the MAC address of the base station.
IP/MAC Binding	Displays the IP address automatically assigned by DHCP. Here you have the option of permanently assigning the displayed IP address to the base station with the displayed MAC address. This option should be activated to enable quick re-login after a functional fault.

The menu **Advanced Settings** consists of the following fields:

Fields in the menu Further Settings

Field	Description
No Hold and Retrieve	The performance features hold a call and retrieve a held call are not available on certain telephones. The function is activated by selecting *Enabled*. The function is disabled by default.

Fields in the menu Codec Settings

Field	Description
Codec Profile	Select the Codec profile to be used. Codec profiles are configured in the **VoIP**->**Settings**->**Codec Profiles** menu.

6.5.1.2.2 Numbers

In the menu **Terminals**->**elmeg System Phones**->**elmeg DECT**->**Numbers** you assign **Internal Numbers** to the mobile parts. You can select from the numbers that you have created for this purpose under **Numbering**->**User Settings**->**Users**.

The system automatically assigns a serial number, the **Mobile Number**, to each mobile part so that you can identify the device. You can then use **Add** to assign a **Internal Number** to a mobile part from the list.

You can delete assigned numbers with ▉.

Values in the list Numbers

Field	Description
Mobile Number	Displays the serial number of the mobile part. This number is permanently assigned to the mobile part so that it can be uniquely identified.
Internal Number	Displays the assigned internal number.
Displayed Description	Displays the description entered for the internal number. In standby mode this description is shown on the mobile part's display.
User	Displays the user's name.

6.5.1.2.3 Settings

In the **Terminals**->**elmeg System Phones**->**elmeg DECT**->**Settings** menu you can reset the administrator password for the base station.

The menu consists of the following fields:

Fields in the menu Basic Settings

Field	Description
Admin Password	Select whether the administrator password should be reset. The function is activated by selecting *Enabled*.

Field	Description
	The function is disabled by default.
	As soon as you select the **OK** button, the password is reset to the default setting.

6.5.2 Other phones

In this menu, you perform assignment of configured internal numbers to the terminals and set additional functions according to terminal type.

Terminals of the corresponding category (Octophon, VoIP, ISDN, or analog) are sorted alphabetically in the **Description** column. Click the column title of any other column to sort entries in ascending or descending order

6.5.2.1 VoIP

In the **Terminals**->**Other phones**->**VoIP** menu, you configure the connected VoIP terminals. For example, you perform assignment of a configured internal number.

6.5.2.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to add VoIP terminals.

The **Terminals**->**Other phones**->**VoIP**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the IP telephone.
Location	Select the location of the IP telephone. You define location in the **VoIP**->**Settings**->**Locations** menu. Depending on the setting in this menu, default behaviour for registration of VoIP subscribers for which no location should be defined is displayed for selection. Possible values: • *Not defined (Unrestricted Registration)*: No location is defined. According to set default behaviour, the subscriber is nevertheless registered. • *Not defined (No Registration)*: No location is defined. According to set default behaviour, the subscriber is

Field	Description
	not registered.
	• *Not defined (Registration for Private Net-works Only)*: No location is defined. According to set default behaviour, the subscriber is only registered if located in a private network.
	• *<Location>*: A defined location is selected. The subscriber is only registered if at this location.

Fields in the Number Settings menu

Field	Description
Internal Numbers	Select the internal number for this terminal You can define several internal numbers.
	Possible values:
	• *No free Extension Available*: All configured internal numbers are already in use. First configure another user with additional numbers.
	• *<Internal Number>*: Select one of the existing numbers of the configured users.

The menu **Advanced Settings** consists of the following fields:

Fields in the SIP Client Settings menu

Field	Description
SIP Client Mode	Select whether a *Dynamic* or a *Static* SIP client is to be used.
	Possible values:
	• *Dynamic* (default value): Your device (e.g. a standard SIP telephone) performs a SIP registration to publish its IP address to the system.
	• *Static*: An incoming call of a (statically configured) SIP client will be accepted without prior registration of this client if the IP address of the client matches the entered IP address under **SIP Client IP Address**. A Microsoft Office Communications Server and other Unified Communication Servers use this mode.
SIP Client IP Address	Only if **SIP Client Mode** = *Static*

Field	Description
	Enter the static local IP address of the SIP client.
Port Number	Only if **SIP Client Mode** = *Static* Enter the number of the port to be used for connection. A 5 digit sequence is possible. For example, the port *5065* must be entered for connection to a Microsoft Exchange Communication Server.
Transport Protocol	Only if **SIP Client Mode** = *Static* Select the transport protocol for the connection. Possible values: • *UDP* (default value) • *TCP* • *TLS* • *Automatic* - With this setting, your device supports automatic negotiation of the protocol with your provider's servers. For this setting to work, this negotiation must also be supported by the provider. For example, the protocol *TCP* must be entered for connection to a Microsoft Exchange Communication Server.

Fields in the Codec Settings menu

Field	Description
Codec Profile	Select the codec profile to be used if the connection is over a VoIP line. Codec profiles are configured in the **VoIP**->**Settings**->**Codec Profiles** menu.
Video	Select if calls between IP telephones are to support the transmission of video data. Video transmission can only be negotiated between the participants if both support this feature.
SRTP	Select if calls via this SIP provider may be secured with SRTP (Secure Real-Time Transport Protocol).

Fields in the Further Settings menu

Field	Description
Multiple SIP Connections (Sub-Exchange)	Select whether multilinks shall be allowed from this terminals. Operation as subsystem: Only in case of connection of a subsystem to a system Here, with a disabled performance feature, only a connection via the subscriber SIP registration is possible. If a second call comes in, it is accepted and the existing call is held. With an enabled performance feature, several SIP connections are possible over the same login. If the performance feature is enabled for as system without subsystem, two simultaneous calls on the phone are not connected to each other after the receiver is replaced but released, for example. Here, the performance feature should not be set. The function is activated by selecting *Enabled*. The function is disabled by default.
No Hold and Retrieve	The performance features hold a call and retrieve a held call are not available on certain telephones. The function is activated by selecting *Enabled*. The function is disabled by default.
T.38 FAX support	Select if you want to transmit FAX documents via Voice over IP using the T.38 standard. *Enabled* activates T.38 support. Per default, the function is disabled. If the function is disabled, FAX documents are transmitted using G.711.

6.5.2.2 ISDN

In the **Terminals**->**Other phones**->**ISDN** menu, you configure the connected ISDN terminals. For example, you perform assignment of a configured internal number.Only for compact systems. Two predefined entries are displayed:

Description	Interface	Terminal Type	Internal Numbers	License Allocation
ISDN 1	S0 1	Telephone	30	Enabled
ISDN 2	S0 2	Telephone	35	Enabled

6.5.2.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to add ISDN terminals.

The **Terminals**->**Other phones**->**ISDN**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the ISDN telephone.
Interface	Select the interface to which the ISDN telephone shall be connected.

Fields in the Basic Phone Settings menu

Field	Description
Terminal Type	Select the terminal type. Possible values: • *Telephone* (default value) • *Answering Machine* • *Voice Mail* • *Emergency Phone*
Internal Numbers	Select the internal number for this terminal You can define several internal numbers. Possible values: • *No free Extension Available*: All configured internal numbers are already in use. First configure another user with additional numbers. • *<Internal Number>*: Select one of the existing numbers of the configured users.

6.5.2.3 analog

In the **Terminals**->**Other phones**->**analog** menu, you configure the connected analogue terminals. For example, you perform assignment of a configured internal number.

Only for compact systems: Two predefined entries are displayed:

Description	Interface	Terminal Type	Internal Numbers	License Allocation
a/b 1	a/b 1	Telephone	10	Enabled
a/b 2	a/b 2	Telephone	11	Enabled
a/b 3	a/b 3	Telephone	12	Enabled
a/b 4	a/b 4	Multi Function Device/Telefax	13	Enabled

6.5.2.3.1 Edit or New

Choose the ✎ icon to edit existing entries.

Choose the ≡₊ icon to copy existing entries. Copying an entry can prove useful if you wish to create an entry only distinguished by a few parameters from an existing entry. In this case, you copy the entry and modify the desired parameters.

The **Terminals**->**Other phones**->**analog**-> ✎ menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the analogue telephone.
Interface	Select the interface to which the telephone shall be connected.

Fields in the Basic Phone Settings menu

Field	Description
Terminal Type	Select the terminal type.
	Possible values:
	• *Multi Function Device/Telefax*
	• *Telephone*
	• *Modem*
	• *Answering Machine*
	• *Emergency Phone*
Internal Number	Select the internal number for this terminal.
	Possible values:
	• *No free Extension Available*: The configured internal

Field	Description
	number is already in use. First configure another user with additional numbers. • *<Internal Number>*: Select one of the existing numbers of the configured users.

Fields in the Phone Settings menu

Field	Description
Call Waiting	Select whether call waiting shall be allowed for this device. The function is activated by selecting *Enabled*. The function is enabled by default.
Do not Disturb	Select whether you wish to use the call protection (do not disturb) performance feature. With this performance feature, you can enable call signalling to your terminal. Analogue terminals use system code numbers for this. Possible values: • *Internal Calls not signaled* • *External Calls not signaled* • *No Calls signaled*

The menu **Advanced Settings** consists of the following fields:

Fields in the CLIP Settings menu

Field	Description
Show incoming Number (CLIP)	Select whether the subscriber's number shall be transmitted. The function is activated by selecting *Enabled*. The function is enabled by default.
Show Date and Time	Only for **Show incoming Number (CLIP)** *Enabled* Select whether the time and date should be taken from your PABX system and displayed on the telephone. The function is activated by selecting *Enabled*.

Field	Description
	The function is enabled by default.
Show incoming Name (CNIP)	Only for **Show incoming Number (CLIP)** *Enabled* Select whether the caller's number shall be displayed. The caller's number can be displayed if an entry exists in the system telephone book. The function is activated by selecting *Enabled*. The function is enabled by default.
Show incoming waiting Number (CLIP off Hook)	Only for **Show incoming Number (CLIP)** *Enabled* Select whether the number of a caller waiting during an existing call shall be displayed. The function is activated by selecting *Enabled*. The function is enabled by default.

Fields in the Further Settings menu

Field	Description
Show new Messages (MWI)	Only for **Show incoming Number (CLIP)** *Enabled* Select whether new messages shall be signalled on a voice mail system. The function is activated by selecting *Enabled*. The function is disabled by default.
Transmit Charges Pulses	Select whether the system shall generate charge pulses for the terminal from the ISDN network charge information. For this purpose, you can define the charge impulse at 12 kHz or 16 kHz. Possible values: • *Off*: Charge information from the ISDN network is not transmitted. • *12 kHz* • *16 kHz*

Field	Description
	The default value is *16 kHz*
FXS Ringing Frequency	Call signalling in analogue terminals occurs by configuring a call switching voltage at the called analogue connections. This call switching voltage is converted into a specific ring tone by the analogue terminal. In the system, for the analogue connections you can set a call switching voltage with a frequency of *25 Hz* or *50 Hz*.
	The default value is *50* Hz.
	The default value is *50* Hz.
Flash Time for DTMF Dialling	When operating analogue terminals with the multifrequency code dialling method, you can set the flashtime that the system detects as maximum flash length. If the terminal flash is longer than the defined period, "replaced receiver" is detected.
	Values from *100 ms* (standard value) to *1000 ms* are possible.
	The default value is *400 ms*.

6.5.3 Overview

6.5.3.1 Overview

In the **Terminals**->**Overview**->**Overview** menu, you get an overview of all configured terminals.

Values in the Overview list

Field	Description
Description	Displays the terminal description.
Phone Type	Displays the telephone type.
Interface / Location	For ISDN, system and analogue terminals, displays the interface at which you're connected to the system. The configured location is displayed for IP terminals.
Internal Numbers	Displays the configured internal number.

6.6 Call Routing

The functions for external calls and automatic route selections for external calls are defined in call routing.

6.6.1 Outgoing Services

In the **Call Routing**->**Outgoing Services** menu you can configure the features **Direct Call**, **Call Forwarding**, **Dial Control** and **Priority Numbers**.

6.6.1.1 Direct Call

In the **Call Routing**->**Outgoing Services**->**Direct Call** menu you configure numbers that are dialled directly without the subscriber needing to dial a number themselves on the phone.

You wish to configure a telephone for which a call to a specific number is set up even without entry of the number (e.g. emergency telephone). You are not at home. However, there is someone at home who needs to be able to reach you quickly and easily by telephone, if required (e.g. children or grandparents). If you have set up the "Direct Call" function for one or more telephones, the receiver of the corresponding telephone only needs to be lifted. After a period without further entries set in configuration, the system automatically dials the configured direct call number.

If you do not dial within the specified period from picking up the receiver, automatic dialling is initiated.

The time for Direct Call is set under **System Management**->**Global Settings**->**Timer**->**Direct Call**.

Note

In the system, up to 10 direct call destinations with names and telephone numbers can be set up by the administrator. These destinations should then only be assigned to the terminals by the user via the user configuration interface. In the configuration, system direct call, or a direct call specifically configured for the terminal, can then be set by the user.

6.6.1.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Outgoing Services**->**Direct Call**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Direct Call Number	Enter the number to be automatically dialled if no number is to be dialled for a certain time after the receiver has been picked up.

6.6.1.2 Call Forwarding

In the **Call Routing**->**Outgoing Services**->**Call Forwarding** menu you configure the call forwarding of external calls for an internal subscriber.

You are temporarily away from your office, but don't want to miss a call. With call forwarding to another number, e.g. your mobile, you can receive your calls even when you are not at your desk. You can forward calls on your number to any call number. It can occur *Immediately*, *On no reply* or *On Busy*. Call forwarding *On no reply* and *On Busy* can exist concurrently. If you are not near your telephone, for example, the call is forwarded to another number (e.g. your mobile phone) after a short period. If you are making a call at your desk, other caller may receive the busy signal. You can forward these callers e.g. to a colleague or the secretary by using call forwarding on busy.

Every internal subscriber to the system can forward her calls to another number. Calls can be forwarded to internal subscriber numbers, internal team numbers or external numbers When the number to which calls shall be forwarded is entered, the system automatically checks whether it's an internal or external number.

In a team, call forwarding can be set up for one subscriber in the team. This call continues to be signalled to the other team subscribers. Call forwarding to an internal or external subscriber is performed in the system.

Call forwarding to an internal number is performed in the system. If an internal call to an external number is to be forwarded, forwarding also occurs in the system. Here, the connection is on the bundle cleared for the subscriber doing the setup.

6.6.1.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Outgoing Services**->**Call Forwarding**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Internal Number	Select the internal number to which the incoming calls shall be forwarded.
Type of Call Forward-ing	Select when incoming calls shall be forwarded to the specified internal number. Possible values: • *Immediately* • *On Busy* • *On no reply* (default value) • *On busy / On no reply*
Target Number (On no reply)	Enter the number to which incoming calls shall be forwarded after time.
Target Number (On busy)	Enter the number to which incoming calls shall be forwarded on busy.
Target Number (Immediate)	Enter the number to which incoming calls shall be forwarded im-mediately.

6.6.1.3 Dial Control

In the **Call Routing**->**Outgoing Services**->**Dial Control** menu you can block particular numbers/partial numbers or permit them.

You wish to prevent dialling of specific numbers in the system, e.g. the numbers of expens-ive value-added services. Enter these numbers or partial numbers into the dial ranges list of blocked numbers. All subscribers subject to dial ranges cannot dial these numbers. However, if you should need specific numbers from a blocked sector, you can clear these via the dial ranges list of cleared numbers.

You can block specific numbers or prefixes with the blocked numbers list. You can clear the blocked numbers or prefixes with the cleared numbers list. If a number entered as a cleared number is longer than one entered as a blocked number, this number can be di-alled. When you dial a number, dialling after the blocked digit is terminated and you hear the busy tone. You can assign each user individually to the dial ranges in the user settings.

Example: Blocked number *01*, all external numbers that begin with *01* are blocked.

Cleared number *012345*, dialling can proceed. All external numbers that begin with *012345* can be dialled. If two identical numbers (same number sequence and same number of digits, e.g. *01234* and *01234*) are entered in the list of cleared numbers as well as the list of blocked numbers, dialling of the number is prevented.

> **Note**
>
> Subscribers who enjoy full or partial dialling access (no outside line access) are authorised for dialling of cleared numbers via the list of cleared numbers.
>
> Please ensure that the area code is entered in the configuration, otherwise, the block can be circumvented in the local network by prefixing the area code.

6.6.1.3.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Outgoing Services**->**Dial Control**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Inhibited number	Enter the number that cannot be dialled.
Enabled number	Enter the number for which dialling is explicitly permitted.

6.6.1.4 Caller number distribution

This menu allows you to specify to which internal number an incoming call is distributed to in dependence from the caller's number. This function can also be used as a blacklist when incoming calls from specific numbers are distributed neither to an internal number nor to an announcement. These calls are rejected.

6.6.1.4.1 Edit or New

Select the ✎ button in order to edit exiting entries, or select the **New** button to add further caller numbers.

The menu consists of the following fields:

Fields in the menu Basic Settings

Field	Description
Caller Number	Specify the caller's number whose calls are to be distributed to a specific internal number. Possible applications are: • complete numbers (0911987654) • area codes (0911) • country codes (001) • prefixes of special numbers (0137) . Numbers from your own public network have to be specified with their area code, the local country code is ignored. **Note** Any incoming number is matched against the specified number starting with the first digit and without considering any possible groups of digits. A single *0*, therefore, matches **all** calls coming in with a leading *0*. This means that a digit sequence matches the more calls the shorter it is. If you select the option *anonymous* instead of specifying a number, all calls are filtered that come in without transmitting a caller number.
Description	Enter a description for the number settings you have just made, e.g., *Family* or *Advertising*.
Assignment	here, you specify how your device is to respond to an incoming call. Possible choices: • *None*: The incoming call is not distributed, at all, and is refused. • *Internal Number*: The call is distributed to an internal number. If you choose this option, another card **(Assignment)** is displayed that allows you to choose from the available internal numbers. • *Announcement*: The caller is played an announcement. You can choose from the available announcements in the card **As-**

Field	Description
	signment.

Note

If you want to assign more than one internal number to an incoming number, create multiple entries for the same incoming number.

6.6.1.5 Priority Numbers

In the **Call Routing**->**Outgoing Services**->**Priority Numbers** menu you configure numbers with particular special functions, e.g. emergency functions.

In your system configuration, you can enter numbers that must be accessible in an emergency. If you now dial one of these priority numbers, it is detected by the system and an ISDN B channel is automatically cleared. If the external ISDN B channels are already in use, one of the ISDN B channels is freed up and the calling subscribers hear the busy tone. An ongoing priority call is not interrupted.

6.6.1.5.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Outgoing Services**->**Priority Numbers**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Priority Number	Enter the number which can even be dialled if all B channels are occupied. In this case, an external B channel is released for this connection and reassigned for the priority call. An ongoing priority call is not interrupted.

6.6.1.6 Special Numbers

At a DDI connection, the called number of an outgoing call is automatically converted to the international E.164 format. This conversion is undesirable for certain numbers. Exceptions from the conversion can be configured here.

6.6.1.6.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Outgoing Services**->**Special Numbers**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Special Number	Specify the number that is to be excepted from E.164 conversion.

6.6.2 Automatic Route Selection

In the **Call Routing**->**Automatic Route Selection** menu you can set up routes for external calls in addition to configured line occupancy. Here, bundles released for users can be selectively assigned to ongoing calls according to the dialled number, or new providers entered along with their network access prefixes. You then specifically define the routing for individually created zones for every weekday.

