COMfortel D-100 / D-110 / D-200 / D-210

Advanced Information

Table of Contents

Information on Release 2.6

The new or revised functions below have been published with this Release.

↗ You will find more information about changes to the firmware and the associated documentation at Auerswald Support/FONtevo Support.

Message key

Pressing the message key displays the messages. If no interaction with the message is intended, pressing the message key again makes the view disappear.

↗ Further help under Notifications.

Name sources, Reverse search

Different name sources can be prioritised in a list per Identity. The prioritisation is used to decide which name is shown in the display for incoming/outgoing calls.

The following parameter is redundant and has been removed: **Reverse search**.

↗ Further help under Name sources.

Pickup

The function Pickup has been extended, the descriptions for the following settings have been newly created: **Pickup-choice on key-press**, **Do not show in pickup list**, **Pickup of connected calls**, App **Pickup**

↗ Further help under Pickup.

SIP message

The phone can send and receive SIP messages. A set message can be sent to a recipient with an appropriately configured function key. A received SIP message is displayed as a notification.

↗ Further help under SIP message.

Internal/external dial tones

By provisioning the dialling tone in the Identities section, it is possible to play different dialling tones to the user.

↗ Further help under Dial tones.

LED signalling

When configuration switchover is enabled in the PBX, the LED of the function key lights up green.

↗ Further help under LED signalling on function keys.

Date and time

Date and time of the phone are displayed in the web interface.

↗ Further help under Date and time.

Refresh register

Re-registration can be triggered at an earlier time than set in the register interval per **Early re-register (sec)**, **Refresh register at 1/2**.

↗ Further help under Registrar.

Removal of settings

With Release 2.6, the following setting is omitted: **Send SIP REGISTER**.

DTMF

Various methods are available for transmitting DTMF signals.

↗ Further help under [DTMF](#).

USB headset

Not all USB connections are suitable for the use of a USB headset. Corresponding notes have been added.

↗ Further help under [Headset mode](#), [Rear View](#).

About these instructions

Gender Notice
Trademarks
Licenses
Instructions

Gender Notice

The English-language documentation does not use gender-specific forms of address.

Any references to persons apply to all people of all genders.

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Instructions

After a firmware update, you usually require a new version of the instructions.

⬈ You can find current instructions on the Internet at Auerswals Support / FONtevo Support.

Functions and Applications

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Information about the section

This section describes how to configure and use the functions on your phone.

Action URL

Action URLs are HTTP GET requests which are either sent to a PC, PBX or server, from your phone, or vice versa. In the latter case, the phone is the HTTP server. Action URLs that are sent to the phone trigger actions (such as starting a call). Action URLs that are sent by the phone report changes to the number and condition of calls to the device (e.g. a newly received call).

↗ Further help in the [Auerswald DokuWiki](#)

Requirement for use

* Web interface is open, user is logged on to it

  or

* valid token (generated using the phone's web interface)

Configuration

Web interface:

* **Access**

  ↗ Details about the settings are in [Access](#)

Phone:

* **Access**

  ↗ Details about the settings are in [Access](#)

Provisioning:

↗ Further help in the [Auerswald DokuWiki](#)

Process

1. In your browser address line, enter the required command.

    The table below displays the possible actions and the corresponding http commands.

    ```
    https://[phoneIP]/api/v1/exec/command?action=*
    ```

`*help`	Displays help texts for the function entered.
`*resume`	Resumes the call
`*log`	Adds an entry to the telephone log.
`*accept`	Accepts a call
`*terminate`	Ends a call
`*call`	Starts a call
`*show`	Displays a brief message in the phone display.
`*hold`	Puts the call on hold
`*transfer`	Connects to a number
`*join`	Connects two existing calls
`*conference`	Conference call
`*dtmf`	DTMF

2. If a token is used for authentication, the token must be sent in the request header.

3. Press Enter to confirm.

    The action you want is carried out.

Note: For full functionality, some commands that are to be executed will need additional parameter settings.

Further help in the Auerswald DokuWiki

Anonymous call

The function **Anonymous call** enables the caller to make a call without revealing their subscriber number to the person being called.

ℹ **Note:** Anonymous calls are not possible with some VoIP providers. If you intend to use this function, first ask your provider if this service attribute is supported. This function can only be used on one of the PBX's land line connections if the CLIR (case-by-case restriction of calling line identification) service feature has been requested from, and activated by, the network provider.

Requirement for use

- The CLIR (suppression of subscriber number display per call) service feature has been enabled by the network operator.

Configuration

Web interface:

- **Telephony** > **Anonymous call**

  ↗ Details about the settings are in Dialled calls

Phone:

- **Settings** > **Telephony** > **Anonymous call**

  ↗ Details about the settings are in Anonymous call

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Note: When you configure an identity, ensure that you select the same CLIR (number presentation suppression) type as is set on the PBX/at the provider.

Details about the settings are in [Options for experts](#)

Details about the settings are in [Function codes](#)

Automatic call acceptance

This function defines the situations in which the telephone automatically accepts calls.

A typical example is dialling via CTI/TAPI. Here, call centre agents are automatically connected with the call you have initiated on the PC. There is no additional confirmation on the telephone.

Configuration

Web interface:

- **Telephony** > **Call settings** > **Automatic call acceptance**

  ↗ Details about the settings are in <u>Automatic call acceptance</u>

Phone:

- **Settings** > **Telephony** > **Automatic call acceptance**

  ↗ Details about the settings are in <u>Telephony</u>

Provisioning:

↗ Details about provisioning this function in the <u>Auerswald DokuWiki</u>

Automatic start of call

When you finish typing a subscriber number, the call starts automatically in the receiver, headset or in hands-free calling mode, after the predefined time, depending on selected speech channel. You do not need to do anything else to start the call, such as pick up the receiver or press the OK button .

Configuration

Web interface:

- **Telephony** > **Dialled calls** > **Automatic start of call**

  ↗ Details about the settings are in [Dialled calls](#)

Phone:

- **Settings** > **Telephony** > **Automatic start of call**

  ↗ Details about the settings are in [Telephony](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

Backup and restore

During a backup, a backup file is created that contains the configuration and other data for the phone. If required, this backup file can be uploaded to the phone again.

A data backup can, for example, be useful if the phone is reset to its factory settings and it is not planned to completely reconfigure it.

ℹ **Note:** Uploading the backup file can involve a data migration. The phone cannot be used while this is happening. A message appears on the display to tell you a data migration is taking place.

↗ Further help under *Reset to default*

Requirement for use

- A data backup involves protected data. Before downloading starts, you must confirm that you want it to do so, in the dialogue window.

⚠ **Important:** Comply with the legal notes below before backing up the data. The data backup includes these elements with special legal protection:

- The phone's configuration
- Names of internal subscribers
- Passwords set for the phone, subscribers, VoIP accounts and VoIP subscribers
- The phone's address books
- Connection data sets with the destination, source, and time information for a telecommunication connection.
- Potentially WLAN passwords
- VPN information and VPN credentials

- Potentially webcam and access data

- LDAP information

- Potentially information about Sync accounts (CalDAV, CardDAV)

All the phone's users must be informed that this content will be transferred, and who will process the data obtained, before a backup is created. It is essential to store the data securely and in accordance with legal requirements. You must ensure that all legal requirements are met if you want to supply the data backup to your specialised dealer or the manufacturer for error analysis.