6.6.2.1 General

In the **Call Routing**->**Automatic Route Selection**->**General** menu, you activate the ARS (Automatic Route Selection) function and select the desired route level.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
ARS	Select whether to enable the ARS performance feature (Automatic Route Selection). The function is activated by selecting *Enabled*. The function is disabled by default.
Routing Stage	Select whether additional routes shall be used if an entered provider or bundle cannot be accessed. Possible values:

Field	Description
	• *1 (No Fallback)*: If the entered provider or the selected bundle (**Call Routing**->**Automatic Route Selection**->**Zones &Routing**-> **Edit/Add**-> **Mo-Su** ->**Routing Stage 1**) is not available, the connection setup is terminated.
	• *2*: If the entered provider or the selected bundle (**Call Routing**->**Automatic Route Selection**->**Zones &Routing**-> **Edit/Add** -> **Mo-Su** ->**Routing Stage 1**) is not available, there is an attempt to initiate the connection over the additional entered routing option (**Call Routing**->**Automatic Route Selection**->**Zones &Routing**-> **Edit/Add** -> **Mo-Su** ->**Routing Stage 2**).
	• *3* (default value): If neither of the two entered providers or bundles (**Call Routing**->**Automatic Route Selection**->**Zones &Routing**-> **Edit/Add** -> **Mo-Su** ->**Routing Stage 1** und **Routing Stage 2**) is available, dialling occurs via the provider entered as default for the user(**Numbering**->**Class of Service**->**Add**->**Basic Settings**->**Trunk Line Selection with Line Access Number**).

6.6.2.2 Interfaces / Provider

In the **Call Routing**->**Automatic Route Selection**->**Interfaces / Provider** menu you enter the routes and providers and their network access prefixes.

6.6.2.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Automatic Route Selection**->**Interfaces / Provider**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Routing Mode	Select how dialling shall be externally routed. Possible values: • *Default* (default value): The default procedure provides that when dialling externally, the prefix entered under **Call-prefix** is placed first.

Field	Description
	• *Route*: External dialling is set up via the bundle selected in **Route**.
Call-prefix	Enter the number to be placed as a prefix when making an external call, e.g. to set up a connection via a call-by-call provider.
Route	Only if **Routing Mode** = *Route* Select a bundle via which the external call shall proceed.

6.6.2.3 Zones &Routing

In the **Call Routing**->**Automatic Route Selection**->**Zones &Routing** menu you define the zones via which dialling shall proceed using specific routes or providers.

Configuration of the routing tables for the defined zones occurs individually for each week-day. For 2 routing tables, routing level 1 and routing level 2 can be created as fallback.

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

6.6.2.3.1 Trunk Numbers

In the **Trunk Numbers** area, enter the numbers or partial numbers of the zones for which you wish to configure the routing tables.

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Zones	Configure the desired external zones which should be dialled via the desired entered provider/routes. Possible values: • *Number/Partial Number*: Enter the number or part of a number identifying a zone. • *Name*: Enter a name for this zone.

6.6.2.3.2 Mon - Sun

In the **Mon** - **Sun** area, select the desired times for each routing level, and the desired route or provider via which outgoing calls shall be routed from the entered time.

Fields in the \<Weekday\> menu

Field	Description
Routing Stage 1	Configure the switching times for routing level 1. For this, first select the **Start Time** from which routing shall occur over a specific interface or a specific network provider, and select the latter under **Interface / Provider**.
Routing Stage 2	Configure the switching times for routing level 2. For this, first select the **Start Time** from which routing shall occur over a specific interface or a specific network provider, and select the latter under **Interface / Provider**.

6.7 Applications

Internal telephone performance features of the system are set up under **Applications**.

6.7.1 Calendar

In the **Applications**->**Calendar** menu, you can decide whether to make new entries or modifications in the calendar.

Every company has fixed business hours. You can enter these in the system's internal calendar. For example, all calls outside of business hours can be signalled to a exchange or an answering machine. During this period, your employees can perform other tasks, without being interrupted by telephone calls. The individual call options of a team are automatically switched through the calendars.

You wish to modify the external calling authorisations after business hours for specific subscribers. In the system configuration, you can set individually for each user whether the authorisation for external calls is automatically switched. The switch occurs according to the data in the assigned calendar.

You can set up five types of calendars in the system. The "Authorisation Class" and "Night Operation" calendars are intended for central switching and can only be set up once. The "Team Signalling" , "Intercom Signalling" and "Redirect to internal/external number" calendars can be set up repeatedly. Several different switching times can be selected for each weekday.

In the configuration, a calendar can be assigned to all performance features for which several options can be defined (e.g. teams) Switching between the individual call options then occurs at the switching times of the assigned calendar.

6.7.1.1 Calendar

In the menu **Applications**->**Calendar**->**Calendar** menu you can view, modify or copy a previously set calendar as well as create new calendars.

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

6.7.1.1.1 General

In the **General** area, you define the name of the calendar to be created.

The menu **Applications**->**Calendar**->**Calendar**->**General** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the calendar.
Application	Select the application for which the calendar shall be used.
	Please note that this field cannot be edited with pre-existing entries. If another application is to be configured, you must create another entry and delete the existing one.
	Possible values:
	• *Team Signalling* (default value): Here, several calendars can be set up.
	• *Doorline Signalling*: Here, several calendars can be set up.
	• *Night Mode*: Here, only one calendar can be set up.
	• *Class of Service*: Here, only one calendar can be set up.
	• *Rerouting for internal/external Number*: Here, several calendars can be set up.
	• *Voice Mail System*: Here, several calendars can be set up.

6.7.1.1.2 Mon - Sun / Exception

In the **Mo - Su** area you set up the switching days and times for this calendar.

The **Applications**->**Calendar**->**Calendar**->**Mo - Su** menu consists of the following fields:

Fields in the <Weekday> menu

Field	Description
Switching Points	Enter the desired switching times. For this, under **Time**, for each weekday select the desired switching points to which switching shall occur from any divergent active switching option in the desired switching options selected under **Action**. Depending on the application, the following switching options are available: • *Team Signalling*: Call option 1 to call option 4 • *Doorcom Signalling*: Door Intercom call option 1 and door intercom call option 2 • *Night Mode*: Night operation on and night operation off • *Class of Service*: Authorisation class by default and authorisation class optional • *Rerouting for internal/external Number*: Redirect option 1 to redirect option 4 • *Voice Mail System*: Action *In the Office* and *Out of Office*
Use settings from	Only if settings have already been performed for a weekday. Select from which weekday the settings should be imported. If you require specific settings for this day, select the option *Individual*.

Exception

In the **Exception** area, select whether holidays shall be taken into account and, if so, how.

The menu **Applications**->**Calendar**->**Calendar**->**Exception** consists of the following fields:

Fields in the Settings holidays menu

Field	Description
Consider public holidays	Select whether appointments entered in the **Applications**->**Calendar**->**Public Holiday** menu shall also be considered in this calendar.

Field	Description
	The function is activated by selecting *Enabled*. The function is disabled by default.
Use settings from	Only if **Consider public holidays** is enabled. Select from which weekday the settings for holidays should be imported. You configure the weekdays in the **Applications**->**Calendar**->**Calendar**->**Mo - Su** menu. If you require specific settings for holidays, select the option *Individual*.
Switching Points	Only for **Use settings from** = *Individual* Enter the desired switching times. For this, under **Time**, select the desired switching points to which switching shall occur from any divergent active switching option in the desired switching options selected under **Action**. Depending on the application, the following switching options are available: • *Team Signalling*: Call option 1 to call option 4 • *Doorcom Signalling*: Door Intercom call option 1 and door intercom call option 2 • *Night Mode*: Night operation on and night operation off • *Class of Service*: Authorisation class by default and authorisation class optional • *Rerouting for internal/external Number*: Redirect option 1 to redirect option 4 • *Voice Mail System*: Action *In the Office* and *Out of Office*

6.7.1.2 Public Holiday

In the **Applications**->**Calendar**->**Public Holiday** menu you can enter public holidays or any special days for which divergent settings should be made via the calendar.

6.7.1.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **Applications**->**Calendar**->**Public Holiday**->**New** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for a holiday.
Date (DD - MM)	Enter the date with day and month in two-digit form. Incorrect entries, e.g. 31.02 are accepted and saved but not executed by the system.

6.7.2 Rerouting

In the **Applications**->**Rerouting** menu, you configure how incoming calls should be handled by default in the system.

6.7.2.1 Rerouting Functions

In the **Applications**->**Rerouting**->**Rerouting Functions** menu you can set up various redirect options for *Immediately*, *On Busy*, *On No Reply* or *On Busy and On No Reply*. You then assign these redirect options to the external connections in the **Numbering**->**Call Distribution**->**Incoming Distribution** menu.

6.7.2.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new rerouting variants.

The **Applications**->**Rerouting**->**Rerouting Functions**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the redirect function.
Type of Rerouting Function	Select the desired exchange function. Possible values:

Field	Description
	• *Immediately* (default value)
	• *On Busy*
	• *On No Reply*
	• *On Busy and On No Reply*

Fields in the On Busy Settings menu

Field	Description
Size of Queue	Only for **Type of Rerouting Function** = *On Busy* or *On Busy and On No Reply* In this field, you can configure the maximum number of subscribers on hold. The queue may include up to 10 subscribers. Additional callers get a busy tone. Possible values are *0* (no queue) to *10*. The default value is *0*.
Take Waiting Calls with	Only for **Type of Rerouting Function** = *On Busy* or *On Busy and On No Reply* Define what callers on hold shall hear (internal or configured music-on-hold, announcement). Possible values: • *MoH Wave 1* to *MoH Wave 18* • *MoH internal 1* (default value) • *MOH internal 2* • *Off*
Max waiting time in the queue	Only for **Type of Rerouting Function** = *On Busy* or *On Busy and On No Reply* Define maximum time a caller can remain on hold. After expiration of this time, the caller shall be transferred to the defined redirect destination. Leave *Endless* for an unlimited queue (equates to the value 0). Disable *Endless* to enter the value you require.

Fields in the On No Reply Settings menu

Field	Description
Time for Rerouting on No Reply	Define the maximum time a caller can remain on hold if they cannot get through to the target number. After expiration of this time, the caller shall be transferred to the defined redirect destination. The default value is *30* seconds.

Fields in the Further Settings menu

Field	Description
Announcement	Select whether the incoming call shall be redirected to an announcement. Possible values: • *Off* (default value): The incoming call is not redirected to an announcement. • *MoH Wave 1* to *MoH Wave 18*
Target Number	Select the internal number to which the incoming call shall be redirected. Possible values: • *No Number (Disconnect)*: The call is terminated, the connection ended. • *<Extension number>*: If a destination number is entered, the call is forwarded.
Transfer with	The caller hears the defined announcement or music while her call is being transferred. Possible values: • *Ring tone* • *MoH Wave 1* to *MoH Wave 18* • *MOH internal 1* • *MOH internal 2* • *<Wave file>*

Announcement before query

You have set up a general information call number which customers with various problems or requests ring up. Naturally, no single employee or team can provide information in every subject areas. So the caller would need to be transferred to the individual departments. If you knew beforehand which requests (subject area) a caller had, you could immediately transfer him to the competent department. Thus, your callers don't have to be initially accepted and transferred by an exchange. Every caller decides for him/herself with which employee he/she wishes to be connected.

With the performance feature **Auto Attendant with DISA** calls are automatically accepted by the system. The caller then hears an announcement with information about which entries are possible during or after the announcement. Once the entry is made, the announcement ends and the caller is transferred to an internal subscriber or team. If the caller provides a false or no entry, he/she is transferred to the defined redirect destination (internal subscriber or team). While being transferred, the caller hears a ring tone or the system's music-on-hold.

> **Note**
>
> DISA - Direct Inward System Access Once a call is received by the system, the caller is automatically transferred after a code number is entered. This code is assigned to an internal number in the system. Entry of a number or code must occur during the announcement. Once the announcement (Wave file) ends, no more entries are accepted. There follows redirect to a defined redirect destination. The performance feature **Auto Attendant with DISA** is an integral part of the system and can accept up to 28 simultaneous calls.

Fields in the Announcement/Auto Attendant Settings menu

Field	Description
Call Switching	Select how incoming calls are to be transferred.
	Possible values:
	• *Announcement without DISA* (default value): The configured announcement is played. There follows either transfer to the configured internal number, or the connection is interrupted and the caller hears the busy tone.
	• *DISA, dial internal numbers*: The caller is prompted to enter an internal number. They are then transferred to that number.
	• *DISA, dial code numbers* : The caller is prompted to enter a code number from 0 to 9. The codes are assigned the desired internal numbers. The caller is then transferred to the

Field	Description
	configured internal number.
Number of playbacks	Select how many times the announcement shall be continuously repeated. At conclusion, the caller hears the busy tone.
Auto Attendant with DISA	Only if **Call Switching** = *DISA, dial code numbers* For ever desired DISA code number, select the desired internal number to which the caller shall be transferred.

6.7.2.2 Rerouting Applications

In the **Applications**->**Rerouting**->**Rerouting Applications** menu you can configure when which rerouting variant is to be enabled. You can switch between the different variants either using a calendar or manually.

Choose the icon to edit existing entries. Select the **New** button to create new rerouting applications.

6.7.2.2.1 General

In the **General** area, you make the basic settings for a rerouting application.

The **Applications**->**Rerouting**->**Rerouting Functions**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the rerouting application.
Type of Rerouting Application	Select the destination to which the incoming call is to be rerouted. Possible values: • *Trunk Number* (default value) • *Extension* • *Global*
Switch call signalling	Select how to switch between the variants. Possible values:

Field	Description
	• *No calendar,only manually* • *<Calendar>*

6.7.2.2.2 Variant 1 - 4

You set up the rerouting variants in the **Variant** area. You can create up to four variants.

The menu **Applications**->**Rerouting**->**Rerouting Applications**->**Variant** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Assignment	Select the rerouting function that you wish to assign to the selected variant.

6.7.3 Voice Applications

You configure your system's Wave files in the **Applications**->**Voice Applications** menu.

You wish to employ special music as music-on-hold, or specific announcements for your clients. You can load your self-produced Wave files to the system.

User-specific voice and music files can be saved in the system. Storage space for 2 MoH melodies is available in the system basic settings. Voice and music data is saved in Wave format.

The following voice applications can be defined in the system:

• Announcement before query

• Announcement without query/Infobox

• Wake-up call

• Music on hold

You can find additional information on function, configuration and operation in the description of the individual performance features.

Basic settings of voice applications

The voice applications can be assigned to individual performance features in two different ways.

Every user employing a voice application with this connection always hears the corres-

ponding voice announcement or music from the start. A newly-arrived user hears the voice announcement or music from the start. The number of users who can simultaneously use such a voice application is limited to 28.

Please note that externally played music or voice application music are free of third-party copyrights (GEMA free). Files in other formats must be converted into the company-specific Wave format before being saved in the system.

Note

Please note that Wave files must be available in the following format:

- Bit rate: 128 kbps
- Sampling size: 16 bit
- Channels 1 (mono)
- Sampling rate: 8 kHz
- Audio format: PCM

6.7.3.1 Wave Files

In the **Applications**->**Voice Applications**->**Wave Files** menu, you can configure your announcement/melody files and volume. You also have the option to play back files or download these to your PC. To save a message, click on the ⊟ icon. The download dialog then opens. To listen to a message, click on the ▶ icon.

There are four voice error messages in wave format on the second page. These messages are played if an internal phone tries to claim a line, but the line is unavailable. Error messages are available for a number of error cases, if no specific message is available, the busy signal is played.

6.7.3.1.1 Edit

Choose the ✎ icon to edit existing entries. Select 🗑 to change the entry.

MoH internal 1 and *MoH internal 2* are files specified in the system and can thus not be deleted.

The **Applications**->**Voice Applications**->**Wave Files**-> **Edit** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the Wave file.
Select file	Click **Browse...** and select the Wave file to be loaded into the system through the Explorer window.
Volume	Select the volume at which the Wave file shall be played by default. Select *0* to play the file at a predefined default volume. You can gradually diminish the volume using the negative values, and increase it with the positive ones. Possible values: • *−5* • *−4* • *−3* • *−2* • *−1* • *0* (default value) • *+1* • *+2* • *+3*

6.7.4 System Phonebook

In the **Applications**->**System Phonebook** menu you can enter and administer numbers in the system phone book.

The employees in your company must phone many customers. This is where the system phone book comes in. You need not enter the customer's number but can extract the name via the system telephone display, and dial. Customer names and telephone numbers can be centrally administered by an employee. If a customer whose number has been entered in the phone book calls, his/her name appears in the system telephone display. The system features an integrated phone book in which you can save phone book entries of up to 24-digits (numbers) and up to 20-character names (text).

When creating a telephone book entry a **Speed Dial Number** code is assigned to each entry. Authorised telephones can initiate speed dial from the phone book via these speed dial numbers.

System telephones

System telephones can dial from the system phone book via a special menu. To search for a telephone entry, enter the first letters (max. 8) of the desired name and confirm the entry. The system always provides 8 phone book entries, which you can view successively. Select the desired entry and confirm with **OK**. You must now begin to dial within 5 seconds. The system telephone redialling list displays the name of the dialled subscriber instead of her number. If a system telephone receives a call whose number and name are saved in the system phone book, the caller's name is indicated in the system telephone display.

Note

The user's other numbers (**Mobile Number** and **Home Number**) are only displayed in the system telephone phone book menu. They are not displayed in the **System Phonebook** of the user interface. Entries in the system telephone phone book menu with the (M) mark refer to an entered **Mobile Number** for a user, those with (H) mark to the **Home Number**.

Note

Your PABX system supports LDAP (Lightweight Directory Access Protocol) for providing the entries of the system phonebook to other devices. Name, Number as well as mobile and private numbers can be transferred this way.

6.7.4.1 Entries

The **Applications**->**System Phonebook**->**Entries** menu displays all the phone book entries that have been set up along with the associated speed dial number. The entries in the **Description** column are sorted alphabetically. In any of the columns you can click the column header and can sort the entries in ascending or descending order.

6.7.4.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **Applications**->**System Phonebook**->**Entries**->**New** consists of the following fields:

Fields in the Phonebook Entry menu

Field	Description
Description	Enter a description for the entry. Subsequent sorting in the phone book follows the initial letters of the entry.

Field	Description
Phone Number	Enter the telephone number (internal or external).
Speed Dial Number	Enter a speed dial code. If no speed dial number is entered, the count is automatically continued, i.e. a speed dial code is assigned automatically. A 3 digit sequence of *000* to *999* is possible.
Call Through	Select whether the phone number is to be authorised for the **Call Through** function. If a telephone number is approved for this, and the caller uses this number for the **Call Through** functions, the caller's authorisation to use the function is checked against the phonebook record. The function is activated by selecting *Enabled*. The function is disabled by default.

6.7.4.2 Import / Export

You can import and export phone book data in the **Applications**->**System Phonebook**->**Import / Export** menu. You can import data exported from Microsoft Outlook, for example. The phone book data stored in your device is exported to a text file.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Action	Select the desired action. Possible values: • *Export* (default value): You can export the names saved in **Applications**->**System Phonebook**->**Entries** into a text file (specifying phone number, speed dial, call through). • *Import*: You can import a text file in the following format: The file imported must consist of individual rows in the following format: name, phone number, speed dial, call through (1 = enabled, 2 = disabled). Example: *Name,Phone Number,Speeddial Number,Call*

Field	Description
	Through
	Hans,123456,1,1
	Klaus,234567,2,2
	Max,345678,3,1
Separator	Only for **Action** = *Import* and Default File Format not enabled
	Enter the separator type in the import file.
	Possible values:
	• *Comma* (default value)
	• *Semicolon*
	• *Space*
	• *Tabulator*
Select file	Only for **Action** = *Import*
	Select the file that is to be imported.