Configuration

Web interface:

- **Backup and restore**

Details about the settings are in Backup and restore

Call

You can use the phone to make internal (on the PBX) and external calls.

Information about a Call

During a call, you see information about the call and your call partner on the screen, as well as general time information (date and time).

DTMF signals

The selected numbers and characters are displayed in the screen for some seconds and you can hear the associated sound.

↗ Description of the operation under [Call]

Accepting a call

A call that has been taken on a different internal phone can be picked up, provided that call pick-up has been permitted on the PBX for the subscriber that is being called.

↗ Description of the operation under [Pickup]

This function is useful, for example, in the case of answering machines.

↗ Description of the operation under [Call]

Ending a call

When your call partner ends the call, your phone goes into its idle state. If you hang up the receiver, you will hear the busy signal through the base unit loudspeaker if the receiver has been hung up incorrectly.

Note: If you want to end the current call, and then immediately make a new call, press the soft key **End** and then start dial the next number.

Description of the operation under [Call](#)

Call bindings

Use call bindings to configure commands that are called directly from a call. For example, a call binding can be used to open a door or switch on a light. Two soft keys can be configured here.

Note: In the COMfortel D-100/D-110/D-200/D-210, a call binding can only be set up using provisioning.

Details about provisioning this function in the Auerswald DokuWiki

Call Completion

Call completion can be used to monitor a busy number.

⬈ Link to video (only in German) https://www.youtube.com/watch?v=Bi3rKTur028

Requirement for use

- Both devices are connected to the same PBX
- The PBX supports SIP Subscribe (BLF)

Configuration

Web interface:

- **Telephony** > **Call signalling** > **Call Completion**
    - ⬈ Details about the settings are in Call signalling

Phone:

- **Settings** > **Telephony** > **Call Completion**
    - ⬈ Details about the settings are in Telephony

Provisioning:

⬈ Details about provisioning this function in the Auerswald DokuWiki

Process

1. You call a subscriber and hear the busy tone.
2. Press the appropriate soft key to activate the automatic callback.

3. A notification is displayed as soon as the called number is no longer busy.

4. Confirm the notification to call the subscriber, who is now free.

5. The callback process is then deleted in the PBX.

Call forwarding

General
Immediately
On busy
On no reply

General

When call forwarding is used, internal and external calls to a single phone can be rerouted to other internal phones or external connections. This enables calls that arrive on this phone to be taken on another phone. Call forwarding ensures that the relevant subscriber can always be reached using their own phone number, even when they cannot take the call on their own phone.

You can make the following settings:

- Type of call forwarding (Immediately, On busy, By time)
- Call forwarding destination
- Waiting time before call forwarding

Immediately

If "Call forwarding immediately" is configured on the phone, the phone does not ring when there is an incoming call. The call is forwarded to the entered call forwarding destination. The call can be taken there.

Configuration

Web interface:

- **Telephony** > **Call forwarding** > **Immediately**

  ↗ Details about the settings are in Call forwarding

Phone:

- **Settings** > **Telephony** > **Call forwarding** > **Immediately**

  ↗ Details about the settings are in Telephony

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

On busy

If "Call forwarding on busy" is configured on the phone, another incoming call during an active call is not signalled. The call is forwarded to the entered call forwarding destination. Even if call waiting (with or without audible indication) is configured on the phone, another incoming call during an active call is not signalled.

Configuration

Web interface:

- **Telephony** > **Call forwarding** > **On busy**

  ↗ Details about the settings are in Call forwarding

Phone:

- **Settings** > **Telephony** > **Call forwarding** > **On busy**

  ↗ Details about the settings are in Telephony

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

On no reply

If "Call forwarding on no reply" is configured on the phone, an incoming call that is not taken within the set signalling time is forwarded to the entered call forwarding destination.

Configuration

Web interface:

- **Telephony** > **Call forwarding** > **On no reply**

  ↗ Details about the settings are in Call forwarding

Phone:

- **Settings** > **Telephony** > **Call forwarding** > **On no reply**

  ↗ Details about the settings are in Telephony

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Call type

During operation on a PBX:

- External call
- Internal call
- Group calling (a call made to a specific group)
- External recall
- Internal recall

Call waiting

Call waiting signals incoming calls during a call.

A call is signalled by

- a tone
- a message on the display

Requirements for use

- If only one identity is set on the phone, call waiting must be switched on, on the PBX, or enabled by the provider.
- Alternatively, you can set up several identities on the phone.

Further help under Identities

Configuration

Web interface:

- **Telephony** > **Call signalling** > **Call waiting**

  Details about the settings are in [Call signalling](Call%20signalling)

Phone:

- **Settings** > **Telephony** > **Call waiting** / **Audible indication**

  Details about the settings are in Telephony

Provisioning:

Details about provisioning this function in the [Auerswald DokuWiki](Auerswald%20DokuWiki)

Process

1. During a conversation, a call waiting call is signalled.

   If the audible indication is switched, you hear a tone. Otherwise, you only see a message on the screen. The call waiting caller hears the ringing tone.

2. You can do one of the following:

   − Reject the call waiting call.

     The call waiting caller hears the busy signal.

   − You take the call waiting call.

     You are then connected with the call waiting call. The previous call partner is held.

   − You disconnect the current call by hanging up.

     Your phone rings. You can accept the call as usual.

   − You forward the call waiting call.

     The call waiting caller also hears the ringing tone until the called person lifts their handset. You are now connected to the first call partner.

Caller list

Overview of call lists
Redial List
Missed calls
Log calls

Overview of call lists

Conversations (whether incoming, outgoing or missed) are stored in the caller list, with the call partner's phone number and the conversation's date and time.

A maximum of 50 phone numbers can be saved. If more than 50 calls from different phone numbers are received, the oldest entry is overwritten.

Note: If several (incoming, outgoing or missed) calls with the same phone number occur one after the other, they are grouped and displayed as a single call.

Details about the settings are in All call lists

Redial List

The last 50 phone numbers dialled are saved in the redial list. The phone numbers are saved with the date and time on which the user dialled them (or, in the case of multiple attempts to dial them, the date and time of the last attempt).

Accessing the redial list

*   Redialling 🔲

    or

*   softkey**Calls**> softkey**Outgoing**

Calling from the redial list

*   Pick up the receiver

    or

*   OK 🔲

    or

*   Headset 🔲

    or

*   Loudspeaker 🔲

    or

*   softkey **Menu** > **Call**> OK🔲

Deleting Entries from the Redial List

*   Entire list: Clear 🔲

or

- Entire list: Clear [⌫] (approx. 2 seconds)

Missed calls

The "Only missed calls" list contains the last 50 missed phone numbers. The phone numbers are saved with the date and time on which the user dialled them (or, in the case of multiple attempts to dial them, the date and time of the last attempt).