You also have the option to import a CSV file.

Example of a CSV file which can be imported

```
"Anrede","Vorname","Nachname","Telefon geschäftlich","Telefon privat"
"Herr","Hans","Meier","+49 (911) 111111","+49 (911) 222222"
"Frau","Emma","will","+49 (911) 333333","+49 (911) 444444"
```

If there are multiple numbers in a data record, you will have the option in the next step to generate two phonebook records from a single data record. To do this, specify the data to be used as the name and phonenumber. If you want to generate only one phonebook entry, select the blank option in all selection fields for the second record **Phonebook Import**.

Fields in the Phonebook Import menu

Field	Description
Phone Number	Select which data is to be used from a data record as the phonenumber.
Name	Select which columns are to be used from a data record as the name. You have the option to use two elements here (e.g. fore-

Field	Description
	name and surname). The middle input field can be used to place a character string between the two elements here. The default separator used is a comma.

Speed dial is automatically assigned. By default, call through is disabled.

6.7.4.3 General

In the **Applications**->**System Phonebook**->**General** menu you define the user name and password for system phone book administration. In the phone book area, the administrator can view and modify the phone book, as well as import and export data.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Web Access Username	Enter a user name for the system telephone book administrator.
Web Access Password	Enter a password for the system telephone book administrator.
Delete Phonebook	If you wish to remove the existing system phone book with all of its entries, enable the option **Delete**. You will then be asked for confirmation **Do you really want to delete all entries of the phonebook?**. Confirm your entry by clicking **OK**. The option **Delete** is disabled by default.
LDAP Service	The LDAP service is needed to make system phonebooks available. If you do not need the function, you can disable the service here. The corresponding protocol port (TCP 389) is then closed then closed, and the system will no longer accept connections on it.

6.7.5 Call Data Records

In the **Applications**->**Call Data Records** menu you configure the recording of incoming and outgoing calls.

The capture of call data records provides an overview of the telephone usage in your company.

All external calls can be saved in the device in the form of call data records. These data records contain important information about the individual calls.

You need to enable the recording of connection data in the **Numbering**->**User Settings**->**Class of Services**->**Applications** menu. The function is not activated in the ex works state.

6.7.5.1 Outgoing

The **Applications**->**Call Data Records**->**Outgoing** menu contains information that permits the monitoring of outgoing activities.

The menu consists of the following fields:

Fields in the Outgoing menu

Field	Description
Date	Displays the connection date.
Time	Displays the time at call start.
Duration	Displays the duration of the connection.
User	Displays the user who called.
Int. No.	Displays the user's internal number.
Called Name	Displays the called name.
Called Number	Displays the dialled number.
Project Code	Displays the call project number, if any.
Interface	Displays the interface over which the external connection was routed.
Costs	Displays the connection charge, but only if the provider transmits the corresponding data.

6.7.5.2 Incoming

The **Applications**->**Call Data Records**->**Incoming** menu contains information that permits the monitoring of incoming activities.

The menu consists of the following fields:

Fields in the Incoming menu

Field	Description
Date	Displays the connection date.
Time	Displays the time at call start.
Duration	Displays the duration of the connection.
User	Displays the user who was called.
Int. No.	Displays the user's internal number.
Caller Name	Displays the caller´s name.
External Number	Displays the caller's number.
Project Code	Displays the call project number, if any.
Interface	Displays the interface over which the connection from outside was routed.

6.7.5.3 General

In the **Applications**->**Call Data Records**->**General** menu, you can define how the connection data is saved in the system.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Web Access Username	Enter a user name for the connection data administrator.
Web Access Password	Enter a password for the connection data administrator.
Save outgoing calls	Select which outgoing connections should be saved. Possible values: • *None* (default value) • *All*

Field	Description
	• *With Project Code only*
Save incoming calls	Select which incoming connections should be saved.
	Possible values:
	• *None* (default value)
	• *All*
	• *With Project Code only*
Privacy Number Truncation	Select whether to save the number in abbreviated form.
	If, for data privacy reasons, the number is to be only partially displayed, you can select the number of positions not to be displayed. You can enter the number of hidden digits separately for **Outgoing Calls** and for **Incoming Calls**. The hiding of digits occurs from right to left.
	Possible values:
	• *No* (default value)
	• *All*
	• *1* to *9*

Fields in the Actions menu

Field	Description
Export call data records	If you wish to save the current connection data record in an external file, click **Export** and save the file under the desired storage location and file name.
Delete call data records	If you wish to delete the current connection data record from the system storage, click **Delete**.

6.7.6 Call List

The menu **Applications**->**Call List** lists details of incoming and outgoing calls. Which kind of calls and how many of them are included can be spcified in the submenu **General**.

6.7.6.1 Incoming

The **Applications**->**Call List**->**Incoming** menu contains information that permits the monitoring of incoming activities.

The menu consists of the following fields:

Fields in the Incoming menu

Field	Description
Date	Displays the connection date.
Time	Displays the time at call start.
Type	Displays the type of the connection.
User	Displays the user who was called.
Int. No.	Displays the user's internal number.
Caller Number	Displays the caller's number.
Trunk Number	Displays the port number.
Interface	Displays the interface over which the connection from outside was routed.
Delete	You can use the **Select all** and **Deselect all** buttons for all the devices displayed.

6.7.6.2 Outgoing

The **Applications**->**Call List**->**Outgoing** menu contains information that permits the monitoring of outgoing activities.

The menu consists of the following fields:

Fields in the Outgoing menu

Field	Description
Date	Displays the connection date.
Time	Displays the time at call start.

Field	Description
Type	Displays the type of the connection.
User	Displays the user who was called.
Int. No.	Displays the user's internal number.
Called Number	Displays the caller's number.
Trunk Number	Displays the port number.
Interface	Displays the interface over which the connection from outside was routed.
Delete	You can use the **Select all** and **Deselect all** buttons for all the devices displayed.

6.7.6.3 General

In the **Applications**->**Call List**->**General** menu, you can define how the connection data is saved in the system.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Record calls	Select which kind of calls are to be included. Possible values: • *None* • *Incoming only* (default value) • *All*
Record connected calls	Select if accepted calls are to be included, too. This can significantly increase the number of included calls and decrease the amount of time the list can cover until the maximum number of calls is reached and the first calls are deleted from the list.
Max Call List entries for System Calls	Specify the maximum amount of system calls that are included in the list. The maximum number is *1000*. System calls include, e.g., call transfers to extern, calls being accepted by an announcement, team calls that are not accepted by a single user.

Field	Description
Max Call List entries per User	Specify the maximum amount of user calls (calls initated of accepted by a configured user) that are included in the list. The maximum number is *200*.

6.7.7 Mini Call Center

The mini call centre is an integrated call centre solution for up to 16 agents. It provides the ideal solution for small groups with high dynamic telecommunication volumes (e.g insides sales, support, order acceptance/processing, customer service). Here, a specific solution with its own administrator has been integrated. The mini call centre is characterised by:

- Flexible allocation of agents and lines
- Dynamic adaptation according to call volume
- Call allocation with off-time for the agent
- Statistical data for agents and lines.

6.7.7.1 Status

In the **Applications**->**Mini Call Center**->**Status** menu, you can view the current status of lines and logged-in agents in a block, along with the subscribers assigned to lines.

The menu consists of the following fields:

Values in the Status list

Field	Description
View	**View** allows you to select which call centre to display.
Line	Displays the mini call centre line.
Agents assigned	Displays the number of agents assigned to this line.
Agents logged on	Displays the number of agents logged-in on this line.
Agents in Wrap-up	Displays the number of agents in post-processing time.
Active Calls	Displays the number of active connections.
Waiting Calls	Displays the number of waiting incoming calls.
Answered of Calls Today	Displays the current number of accepted calls for this day.

Field	Description
Lost Calls Today	Displays the current number of missed calls for this day.

6.7.7.2   Lines

In the **Applications**->**Mini Call Center**->**Lines** menu, lines are assigned to external and internal numbers, and the name of the call centre to which the line belongs is displayed.

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

6.7.7.2.1   General

In the **General** area, you perform basic settings for a line.

The menu **Applications**->**Mini Call Center**->**Lines**->**General** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the line.
External Number	Select a number configured as mini call centre for the external connection of this call centre line.
Internal Number	Enter the desired internal number for this line.
Call Center Description	Select *New* and enter a name for the new mini call centre. Or select the name of a mini call centre which has already been generated.

Fields in the Further Settings menu

Field	Description
Switch call signalling	Select whether the call options for this line shall be switched over a configured calendar and, if so, over which. Possible values: • *No calendar,only manually* • *<Calendar>*
Active Variant	Select which call option shall be enabled by default after config-

Field	Description
	uration for this line.

The menu **Advanced Settings** consists of the following fields:

Fields in the Advanced Settings menu

Field	Description
Team Speed Timer	Enter the time after which call forwarding to the next free agent assigned to this line shall occur.

6.7.7.2.2 Variant 1 - 4

In the area **Variant**, you set up call options for the mini call centre.

The menu **Applications**->**Mini Call Center**->**Lines**->**Variant** consists of the following fields:

Fields in the Settings menu

Field	Description
Automatic Call Pick-up with	Select whether an incoming call shall be automatically accepted and, if so, with which announcement or melody.
	The function is activated by selecting $Enabled$.
	The function is disabled by default.
	Select the Wave file to be used for the call acceptance. All preset and additionally-loaded Wave files in the system can be selected.

Fields in the Rerouting Functions menu

Field	Description
Rerouting on no response	Select whether and, if so, with which option an incoming call shall be redirected after the entered time.
	Possible values:
	• $None$: There shall be no redirect on no-reply.
	• $<Team>$: The incoming call is forwarded to the selected team after the time specified in **Time until rerouting:**.
Further Rerouting	Select additional redirect functions. You must first configure

Field	Description
	these in **Applications**->**Rerouting**->**Rerouting Functions**. Then, the following values may be selected: • *Off*: No additional redirect functions. • *Immediately*: Immediately transfers the call according to a configured redirect function . • *On Busy*: Transfers the call according to a configured redirect function on engaged.
Rerouting Function	Only for **Further Rerouting** = *Immediately* or **Further Rerouting** = *On Busy* Select a configured redirect option for immediate redirect, or on busy.
Busy when	Only for **Further Rerouting** = *On Busy* Select from how many busy agents the lines shall be considered busy.

6.7.7.2.3 Log on / Log off

In the **Log on / Log off** area, select which of the assigned agents shall be logged into the line.

The menu **Applications**->**Mini Call Center**->**Lines**->**Log on / Log off** consists of the following fields:

Fields in the Log on / Log off menu

Field	Description
Numbers	Displays the internal number and description of the assigned agent.
Status	Select whether the agent is logged into the line. The agent is logged in by selecting *Logged on*.

6.7.7.3 Agents

In the **Applications**->**Mini Call Center**->**Agents** menu, lines are assigned to agents. An agent can operate one or more mini call centre lines.

6.7.7.3.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **Applications**->**Mini Call Center**->**Agents**->**New** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
User	Select the configured user who shall serve as a call centre agent. You configure required users in the **Numbering**->**User Settings**->**User** menu. The following users are already created: • *User 1 to User 4 analog phones* • *User 5 and User 6 Sys Tel* • *User 7 DECT* • *User 8 and User 9 ISDN*
Internal Number	Select the user's internal number to be used for the call centre.

Fields in the Assigned Lines menu

Field	Description
Select lines	Select the lines for which the agent shall be responsible. The name of the corresponding call centre is displayed again when lines are selected in order to improve the overview. Under **Assign** select whether the entry should be enabled.

Fields in the Wrap-up Settings menu

Field	Description
Wrap-up Time	Enter the time available to this agent for post-processing after concluding a call. No further call can be assigned to this agent during this period. The agent has the option of temporarily extending the period with a telephone procedure.

6.7.7.4 General

In the **Applications**->**Mini Call Center**->**General** menu, you can set up an HTML web interface access for the mini call centre manager. The latter can then monitor the status of lines and agents, and modify the settings for lines and agents.

The menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Web Access Username	Enter a user name for the mini call centre administrator. When a user logs into the user interface under this name, he/she has access to the user interface with selected parameters for administration of the call centre.
Web Access Password	Enter a password for the mini call centre administrator.

6.7.8 Doorcom Units

You can connect a door intercom as an intercom adapter to an analogue connection of your system.

If a door intercom adapter is connected to your system, you can speak with a visitor at the door from every authorised telephone. You can assign particular telephones to each ring button. These phones then ring if the ring button is pressed. On analogue telephones, the signal on the telephone matches the intercom call. In place of the internal telephones, an external telephone can also be configured as the call destination for the ring button. Your door intercom can have up to 4 ring buttons. The door opener can be pressed during an intercom call. It is not possible activate the door opener if an intercom call is not taking place.

> **Note**
>
> All functions of the door intercom (intercom adapter) are controlled via the code numbers indicated in the intercom user's manual. The system does not support the intercom with specific codes.

6.7.8.1 Doorcom Units

In the **Applications**->**Doorcom Units**->**Doorcom Units** menu, select the internal analogue connection (FXS) to which a doorcom unit shall be connected. Then dial the internal number for the connection, and optionally the codes for call acceptance.

6.7.8.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

If you intend to add **Doorcom Units**, you may first have to free an interface in the menu **Terminals**->**Other phones**->**Analogue**, i.e. delete one of the preconfigured entries with the 🗑 button.

The menu **Applications**->**Doorcom Units**->**Doorcom Units**->**New** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Interface	Select the interface to which an intercom adapter shall be connected. All free FXS interfaces are available.
Internal Number	Select the configured internal number to be assigned to the intercom adapter. The number is set up in the **Numbering**->**User Settings**->**User** menu.
Code for Doorcom Call Acceptance	Pressing a bell button on the intercom sets off a call in the system. To establish a connection between a called subscriber and the intercom adapter, that subscriber must pick up the receiver and dial the code number for call acceptance. Enter this code for call acceptance. If a subscriber accepts a call from the intercom adapter, the PABX automatically dials the code number required to set up the connection. The subscriber need not make any more entries.

6.7.8.2 Doorcom Signalling

In the **Applications**->**Doorcom Units**->**Doorcom Signalling** you configure the signalling variants for receiving calls via a doorcom unit. Two intercom call options are available.

The code number for the bell button is the number the intercom adapter dials into the system when the bell button is pressed. You can perform an internal call allocation for each bell button. Please note that guidelines for connecting the intercom adapter depend on the respective manufacturer. For this, read the operating instructions provided by the manufacturer of the intercom adapter.

6.7.8.2.1 General

In the **General** area you set up the basic features of intercom signalling.

The menu **Applications**->**Doorcom Units**->**Doorcom Signalling**->**General** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Select one of the configured intercom settings previously created in the **Applications**->**Doorcom Units**->**Doorcom Units** menu.
Bell ID	Enter an unambiguous four-digit code for the bell. Pressing a bell button on the intercom adapter initiates a call to the terminals entered in the assigned intercom call option.
Bell Name	Enter a name for the bell.
Switch signalling	Select whether the intercom call options for this bell shall be switched over a configured calendar and, if so, over which. For each bell, you can create up to two doorcom signalling variants in the **Applications**->**Doorcom Units**->**Doorcom Signalling**->**New**->**Variant** menu. Possible values: • *No calendar,only manually* • *<Calendar>*
Active Doorcom Variant	Select which intercom call option shall be enabled by default for this bell after configuration.

The menu **Advanced Settings** consists of the following fields:

Fields in the Advanced Settings menu

Field	Description
Call Signalisation Timer	Enter the time in seconds for which the door intercom call shall be signalled. The default value is *40* seconds.
Team Speed Timer	Here, enter the **Team Speed Timer** following which call forwarding after time shall be performed. The default value is *15* seconds.
Simultaneous after time	It is possible for all numbers assigned to this door intercom signalling to be called simultaneously after a specified time.

Field	Description
	The default value is *60* seconds.

6.7.8.2.2 Doorcom Signalling Variant 1 and 2

In the **Doorcom Signalling Variant** area you configure the two doorcom call options for this signalling profile.

The menu **Applications**->**Doorcom Units**->**Doorcom Signalling**->**Doorcom Signalling Variant** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Assignment	Select where pressing of the bell button shall be signalled. Possible values: • *Internal*: Signalling occurs on an internal number. • *External*: Signalling occurs to an external number.
Internal Assignment	Select the internal numbers on which pressing of the door bell shall be signalled. With **Add** you add an internal number.
External Assignment	Enter the external telephone number to which pressing the door bell shall be signalled.
Signalling	You can call the internal number with a broadcast call. Possible values: • *Simultaneous* (default value): All assigned terminals are called simultaneously. If a telephone is busy, call waiting can be used. • *Linear*: All assigned terminals are called in the sequence of their entry in the configuration. If a terminal is engaged, the next free terminal is called. The call is signalled ca. 15 seconds per subscriber. The period can be set between 1 and 99 seconds (per bell) in the configuration. If subscribers are on the phone or logged out, there is not forwarding time for these. • *Rotating*: This call is a special case of the linear call. After all terminals are called, call signalling begins again with the first entered terminal. The call is signalled until the caller re-

Field	Description
	places the receiver or the call is ended by the intercom adapter (after ca. 2 minutes).
	• *Adding*: The terminals are called in the sequence of their entry in the configuration subscriber list. Every terminal that has already been called is called again, until all entered terminals are called. In the configuration, you can define when each next terminal is called.
	• *Linear,Simultaneous after time*: You have set linear for the door intercom call. After the defined time has run out, you can also set in the configuration that all team subscribers are then called in parallel (simultaneously).
	• *Rotating,Simultaneous after time*: You have set rotating for the door intercom call. After the defined time has run out, you can also set in the configuration that all intercom subscribers are then called in parallel (simultaneously).

6.7.9 Voice Mail System

The voicemail system is an intelligent answering machine for those who use your PABX. An individual voicemail box can be configured for each extension. All subscribers can hear, save or delete their messages from any telephone using a personal PIN code.

Subscribers can have themselves informed of incoming e-mails. Recorded messages can be automatically transferred to any e-mail address.

General settings of the voicemail system are performed on your PABX. Operation of the individual voicemail boxes occurs via telephone.

Every subscriber can use her individual voicemail box by transferring calls to her voicemail box.

> **Note**
>
> Choose in the **Maintenance**->**Software &Configuration** menu the option *Import Voice Mail Wave Files*.

6.7.9.1 Voice Mail Boxes

The **Applications**->**Voice Mail System**->**Voice Mail Boxes** menu displays a list with the individual voicemail boxes of the individual subscribers, where voicemail boxes have been configured.

Values in the Voice Mail Boxes list

Field	Description
Internal Number	Displays the number of the individual subscriber for which the voicemail box is configured.
User	Displays the name of the individual subscriber for which the voicemail box is configured.
Language	Displays the language of the announcement text on the voice-mail box. *Default* means that the centrally-set language, defined for the entire voicemail system in the **Applications**->**Voice Mail System**->**General** menu, is used.
Notification	Indicates whether the subscriber is informed of missed calls.
Active Variant	Displays the current status of the voicemail box (*In the Office* or *Out of Office*.