Accessing the List of Missed Calls

- Messages [icon]

  or

- softkey**Calls**> softkey**Missed**

Calling from the List of Missed Calls

- Pick up the receiver

  or

- OK [ok]

  or

- Headset [icon]

  or

- Loudspeaker [icon]

  or

- softkey **Menu** > **Call**> OK[ok]

Deleting Entries from the List of Missed Calls

- Entire list: Clear [icon]

or

- Entire list: Clear ⌫ (approx. 2 seconds)

ℹ **Note:** You cannot use the Message button to display seen missed calls if the display has been set to Disabled in the **Display and LED** > **Seen missed calls** menu option.

Log calls

When this function is active, incoming and outgoing calls are saved in the Call history.

ℹ️ **Note:** If the function is switched off, calls cannot be started by redialling the number.

Ein möglicher Anwendungsfall ist der Einsatz von COMfortel D-Serien-Geräten in Hotels: Bei ausgeschalteter Funktion wird verhindert, dass Gäste Rufnummern von zuvor getätigten Anrufen anderer Gäste sehen.

The Telephony Settings should be blocked, to prevent the function from being activated by unauthorised persons.

↗ Further help under [User roles and permissions](#)

Configuration

Web interface:

- **Telephony** > **Call settings** > **Caller list**

  ↗ Details about the settings are in [Call history](#)

Phone:

- **Settings** > **Telephony** > **Log calls**

  ↗ Details about the settings are in [Log calls](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

Central caller list

"Central caller list" is a PBX COMtrexx function.

The following data for inbound and dialled calls is saved in the central caller list on the PBX:

- Name (Last name, First name) if known in the central caller list
- Subscriber number
- Identity (user: local phone number, groups: local group name)
- Date
- Time
- Conversation duration

The PBX stores up to 50 entries per user or group.

The PBX automatically updates this list of stored calls in the user's telephone.

The entries can be displayed in a sorted sequence (inbound, missed, all calls) in the telephone.

If this function was not enabled (or temporarily disabled) for the user, none of the calls made in this time period are saved in the list. They are not displayed on the telephone when the function is enabled again.

If this function is disabled, the telephone's local caller list is displayed. If this function is enabled, the local caller list is replaced by the central list, but is not deleted.

If the telephone is not registered when it receives an incoming call, or if an update is being performed, the list in the telephone is updated when the connection is restored.

If a user who is authorised to use the Roaming User function changes telephone, either the local caller list entries or the central caller list entries are displayed on the new telephone (if this function is enabled).

Further help under Roaming User

Requirement for use

The function is enabled in the PBX.

Certificate

External connections are encrypted to prevent listening in. The processes used to set up and shut down a connection, and signalling, are all encrypted by SIPS.

Configuration

Web interface:

- **Identities** > **Options for experts** > **SIPS**
- **Identities** > **Options for experts** > **Certificate**
- **Network** > **Topology** > **Basic settings** > **Check hostname**

  ↗ Details about the settings are in Options for experts

  ↗ Details about the settings are in Topology

Phone:

- **Settings** > **Identities** > **Security** > **SIPS**
- **Settings** > **Identities** > **Security** > **Certificate**
- **Network** > **Topology** > **Check hostname**

  ↗ Details about the settings are in Security

  ↗ Details about the settings are in Topology

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Codecs

A codec is used to code and decode data or signals. The purpose of codecs is to reduce the amount of data to be transferred. Different codecs have different characteristics. Using them, the bandwidth, voice quality or compression are synchronised between the VoIP provider/PBX and telephone. The codec used has a direct effect on call quality over VoIP.

Configuration

Web interface:

- **Identities** > **Codecs**

  ↗ Details about the settings are in [Codecs](#)

Phone:

- **Settings** > **Identities** > **Options for experts** > **Audio**

  ↗ Details about the settings are in [Audio](#)

Provisioning:

↗ Further help in the [Auerswald DokuWiki](#)

Conference call

During a conference call you are talking with two subscribers at once.

All the options listed above are for a conference call that you have initiated yourself.

⤢ Description of the operation under Conference call

Splitting a call

During a conference call, you can talk to one of the two subscribers in turn, and put the other subscriber on hold in the background. You do not have to end the conference call, to do so.

Connecting the other two conference call partners to each other

If you want to leave a conference call, but the conference call partners would like to continue talking with each other, you can connect the two parties with each other.

ℹ **Note:** To enable you to connect two external subscribers with each other, "Transfer of external calls to external" authorisation must be granted on the PBX.

Splitting a conference call

The conference call is cancelled. One subscriber is put on hold and there is an active connection with the other subscriber. You can change between holding a call and an active call.

Holding a conference call and initiating an additional call

The entire conference call is held. The conference call partner hears music on hold. When the call ends, the conference call can be continued.

Ending a call with a conference call partner

You can end the call with a specific conference call subscriber and then keep talking to the remaining subscribers. This also applies if one of the conference call partners hangs up.

Ending a conference call completely

If you initiated the conference call, you can end the conference call completely.

Contact

Via the soft key **Contact**to display saved subscriber numbers with names.

If a subscriber number associated with a contact is transmitted during a call, the assigned name will be displayed (instead of the subscriber number).

Additional subscriber number can be saved for each contact.

In addition, the internal subscriber numbers (subscribers, groups, emergency call) are displayed.

All contacts are displayed in alphabetical order in the contact list.

Contacts to whom calls are made frequently can also be displayed in the Favorites list.

Contacts can be saved using backup.

↗ Details about the settings are in Backup and restore

Overview Contact / Favourites
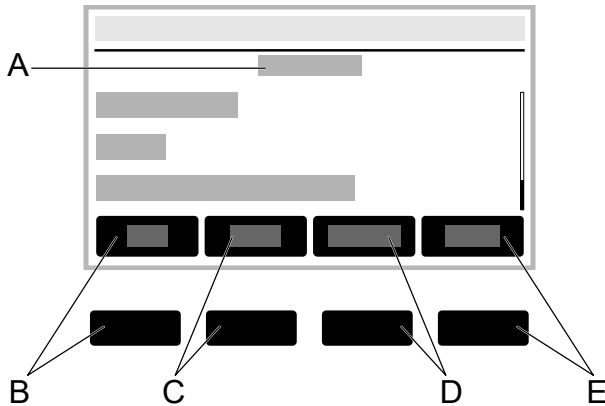
A   Title of the current application: **Contact** / **Favourites**

B   softkey**Fav.** / **All:**Switches between the lists **Contact** / **Favourites**.

C   softkey**Delete**Deletes a saved contact.

D   softkey**Details:**Displays more saved subscriber numbers for the contact. The subscriber number type (Private, Mobile etc.) is briefly displayed. Via soft key **Default** one of the displayed numbers can be set as the number that is usually used. You can also use these soft keys:

   Call: Dials the marked subscriber number. You can also initiate the call by picking up the receiver.

   Edit: Displays the menu in which you edit a saved contact.