6.7.9.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **Applications**->**Voice Mail System**->**Voice Mail Boxes**->**New** consists of the following fields:

Fields in the Basic Setup menu

Field	Description
Internal Number	Select the internal number of the subscriber for which you wish to set up a voicemail box. You may choose among the numbers configured in the **Numbering**->**User Settings**->**User** menu.
Status	Select when incoming calls shall be forwarded to the specified internal number. Possible values:

Field	Description
	• *Off*
	• *Immediately*
	• *On Busy*
	• *On no reply*
	• *On busy / On no reply*
No Reply Time	Only for **Status** = *On no reply* and *On busy / On no reply*
	Define the maximum time a caller can remain on hold if they cannot get through to the target number. After expiration of this time, the caller shall be transferred to the defined redirect destination.
	The default value is *15* seconds.
Voice Mail Language	Select the desired language for the voicemail box announcements.
	Possible values:
	• *Deutsch*: The voicemail box uses German texts.
	• *Dutch*: The voicemail box uses Dutch texts.
	• *English*: The voicemail boxe uses English texts.
	• *Italian*: The voicemail box uses Italian texts.
	• *Spanish*: The voicemail box uses Spanish texts.
	• *French*: The voicemail box uses French texts.
	• *Portugues*: The voicemail box uses Portugues texts.
	• *Turkish*: The voicemail box uses Turkish texts.
	• *Default* (default value): The voicemail box uses the language centrally defined for the entire voicemail system in the **Applications**->**Voice Mail System**->**General** menu.
	Note
	You'll only require a setting that departs from *Default* if you wish to operate voicemail boxes with various languages within your voicemail system.

Field	Description
E-Mail Address	Here is displayed the user e-mail address to which a notification shall be sent if a message has been left on the voicemail box. The email address is created in the **Numbering**->**User Settings**->**User**->**Basic Settings** menu.
E-Mail Notification	Once a message has been left on the voicemail box, the subscriber can be notified. Possible values: • *None* (default value): The subscriber is not notified. • *E-Mail*: The subscriber is informed of a present message via e-mail. • *E-Mail with Attachment*: Once a caller has left a message, the subscriber receives an e-mail with a recording of the message in the attachment. • *User defined*: If the administrator activates the *User defined* function, the e-mail alert settings can be changed by the user in the **User Access**. If the administrator sets a different value, a block is placed on changes from the user.

> **Note**
>
> Once a subscriber has received notification of a new message in an e-mail, the **Status** of the notification is changed according to the settings in the **User Access**. You can configure the status behaviour in the **User Access**->**Voice Mail System**->**Settings** menu under **E-Mail forwarding behavior**.

Field	Description
Max Recording Time	Enter the maximum recording time per message. The possible values are *5* to *300* seconds, the default value is *180* seconds.
Calendar for status "Out of Office"	When the subscriber is out, the voicemail box can be switched over a calendar. If a calendar is to be used, it needs to be configured in the **Applications**->**Calendar** menu with the setting **Application** = *Voice Mail System* Possible values:

Field	Description
	• *No calendar,only manually* (default value): The subscriber can manually switch the voicemail box on and off.
	• *<Calendar>*: Using the selected calendar, the voicemail box can be switched on or off at the times defined there.

Fields in the User Settings menu

Field	Description
Status of Mail Box Owner	Define in which mode the mailbox shall be used when starting the voicemail system. Possible values: • *In the Office* (default value): Select this setting if the subscriber is in the office when the voicemail system is started. • *Out of Office*: Select this setting if the subscriber is out of office when the voicemail system is started.
Check PIN	Select whether the currently configured voicemail box should be protected with a PIN. The function is enabled with *Enabled*. The function is disabled by default. The PIN for the personal Voice Mail Box you can change in the **Numbering**->**User Settings**->**Users**->**Authorizations** menu under **PIN for Phone Access**.
PIN	Only when setting up a Team Voice Mail Box. The PIN is required to set up the voice mail box. If the voice mailbox is being monitored over a telephone, this PIN must be entered.
Mode for status "In the Office"	The voicemail box can be operated with two different settings during office hours. Possible values: • *Announcement and Record* (default value): A caller hears an announcement and can leave a message. • *Announcement only*: A caller hears an announcement, but

Field	Description
	cannot leave a message.
Mode for status "Out of Office"	The voicemail box can be operated with two different settings outside of office hours. Possible values: • *Announcement only* (default value): A caller hears an announcement, but cannot leave a message. • *Announcement and Record*: A caller hears an announcement and can leave a message.

Fields in the Authorizations menu

Feld	Beschreibung
Login Name	Here you can change the permissions for external calls. The user selected here also receives user access to his personal voice mail box.

Select the ✎ icon to set up your own voice announcements for the selected voice mail box in addition to the settings above.

The **Applications**->**Voice Mail System** ->**Voice Mail Boxes** -> ✎ menu consists of the following fields:

Fields in the menu Voice Announcement

Feld	Beschreibung
In the Office	You can upload your own announcement for the **In the Office** state. Use the WAV file format for this annoncement. Click **New Message** to upload the file. The window **Annoucement Options** opens. If an announcement has been stored, use the ▶ icon to play the announcement, the 🗑 icon to delete it.
Out of Office	You can upload your own announcement for the **Out of Office** state. Use the WAV file format for this annoncement. Click **New Message** to upload the file. The window **Annoucement Options** opens.

Feld	Beschreibung
	If an announcement has been stored, use the ▶ icon to play the announcement, the 🗑 icon to delete it.

Fields in the menu Annoucement Options

Feld	Beschreibung
Action	Displays *Update announcement*.
Source Location	For **Action** = *Update annoncement* Select the WAV file to be used for the announcement and click **Start** to upload.

> **Note**
>
> Please note that Wave files must be available in the following format:
>
> • Bit rate: 128 kbps
> • Sampling size: 16 bit
> • Channels 1 (mono)
> • Sampling rate: 8 kHz
> • Audio format: PCM

6.7.9.2 Status

In the **Applications**->**Voice Mail**->**Status** menu, the status of the individual voicemail boxes for specific subscribers is indicated. You can see how many calls have gone into which voicemail box, and how many "old" calls are already present.

Values in the System Messages list

Field	Description
Internal Number	Displays the number of the individual subscriber for which the voicemail box is configured.
User	Displays the name of the individual subscriber for which the voicemail box is configured.
New Calls	Displays the calls which have not yet been listened to by the subscriber.

Field	Description
Old Calls	Displays the calls which have already been listened to by the subscriber.

Note

By default, the system can record a maximum of 59 calls per voicemail box. You cannot change this value in the GUI.

6.7.9.3 General

In this menu, you can configure the general settings for your voicemail system. In the section **Import voice mail wave files** , you can upload a ZIP file with predefined greetings to control the voice mail system. Voice Mail system. You can find the corresponding one in the download section of your device. Individual announcements can be created when setting up the individual voice mail box.

The menu **Applications**->**Voice Mail**->**General** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Voice Mail System	Select whether to activate your voicemail system. The function is enabled with *Enabled*. The function is disabled by default.
Description	Only for **Voice Mail System** enabled. Enter a description for your voicemail system. This description is displayed on the telephone when a call goes in to the voice mail system. The default value is *Voice Mail*.
Internal Number	Only for **Voice Mail System** enabled. Enter the internal number under which to access your voicemail system. The default value is *50*.

Field	Description
Language	Select the language for the entire voicemail system. Possible values: • *Deutsch* (default value) • *Dutch* • *English* • *Italian* • *Spanish* • *French* • *Portugues* • *Turkish*h Diverging from the language set here, a language can be individually set for each voice mail box in the **Applications**->**Voice Mail System**->**Voice Mail Boxes**->**New** menu.

Fields in the Mail Settings menu

Field	Description
SMTP Server	Enter the address (IP address or valid DNS name) of the e-mail server to be used for sending the e-mails.
Return Address	Enter any address to be used as sender when sending e-mails. This address merely serves to identify e-mails in the inbox.
SMTP User Name	Enter the user name for the SMTP server.
SMTP Password	Enter the password for the SNMP server user.

The menu **Advanced Settings** consists of the following fields:

Fields in the Advanced Settings menu

Field	Description
Lifetime	Voicemail messages are deleted after an adjustable period of time if the maximum number of messages has been reached for a voice mail box. If the maximum number of messages is reached before the specified time has expired, the next caller cannot leave a message and will be notified if applicable.

Field	Description
	Possible values are *10* to *60* days. The default value is *60*.
Non-standard SMTP Server port	Enter the port to be used for sending e-mails.

The default value is *25*. |

6.8 Maintenance

6.8.1 Update System Phones

In the **Maintenance**->**Update System Phones** menu you can update the software for your system telephones.

6.8.1.1 elmeg OEM

In the **Maintenance**->**Update System Phones**->**elmeg OEM** menu, you will see a list of the connected elmeg OEM telephones or base stations. This view displays both elmeg IP1x telephones and elmeg DECT base stations, if there are any. You can select devices to have their software updated immediately or allow them to download completely new software from the system.

In the case of immediate updating, there is no version control.

> **Note**
>
> Note that immediate software updates for DECT multi-cell systems are only available via the system's web configurator and that they cannot be initiated by the PABX GUI.

Values in the list Update from external Server

Field	Description
Automatic Update from external Server	Enable or disable the automatic update from external server function.

The function is enabled with *Enabled*.

The function is enabled by default. |
| **Description** | Displays the description entered for the system telephone. |

Field	Description
Phone Type	Displays the system telephone type.
MAC Address	Shows the system telephone's MAC address.
Phone Version	Displays the software version of the telephone.
Status	Displays the system telephone status, or a progress bar during the update progress. ⬤ identifies a connected system telephone whose system software is supported by your PABX. ⬤ identifies a system telephone that is either not connected, or whose system software is not supported by your PABX.
Update immediately	Displays whether the system telephone software should be updated immediately. This function is enabled on an individual device by setting a checkmark. The function is disabled by default. You can use the **Select all** and **Deselect all** buttons for all the devices displayed.

6.9 Monitoring

This menu contains information that enable you to locate problems in your network and monitor activities, e.g. at your device's WAN interface.

6.9.1 Status Information

This menu displays current settings for terminals and team subscribers. This data is continuously read out.

6.9.1.1 Users

In the **Monitoring**->**Status Information**->**Users** menu the current settings for a user's internal number (MSN) are displayed.

6.9.1.1.1 Users - Details

By pressing the ⌕ button, you display detailed statistics on the respective user.

Values in the Extension Status list

Field	Description
Number	Displays the user's internal number.
Name	Displays the name assigned to the user. If a voicemail system is active, *Voice Mail System* is displayed.
Current Class of Service	Displays the all of the authorisation classes assigned to the user. The currently enabled authorisation class is marked appropriately with a green arrow ( ).
Terminal	Displays the interface assigned to this subscriber.
IP Address	Displays the IP address.
Registration	Displays whether the user is registered.
Charges	Displays calculated charges for accrued connection units.

Values in the System Settings list

Field	Description
Parallel Ringing	Displays whether parallel call is set up for the user.
Call Forwarding	Displays current call forwarding for this user.
Direct Call	Displays whether direct call on receiver pickup is configured for the user.
Room Monitoring	Displays whether room monitoring is enabled for the user.

6.9.1.2 Teams

In the **Monitoring**->**Status Information**->**Teams** menu, current team settings are displayed.

6.9.1.2.1 Teams - Details

By pressing the ⌕ button, you display detailed statistics for the respective team.

Values in the Team Status list

Field	Description
Name	Displays the name assigned to the team.

Field	Description
Number	Displays the team's internal number.
Users assigned/Users logged on	Displays the users assigned to the team, and how many of these users are logged in.
Call Forwarding	Displays current call forwarding for this team.

Values in the System Settings list

Field	Description
Active Variant (Day)	Displays the currently enabled call option for this team.
Switch call signalling	Displays whether the call option can be switched manually, over the calendar or manually and over the calendar.
Signalling	Displays the type of call signalling in the team.
Busy on busy	Displays whether busy on busy is configured for the team.
Automatic Call Pick-up	Displays whether automatic call acceptance is configured, and which melody is played.
Rerouting on no re-sponse	Displays whether redirect on no reply is enabled and, if so, the time period after which it occurs and the destination team.
Further Rerouting	Displays which of the redirect functions are enabled and which subscriber is the redirect destination.

Chapter 7 Telephony (Media Gateway)

7.1 Physical Interfaces

7.1.1 ISDN Ports (Media Gateway)

7.1.1.1 ISDN Internal

In the **Telephony**->**Physical interfaces**->**ISDN ports**->**ISDN Internal** menu, you can see an overview of the settings for the two internal ISDN ports. You can edit the entries and operate the ports either as point-to-multipoint or point-to-point connections.

7.1.1.1.1 Edit

In the **Edit** menu, you simply set the **ISDN Configuration Type** of the selected port.

7.2 VoIP (Media Gateway)

Voice over IP (VoIP) uses the IP protocol for voice and video transmission.

The main difference compared with conventional telephony is that the voice information is not transmitted over a switched connection in a telephone network, but divided into data packets by the Internet protocol and these packets are then passed to the destination over undefined paths in a network. This technology uses the existing network infrastructure for voice transmission and shares this with other communication services.

The Session Initiation Protocol (SIP) is used to establish, clear and control a communication session.

7.2.1 Settings

7.2.1.1 SIP Provider

If you want to connect your device to other SIP servers (e.g. servers of Internet SIP service providers), you can configure the necessary entries here. In this case, the media gateway acts as a SIP client.

You can also configure the entries for SIP trunking scenarios here. In this case, the media gateway acts as a SIP server for other SIP servers. An example of this is the connection of

a SIP PBX (e.g. Asterisk) to the media gateway.

This means that all SIP provider accounts are configured here as well as direct dial-in tele-phone systems connected to the media gateway.

Note

Do not use this menu to configure SIP clients or PSTN clients such as SIP phones, ter-minal adapters or ISDN phones. These must be set up on the connected telephone system.

The **VoIP**->**Settings**->**SIP Provider** menu diusplays a list of all available SIP Providers (SIP Client Mode and SIP Server Mode).

7.2.1.1.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **VoIP**->**Settings**->**SIP Provider**->**New** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	You can enter a designation for the SIP provider. A 20 digit al-pha-numeric sequence is possible.
Provider Status	Select whether this VoIP provider entry is enabled (*Enabled*, standard value) or not (*disabled*).
SIP Mode	Select whether the SIP account should be operated and in which SIP mode. Which setting can be used depends on the provider. Possible values: • *Client*: The media gateway is operated as a SIP client. It re-gisters with a registrar and is assigned an individual call num-ber or a system connection number. • *Server*: The media gateway is operated as a SIP server so that SIP clients can connect.
Access Type	Select whether the SIP provider provides a single number port or an extension port.

Field	Description
Authentication ID	Enter your provider's authentication ID. A 64 digit alpha-numeric sequence is possible.
Password	At this point, you can assign a password. A 64 digit alpha-numeric sequence is possible.
User Name	Enter the user name you received from your VoIP provider. A 64 digit alpha-numeric sequence is possible.
Domain	Enter a new domain name or a new IP address for the SIP proxy server. If you do not make an entry, the entry in the **Registrar** field is used. Note: Enter a name or IP address only if this is explicitly specified by the provider.

Fields in the Registrar menu

Field	Description
Registrar	Enter the DNS name or IP address of the SIP server. A 26 digit alpha-numeric sequence is possible.
Registrar Port	Enter the number of the port to be used for connection to the server. The default value is *5060*. A 5 digit sequence is possible. If you prefer a DNS SRV request instead of a DNS A record request for this provider, enter port *0* here. For connections offered by Telekom this is a required setting because the SRV entry provides additional server addresses which may provide a better service quality. SIP provider that are created with the Initial Operation menu or with the Telephony assistant automatically use the correct port number.
Transport Protocol	Select the transport protocol for the connection. Possible values: • *UDP* (default value) • *TCP*

Field	Description
	• *TLS*
	• *Automatic* - With this setting, your device supports automatic negotiation of the protocol with your provider's servers. For this setting to work, this negotiation must also be supported by the provider.

Fields in the STUN menu

Field	Description
STUN server	Enter the name or the IP address of the STUN server.
	STUN = Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
	A STUN server is required to allow VoIP devices access to the internet behind an active NAT. This determines the current public IP address for the connection and uses this for remote addressing.
	Maximum number of characters: 32.
Port STUN server	Enter the number of the port to be used for the connection to the STUN server.
	The default value is *3478*. A 5 digit sequence is possible.

Fields in the Timer menu

Field	Description
Registration Timer	Enter the time in seconds within which the SIP client must re-register to prevent the connection from disconnecting automatically.
	The default value is *600*.

The menu **Show more** consists of the following fields:

Fields in the Proxy menu

Field	Description
Proxy	Enter the DNS name or IP address of the SIP server. A 26 digit alpha-numeric sequence is possible.

Field	Description
Proxy Port	Enter the number of the port to be used for connection to the proxy. The default value is *5060*. A 5 digit sequence is possible.
Transport Protocol	Select the transport protocol for the connection. Possible values: • *UDP* (default value) • *TCP* • *TLS* • *Automatic* - With this setting, your device supports automatic negotiation of the protocol with your provider's servers. For this setting to work, this negotiation must also be supported by the provider.

Fields in the Codec Settings menu

Field	Description
Codec Profiles	Select the location of the SIP server. Locations are defined in the**VoIP**->**Settings**->**Codec Profiles** menu. Possible values: • *System Default* (default value): The server is not operated at any defined location. • *<Codec-Profil-Name>*
Video	Select if calls between IP telephones are to support the transmission of video data. Video transmission can only be negotiated between the participants if both support this feature.
SRTP	Select if calls via this SIP provider may be secured with SRTP (Secure Real-Time Transport Protocol).
MediaSec	*MediaSec*: MediaSec regulates the protection of SIP data between the SIP server and your system. For seamless support, automatic negotiation of the transport protocol is mandatory. Fixed transport protocol settings (UDP and TCP) may cause problems during registration. Additionally, the use of SRTP must be allowed. Your VoIP provider must support MediaSec.