E   →**Fav.** / →**Fav.:**Turns a contact into a favourite or favourite into a normal contact.

Note: If the LDAP function is activated, the LDAP contacts are listed below the list of contacts that have been created locally. Special features of this service:

- The number of displayed contacts differs according to the **LDAP** > **Max. LDAP contacts**.

- You cannot delete or edit LDAP contacts, or mark them as a favourite. 2 soft keys**Delete** , →**Fav.** and **Edit**are not offered.

CTI

CTI (Computer Telephony Integration) is a communication solution with which the phone can be controlled from a PC.

Coupling the phone and PC enables additional functions to be used. These improve the efficiency of communications in a company and ensure improved exchange of information.

With it, for example, contacts can be selected from a database and called directly by clicking with the mouse. Missed calls are displayed directly on the PC, and a recall can be triggered by clicking with the mouse.

The implementation of CTI takes place in the devices of the COMfortel D-series via Action-URL.

↗ Further help under Action URL

Date and time

The date and time are displayed in the phone's status line.

he date and time are needed to log incoming, outgoing and missed calls and to control various functions on the phone, such as automatic updates.

Note: If you change the time zone, the time stamp assigned to all data on the phone is adjusted to suit the new time zone.

Configuration

Web interface:

- **Date and time** > **Basic Settings**

    Details about the settings are in Date and time

Phone:

- **Settings** > **Date and time**

    Details about the settings are in Date and time

Provisioning:

Details about provisioning this function in the Auerswald DokuWiki

Device Information

If required, you can display detailed information about the phone and assign a device name to it.

The device name you assign is:

- displayed in the browser tab after you enter the device type
- displayed on the idle screen, unless a name with higher priority has been assigned (SIP display name (on which provisioning can be performed), display name from identities)
- used as the identifier for a backup file.

  Example: `[Device name]_ [device type]_ config_ [date]_ [time]`.xml

Display Options

Web interface:

- **Dashboard**

  ↗ Details about the settings are in Status information

Phone:

- **Settings** > **About phone**

  ↗ Details about the settings are in About phone

Dial a Subscriber Number Manually

ⓘ **Important:** Always enter external subscriber numbers with a leading exchange line access number. In the default settings, the PBX's exchange line access number is "0", but it can be changed in some PBXs (refer to Advanced Information of the PBX).

ⓘ **Notices:**When dialling through a standard SIP account, you can prevent automatic output of the area code by entering a leading "#" and thereby reach for example the VoIP provider's voice mailbox. If you want to exit dial preparation without dial the subscriber number entered, press Back ⬅.

↗ Description of the operation under Dialled calls

Direct call

The Direct call function supports people who need assistance or children who are unable to dial a specific number or perform other actions on the telephone. To use this function, both the Destination and the Waiting time before call starts in s must be defined. The function will not become active unless a Destination is entered. As soon as the receiver is picked up or placed next to the device, the Destination is called after the defined Waiting time before call starts in s has elapsed.

The direct call can be started immediately by pressing one of the following keys

- ⏎ OK
- 🔊
- 🎧

The Direct call countdown is cancelled if:

- a different phone number or digit is entered
- the receiver is hung up

Configuration

Web interface:

- **Telephony** > **Call settings** > **Direct call**

  ↗ Details about the settings are in [Direct call](#)

Phone:

- **Settings** > **Telephony** > **Direct call**

  ↗ Details about the settings are in [Direct call](#)

Direct call

Provisioning:

⬈ Details about provisioning this function in the Auerswald DokuWiki

Display and LED

Settings made in the Brightness and Contrast menu options affect the following areas:

- Potential device idle state

- The brightness and contrast on the phone display

- The brightness and contrast on all key extension modules that are connected.

- The way the Message LED behaves.

Configuration

Web interface:

- **Display and LED** > **Basic Settings**

  ↗ Details about the settings are in Display and LED

Phone:

- **Settings** > **Display and LED**

  ⓘ **Note:** You can only make settings that control LED behaviour on the phone.

  ↗ Details about the settings are in Display and LED

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Do not disturb

If Do-not-disturb is enabled, inbound calls are rejected.

The caller hears the busy tone.

The Do not disturb function has no effect on dialled calls and callbacks.

Configuration

Web interface:

- **Telephony** > **Call signalling** > **Do not disturb**

  ↗ Details about the settings are in [Call signalling](#)

Phone:

- **Settings** > **Telephony** > **Do not disturb**

  ↗ Details about the settings are in [Telephony](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

DTMF

Press the number block keys on the telephone to generate what are known as dual tone multifrequencies (DTMFs). These are used to transmit specific information to the PBX, provider or destination.

Configuration

Web interface:

- **Identities** > **Options for experts**

  Details about the settings are in Options for experts

Phone:

- **Settings** > **Identities** > **Options for experts** > **SIP**

  Details about the settings are in SIP

Provisioning:

Further help in the Auerswald DokuWiki

Fallback identity

Link to video (only in German) https://www.youtube.com/watch?v=3aJlpAHd0Zg

A configured fallback identity supports system operation. If the main identity fails, the fallback identity is automatically registered on the PBX/at the provider.

Note: This process starts automatically and may take a few minutes.

The fallback identity only has reduced functionality. For example, functions cannot be assigned to some function keys.

When a fallback identity is active, the message key flashes and informs the user about the current status.

When the main identity is restored, the fallback solution is automatically disabled and operation starts again, as it was before.

Each main identity can only have one fallback identity and vice versa. You cannot set up a chain or ring.

Requirements for use

• The fallback identity must be configured on a different PBX or at a different provider.

Configuration

Web interface:

• **Identities** > **Options for experts** > **Fallback for**

⬈ Details about the settings are in Options for experts

Phone:

- **Settings** > **Identities** > **Options for experts** > **Fallback for**

  ⬈ Details about the settings are in Fallback for

Provisioning:

⬈ Further help in the Auerswald DokuWiki

Process

You can decide whether a configured identity is to function as a main or fallback identity.

- No other settings are necessary for a main identity.

- This setting is necessary for a fallback identity:

1. In the "Fallback for" list, select the identity that is to be covered by a second if it becomes necessary.

2. **Save**

Function keys

The basic version of the device has six function keys which you can program to suit your needs. Each function key has a multi-colour LED that indicates its status, based on its assignment.

The number of programmable function keys on the COMfortel D-200 or D-210 can be extended by adding up to three COMfortel D-XT20 or D-XT20i key extension modules (not included in the scope of delivery). Modules are attached to the right-hand side of the phone.

Note: If you connect more than one key extension module, use a COMfortel D-XT-PS power supply unit (not in the scope of delivery).

The programmable function keys can each be defined three times (on three levels). Consequently, in the fully extended system, there are 198 programmable settings on 66 buttons.

Note: New parameters have been defined for some function keys (e.g. Speed dial) from firmware version 1.10. If functionality is restricted following an update to a higher firmware version, it might be sensible to delete the particular key and then configure it again.