Fields in the Further Settings menu

Field	Description
From Domain	Enter the SIP provider's "From Domain". It is used after the @ as sender data in the SIP header of the SIP data packages.
Number of allowed simultaneous Calls	Select the maximum number of calls that shall be simultaneously possible Please also note the settings for bandwidth management here. Possible values: • *International* (default value): An unlimited number of simultaneous calls is possible. • *1* • *2* • *3* • *4* • *5* • *10*
SIP interface binding	Select the location of the SIP server. Locations are defined in the **VoIP**->**Settings**->**Locations** menu. Possible values: • *Any Location* (default value): The server is not operated at any defined location. • *<Location Name>*
Dial End Monitoring Time	Select the time in seconds (after dialling the last digit of a call number) after which the system begins external dialling. The default value is *5*.
Call Forwarding extern (SIP 302)	Select whether calls are to be redirected externally with the SIP provider. The call is forwarded using SIP status code 302. The function is activated by selecting *Enabled*. The function is disabled by default.
SIP-Header for anonymous call signaling	In order to ensure that your system can forward anonymous calls with SIP connections you can specify in which part of the SIP header information the string "anonymous call" is transferred. The information can be transferred in multiple parts. For

Field	Description
	most prviders you can simply keep the preconfigured setting *Privacy ID* = *Enabled* . For the service provider 1 & 1 you need to additionally enable *Privacy Header* . Possible values: • *Display* • *Users* • *Domain* • *Privacy Header* • *Privacy User* • *Privacy ID*
SIP Header Field: FROM Display	Not for **Trunk Mode** = *Off* The sender ID is placed in the "Display" field of the SIP header. Possible values: • *None* (default value): The sender ID is not sent. • *Username*: The user-configured user name is displayed. • *Caller Address*: The user-configured number the called party is displayed. • *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.
SIP Header Field: FROM User	Not for **Trunk Mode** = *Off* The sender ID is sent in the "User" field of the SIP header. Possible values: • *Username*(default value): The user-configured user name is displayed. • *Caller Address*: The user-configured number the called party is displayed. • *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.
SIP Header Field: P-Preferred	Not for **Trunk Mode** = *Off* The so-called "p-preferred-identity" field is added to the SIP

Field	Description
	header and contains the sender ID.
	Possible values:
	• *None* (default value): The sender ID is not sent.
	• *Username*: The user-configured user name is displayed.
	• *Caller Address*: The user-configured number the called party is displayed.
	• *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.
SIP Header Field: P-Asserted	Not for **Trunk Mode** = *Off*
	The so-called "p-asserted-identity" field is added to the SIP header and contains the sender ID.
	Possible values:
	• *None* (default value): The sender ID is not sent.
	• *Username*: The user-configured user name is displayed.
	• *Caller Address*: The user-configured number the called party is displayed.
	• *Billing Number*: The actual phone number from which the calls is initiated (e.g. for billing purposes) is displayed.
SIP Registration behavior after Restart	If after registering with a provider a reset of the system should occur, for example, or a power failure, depending on the provider, another registration may prove impossible. By enabling **Clear Bindings** or **Fetch Bindings**, you can enable re-registration.
Upstreaming Device with NAT	If you enable this function, you can use a gateway with NAT and still make VoIP calls. Without this function, it may not be possible to call you with VoIP if you use a gateway with NAT.
	The function is activated by selecting *Enabled*.
	The function is disabled by default.
Early media support	Select whether you'll allow exchange of voice and audio data before a receiver accepts a call.
	The function is enabled with *Enabled*.

Field	Description
	The function is enabled by default.
Registration type	Specify how registration and authentication at a provider are to be handled, or if they can omitted completely. In the latter case, the relevant data are sent to a particular IP address that is already known to the correspondent. Registration and authentication are not then needed and the Registration function is disabled. An example of this method is Microsoft Exchange SIP. If no registration is required, the device can send **Periodic Options** to maintain the connection. If a registration is required, it can be carried out in either of two ways: • *Single*: With this option, a single MSN is registered with the SIP provider. • *Bulk (BNC)*: With this option, a SIP Trunk (DDI) is registered with the SIP provider, i.e. several numbers are registered under a single address.
T.38 FAX support	Select whether faxes shall be transmitted with T.38. The function is enabled with *Enabled*. The function is enabled by default. If the function is disabled, faxes are transmitted with G.711.
Send SIP UPDATE	This function ensures that after a call transfer the number of the new call partner is displayed for the initial calling party. **Note** Note that this function is not supported by all service providers. *Enabled* activates the functions. The function is not enabled by default.
Called number retrieved over	In some applications (especially in DDI connections) the target address of a SIP call needs to be extracted from the **Request**

Field	Description
	URI. By activating this option the address is preferably read from this field of the invite.
Check Source IP	As a response to a DNS SRV request, your SIP provider transmits the addresses of valid registration servers. If you activate this option, each SIP invite has its source IP checked against these valid addresses. If it does not originate from one of them, the invite is ignored. The option is not active per default.
TLS certificate check	Only for DDI / SIP trunk connections. If a connection is encrypted using TLS (Transport Layer Security) a validity check on the server certificate of the remote station is performed. The option is not active per default.
Special SIP REFER Handling	Some configurations - e.g. to set up a SWXY connection - require specific handling of SIP REFER methods. You can activate these here.

7.2.1.2 Locations

In the **VoIP**->**Settings**->**Locations** menu you configure the locations of the VoIP subscribers who have been configured on your system, and define the bandwidth management for the VoIP traffic.

Individual locations can be set up for using the bandwidth management. A location is identified from its fixed IP address or DynDNS address or from the interface to which the device is connected. The available VoIP bandwidth (up- and downstream) can be set up for each location.

Fields in the Registration behavior for VoIP subscribers without assigned location menu.

Field	Description
Default Behavior	Specify how the system is to proceed when registering VoIP subscribers for whom no location has been defined.
	Possible values:
	• *No Registration*: The VoIP subscriber is never registered.
	• *Registration for Private Networks Only* (default value): The VoIP subscriber is only registered if located within the private network.
	• *Unrestricted Registration*: The VoIP subscriber is al-

Field	Description
	ways registered.

Fiels in the Registration behavior for VoIP providers without assigned location menu

Feld	Beschreibung
Default Interface	VoIP providers that are not assigned a location require an interface for connection setup. Here you can set one of the created interfaces as default for this case.
Standard interface for VoIP backup	If you have set up a VoIP backup connection, you can specify here which interface should be used to establish the connection.

> **Note**
>
> Note that the VoIP backup setting will not be automatically reset if the corresponding Internet connection is no longer available. You can then switch it to the primary connection here.

7.2.1.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **VoIP**->**Settings**->**Locations**->**New** consists of the following fields:

Fields in the Basic Settings menu.

Field	Description
Description	Enter the description of the entry.
Parent Location	You can cascade the SIP locations as you wish. Define here which SIP location that has been defined constitutes the high-level node for the SIP location to be configured here.
Type	Select whether the location is to be defined through IP addresses/DNS names or interfaces. Possible values: • *Addresses* (default value): The SIP location is defined via IP addresses or DNS names. • *Interfaces*: The SIP location is defined via the available in-

Field	Description
	terfaces.
Addresses	Only for **Type** = *Addresses* Enter the IP addresses of the devices at the SIP locations. Click **Add** to configure new addresses. Enter the IP address or DNS name that you want under **IP Address/DNS Name**. Also enter the required **Netmask**.
Interfaces	Only for **Type** = *Interfaces* Indicate the interfaces to which the devices of a SIP location are connected. Click **Add** to select a new interface. Under **Interface**, select the interface you want.
Upstream Bandwidth Limitation	Determine whether the upstream bandwidth is to be restricted. The bandwidth is reduced with *Enabled*. The function is disabled by default.
Maximum Upstream Bandwidth	Enter the maximum data rate in the send direction in kBits per second.
Downstream Bandwidth Limitation	Determine whether the downstream bandwidth is to be restricted. The bandwidth is reduced with *Enabled*. The function is disabled by default.
Maximum Downstream Bandwidth	Enter the maximum data rate in the receive direction in kBits per second.

The menu **Advanced Settings** consists of the following fields:

Fields in the DSCP menu.

Field	Description
DSCP Settings for rtp Traffic	Select the Type of Service (TOS) for RTP data. Possible values:
	• *DSCP Binary Value* (default value): Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in binary format, 6 bit). The pre-configured value is *101110*.
	• *DSCP Decimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in decimal format).
	• *DSCP Hexadecimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in hexadecimal format).
	• *TOS Binary Value*: The TOS value is specified in binary format, e.g. 00111111.
	• *TOS Decimal Value*: The TOS value is specified in decimal format, e.g. 63.
	• *TOS Hexadecimal Value*: The TOS value is specified in hexadecimal format, e.g. 3F.

7.2.1.3 Codec Profiles

In the **VoIP**->**Settings**->**Codec Profiles**, you can define the various codec profiles to control voice quality and set up specific provider-dependent default settings.

When creating codecs, please bear in mind that good voice quality requires corresponding bandwidth, which will limit the number of possible simultaneous calls. Moreover, the remote station must also support the corresponding codec selection.

7.2.1.3.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **VoIP**->**Settings**->**Codec Profiles**->**New** consists of the following fields:

Fields in the Basic Parameters menu

Field	Description
Description	Enter a description for the entry.
Codec Proposal Se-	Choose the order in which the codecs are offered for use by the

Field	Description
quence	system. If the first codec cannot be used, the second is tried and so on. Possible values: • *Default* (default value): the codec in the first position in the menu will be used if possible. • *Quality*: The codecs are sorted by quality. The codec with the best quality is used if possible. • *Low Bandwidth*: The codecs are sorted by required bandwidth. If possible, the codec with the lowest bandwidth requirement is used. • *High Bandwidth*: The codecs are sorted by required bandwidth. If possible, the codec with the highest bandwidth requirement is used.
G.711 uLaw	Not for **Codec Proposal Sequence** = *default* ISDN codec with US characteristic G.711 uLaw passes audio signals in the range of 300-3500 Hz and samples them at the rate of 8,000 samples per second. At 64 kbit/s bit rate the mean opinion score (MOS) is 4,4. This audio codec uses µlaw quantization.
G.711 aLaw	Not for **Codec Proposal Sequence** = *default* ISDN codec with EU characteristic G.711 aLaw passes audio signals in the range of 300-3400 Hz and samples them at the rate of 8,000 samples per second. At 64 kbit/s bit rate the mean opinion score (MOS) is 4,4. This audio codec uses alaw quantization.
G.722	Not for **Codec Proposal Sequence** = *default* G.722 passes audio signals in the range of 50-7000 Hz and samples them at the rate of 16,000 samples per second. At 64 kbit/s bit rate the mean opinion score (MOS) is 4,5.
G.729	Not for **Codec Proposal Sequence** = *default* G.729 passes audio signals in the range of 300-2400 Hz and

Field	Description
	samples them at the rate of 8,000 samples per second. At 8 kbit/s bit rate the mean opinion score (MOS) is 3,9.
G.726 (32 Kbit/s)	Only if **Codec Proposal Sequence** is not *Default* G.726 supports bit rates of 16, 24, 32, and 40 kbit/s. G.726 achieves a Mean Opinion Score (MOS) of about 4.2 for the 40 kbit/s variant and about 3.85 for the 32 kbit/s variant.
G.726 Codec Settings	G.726 can be used with different byte orders according to ITU-T. Select the variant you want to use.
DTMF	Not for **Codec Proposal Sequence** = *default* Select whether the DTMF Outband codec is to be used. First the system attempts to use RFC 2833. If the remote terminal does not use this standard, SIP Info is used. The function is activated by selecting *Enabled*. The function is enabled by default.

7.2.1.4 Options

In the **VoIP**->**Settings**->**Options** menu you can perform global VoIP settings.

The menu consists of the following fields:

Fields in the Basic Parameters menu

Field	Description
RTP Port	Specify the port through which the RTP data should be routed. By default the value *10000* is set.

Fields in the VoIP Provider Settings menu

Field	Description
DSCP Settings for sip Traffic	Select the Type of Service (TOS) for SIP data. Possible values: • *DSCP Binary Value* (default value): Differentiated Services Code Point according to RFC 3260 is used to signal the prior-

Field	Description
	ity of IP packets (indicated in binary format, 6 bit). The default value is *110000*.
	• *DSCP Decimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in decimal format).
	• *DSCP Hexadecimal Value*: Differentiated Services Code Point according to RFC 3260 is used to signal the priority of IP packets (indicated in hexadecimal format).
	• *TOS Binary Value*: The TOS value is specified in binary format, e.g. 00111111.
	• *TOS Decimal Value*: The TOS value is specified in decimal format, e.g. 63.
	• *TOS Hexadecimal Value*: The TOS value is specified in hexadecimal format, e.g. 3F.
SIP Port	Specify the port SIP data are to be transferred through. The default value is *5060*.

> **Note**
>
> If you change the port during operation, the change only becomes effective after the next reboot of your device.

Fields in the SIP dual Stack (IPv4/IPv6) menu

Field	Description
SIP dual Stack (IPv4/IPv6)	Enable this option if you want to support IPv6 for VoIP. Both, IPv4 as well as IPv6 are supported, and if a VoIP provider supports IPv6, IPv6 is preferred. If a VoIP provides does not support IPv6, IPv4 is used. With selection of *Activated* the option is anabled. The function is not enabled by default. This means that only IPv4 is used until you enable dual stack SIP.
SIP Service	This service is the central service for controlling incoming and outgoing calls. If it is deactivated, telephony is no longer possible. If you do not need the functions, you can disable the ser-

Field	Description
	vice here . The corresponding protocol ports (UDP/TCP 5060) will then be closed and the system will no longer accept connections to them.

7.2.2 Media Gateway

A media gateway serves as a translation instance between different telecommunications networks, e.g between the plain old phone network and the next generation networks (IP networks).

With the **be.IP swift** Media Gateway, a company equipped with an automatic PBX on a wired telephone network can be connected to a SIP Trunking Service Provider on the Internet in order to use IP telephony.

The **be.IP swift** Media Gateway supports the binding of several SIP Provider Accounts. With this gateway, you can set up extensions, create an extension number plan and configure exchange functions and optimise voice data transmission for low bandwidth of the upload connection.

7.2.2.1 Call Routing

Here you can define the conditions for the routing of calls. Define a list with rules or rule chains that are used to manipulate the indicated destination numbers.

A list of all existing entries is displayed in the **VoIP**->**Media Gateway**->**Call Routing** menu.

7.2.2.1.1 Edit or New

Choose the icon to edit existing entries. Select the **New** button to create new entries.

The **VoIP**->**Media Gateway**->**Call Routing**-> ->**New** menu consists of the following fields:

Fields in the Basic Parameters menu

Field	Description
Description	Enter the name of the entry.
Administrative Status	Select whether the entry should be activated. The function is enabled with *Enable*. The function is enabled by default.

Field	Description
Type	Specify how calls are to be routed. Possible values: • *Accept Rule*: For calls forwarded by the media gateway to a PBX or an ISDN TE connector or a SIP DDI client. For this, the following can be used: PRI interfaces in NT mode, BRI interfaces in NT mode, SIP accounts in trunk mode (server mode). • *Deny*: For calls that are not to be routed (to be blocked).
Calling Line	You can restrict the application of the entry to the line on which the call comes in. The selection depends on the interfaces available and on the SIP accounts that have been created. Possible values: • *pri<Interface Index>*: restricts the routing entry to the selected PRI interface. • *bri<Interface Index>*: restricts the routing entry to the selected BRI interface. • *<SIP Account>*: restricts the routing entry to the selected SIP account. • *Any*: No restriction of the entry.
Calling Address	You can restrict the application of the entry to a particular caller. To do this, you must specify the subscriber number exactly (no wildcards).
Called Address	Enter the called address to which the rule is to be applied. To do this, enter an address numerically (e.g. a subscriber number) or alphanumerically (e.g. for a trunk) that is to be compared with a dialled address. The following wildcards can be used: • * means that at the end of a character string any number of characters may follow, • ? is a placeholder for an arbitrary character.

Field	Description
	If the configured address agrees with the signalled address, the entry is used.

In the **Routing Rules** menu you can define rules to determine how the subscriber number is manipulated before it is used for dialling.

Use **Add** to create more entries.

Fields in the Routing Rules menu (For Type = Accept Rule only)

Field	Description
Administrative Status	Select whether the rule should be activated. The rule is enabled with *Enable*. The rule is active by default.
Line	Choose the ISDN line (PRI, BRI) or SIP account used for the outgoing call.
Called Address Translation	Enter how the subscriber number is manipulated before it is used for dialling. Notation: <a:b>; i.e. a is replaced by b. Every rule must be ended with a semicolon. A number of rules can be chained together using semicolons as separators, e.g. <a:b>;<c:d>;<e:f>. After confirmation of entry, the rule chain is automatically sorted by the "best match" method. Numerical and alphanumerical values are permissible. ? is a placeholder for an arbitrary character. **Example 7.1. Example of a rule** • Rule: <:+49911>; • number dialled: 96731234 • manipulated number: +4991196731234

7.2.2.2 Calling Party Format

Here you define the processing of the calling party number for incoming calls. You can, for example, add a prefix to a received call number in order to route corresponding outgoing calls via a particular SIP account.

In the **VoIP**->**Media Gateway**->**CLID Translation** menu, a list of all existing entries is shown on which the received number is edited.

7.2.2.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create entries for CLID translation.

The **VoIP**->**Media Gateway**->**CLID Translation**->✎->**New** menu consists of the following fields:

Fields in the Basic Parameters menu

Field	Description
Description	Enter the name of the entry.
Calling Line	Select the ISDN line or SIP account from which the call comes.
	The selection depends on the interfaces available and on the SIP accounts that have been created.
	Possible values:
	• *pri<Interface Index>*: Restricts the entry to the selected PRI interface.
	• *bri<Interface Index>*: Restricts the entry to the selected BRI interface.
	• *<SIP Account>*: Restricts the entry to the selected SIP account.
	• *Any*: No restriction of the entry.
Called Line	Here you have the option of entering the destination line of the call.
	Possible values:
	• *pri<Interface Index>*: Restricts the entry to the selected PRI interface.
	• *bri<Interface Index>*: Restricts the entry to the selected

Field	Description
	BRI interface.
	• *<SIP Account>*: Restricts the entry to the selected SIP account.
	• *Any*: No restriction of the entry.
	Enter either **Called Line** or **Called Address**.
	If a value other than *Any* is selected, **Called Address** should not be used. If **Called Line** = *Any* and **Called Address** is not used, all calls for **Called Line** are processed.
Called Address	Here you have the option of entering the destination address of the call.
	Enter either **Called Line** or **Called Address**. If **Called Address** is used, then **Called Line** = *Any* can be set .
Calling Address Translation	Enter the transformation rule applied to the call numbers.
	Notation: <a:b>; i.e. a is replaced by b. Every rule must be ended with a semicolon. A number of rules can be chained together using semicolons as separators, e.g. <a:b>;<c:d>;<e:f>;. After confirmation of entry, the rule chain is automatically sorted by the "best match" method.
	? is a placeholder for an arbitrary digit.
	Example 7.2. Example of a rule
	• Rule: <:+49911>;
	• number dialled: 96731234
	• manipulated number: +4991196731234

7.2.2.3 Priority Numbers

In the **Call Routing**->**Outgoing Services**->**Priority Numbers** menu you configure numbers with particular special functions, e.g. emergency functions.

In your system configuration, you can enter numbers that must be accessible in an emergency. If you now dial one of these priority numbers, it is detected by the system and an ISDN B channel is automatically cleared. If the external ISDN B channels are already in use, one of the ISDN B channels is freed up and the calling subscribers hear the busy tone.

An ongoing priority call is not interrupted.

7.2.2.3.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Outgoing Services**->**Priority Numbers**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Priority Number	Enter the number which can even be dialled if all B channels are occupied. In this case, an external B channel is released for this connection and reassigned for the priority call. An ongoing priority call is not interrupted.

7.2.2.4 Special Numbers

At a DDI connection, the called number of an outgoing call is automatically converted to the international E.164 format. This conversion is undesirable for certain numbers. Exceptions from the conversion can be configured here.

7.2.2.4.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The **Call Routing**->**Outgoing Services**->**Special Numbers**->**New** menu consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Description	Enter a description for the entry.
Special Number	Specify the number that is to be excepted from E.164 conversion.

Chapter 8 WLAN

8.1 WLAN

8.1.1 WLAN

8.1.1.1 Reviewing and changing WLAN settings for the 2.4 GHz frequency band

Base Station

The menu consists of the following fields:

Fields in the Base station menu

Field	Description
Enable base station	The wireless network in the 2.4 GHz frequency band can be (de)activated here.
SSID	The name of the WLAN base station is an ID for easy identification. A name has already been configured at the factory. The wireless network's default name must be entered in the router pass and is located on the nameplate on the back of the **be.IP swift**.
	If there is a network with the same name within range, enter a new name in the **SSID (Wireless network name)** field in order to better distinguish it. **Save** the new settings and use it in your WLAN devices.
	You can prevent the name from being displayed by selecting **Hide SSID**. Before you activate this option, **be.IP swift** and the computer must have established a WLAN connection once. If you afterwards hide the name, then the connection is automatically found, but the SSID is no longer displayed in the computer's list of available wireless networks.

⚠️ **Caution**

Please make sure to change the **Network key** and use a unique, secure password for it! The default settings for the **SSID** and **Network key** are intended only to make it easy to establish a WLAN connection for the first time.

⚠️ **Caution**

Please note that hiding the name (**Hide SSID**) is not a security feature since the name can still be easily read using the appropriate software.

📝 **Note**

The WLAN SSID is replaced by the factory settings when reset (see *Restarting or resetting a device* on page 8). Regardless of whether you use the default or a personal name for the wireless network: The name (SSID) must be entered in all clients that are intended to be used with the wireless network; otherwise, the connection cannot be established.

Settings

The menu consists of the following fields:

Fields in the Settings menu

Field	Description
Transmission power	If the **be.IP swift** is relatively close to the computer (and other WLAN devices), you can reduce the **Transmission power** without deteriorating transmission. You can gradually change the transmission power between *6 %* and *100 %*.
	By reducing the transmission power, you increase the security of the WLAN. As a result, only the network devices that you want can be found in the transmission radius of the **be.IP swift**. This makes it difficult for third parties to access. Furthermore, it also reduces the **be.IP swift**'s power consumption.
Reduce transmission power automatically	Controls the transmission power based on need.
Transmission mode	Defines the WLAN standard that is used between the WLAN

Field	Description
	base station and the WLAN client. Both devices must mutually use at least one WLAN standard to exchange data. The WLAN base station supports the following WLAN standards in the 2.4 GHz frequency band: • **802.11b** with speeds of up to 11 Mbit/s • **802.11g** with speeds of up to 54 Mbit/s • **802.11n** with speeds of up to 130 Mbit/s • 802.11ax with speeds of up to 1148 Mbit/s If you are unfamiliar with the WLAN client's WLAN standard, simply choose one of the following methods: • **Mixed 802.11g and 802.11b** • **Mixed 802.11n and 802.11g** • **Mixed 802.11ax, 802.11n, 802.11g and 802.11b** The WLAN client and the WLAN base station automatically select the best WLAN standard.
Channel bandwidth	A single channel has a **Channel bandwidth** of 20 MHz. By bonding a second channel, you can create a 40 MHz bandwidth and therefore achieve significantly higher transmission rates in the *801.11n* WLAN standard. The default setting is *20/40 MHz*. At 40 MHz, you must expect a higher probability of error due to greater overlap with other users. This is why the *20 MHz* setting is better if you have many wireless neighbors. *20/40 MHz* can also be used if there are little to no wireless neighbors.
Wireless channel	If there are several WLAN base stations in the area, they often transmit at the same *Funkkanal*, which can affect transmission. The *Auto* setting automatically searches for an available wireless channel. If necessary, change the *Funkkanal* of the **be.IP swift**. Channels *1* to *13* are available to you. Check under "Channel usage in the surroundings" to verify which wireless channels are already being used.
Protected mode	Protected mode improves the quality of the signal by reducing the transmission rate.

Field	Description
802.11e/WMM QoS	This option can be used to improve the quality of service for streaming applications.

Encryption

To protect the WLAN from unauthorized use, you should encrypt the WLAN data transmission. The factory default encryption type is *WPA2*. *WPA2* is currently the most modern security standard for WLAN and is required for WPS. You can choose one of the following options as the **Encryption method**.

The menu consists of the following fields:

Fields in the Encryption menu

Field	Description
WPA/WPA2	The *WPA* and *WPA2* encryption methods can be simultaneously used on your device.
WPA2 only	Only WLAN devices that support WPA2 encryption can connect to the **be.IP swift** (default). WPA2 (Wi-Fi Protected Access 2) is currently the most modern security standard for WLAN according to the IEEE 802.11a/b/g/n standards and is based on the Advanced Encryption Standard (AES). It is the successor to WPA and implements the fundamental functions of the new IEEE 802.11i security standard. WPA2 encryption is the factory default. The default key must be entered in the router pass and is located on the nameplate on the back of the **be.IP swift**. However, it is recommended that you use a personal, secure WPA2 key that you can enter in the **Network key** field. It should have at least 8 but no more than 63 alphanumerical characters (A-z 0-9). Special characters *. _ - ! ~ * \ ' ( ) = + $, \ ? /* can also be used.
WPA only	Only WLAN clients that support WPA encryption can connect to the **be.IP swift**. WPA includes WEP's architecture, although it provides additional protection via a dynamic key that is based on the Temporal Key Integrity Protocol (TKIP). WPA offers PSK (pre-shared keys) to authenticate users. We recommend that you use this encryption mode if the connected WLAN adapters do not support *WPA2* mode. To use WPA encryption, enter a key in the **Network key** field. It should have at least 8 but no more than 63 alphanumerical characters (A-z 0-9). Special

Field	Description
	characters $. \_ - ! \sim * \setminus \prime ( ) = + \$, \setminus ? /$ can also be used.
WEP	Only WLAN clients that support WEP encryption can connect to the **be.IP swift**. WEP (Wired Equivalent Privacy) is a former encryption algorithm for WLAN. Due to different interfaces, the method is now deemed insecure. Now the key can be calculated after recording the data. This process is used only if necessary with older devices for compatibility reasons.
	Caution For security reasons, we recommend that you do not use WEP encryption! According to current findings, this method is insecure and should be used only as a last resort.
	WEP mode: Determines the length of the key. **128-bit**: 13 alphanumeric characters (A-z 0-9). **64-bit**: 5 alphanumeric characters (A-z 0-9). In the **Network key** field, enter the corresponding key with precisely [13 or 5] characters.
Unencrypted	**be.IP swift** does not use any encryption. This setting should be used only for testing purposes since this option also gives third-parties access to the WLAN.
	Note The encryption method configured on the **be.IP swift** must be supported and used by all the WLAN clients on the network.
	Note The **Network key** and the **Encryption method** will be restored to factory settings when reset (see *Restarting or resetting a device* on page 8). If you wish to use a personal key instead of the default key, you must re-enter it after the reset.

Field	Description
Authentication	*PSK* specifies the encryption in the router (default setting). *802.1x* allows you to administrate the passwords via a RADIUS server.

Save the new settings and use them your WLAN devices.

WPS -- Wi-Fi Protected Setup

Using the **Activate WPS Push Button** option, the WPS function can be started using the *WPS* button on the **be.IP swift**. If the WPS function generally cannot be activated via the *WPS* button, the checkmark next to the (2.4 GHz) frequency must be removed.

> **Note**
>
> This option can be activated only if the SSID is not hidden.

> **Caution**
>
> To prevent unauthorized login of guests, e.g. in lodging homes or restaurants, the *WPS* key should be deactivated after registering new devices!

> **Note**
>
> WPS can be used only in one frequency band. If this function is already used in the 5 GHz frequency band, the function is deactivated here.

Click on **Save**.

8.1.1.2 Reviewing and changing WLAN settings for the 5 GHz frequency band

Base Station

The menu consists of the following fields:

Fields in the Base station menu

Field	Description
Enable base station	The wireless network in the 5 GHz frequency band can be (de)activated here.

Field	Description
SSID	The name of the WLAN base station is an ID for easy identification. A name has already been configured at the factory. The wireless network's default name must be entered in the router pass and is located on the nameplate on the back of the **be.IP swift**. If there is a network with the same name within range, enter a new name in the **SSID (Wireless network name)** field in order to better distinguish it. **Save** the new settings and use it in your WLAN devices. You can prevent the name from being displayed by selecting *SSID verbergen*. Before you activate this option, **be.IP swift** and the computer must have established a WLAN connection once. If you afterwards hide the name, then the connection is automatically found, but the SSID is no longer displayed in the computer's list of available wireless networks.

Caution

Please make sure to change the **Network key** and use a unique, secure password for it! The default settings for the **SSID** and **Network key** are intended only to make it easy to establish a WLAN connection for the first time.

Caution

Please note that hiding the name (**Hide SSID**) is not a security feature since the name can still be easily read using the appropriate software.

Note

The WLAN **SSID** is replaced by the factory settings when reset. Regardless of whether you use the default or a personal name for the wireless network: The name (**SSID**) must be entered in all clients that are intended to be used with the wireless network; otherwise, the connection cannot be established.

Settings

The menu consists of the following fields:

Fields in the Settings menu

Field	Description
Transmission power	If the **be.IP swift** is relatively close to the computer, you can reduce the **Transmission power** without deteriorating the transmission. You can gradually change the transmission power between *6 %* and *100 %*. By reducing the transmission power, you increase the security of the WLAN. As a result, only the network devices that you want can be found in the transmission radius of the **be.IP swift**. This makes it difficult for third parties to access. Furthermore, it also reduces the **be.IP swift**'s power consumption.
Reduce transmission power automatically	Controls the transmission power based on need.
Transmission mode	Defines the WLAN standard that is used between the WLAN base station and the WLAN client. Both devices must mutually use at least one WLAN standard to exchange data. The WLAN base station supports the following WLAN standards in the 5 GHz frequency band: • **802.11a** with speeds of up to 54 Mbit/s • **802.11n** with speeds of up to 300 Mbit/s (only with 2.4 GHz) • **802.11ax** with speeds of up to 4804 Mbit/s If you are unfamiliar with the WLAN device's WLAN standard, simply choose one of the following methods: • **Mixed 802.11a, 802.11n, 802.11ac and 802.11ax** The WLAN client and the WLAN base station automatically select the best WLAN standard.
Channel bandwidth	A single channel has a **Channel bandwidth** of 20 MHz. By bonding four channels, you can create a 80 MHz bandwidth and therefore achieve significantly higher transmission rates with the **801.11ax** WLAN standard.
Wireless channel	Depending on the configuration in the **Frequency band mode** menu, here you can have the WLAN wireless channels automatically selected (recommended) or assign a fixed channel to the **be.IP swift** (recommended). Please note the following information if you choose the wireless channel yourself:

Field	Description
	• Select a channel with as little interference as possible. In the 5 GHz band, many WLAN devices support only channels 36 to 48 (**Compatibility mode**). When using a channel from 52 – 140 (**Complete mode**), the **be.IP swift** regularly checks if the channel that is used is being used by a preferential radar (e.g., weather radar). If such a preferential radar is detected, then this channel may not be used for 30 minutes. In this case, the **be.IP swift** automatically switches the channel. At the same time, WLAN devices can lose their connection to **be.IP swift**. • Due to the check for the aforementioned primary sender (e.g. weather radar), the channel availability check (CAC) for the critical wireless channel ranges of 120, 124, 128, and 132 requires up to 10 minutes. • The channel availability check (CAC) for wireless channels 100, 104, 108, 112, 116, 136, and 140 is generally completed after approx. 1 minute.
Frequency band mode	This setting affects the possible selection of wireless channels in the **Wireless channel** menu. The following settings are available: • **Frequency band mode**: Complete (Wireless channels 36 – 64 , 100 – 140) **Wireless channel:** Auto or 36 – 64 , 100 – 140 • **Frequency band mode**: Compatibility mode (wireless channels 36, 40, 44, 48) **Wireless channel**: Auto or 36 – 48
802.11e/WMM QoS	This option can be used to improve the quality of service for streaming applications.

Encryption

Only the WPA2 encryption method and no encryption are available in 5 GHz.

WPS -- Wi-Fi Protected Setup

Using the **Activate WPS Push Button** option, the WPS function can be started using the *WPS* button on the **be.IP swift**. The WPS function can be used only on one frequency band (2.4 GHz or 5 GHz). In this case, if you wish to activate the WPS function for 5 GHz, it must

be deactivated for 2.4 GHz beforehand. If the WPS function generally cannot be activated via the `WPS` button, the checkmark next to both frequency bands (2.4 GHz, 5 GHz) must be removed.

This option can be activated only if the SSID is not hidden.

⚠ **Caution**

To prevent unauthorized login of guests, e.g. in lodging homes or restaurants, the WPS function must be deactivated after registering new devices!

Note

WPS can be used only in one frequency band. If this function is already used in the 2.4 GHz frequency band, the function is deactivated here.

Click on **Save**.

8.2 Monitoring

In the **Connected WLAN devices** overview, you can see all the WLAN devices that are connected to the wireless network.

8.2.1 WLAN Monitoring

8.2.1.1 Checking channel usage in the surroundings

This screen dialog graphically illustrates **Channel usage in the surroundings**. It gives you an overview of which WLAN networks are active in the surroundings and which **Wireless channel** per frequency band is available. It also displays which WLAn network is assigned to which wireless channel along with reception quality.

This display is continuously updated with the **Activate continuous monitoring** option. If you do not have the **Wireless channel** setting automatically configured by the **be.IP swift** (as described in **Reviewing and changing WLAN settings for the 2.4 GHz frequency band**), you should choose a range in the graphics for your **Wireless channel** that has the least amount of overlap with other WLAN base stations as possible.

Chapter 9 Internet and network

9.1 Network

9.1.1 Static IP addresses

9.1.1.1 Using fixed IP addresses

By reserving fixed IP addresses, the IP addresses are permanently assigned to certain MAC addresses on the DHCP server. The MAC address is a unique address that is assigned to the hardware. IP addresses are assigned to MAC addresses for an indefinite period of time. Permanent assignments are mainly made if, for instance, the DHCP client provides server services and therefore needs to be available under a fixed IP address. Even port forwardings from a router to a client generally require a fixed IP address.

Fixed IP addresses are assigned from an IP address range that is managed from the DHCP server.

In the **List of static IP addresses** table, you can see the clients for which fixed IP addresses have been assigned. Using the pen symbol, you can edit the respective settings and use to remove the corresponding entry.

Using the **New** button, you can define fixed IP addresses. Here, next to the **MAC address** of the device to which you wish to assign the static **IP address**, enter an IPv4 address in the defined address range and enter a **Name**. If you make a selection beforehand using the **List of connected clients**, the **MAC address**, **IP address** and **Name** of the selected device are used. Clicking on confirms the settings.

9.1.2 Static ARP addresses

9.1.2.1 List of static ARP entries, network settings

ARP is the link between the IP and MAC address. In this screen view, you can create static links between the IP and MAC addresses in your LAN yourself if necessary.

9.1.3 Advanced Access

9.1.3.1 Displaying advanced internet connection data

In this screen dialog, you can see an overview of advanced settings for your internet connection.

Here you can make additional settings for time-based or volume-based internet connections.

9.1.4 Load distribution

9.1.4.1 Configuring load balancing

Here you can define rules for data traffic on the WAN interfaces. It requires you to have configured the LTE backup as well as an available, active DSL connection. Depending on the protocol, load balancing automatically selects the best line for transferring sessions.

Activate load balancing: Activate this option in order to use load balancing with this site's settings.

Selecting the WAN interface

Select which interface you wish to use as the primary and the secondary WAN interface for the upstream.

Selecting the downstream LAN

Select the LAN interface that you wish to subject to load balancing. The other interface automatically uses the primary WAN interface.

The tables define the rules for load balancing:

- With **New**, you can add an additional rule to the corresponding list. You can change existing entries using the pen symbol ✎ and use the trash can symbol to erase existing 🗑 rules.
- Using **LAN Host**, you can define the IP address in the internal network while you can use **Target host** to determine the IP address of the corresponding device on the internet. IP address 0.0.0.0 means that this rule is applied to all IP addresses and is displayed as *Alle* in the overview table. Under **Port**, you determine the respective port, and under **Protocol**, you can select between *UDP*, *TCP* and *Alle* (chooses both UDP and TCP).

Specifically transmitting data traffic via the primary WAN interface

Specify which services are to always use the primary WAN interface and therefore be excluded from load balancing.

The default setting is that all connections use the primary WAN interface via port 443 using the TCP protocol (https connections). This is due to the fact that both WAN interfaces have different externally visible IP addresses. If websites use cookies in which the IP address is saved for user identification, among other things, a session may be terminated on the server end, e.g. during online banking if the IP changes during a session. Since such websites mostly use secured connections (https), this rule prevents such behavior.

Note

Please consult the corresponding professional literature for rules on standardized ports and logging for different internet services.

If you have difficulties with maintaining sessions on websites, please review the **Port** that is being used and add it to this list if necessary.

Specifically transmitting data traffic via the secondary WAN interface

Specify which services are to always use the secondary WAN interface and therefore be excluded from load balancing.

By default there are no connections. If, for instance, the second WAN interface had a higher downstream speed compared to the primary WAN interface as the case may be when using UMTS via WAN2 and ADSL via WAN1, then it is worthwhile, for instance, to always send emails via the secondary port. To do so, the TCP protocol must be selected and 25 must be entered as the port.

Simultaneously distributing data traffic via both WAN interfaces

At this point, the rules for load balancing are defined.

Save accepts the configuration for the load balancing.

9.1.5 IPTV

9.1.5.1 Configuring IPTV

Once you have booked a package, you can activate and configure this service via the screen dialog.

The menu consists of the following fields:

Fields in the IPTV menu

Field	Description
Activate IPTV	(De)activates IPTV.

DSCP class of IGMP packets	Defines the prioritization of the IPTV data packets. The recommended default setting is BE. Further relevant settings are:
	
	• **BE**
	
	  Best effort forwarding.
	• **AFxy**
	
	  Assured forwarding provides the delivery of four independent classes. Within a class, an IP packet can be assigned one of three different levels of drop precedence.
	• **CSx**
	
	  The Class Selector (CS) settings are used for IP priorities (IP precedence) of the older Type of Service (ToS)-based QoS to ensure backward compatibility and are also used for data flow prioritization.
	• **EF**
	
	  Expedited Forwarding is intended to provide low delay, low jitter and low loss in the delivery of packets.

IGMP Version	Selection of the Internet Group Management Protocol version that is required for supporting multicasting. IGMPv3 offers greater security than IGMPv2.
IPTV interface	Determine the interface through which you would like to receive IPTV data.

Click on **Save** to accept the settings.

9.1.6 Fiber optic connection

9.1.6.1 Configuring the fiber optic connection

Provides you with an overview of your connected fiber optic connection.

Furthermore, this is where you can edit the Subscriber Location Identification (SLID) that is used for registration.

9.1.7 NAT assignment

9.1.7.1 NAT assignment

Self-configured NAT address mappings are displayed in the NAT mapping table.

9.2 Local services

9.2.1 DNS

9.2.1.1 Configuring the DNS Relay

A DNS (Domain Name Server) is required in order to resolve address queries in IP addresses. For instance, in response to a query for "google.com", the DNS would reply with 173.194.70.101 for IPv4 or 2a00:1450:4005:800::1013 for IPv6. In this screen dialog, you can set up and configure the preferred DNS server.

The menu consists of the following fields:

Fields in the DNS relay menu

Field	Description
Activate DNS Relay	The **be.IP swift** operates as a DNS relay. If the option is deactivated, the DNS relay is disabled. In this case, a DNS server must be manually entered on each client. The option is active by default.
Activate user-defined DNS Server	This setting is required only if you do not wish to use the internet service provider's default settings. If necessary, enter the IPv4 or IPv6 address of the DNS server. The **DNS server 1** replaces the *Primären DNS* from the internet settings, and **DNS server 2** replaces the *Sekundären DNS*.