Configuration

Web interface:

- **Function keys**

    ↗ Details about the settings are in Function keys

Phone:

- **Settings** > **Function keys**

    ↗ Details about the settings are in Function keys

Provisioning:

⬈ Further help in the [Auerswald DokuWiki](#)

Process for assigning function keys using the phone

1. Select a button to which a function has not already been assigned. All keys that have a function assigned to them shine orange.

2. If necessary, use the directional pad button to switch the key level or use the soft key**Level**.

3. Press the key to which a function is to be assigned.

   ℹ **Note:** If a key to which a function has not been assigned is pressed for a longer time (long press), the menu for configuring the selected key opens automatically.

   The selected key flashes orange.

4. Select the function that is to be assigned to the key.

5. Set other necessary parameters, if necessary.

6. Confirm your entries with the soft key**OK** or with OK ⌷ ᴼᴷ ⌷.

   The assigned key label is displayed on the display.

   The key can be used with the function that has been assigned to it.

Selecting a Key Level

On the home screen: Change between key levels by pressing the following keys:

- directional pad Left and Right keys

- softkey**Level**

in the Choose a key: Change between key levels by pressing the following key:

- softkey**Level**

On the key extension module: Change between key levels by pressing the following key:

- **control rocker**

On the basic device, you can see the current key level on the horizontal scroll bar under the date display.

On the key extension module, you can see the current key level on the horizontal scroll bar on the top and bottom edge of the display.

The display content changes according to the selected key level. The function key names are also displayed.

If no function keys have functions assigned to them, the display does not change.

Process for assigning function keys using the Web Interface

1. Select a button to which a function has not already been assigned.

2. Change to the right level, if required **(Level 1– 3 tab)**.

3. Select the function that is to be assigned to the key.

4. Set other necessary parameters, if necessary.

5. Save your settings.

   The assigned key label is displayed on the display.

   The key can be used with the function that has been assigned to it.

Extending the range of functionality using provisioning

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

Requirement for use

• Provisioning from PBX search enabled

⤴ Further help under Using provisioning

• PBX firmware from Version 7.4

• Suitable template < > phone assignment on the PBX

During the next provisioning process, the configurations in the template file are loaded into the phone. The additional PBX functions are in the **Choose a function** (phone) or **Function** menu (web interface) and can be assigned to the function keys.

⤴ Further help in the Auerswald DokuWiki

Function templates

If a device in the COMfortel® D-series is used on an Auerswald/FONtevo PBX, this function is used to provide access to templates and scripts that are specially designed for running on Auerswald/FONtevo PBXs. It is not necessary to have correctly configured provisioning to use it.

Configuration

Web interface:

- **Using provisioning** > **Additional function templates**

  ↗ Details about the settings are in <u>Additional function templates</u>

Phone:

- **Settings** > **Using provisioning**

  ↗ Details about the settings are in <u>Using provisioning</u>

Provisioning:

↗ Further help in the <u>Auerswald DokuWiki</u>

Hands-free calling mode

During hands-free calling, the loudspeaker and microphone on the device are switched on. This enables you to phone if you don't have a hand free. Other people in the room can take part in the call.

↗ Description of the operation under Hands-Free Calling

Headset mode

A headset enables you to participate in calls without actually holding the receiver, unlike when you use a receiver. Corded and cordless headsets are supported. Headsets that are connected to the telephone are prioritised as follows:

1. USB headsets (COMfortel D-2x0 only)

2. Headsets with a DHSG interface

Caution: Disconnect the telephone from the power supply before plugging in a USB headset.

Note: To connect an USB headset, only use the USB socket on the telephone and not on the COMfortel D-XT20/D-XT20i key extension module.

A headset must be plugged in for these functions to work:

- Initiating headset calls

- Transferring an active call to the headset

- Ending a call in headset operation

- Initiating the listen-in function on the receiver

- Ending the listen-in function through the receiver

Description of the operation under Headset Call

Identities

General
PBX
VoIP provider

General

An identity contains all the configuration data needed to register on a PBX/ with the provider and to enable telephoning. Up to 6 identities can be configured for the phone. One of these identities is the default identity.

The following applies to the default identity:

- There is only one default identity.

- The default identity is automatically used for dialled calls.

- If only one identity has been configured, it is automatically the default identity.

- If more than one identity has been set up, you can configure an Identity function key for each identity. If necessary, (e.g. if there is a problem with the default identity), you can simply press the appropriate function key to set another current identity as the default identity.

  Note: The exception to this are the fallback identities. Fallback identities inherit their status from the main identity.

Caution: The default identity does not switch automatically. If a default identity fails, the LED on the appropriate function key lights up red.

PBX

Requirements for configuration

- You need to know the internal subscriber number for the phone entered in the PBX configuration

- You need to know the password entered in the PBX configuration

- You need to know the IP address set for the phone on the PBX

- The PBX must support the different codecs.

⬈ Further help under [Codecs](#)

Configuration

1. Enter the data specified by the PBX and make settings

   SIP Username = internal number as assigned on the PBX

   SIP Registrar = The PBX's IP address

2. Enable an identity

Web interface:

- **Identities** > Display settings ⌄

  ⬈ Details about the settings are in [Identities](#)

Phone:

- **Settings** > **Identities**

  ⬈ Details about the settings are in [Identities](#)

Provisioning:

⬈ Details about provisioning this function in the [Auerswald DokuWiki](#)

VoIP provider

↗ Link to video (only in German) https://www.youtube.com/watch?v=v5v923ASa2M

Requirements for configuring an identity

- You need to know the user name assigned by the VoIP provider
- You need to know the password assigned by the VoIP provider
- You need to know the Authentication ID assigned by the VoIP provider
- You need to know the IP address assigned by the VoIP provider
- The VoIP provider must support the different codecs.

↗ Further help under Codecs

Configuration

1. Enter the data specified by the PBX and make settings

   SIP Username = internal number as assigned on the PBX

   SIP Registrar = The PBX's IP address

2. Enable an identity

Web interface:

- **Identities** > Display settings ⌄

  ↗ Details about the settings are in Identities

Phone:

- **Settings** > **Identities**

  ↗ Details about the settings are in Identities

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Idle state

The telephone enters its idle state if it is not used before the time set here elapses.

The idle state display is displayed when the phone is in its idle state.

The idle state display has a reduced brightness level.

When the device enters its idle state, the display returns to the home screen. All previously opened menus are closed.

Press any key to bring the phone out of its idle state.

ℹ️ **Notices:**"No automatic" is set as the default value. The idle state does not affect the LED displays on assigned function keys.

Configuration

Web interface:

- **Display** > **Basic Settings** > **Idle state**

  ↗ Details about the settings are in Display and LED

Phone:

- **Settings** > **Display and LED** > **Idle state**

  ↗ Details about the settings are in Display and LED

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

InterCom

The InterCom function enables an audio connection to be established to a phone without someone having to actively take a call (for example, in a medical practice). Communications takes place bi-directionally so that a person who is close by can use the intercom created to talk to the caller.

Requirement for use

- PBX firmware from Version 8.2 with activation for **D-series convenience package**
- For the InterCom destination:
    - InterCom permission granted in the PBX
    - Activated function: **Automatic call acceptance** > **On SIP request**
    - Optional: **Beep when connecting**, **Interrupt active calls**
- For the InterCom source:
    - Configured speed dialling key with **InterCom** parameter activated

Configuration (InterCom destination)

Web interface:

- **Telephony** > **Call settings** > **Automatic call acceptance**
    - ↗ Details about the settings are in Automatic call acceptance

Phone:

- **Settings** > **Telephony** > **Automatic call acceptance**
    - ↗ Details about the settings are in Telephony

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Configuration (InterCom source)

Web interface:

- **Function keys** > new function key > **Speed dial**

  ↗ Details about the settings are in Function keys

Phone:

- **Settings** > **Function keys** > new function key > **Speed dial**

  ↗ Details about the settings are in Function keys

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Process

1. Press a speed dialling key configured for InterCom.

   The function key's LED flashes red. If a call connection has been made, the LED starts shining red.

2. If the **Beep when connecting** setting is activated, a short signal tone is output at the InterCom destination.

3. A bi-directional connection is then present between the InterCom source and InterCom destination without a call being actively taken.

Jitter buffer

Specifies how many RTP packets can be cached to buffer or compensate for disruptions.

Configuration

Web interface:

- **Network** > **Topology** > **Jitter buffer**

    ↗ Details about the settings are in <u>Jitter buffer</u>

Phone:

- **Network** > **Topology** > **Basic Settings**

    ↗ Details about the settings are in <u>Topology</u>

Provisioning:

↗ Details about provisioning this function in the <u>Auerswald DokuWiki</u>

Language and location

You can configure the phone to suit the particular locale by setting the language for the screen and the keyboard.

Configuration

Web interface:

- **Start** > **Language and location**

  ↗ Details about the settings are in <u>Language and location</u>

Phone:

- **Settings** > **Language and location**

  ↗ Details about the settings are in <u>Language and location</u>

Provisioning:

↗ Details about provisioning this function in the <u>Auerswald DokuWiki</u>

ⓘ **Note:** In the web interface, you can set both the language and location, both for the web interface and for the phone.

LDAP

Using LDAP (Lightweight Directory Access Protocol), you can, for example, make phone books on a PBX available to the phone. Stored contact information is called and displayed in

- All call lists

- contact lists

- calls

- suggestions lists in dial preparation.

None of the information for a contact is stored on the device.

Requirement for configuration

- An LDAP server (for example, a PBX) and a minimum of one configured LDAP account.

Configuration

Web interface:

- **LDAP**

    ↗ Details about the settings are in [LDAP](#)

  ℹ **Note:** The settings under **LDAP** > **LDAP filter** should only be carried out by experts. LDAP filter syntax is used. RFC 2254 and RFC 4515 are used.

Phone:

- **Settings** > **LDAP**

    ↗ Details about the settings are in [Contacts/LDAP](#)

LDAP

Provisioning:

⬈ Details about provisioning this function in the [Auerswald DokuWiki](#)

LLDP

IP phones use LLDP to collect and query device-specific information from devices that are directly connected in the network. This automatically updates the VLAN configurations contained in the LLDP packages. The VLAN configurations are uploaded from the switch to the IP phone. An IP phone can therefore be connected to any LLDP-enabled switch, receive its VLAN configurations (internal port) and then start communications via the VLAN (plug and play networks). To do this, the IP phone sends its device information using the LLDP-MED software upgrade.

Requirements for use

- Devices that are directly connected in the network must also use LLDP.

Configuration

Web interface:

- **Network** > **Topology** > **Basic Settings**

    ⤴ Details about the settings are in Topology

Phone:

- **Network** > **Topology** > **Basic Settings**

    ⤴ Details about the settings are in Topology

Provisioning:

⤴ Details about provisioning this function in the Auerswald DokuWiki

Logging

The logging process records all important events, errors and warnings.

A difference is made between:

- logging (errors and warnings) and
- extended logging (debug info).

To display the logged data, press **Web interface** > **Dashboard** > **Status Info** > 📄.

Configuration

Web interface:

- **Access** > **Logging**

  ↗ Details about the settings are in [Logging](#)

Phone:

- **Settings** > **Access**

  ↗ Details about the settings are in [Access](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

Loudspeaker listening mode

In loudspeaker listening mode, both the loudspeaker in the receiver and the loudspeaker in the device are switched on. This enables other people in the room to listen in on the call. The call is continued using the microphone in the receiver (not hands-free operation).

Description of the operation under Loudspeaker listening

Music on hold

When an incoming call is on hold, the "music on hold" is played.

ℹ **Note:** If "Music On Hold" has already been enabled on the PBX/at the provider, no additional settings are necessary here.

Configuration

Web interface:

- **Identities** > **Options for experts** > **Music on hold**

  ↗ Details about the settings are in Options for experts

Phone:

- **Settings** > **Identities** > **Options for experts** > **Audio** > **Music on hold**

  ↗ Details about the settings are in Audio

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Name sources

Different name sources can be prioritised in a list per Identity. The prioritisation is used to decide which name is shown in the display for incoming/outgoing calls.

These name sources are available:

- **Speed dial**
- **Contacts**
- **SIP-P-Asserted-Identity**
- **SIP-To-From**
- **SIP-Contact**
- **LDAP**

Note: The sequence in which the name sources are displayed corresponds to their default settings.

For SIP-P-Asserted-Identity, SIP-To-From and SIP-Contact: if the SIP contact in the SIP header contains a display name, this is the name used in the display.

For speed dialling keys, contacts and LDAP: The phone number in memory is shown on the display. If an entry with a suitable number is found, its name is displayed. If a label has been set for a speed dialling key, this label is displayed.

Note: If the list is empty, only the phone numbers are displayed.

Configuration

Web interface:

- **Identities** > **Options for experts** > **Name sources**

⤢ Details about the settings are in <u>Options for experts</u>

Phone:

- **Settings** > **Identities** > **Options for experts** > **Name sources**

  ⤢ Details about the settings are in <u>Name sources</u>

Provisioning:

⤢ Details about provisioning this function in the <u>Auerswald DokuWiki</u>

NAT

NAT (Network Address Translation) converts an IP address, for example of a private network, into an IP address on the public network. This gives all computers that communicate with each other in the private network access to the Internet via a single IP address. However, the internal IP addresses on the private network are not accessible for the Internet.

Requirements for configuration

- Configured DNS server
- When selecting the option **Active:** the STUN server set up for the provider/PBX

❗ **Important:** Every open port on the NAT router is a security risk. Apply the appropriate security measures.

Configuration

Web interface:

- **Identities** > **NAT**

  ↗ Details about the settings are in <u>NAT</u>

Phone:

- **Settings** > **Identities** > **Options for experts** > **NAT**

  ↗ Details about the settings are in <u>NAT</u>

Provisioning:

↗ Details about provisioning this function in the <u>Auerswald DokuWiki</u>

Network

General
IPv4
IPv6
VLAN
VPN
Topology

General

A network is a group of more than one computer and other communication devices, which are connected together. It provides a way for multiple users to access shared resources.