9.2.2 DynDNS

9.2.2.1 Setting up dynamic DNS

In this screen dialog, you can see the list of configured dynamic DNS accounts. Dynamic DNS is an internet service that allows you to configure a fixed name as a pseudonym for a dynamically assigned IP address as is typically the case for internet addresses. Using Dynamic DNS and what is known as **Port forwardings** (see *UPnP Port forwarding* on page 191), you can make servers from the home network statically visible on the internet.

For dynamic DNS, you must set up an account with a dynamic DNS service provider (e.g., dyndns.org). With this account, you can configure a new dynamic DNS account on the **be.IP swift**.

Proceed as follows:

(1)   Click on **New**.

(2)   Select the right service provider in the **Service provider** drop-down list. If your **Service provider** is not in the list, you can create an individual entry using the **Update URL** entry.

(3)   In the following fields, enter the data shared by the service provider.

(4)   Click on **Save**.

> ⚠ **Caution**
>
> Please note that the home network server that is statically visible on the internet due to dynamic DNS can be an attack target for abuse by third parties!
>
> Therefore, protect the dynamic DNS account with a secure password. Keep the security software in your network up to date. Immediately follow up on unusual events. Regularly inform yourself about system security.

9.2.3 UPnP

9.2.3.1 Defining settings

Universal Plug-and-Play (UPnP) provides the Internet Gateway Device (IGD) protocol with a simple way to open ports and forward queries from the internet to a computer.

These port forwardings are necessary, e.g., for data transfers in instant messaging programs and filesharing (e.g., Skype, eMule). Furthermore, UPnP enables the detection and integration of new hardware and the distribution of multimedia content in a local network.

The **Internet Gateway Device (IGD)** provides status information about your **be.IP swift** in your home network, such as the public (external) IP address and existing port mappings

and allows UPnP clients to add or remove port mappings.

Using the **Enable UPnP IGD Status Information** and **Enable UPnP IGD Port Forward-ings** options, you can specify whether only status information (default = activated) is provided in the network and if IGD port forwardings may also be used along with it.

> **Caution**
>
> Depending on the operating system of the connected computer, automatic port config-uration poses a potential security risk. As a result, the **be.IP swift**'s firewall can be de-activated by malware that has infected the computer.

9.2.3.2 UPnP Port forwarding

Port forwarding set up using UPnP is displayed in this screen dialog. These These forward-ings are automated and are not without risk.

Port forwarding allows computers within a local network to be reached from the Internet as well. This is important, for example, to exchange data over the Internet or to make a Web server accessible from the Internet. from the Internet.

> **Note**
>
> Network clients (computers, NAS, etc. in the home network) that can be reached from the Internet via port forwarding are also exposed to attacks from the Internet. Install software on these network clients that protects against Internet attacks and use strong passwords that are changed regularly.

9.3 Routing

9.3.1 Routing

9.3.1.1 Configuring static routing via LAN interfaces (network)

Her you can enter static routes for networks that can be accessed via LAN interfaces. Con-nections towards WAN are configured under **Internet & network**->**Routing**->**Static rout-ing** (see *Configuring static routing via WAN interfaces (internet)* on page 192).

Using the **New** button, you can open a screen dialog with the following configuration op-tions:

Fields in the Static Routing menu

Field	Description
List of connected clients	Select an entry from this list and the **Name** and **Gateway** fields below will be filled out accordingly if $IPv4$ is selected as the **Address type**.
Name	Any name for the route.
Address type	IPv4 or IPv6 depending on the address range of the third-party network.
Destination Network	Address range of the network devices that are to be accessed.
Gateway	The router's address for the destination network.
Interface	The selection depends on the configured DSL or WAN interface.

Save accepts the route data.

9.3.1.2 Configuring static routing via WAN interfaces (internet)

Here you can enter static routes for networks that can be accessed via the WAN interface. Connections towards the LAN interface are configured under **Internet & network**->**Routing**->**Configure static routing** (see *Configuring static routing via LAN interfaces (network)* on page 191).

Using the **New** button, you can open a screen dialog with the following configuration options:

Fields in the Static Routing menu

Field	Description
Name	Any name for the route.
Address type	IPv4 or IPv6 depending on the address range of the third-party network.
Destination Network	Address range of the network devices that are to be accessed.
Gateway	The router's address for the destination network.

Field	Description
Interface	The selection depends on the configured DSL or WAN interface.

Save accepts the route data.

9.4 VPN

9.4.1 IPsec

IPsec offers secure data transmission via potentially insecure networks such as the internet and enables the configuration of a virtual private network (VPN).

9.4.1.1 Activating IPsec

Here you can enable or disable the function as needed. Your **be.IP swift** supports the IKEv2 protocol. The **IPsec** function is deactivated in **be.IP swift** by default.

Using the **Save** button, you can activate the selected settings.

9.4.1.2 IPsec address pools

This page shows an overview of all configured pools of virtual IP addresses that can be set up and given to remote IPsec clients.

List of configured IP address pools

In the "Single remote client" scenario, a remote client (initiator) can establish an IPsec connection and request an additional IP address for this device (responder) for use within the IPsec tunnel. The requested address is issued from a pool defined on this page and is either defined as an IP subnet or as an address range. A pool can be used by several IPsec profiles at the same time.

With **New**, you can define a new address pool; you can use this symbol ✏ to change or erase 🗑 existing entries. The **In use** column provides the status of the corresponding address pool.

If you create a new address pool, you must enter a freely definable name in **Name**. Under **Type of pool**, select the **Subnet** application specifying the corresponding network address with a suffix for the netmask or an **IP address range** specifying the **IP start address** and **IP end address**.

Do not erase any entries that are currently used by a profile. Please be aware of the column **In use**.

By pressing the **Save** button, you transfer the data to the **List of configured IP address pools**.

9.4.1.3 IPsec profiles

This page shows an overview of all configured IPsec profiles.

With **New**, you can create an IPsec profile; you can use this symbol ✎ to change or erase 🗑 existing entries. The **Activated** column displays the status.

List of configured IP address pools

A separate profile is required for each IPsec connection. The identity of each IPsec partner is defined by the identification IDs. A key/pre-shared key (PSK) is exchanged between the participants in order to encrypt and decrypt the data.

Fields in the List of configured IP address pools menu

Field	Description
Activate	By setting or removing the checkmark, you can activate or de-activate the IPsec profile.
Name	Enter a freely definable name.
Local ID	Enter a local ID here. To do so, enter a unique name.
Remote ID	Enter the ID of the remote terminal to which the profile will be applied.
Preshared key	Enter a letter-number-character combination consisting of 8 to 64 letters, numbers, and the following special characters: -*@#$/%_+.: \, and spaces are allowed.
Preshared key (confirmation)	Re-enter the key that was just entered.
Scenario	Select the type of network connection **Single remote client** or **LAN-to-LAN connection**. Depending on the selected scenario, you will be asked for the following information:

Single remote client configuration

Fields in the Single remote client configuration menu

Field	Description
IP address pool	Select an address pool that has been previously issued under **List of configured IP address pools**.

or

LAN-to-LAN connection

Fields in the LAN-to-LAN connection settings menu

Field	Description
Type of remote end-point address	They decide on the version of the IP protocol for the IPsec tunnel connection (**IPv4**, **IPv6** or any with **Domain name**).With **Domain name**, the address type depends on the DNS name resolution.
Remote endpoint domain name	Enter the domain name to be connected here.
Local IPv4 network	Enter the local IPv4 network address with the suffix for the network mask.
Remote IPv4 network	Enter the remote IPv4 network address with the suffix for the network mask.
Local IPv6 network	Enter the local IPv6 network address with the suffix for the network mask.
Remote IPv6 network	Enter the remote IPv6 network address with the suffix for the network mask.

The information on the local and remote network is what are known as traffic selectors and are important for routing decisions. It manages which packet is transferred with which source and target address via the respective VPN tunnel.

Using the **Save** button, you create an IP address pool in the **List of configured IP address pools**.

9.5 Firewall

9.5.1 Firewall

9.5.1.1 Using a firewall

To protect your home network and your internet connection from unauthorized use, a **Firewall** is integrated in the **be.IP swift**. It offers comprehensive protective measures through which internet access can be externally limited to mitigate the risk of hacker attacks and to prevent a wide range of common attacks.

The **be.IP swift**'s firewall operates according to the principle of SPI (Stateful Packet Inspection). It is a dynamic packet filtering technique in which each data packet is assigned to a particular, active session. At the same time, the data packets are evaluated together with the connection status, and during this process, the data packets are analyzed during transmission and saved in dynamic state tables. The decision to forward a data package is made based on the state of the data connection. During the process, any data packets that cannot be assigned to the defined criteria or may be part of a DoS attack are discarded. This is why the **be.IP swift**'s firewall with SPI technology is superior to pure packet filter firewalls in security-related applications.

Furthermore, you can configure exceptions for individual applications under "Port forwardings" and forward the data traffic for these applications.

The menu consists of the following fields:

Fields in the Use firewall menu

Field	Description
Enabling IPv4 SPI	This option cannot be deactivated.
Enabling IPv6 SPI	This option can be used to activate the SPI (Stateful Packet Inspection) firewall for IPv6 connections.
	⚠ Caution If you **Deactivate the IPv6 SPI firewall**, all the devices on the home network are directly accessible from the internet. Use this option only if you are familiar with the possible outcomes!
Activate firewall logging	This option logs all the registered firewall attacks. You can open the protocol via the **System**->**Diagnostics**->**System notifications** menu (see *System Messages* on page 9).

9.5.1.2 Creating schedules for firewall rules

To create a new schedule, click on **New**. You can also configure several schedules. Schedules that have already been set up are displayed in the overview. To edit a schedule that has already been configured, click on the pen symbol at the end of the overview ✏ .

The menu consists of the following fields:

Fields in the Rules for schedules menu

Field	Description
Name	You can enter any name for the schedule.
Time period	Enter the desired time period here. A **Start time (hh:mm)** and **End time (hh:mm)** can be entered for every weekday. If a continuous schedule extends over several days, each intermediate day must be specified as an entire day. The start time is 00:00 and the end time is 23:59.

9.5.1.3 Defining firewall rules

To create a new firewall rule, click on **New**. Firewall rules that have already been configured are displayed in the overview. To edit a firewall rule that has already been configured, click on the respective pen symbol at the end of the overview ✏ .

You can define the rules for clients via the IP address and select services that have already defined for that purpose or determine your own criteria. To that end, use the following fields:

Fields in the Firewall rules menu

Field	Description
Automatically refresh provider prefix	Valid only for IPv6. If the computer in your local network has IPv6 addresses, they consist of a local part and a provider prefix. This prefix is assigned to you and can be automatically updated by the **be.IP swift**.
Client name	Enter any name for the rule.
Address type	Select the format of the IP address. Accordingly, the rule can be created for an IPv4 or IPv6 address.

Field	Description
Client address	Enter the IP address of the network device to which the rule will be applied. Use the format of the previously selected **Address type**. For IPv4, the rule can be applied to all the devices in the home network with the address.
Assign the following schedule rules	If you have created a schedule under "Creating schedules for firewall rules", you can select this option. Otherwise, the selected services will always be blocked according to this setting.
Status	The selection defines whether the rule is activated.

Client services

Using the **WWW**, **Send e-mail**, **News Forums**, **Receive e-mail**, **Secure**, **HTTP**, **File Transfer**, **Telnet Service**, **NetMeeting**, **DNS**, **SNMP**, **VPN-PPTP**, **VPN-L2TP**, **TCP**, **all UDP ports** service names, you can define which services are blocked by activating the option.

If you block **User-defined services**, the following section is activated, in which you can individually define the *TCP* or *UDP* protocol and the *Port-Bereich*.

⚠️ **Caution**

Do not change the default values if you do not have fundamental knowledge about the network settings. Changes to this screen dialog can result in the **be.IP swift** being inaccessible. If this happens, reset the **be.IP swift** to the factory settings.

9.5.1.4 Configuring network zones

Here you can find an overview of preconfigured zones. A zone can always contain only one interface. Each zone's policy describes how data packets for which no extra rules exist are handled. NAT (Network Address Translation) allows you to replace your PC's IP address with the **be.IP swift**'s IP address.

ICMPv6 Filter

Here you can configure the ICMPv6 filter for which incoming and outgoing packets of the ICMPv6 protocol are used. The following filters can be selected:

Fields in the ICMPv6 filter menu

Field	Description
destination-unreach-able	Data packet could not be delivered.
packet-too-big	Data packet could not be delivered since it is larger than the permissible size.
parameter-problem	An error has occurred when processing a data packet and it was discarded.
time-exceeded	The data packet could not be delivered since it is either in an endless loop or the number of permissible intermediate stations was exceeded.
router-solicitation	A "router advertisement" message is being requested by a client.
router-advertisement	The router distributes IP addresses and additional information.
neighbor-solicitation	Requests a MAC address for an IP address.
neighbor-advert-isement	Sends the MAC address to an IP address in response to a "neighbor-solicitation" message.
redirect	Information that there is a better intermediate station for reaching a particular destination.

To edit a zone that has already been configured, click on the respective pen symbol at the end of the overview ✎ to open the **Edit entry** screen dialog.

Each zone's **Policy** describes how data packets for which no extra rules exist are handled.

The menu consists of the following fields:

Fields in the Policy menu

Field	Description
Name	Name of the internet connection
NAT	Network address translation replaces a network device's local IP address with your device's public IP address. This is required if, for instance, the network device has a private IP address but you wish to communicate with servers on the internet.

Field	Description
	This function is available only for IPv4 connections.

ICMPv4 Filter

The menu consists of the following fields:

Fields in the ICMPv4 filter menu

Field	Description
Allow ping to WAN Interface from the Internet	ICMP is used to exchange error and information messages. However, since this function can also be misused for attacks on the network, this option can be explicitly activated.

ICMPv6 Filter

You can generally allow or prohibit incoming and outgoing ICMPv6 protocol packets.

The menu consists of the following fields:

Fields in the ICMPv6 filter menu

Field	Description
Allow ping to WAN Interface from the Internet	If this function is activated, then ICMPv6 messages can be sent and accepted from the internet and to the network.
Allow ping to LAN Clients from the Internet	If this function is activated, then ICMPv6 messages can be sent and accepted from the internet and to the clients on the local network.

9.5.1.5 IP Parameter Configuration

In this menu, you can set up advanced parameters to precisely configure the firewall.

> **Warning**
>
> Change the settings here only if you have advanced knowledge of this subject matter. Incorrect settings can severely affect the functionality of the **be.IP swift**.

The following parameters are available here:

• **ICMP Echo Ignore:** Die **be.IP swift** BASIC no longer responds to a ping.

- **ICMP Ignore Broadcast Echo:** Die **be.IP swift** BASIC no longer responds to a broad-cast ping.
- **RIP Defect:** Ignores packets that are received via a network interface and whose sender IP address does not match the subnet behind the network interface.
- **TCP ECN:** Optimizes TCP connections and are set up to and from the IAD itself (GUI, TR069, etc.).
- **TCP Handshake Timeout:** If a connection has been improperly established, then it still remains in the firewall's connection tracking for as many seconds as specified here.
- **TCP Fin Timeout:** If a connection has been improperly terminated, then it still remains in the firewall's connection tracking for as many seconds as specified here.
- **TCP/IP Conntrack timeout:** After an external TCP connection has been established, in-coming packets that belong to this connection are still accepted for as many seconds as specified here.
- **UDP/IP Conntrack timeout:** After an external UDP connection has been established, in-coming packets that belong to this connection are still accepted for as many seconds as specified here.
- **ICMP Rate Limit:** Timer in milliseconds; the minimum amount that must elapse before a new ICMP message is sent to a particular host.
- **TCP Syn Retries:** Max. number of how often a SYN message is resent before the con-nection setup is terminated as unsuccessful.
- **TCP Syn Ack Retries:** Max. number of how often a SYN-ACK message is resent before the established connection is terminated.

9.6 Monitoring

9.6.1 DSL

9.6.1.1 DSL status overview

Status information regarding your DSL connection, the data rate, and the operating data and error indicators are displayed in this screen dialog. This is where experts receive tech-nical details on the DSL connection. The displayed information may be required in case of support inquiries.

The error indicators relate to G.997.1 (physical layer management for digital subscriber line transceivers), and it does not provide any direct information on physical problems. A certain number of error indicators is deemed a normal condition.

9.6.2 Internet

9.6.2.1 Internet status

Here you can view detailed information about your internet connection:

- **DSL status** (status, mode, connection type, and connection retries)
- **Data rate** (DSL data rate, line capacity)
- **Internet connection** (IP address, gateway, DNS server, and connection duration)

Furthermore, you can manually establish or disconnect internet connections here if they have been properly configured.

9.6.3 Network

9.6.3.1 Network status

Here you can view basic information about your network:

- **Port allocations**
- **IP address and subnet mask**
- Status of the **DHCP server** with information about the DHCP gateway's IP, the configured IP address range, and the allocation duration)
- **List of connected clients**

  Individual devices that are currently no longer connected to the **be.IP swift** are shown in gray and 🗑 can be used to remove them from the list. With ✎ you can edit the displayed name of the client.

9.6.4 VPN

9.6.4.1 VPN

In this menu you can see an overview of the VPN connections that have been set up. If you click the 🔍 Icon, detailed information about the corresponding connection is displayed.

Chapter 10 User Access

The system administrator can set up an individual configuration access interface for the users. You, the user, can thus display your most important personal settings and individually customise some of these.

To log in to the configuration interface with your assigned access data, enter your **User Name** and **Password** in the login window.

After successful login, the **Status** page is displayed. It includes an overview of your most important settings.

In the **Phonebook** menu, you can access the **System Phonebook** and create, edit as well as delete entries in a user-specific telephone book.

In the **Call Data Records** menu, you get a detailed overview of the calls you have conducted and accepted.

The **Settings** menu contains an overview of current settings of performance features **Direct Call**, **Call Forwarding** and **Parallel Ringing**. You can individually customise these here. In addition, you can view general settings and customise access and contact data.

You can also view the settings of the **System Phone** assigned to you, and modify these to your needs.

In the **Voice Mail System** ->**Settings** menu, you'll see the current configuration of your individual voicemail box, as well as the number of messages present. You can modify several frequently used voicemail box parameters here. The **Voice Mail System** ->**Messages** menu displays a detailed overview of all received calls.

10.1 Settings

In the **Settings** menu, you can perform personal settings to performance features direct call, call forwarding (CF), parallel call and do not disturb, as well as customise general settings.

10.1.1 Feature Settings

In the **Settings**->**Feature Settings** menu, the settings for performance features direct call, call forwarding (CF), parallel call and do not disturb can be customised.

10.1.1.1 Call Forwarding

In the **Settings**->**Feature Settings**->**Call Forwarding** menu, you configure forwarding of incoming calls to your internal number onto the entered destination number.

You are temporarily away from your office, but don't want to miss a call. With call forwarding to another number, e.g. your mobile, you can receive your calls even when you are not at your desk. You can forward calls on your number to any call number. It can occur *Immediately*, *On no reply* or *On Busy*. Call forwarding *On no reply* and *On Busy* can exist concurrently. If you are not near your telephone, for example, the call is forwarded to another number (e.g. your mobile phone) after a short period. If you are making a call at your desk, other caller may receive the busy signal. You can forward these callers e.g. to a colleague or the secretary by using call forwarding on busy.

Calls can be forwarded to internal subscriber numbers, internal team numbers or external numbers When the number to which calls shall be forwarded is entered, the system automatically checks whether it's an internal or external number.