These protocols can be set up for the device:

- IPv4
- IPv6
- VLAN
- VPN

IPv4

A minimum of one IPv4 address and one subnet mask is required, to enable a device to communicate on the network.

Requirement for configuration

- There must be a connection to the network

Configuration

Web interface:

- **Network** > **IPv4**

    ↗ Details about the settings are in IPv4

Phone:

- **Settings** > **Network** > **IPv4**

    ↗ Details about the settings are in IPv4 settings

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

IPv6

A minimum of one IPv4 address and one subnet mask is required, to enable a device to communicate on the network.

Thanks to the longer IPv6 addresses, IPv6 offers a greater number of possible IP addresses than IPv4.

An IPv6 address is 32 characters long and consists of 8 blocks, each containing 4 characters.

Example: 2001:0db8:85a3:08d3:1319:8a2e:0370:7344

Requirement for configuration

- There must be a connection to the network

Configuration

Web interface:

- **Network** > **IPv6**

    ↗ Details about the settings are in IPv6

Phone:

- **Settings** > **Network** > **IPv6**

    ↗ Details about the settings are in IPv6 Settings

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

VLAN

A VLAN (Virtual Local Area Network) is a logical network within a physical network.

Requirements for configuration

- VLAN is provided in the network
- The network interface type is "Enable LAN Port VLAN"

Configuration

Web interface:

- **Network** > **VLAN**

    ↗ Details about the settings are in [VLAN](#)

Phone:

- **Settings** > **Network** > **VLAN**

    ↗ Details about the settings are in [VLAN](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

VPN

A VPN is a logical private network on a publicly accessible infrastructure. Only communications partners that are on this private network can exchange data and information. Data is exchanged via the VPN tunnel, where the tunnel acts as the envelope in which the data is transported.

Requirement for use

- Each end point must support the specific tunnelling protocol that is being used.

- You need to know the access data required to configure a VPN connection.

Configuration

Web interface:

- **Network** > **VPN**

  ↗ Details about the settings are in VPN

Phone:

- **Settings** > **Network** > **VPN**

ℹ **Note:** The VPN can merely be enabled/disabled on the phone. It can only be configured in the web interface or by provisioning.

↗ Details about the settings are in VPN

Provisioning:

ℹ **Note:** An example configuration, for a FRITZ!box (a German DSL modem/router), is described in the Auerswald DokuWiki. This

configuration cannot be applied to VPN connections that are to be set up in a company environment.

↗ Details about provisioning this function in the Auerswald DokuWiki

Process

1. Activate **VPN** by dragging the slider to the right.

   VPN can alternatively be enabled on the phone.

2. Select the VPN type.

3. Enter the following parameters:

   – **Name**

   – **Server**

   – **User name**

   – **Password**

   – **DNS server**

   – **Search domain**

   – **IPsecIdentifier**

   – **Forwarding routes**

   – **preinstalled IPsecSecret**

4. Load files for these parameters:

   – **IPsecUserCert**

   – **IPsecUserPrivateKey**

   – **IPsecCaCert**

   – **IPsecServerCert**

- **OpenVPN Config**

Follow these steps: Via **Browse** select the corresponding file > **Save**.

Other options in the upload fields and in the context menu which is now displayed are:

- **View file**
- **Download**
- **Close**
- 🗑 Delete
- 📋 Copy

⚠ **Caution:** Settings for VPN connections of the type **IPsec Xauth RSA** and **IPsec Hybrid RSA** can vary greatly, depending on which server is in use. The network administrator must predefine the configuration.

Topology

Topology settings are non-identity-specific.

Configuration

Web interface:

- **Network** > **Topology**

  ↗ Details about the settings are in [Topology](#)

Phone:

- **Network** > **Topology**

  ↗ Details about the settings are in [Topology](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

Notifications

The descriptions of particular events, such as healthcare-related alarm messages, can be displayed as popup messages. In addition, new popup messages can be announced with an audible tone. The following functions are available for popup messages:

- **Clear**

- **Ignore**

    Hides the popup message. Press 🔲 to display previous notifications and delete them if required.

Requirement for use

- Device must be operating on a PBX/provider with a connected alarm server

- Optional: The **Play popup notification sound** setting can be selected.

Configuration

Phone:

- **Settings** > **Identities** > select an identity > **Options for experts** > **Audio**

    ↗ Details about the settings are in [Identities](#)

Provisioning:

↗ Further help in the [Auerswald DokuWiki](#)

Outbound proxy

The outbound proxy is an intermediate server that processes all VoIP requests and connections going to the provider (except for registration).

You must make settings in the outbound proxy range if:

- the IP address cannot be resolved
- an outbound proxy is present in your own network
- the provider requires an outbound proxy

Requirements for configuration

- Data provided by the VoIP provider/administrator for the **Outbound Proxy host** and **Outbound Proxy port**

Configuration

Web interface:

- **Identities** > **Options for experts** > **Outbound proxy**
    - ↗ Details about the settings are in Outbound proxy

Phone:

- **Settings** > **Identities** > **Options for experts** > **Outbound proxy**
    - ↗ Details about the settings are in Outbound proxy

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Park position

If a park position key has been configured, press this key to park a call or unpark it again. By default, you can configure up to ten park position keys.

Note: By activating specific extra features the system, it can be extended to 30 park positions for specific countries.

The hold time is configured in the PBX and displays the maximum number of minutes that a call can be held in the internal park zone.

Requirement for use

- **Call parking** configured on the PBX
- Configured park position key

Configuration

Web interface:

- **Function keys** > **Park position**

  Note: The phone number/URI to be assigned consists of the internal base phone number assigned in the PBX for the open query call and the park position (0 to 9).

  Details about the settings are in Park position

Phone:

- **Settings** > **Function keys** > **Park position**

  Note: The phone number/URI to be assigned consists of the internal base phone number assigned in the PBX for the open query call and the park position (0 to 9).

⤴ Details about the settings are in [Park position](#)

Provisioning:

⤴ Further help in the [Auerswald DokuWiki](#)

Process

- An active call is in progress.

- Press a free park position key to park the call (LED stops shining on the key).

  No music on hold is played.

  The LED on the park position key shines red.

- Press an in-use park position key (key's LED is shining red) to unpark a call from that park position.

  The park position key's LED then stops shining.

ℹ **Note:** If the call is parked for longer than the hold time specified in the PBX, you have these options:

  - A signal is sent to the user who parked the call in the parking zone.

  - If this user is busy, the call is kept in the park zone again for the set time.

  - If the hold time expires a second time, without the call being taken, the connection is disconnected and the caller hears the busy signal.

Passwords

Passwords are used for authentication. The user uses the agreed character string to log on to the web interface.

When you set a new password, comply with the conventions below. The password should:

- be different from the previous one used
- contain at least one number
- contain at least one capital letter
- contain at least one lower case letter
- contain at least one special character
- be at least 9 characters long
- be different from the user name

ℹ️ **Note:** In general: If the password is complicated, it can be shorter. If a password is longer, it can use simpler character strings. Coloured bars provide direct visual feedback about the password strength when a password is entered.