To continue with configuring, click the ✎ symbol.

Select the ⌨ button to go to the **IP1x0** telephone user interface administrator page. This is described in the telephone user guide!

The **Settings**->**Feature Settings**->**Call Forwarding** menu consists of the following fields:

Fields in the Call Forwarding menu

Field	Description
Active Function	Select whether to enable the call forwarding (CF) function for your telephone.
	The function is enabled with *Enabled*
	The function is disabled by default.
Type	Select when incoming calls shall be forwarded to the specified internal number.
	Possible values:
	• *Immediately*
	• *On Busy*
	• *On no reply* (default value)
	• *On busy / On no reply*

Field	Description
Destination on no Reply	Enter the number to which incoming calls shall be forwarded after time.
Destination on Busy	Enter the number to which incoming calls shall be forwarded on busy.
Destination immediately	Enter the number to which incoming calls shall be forwarded immediately.

10.1.1.2 Log on / Log off

With system telephones, it is possible to log out of a team using the **Log on / Log off** function key. The team administrator must run this function manually if standard telephones are used.

The **Settings**->**Feature Settings**->**Log on / Log off** menu consists of the following fields:

Fields in the Log on / Log off menu

Field	Description
Description	Indicates the teams the user belongs to.
Status	Select whether the team member shall be logged in or out of the team. The function is activates by selecting *Logged on*. The function is enabled by default.

10.1.2 General Settings

Your user's most important settings are listed in the **Settings**->**General Settings** menu. Personal access data (configuration password, IP phone password) as well as mobile and home office numbers can be customised.

The menu consists of the following fields:

Fields in the User Data menu.

Field	Description
Name	Displays your user's name.

Field	Description
Description	Displays the additional description for your user.
Login Name	Displays your user name for login to the user configuration interface.
Password for HTML Configuration Access	If you wish to change your password for access to the user configuration interface, enter a new password here. To check, you can display the password in plain text by clicking the **Show** option.
Password for IP Phone Registration	If you wish to change your password for IP telephone login, enter a new password here. To check, you can display the password in plain text by clicking the **Show** option.
PIN for Phone Access	If you wish to change the PIN for your personal voicebox, enter a new PIN here. To check, you can display the password in plain text by clicking the **Show** option.
Mobile Number	Here, you can enter the mobile number under which you can be reached.
Home Office Number	Here, you can enter the home office number under which you can be reached.
Busy on busy	Indicates whether performance feature Busy on Busy is enabled for the currently selected user.
	If a subscriber for whom multiple telephone numbers have been configured makes a call, you can decide whether additional calls for this user shall be signalled. If "Busy on Busy" is set for this user, other callers get an **Engaged** signal if the user is calling on one of her numbers.
	The function is activated by selecting *Enabled*.
	The function is disabled by default.

Fields in the Status Information menu.

Field	Description
Extension Numbers	Displays the internal numbers assigned to you.

Field	Description
Current Class of Ser-vice	Displays the authorisation class to which you're currently as-signed.
Dialling Authorization	Displays your dial permission.
Allow manual trunk group selection	Displays whether you're allowed to assign additional bundles to outside lines and, if so, which.
Pick-up Group	Displays the number of the group in which calls may be picked up.

10.2 Status

The **User Access**->**Status** menu displays the most important settings performed for you by the system administrator.

The menu consists of the following fields:

Values in the User Data list

Field	Description
Name, First Name	Displays the configured surname and name, if applicable, of your user.
Description	Displays the configured additional description for your user.

Values in the Internal Numbers &Communication Cost list

Field	Description
<Internal Number>	Displays the connection charges for the internal numbers as-signed to your user.

Values in the Further Settings list

Field	Description
Current Class of Ser-vice	Displays the name of the authorisation class to which your user is assigned.
Dialling Authorization	Displays the dial permission for your telephones. This derives from the setting for the corresponding user class.

Field	Description
	Possible values:
	• *International*: The telephones have unrestricted dialling authorisations and can initiate all connections.
	• *National*: The telephones can initiate all calls except international calls. If a number starts with the code for international dialling, the number cannot be dialled.
	• *Incoming*: The telephones can receive incoming external calls, but cannot initiate any external calls. Internal calls are possible.
	• *Region*: The telephones cannot make any national or international calls. For this dial permission, 10 exception numbers allowing national or international dialling can be configured. An exception number can consist of complete call numbers or sections thereof (e. g. the first numerals).
	• *Local*: The telephones can make local calls. National and international calls are not possible.
	• *Internal*: The telephones do not have authorisation for incoming or outgoing external calls. Only internal telephone calls are possible.
Allow manual trunk group selection	Indicates whether your user is assigned to an authorisation class for which manual bundle assignment is allowed. If so, authorised bundles or external connections are displayed.
	Besides general exchange access, a telephone can also selectively use a bundle. Here an external connection is initiated with the corresponding code for the target assignment of the bundle and not by dialling the dialling code.
	To be able to perform a selective bundle assignment, the authorisation class must possess the appropriate authorisation. The authorisation can also include bundles that the authorisation class can otherwise not assign. If a telephone does not possess the authorisation for selective bundle assignment, or if the selected bundle is in use, the busy tone is heard after dialling the code. If **Automatic Outside Line** is set up for an authorisation class, users of this authorisation class must press the star key before selective bundle assignment, then initiate external dialling with the code for bundle assignment.

Field	Description
Pick-up Group	Displays the number of the group in which calls may be picked up.

10.3 Phonebook

In the **Phonebook** menu, telephone book entries are displayed separately according to **System Phonebook** and **User Phonebook**. In **User Phonebook** the user can create, modify or delete up to 50 own entries. These entries can only be viewed by the respective user. These entries are updated via the **GUI**.

10.3.1 System Phonebook

In the **System Phonebook**, entries of the overall system created by the administrators are displayed You cannot modify these.

Values in the System telephone book list

Field	Description
Description	Displays the subscriber's description. The **System Phonebook** is sorted according to these entries.
Phone Number	Displays the telephone number.
Speed Dial Number	Displays the speed-dial number.
Call Through	Indicates whether the telephone number for the **Call Through** function is activated.

10.3.2 User Phonebook

In the **User Phonebook**, your user entries are displayed. You can add, edit or delete entries.

10.3.2.1 Edit or New

Choose the ✎ icon to edit existing entries. Select the **New** button to create new entries.

The menu **User Access**->**Phonebook**->**User Phonebook**->**New** consists of the following fields:

Fields in the Phonebook Entry menu

Field	Description
Description	Enter a description for the entry. Sorting in **User Phonebook** follows the initial letters of the entry.
Phone Number	Enter the telephone number (internal or external).

10.4 Call Data Records

The **Call Data Records** menu displays your user's incoming and outgoing connections recorded to date.

10.4.1 Outgoing

The **Call Data Records**->**Outgoing** menu consists of the following fields:

Values in the Outgoing list

Field	Description
Date	Displays the connection date.
Time	Displays the time at call start.
Duration	Displays the duration of the connection.
User	Displays the user who called.
Int. No.	Displays the user's internal number.
Called Number	Displays the dialled number.
Project Code	Displays the call project number, if any.
Interface	Displays the interface over which the external connection was routed.
Costs	Displays the connection charge, but only if the provider transmits the corresponding data.

10.4.2 Incoming

The **Call Data Records**->**Incoming**menu consists of the following fields:

Values in the Incoming list

Field	Description
Date	Displays the connection date.
Time	Displays the time at call start.
Duration	Displays the duration of the connection.
User	Displays the user who was called.
Int. No.	Displays the user's internal number.
External Number	Displays the caller's number.
Project Code	Displays the call project number, if any.
Interface	Displays the interface over which the connection from outside was routed.

10.5 Call List

The menu **Applications**->**Call List** lists details of incoming and outgoing calls. Which kind of calls and how many of them are included can be spcified in the submenu **General**.

10.5.1 Incoming

The **Applications**->**Call List**->**Incoming** menu contains information that permits the monit-oring of incoming activities.

The menu consists of the following fields:

Fields in the Incoming menu

Field	Description
Date	Displays the connection date.

Field	Description
Time	Displays the time at call start.
Type	Displays the type of the connection.
User	Displays the user who was called.
Int. No.	Displays the user's internal number.
Caller Number	Displays the caller's number.
Trunk Number	Displays the port number.
Interface	Displays the interface over which the connection from outside was routed.
Delete	You can use the **Select all** and **Deselect all** buttons for all the devices displayed.

10.5.2 Outgoing

The **Applications**->**Call List**->**Outgoing** menu contains information that permits the monitoring of outgoing activities.

The menu consists of the following fields:

Fields in the Outgoing menu

Field	Description
Date	Displays the connection date.
Time	Displays the time at call start.
Type	Displays the type of the connection.
User	Displays the user who was called.
Int. No.	Displays the user's internal number.
Called Number	Displays the caller's number.
Trunk Number	Displays the port number.

Field	Description
Interface	Displays the interface over which the connection from outside was routed.
Delete	You can use the **Select all** and **Deselect all** buttons for all the devices displayed.

10.6 Assigned elmeg Phones

The **Assigned elmeg Phones** menu shows the telephones that the system administrator has assigned to you.

> **Note**
>
> The **Assigned elmeg Phones** menu is only displayed if you've already been assigned system telephones by the administrator.

10.6.1 Assigned elmeg Phones

The **Assigned elmeg Phones**->**Assigned elmeg Phones** menu shows a list with the key information about your telephone. The symbol takes you to the phone's configuration interface.

Select the symbol to select the display language and the key settings of the phone.

10.6.1.1 Settings

The **Settings** menu allows you to select the desired display language from a drop-down menu.

10.6.1.2 Keys

The menu **Keys** displays the configuration of your system telephone's keys.

> **Note**
>
> You can configure the key assigment either through your PABX system or on the telephone itself. We recommend using your PABX system for this, since it overwrites the telephone configuration.

You can avoid the overwriting for individual keys that have already been configured on the telephone by choosing *Not configured* on the PABX system.

Your telephone is equipped with several function keys that allow the assignment of different functions. The functions available for programming are different across different types of telephones.

Values in the list Keys

Field	Description
Key	Displays the name of the key.
Label Description	Displays the configured key name. This appears on the labelling page (label strips).
Key Type	Displays the key type.
Settings	Displays the additional settings with a summary

Print allows you to print out a label sheet for the description field of your system phone or key extension.

Edit

Choose the ✎ icon to edit existing entries. In the pop-up menu, you configure the functions of your system telephone keys.

You can use the following functions with system telephones:

- *Dial Key (Standard)*: You can store a number on each function key. External numbers have to be prefixed by the exchange code *0* if *no automatic outside line* has been configured for your **Class of Service** on the telephone.
- *Dial Key (DTMF)*: You can store a DTMF sequence on every function key.
- *Extension Key (User)*: You can set up dialling to an internal extension using a line key. After pressing the corresponding key, hands free is switched on and the internal extension entered is selected. If a call is signalled on the internal extension you have entered, you can pick this up by pressing the line key.
- *MSN Selection Key*: Assigns a specific connection (i.e. a specific SIP account) to the function key. You can use this key to initiate a call via this connection, or you can accept a call coming in via this connection. The key flashes if a call is received, it is lit if the connection is busy. Select the desired connection. All configured connections are available. Configure SIP accounts exclusively on your PABX system.

- *System Parking (Open Enquiry)*: The called extension enters an enquiry and dials a code. The telephone is now open for additional operations like e.g.an announcement. A second subscriber can accept the call by picking up the receiver and dialing the code corresponding to the call. The codes are determined by the PABX, but can also be assigned to the functions keys of one or more system phones. If a call is put into open enquiry by pressing a function key, this is indicated by the flashing of the respective function key LED on all system phones with a corresponding configuration. Pressing the function key accepts the call. This function is only available if a call has been parked.

- *Login / Log Out, Team*: You can log into out out of one of the configured teams.

- *Call Forwarding (enable)*: Assigns activating or deactivating a call forwarding that has been configured on the telephone. You can only store a single call forwarding on the device; it is applied to all calls.

- *Parallel Ringing*: Incoming calls to an internal number are signaled at another external number in parallel.

- *Night Mode*: You can set the phone into night mode oprtation.

- *Not configured*: The function key is managed by the telephone itself and not by the PABX system.This options locks the key for the provisioning by your PABX system.

The menu **Terminals**->**elmeg System Phones**->**elmeg IP**->**Keys**->**Edit** consists of the following fields:

Fields in the menu Keys

Field	Description
Key name	Enter a name for the key to be used as text for the corresponding key when the ID labels are printed.
Key Type	Depending on the model, telephones have seven or 14 keys that can have functions assigned to them. Optional key extension modules extend the number of available functions keys. Possible values: • *Dial Key (Standard)* • *Dial Key (DTMF)* • *Extension Key (User)* • *MSN Selection Key* • *System Parking* • Login / Log out Team • *Call Forwarding (enable)* • *Parallel Ringing* • Night Mode

Field	Description
	• *Not configured*
Internal MSN	Only for **Key Type** = *Dial Key (Standard)*, *Extension Key (User)*, *MSN Selection Key*, *Call Forwarding (enable)* or *System Parking* You can select one of the internal MSNs configured in the menu **Terminals**->**elmeg System Phones**->**elmeg IP**->**Numbers**.
Number	Only for**Key Type** = *Dial Key (Standard)* or *Dial Key (DTMF)* You can save a number or a DTMF sequence to any function kye. Specify the number or the characters for the DTMF sequence.
Internal Number	Only for **Key Type** = *Extension Key (User)* Select the internal number of the subscriber that is to be called when pressing this key.
Pick-Up Code	Only for**Key Type** = *Extension Key (User)* The code that is required for the busy lamp field to allow you picking up a call on an IP telephone when the LED is flashing. The default value is *#0*.
Waiting Queue	Only for**Key Type** = *System Parking (Open Enquiry)* Select the waiting queue to which the currect connection is to be added.
URL	Only for**Key Type** = *XML-Content* For this function you can store the URL to a server which hosts the desired information. This function is currently not supported by your PABX system.

Transfer key

Select the ↑↓ icon to move configured function keys.

Fields in the menu Move to

Field	Description
Phone	Select one of the connected telephones.
Module	Select *Telephone* or a key extension.
Key	Select the key to which you wish to transfer the configured function.

10.7 Voice Mail System

You can access information on you voicemail box in the **Voice Mail System** menu.

> **Note**
>
> The **Voice Mail System** menu is only displayed once a personal voicemail box has been set up for you.
>
> If you are registered as an administrator for a Team Voice Mail Box, you also have access to the configuration of the Voice Mail Box here.

10.7.1 Settings

In the **Voice Mail System** ->**Settings** menu, your voicemail box settings are displayed.

Values in the Settings list

Field	Description
Internal Number	Displays your internal number.
User	Displays your user name.
Status of Mail Box Owner	Displays your status.
Check PIN	Indicates whether access to your voicemail box is protected by a PIN.
Mode for status "In the Office"	Indicates in which mode your voicemail box operates for the "In Office" status.

Field	Description
Mode for status "Out of Office"	Indicates in which mode your voicemail box operates for the "Out of Office" status.
New Calls	Indicates the number of new calls.
Old Calls	Indicates the number of old calls.
Saved Calls	Indicates the number of saved calls.

10.7.1.1 Edit

Choose the ✎ icon to edit existing entries. You can change the settings of selected para-
meters.

The menu **Voice Mail System** ->**Settings** consists of the following fields:

Fields in the Basic Settings menu

Field	Description
Status of Mail Box Owner	Define in which mode your mailbox shall be used when starting the voicemail system. Possible values: • *In the Office* (default value): Select this setting if you're in the office when the voicemail system is started. • *Out of Office* : Select this setting if you're out of the office when the voicemail system is started.
Check PIN	Select whether your voicemail box should be protected with a PIN.
Mode for status "In the Office"	Your voicemail box can be operated with two different settings during office hours. Possible values: • *Announcement only*: A caller hears an announcement, but cannot leave a message. • *Announcement and Record*: A caller hears an announce-ment and can leave a message.
Mode for status "Out	Your voicemail box can be operated with two different settings

Field	Description
of Office"	outside office hours. Possible values: • *Announcement only*: A caller hears an announcement, but cannot leave a message. • *Announcement and Record*: A caller hears an announcement and can leave a message.

Fields in the Voice Mail Message via E-Mail menu

Field	Description
E-Mail Notification	Once a message has been left on the voicemail box, the subscriber can be notified. Possible values: • *None* (default value): The subscriber is not notified. • *E-Mail*: The subscriber is informed of a present message via e-mail. • *E-Mail with Attachment*: Once a caller has left a message, the subscriber receives an e-mail with a recording of the message in the attachment.
	Note After a subscriber has been informed about a new message through an E-Mail, the **Status** of the message is chnaged according to the configuration in **User Access**->**Voice Mail System** ->**Settings** under **E-Mail forwarding behavior**.
E-Mail forwarding behavior	Only for **E-Mail Notification** = *E-Mail* or *E-Mail with Attachment* Select an option for forwarded messages. Possible values: • *Keep Message in 'new' after forwarding*: Following an e-mail alert or forwarding, the status of the voicemail message is set to *New*.

Field	Description
	• *Move Message to 'old' after forwarding*: Following an e-mail alert or forwarding, the status of the voicemail message is set to *Old*.
	• *Remove Message after forwarding*: The voicemail message is deleted following an e-mail alert or forwarding.

Select the ✎ icon to set up your own voice announcements for the selected voice mail box in addition to the settings above.

The **Applications**->**Voice Mail System** ->**Voice Mail Boxes** -> ✎ menu consists of the following fields:

Fields in the menu Voice Announcement

Feld	Beschreibung
In the Office	You can upload your own announcement for the **In the Office** state. Use the WAV file format for this annoncement. Click **New Message** to upload the file. The window **Annouacement Options** opens. If an announcement has been stored, use the ▶ icon to play the announcement, the 🗑 icon to delete it.
Out of Office	You can upload your own announcement for the **Out of Office** state. Use the WAV file format for this annoncement. Click **New Message** to upload the file. The window **Annouacement Options** opens. If an announcement has been stored, use the ▶ icon to play the announcement, the 🗑 icon to delete it.

Fields in the menu Annouecement Options

Feld	Beschreibung
Action	Displays *Update announcement* .
Source Location	For **Action** = *Update annoncement* Select the WAV file to be used for the announcement and click **Start** to upload.

> **Note**
>
> Please note that Wave files must be available in the following format:
>
> • Bit rate: 128 kbps
> • Sampling size: 16 bit
> • Channels 1 (mono)
> • Sampling rate: 8 kHz
> • Audio format: PCM

10.7.2 Messages

In the **Voice Mail System** ->**Messages** menu, a list of your messages is displayed. You also have the option to play back voicemail messages or download these to your PC. To save a message, click on the ⊟ icon. The download dialog then opens. To listen to a message, click on the ▶ icon.

Individual or all wave files can be deleted by clicking the **Select all** / **Deselect all** checkbox then pressing **Delete Selected**.

Values in the Messages list

Field	Description
Internal Number	Displays the internal number of a voicemail box. Several internal numbers can be assigned to a single user. The user can operate a separate voicemail box under each internal number.
User	Displays the name of the voicemail box user.
Call from	Displays the caller's number.
Date/Time	Displays the date and time of the call.
Call Status	Indicates whether the call is *New*, *Old* or *Saved*.
Select all / **Deselect all**	You can select individual entries using the checkbox in the corresponding line, or select them all using the **Select all** button or **Deselect all**. You can delete the selected entries by pressing the **Delete Selected** option.

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