Configuration

Web interface:

- From the factory settings: Password dialogue is displayed when user enters phone's IP address in web browser
- **User**

  ↗ Details about the settings are in <u>User</u>

Phone:

Passwords

- **Settings** > **Access** > **Password web interface**

  ⬀ Details about the settings are in Access

Provisioning:

⬀ Details about provisioning this function in the Auerswald DokuWiki

Pickup

You can use the pick-up function to take a call on your own phone that was being made to another phone.

↗ Description of the operation under [Pickup](#)

Requirements for use:

- Permission granted in the PBX for **Pickup**
- Function key with the speed dial for the phone that is to be called configured on it.

Configuration

Web interface:

- **Function keys** > **Speed dial** > **Pickup-choice on key-press**
- **Function keys** > **Speed dial** > **Do not show in pickup list**
- **Function keys** > **Speed dial** > **Pickup of connected calls**

  ↗ Details about the settings are in [Speed dial](#)

Phone:

- **Settings** > **Function keys** > **Speed dial** > **Pickup-choice on key-press**
- **Settings** > **Function keys** > **Speed dial** > **Do not show in pickup list**
- **Settings** > **Function keys** > **Speed dial** > **Pickup of connected calls**

  ↗ Details about the settings are in [Speed dial](#)

- App**Pickup**

  ↗ Details about the settings are in [Pickup](#)

Query Calls

During a query call you are talking with one subscriber while the previous call partner is put on hold by the PBX in the background, and hears the on-hold music.

The phone supports 3 call channels. When the channels in the phone are put on hold in the PBX, they become inactive, so up to 2 calls can be on hold in addition to one active call.

A query call is initiated either by calling a query call subscriber, by picking up during a call, or by accepting a call signalled by the audible indication.

↗ Description of the operation under Pickup

↗ Further help under Call waiting

Options during a query call

- **Splitting a call**

  You can use the transfer function to alternate between the call partners.

  ↗ Description of the operation under Query Calls

- **Ending one of the two calls**

  When you end a query call and would like to continue talking with one of the call partners, you can end the call with the other call partner separately.

  Alternatively, one of the call partners can simply hang up so that you can continue talking with the other call partner.

  ↗ Description of the operation under Query Calls

- **Connecting the two call partners to each other**

    If you would like to let your call partners speak with each other, you can connect them.

↗ Description of the operation under Query Calls

Redial List

↗ Description of the operation under [Redial List](#)

Reset to default

If you reset the phone to default settings, you can delete any data and settings saved on it. This includes, for example, deleting personal data (such as your contacts), function key assignments and identities.

The phone is shut down and then restarted. This can take some time.

ⓘ **Note:** If you reset the phone to its factory settings, you can delete any data and settings saved on it. All personal data is deleted. You cannot undo this action. To save particular settings such as a configured identity, you can first perform a data backup.

↗ Further help under Backup and restore

Requirement for use

- If needed: Back up your personal data and settings

Configuration

Web interface:

- **Update** > **Restart and factory reset** > **Reset to default**

  ↗ Details about the settings are in Update

Phone:

- **Settings** > **Update** > **Reset to default**

  or

- Follow the instructions in **Procedure directly on the device**

↗ Details about the settings are in Update

Procedure directly on the device

1. Disconnect the network cable that supplies the phone with current.

   The power supply is no longer provided.

2. Press and hold these two buttons at the same time:

   - Hold 📞

   - Directional pad ▶

3. Plug the network cable in again.

   The power supply is restored.

   The housing LED first flashes quickly, then more slowly.

4. As soon as the casing LED is flashing more slowly, release both buttons.

   You have reset the phone to its default settings.

ℹ **Note:** If you do not release the Hold button 📞 and the directional pad ▶ in the predefined time window, the phone boots in the normal way. The phone is not reset to its default settings.

Roaming User

Roaming User is a PBX COMtrexx function.

This function enables users who work, for example, in a large office with workplaces that are not assigned to specific people (i. e. "free seating" or "hot desking") to access any telephone that is configured in the PBX. In this way, the user can access the telephone system in the usual way, with their personal settings and authorisations.

Large office use case

The office has workplaces (with telephones) that are not assigned to specific people. These telephones are not assigned to a particular user or roaming user. A roaming user can now log on to one of these telephones. When they change workplace, they simply need to log on to a different telephone.

Office/home office use case

A roaming user has a telephone that is configured in the PBX at their office workplace and in their home office. To move from their office to their home office, the roaming user simply has to log on to their home office telephone.

Requirement for use

Roaming User authorisation must be assigned to the user in the PBX.

Process for logging onto/logging off the telephone

1. The function key →| Login / ←| Logout is provisioned by the PBX.

> ℹ **Note:** If the user is logged out, the PBX provisions the Login function key →|. After the user logs in, the PBX uses provisioning to replace the Login function key with the Logout function key.

2. →| Login: The user uses the Login function key to log on to the required phone with their PIN and user phone number.

3. All the user needs to do is log on to a different telephone.

   The following regulations/restrictions apply:

   – When the user starts using a different telephone, they are automatically logged out of the telephone they were using previously.

   – A user can only ever use one telephone at once.

   – If a user who is authorised to use the Roaming User function changes to a different telephone, they will either find the entries for the local caller list or the central caller list on it, as before (if the corresponding function is enabled).

↗ Further help under Central caller list

↗ Details about provisioning this function in the Auerswald DokuWiki

Silent Alert

The silent alarm function can trigger an instant, unobtrusive action by the destination on the specified URL (example: alarm server). This action might be necessary if, for example, the call recipient is threatened, the tone of the conversation becomes angry, or there is a medical emergency or a hostage situation.

Configuration

Web interface:

- **Function keys** > **Silent Alert**

  ↗ Details about the settings are in Silent Alert

Phone:

- **Settings** > **Function keys** > **Silent Alert**

  ↗ Details about the settings are in Silent Alert

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Process

- A dangerous situation has occurred and the silent alarm function key must be pressed.

- Depending on the predefined delay, a number of different scenarios are possible:

  – Delay 0 sec: The silent alarm is sent after the delay period comes to an end.

- Delay min. 1 sec: A countdown counter appears on the display. The following options are also available

  Dismiss: Cancels the silent alarm.

  Hide: Displays the Idle screen. The countdown counter continues running in the background.

  Start immediately!: Cancels the countdown counter and the alarm is sent immediately.

- An action is triggered at the destination (an alarm server etc.), depending on which URL has been entered.

- Alternatively, a warning can be sent by SIP message. Use the SIP message function key to do this.

Further help under [SIP message](#)

Silent Call

In a silent call, you can trigger a function by pressing the key to which it has been assigned without displaying the call on the telephone display.

Requirement for configuration

- PBX's valid known URI

- Provisioning from PBX search is enabled

  ↗ Details about the settings are in Using provisioning

- adapted .xml file with which the PBX function can be triggered.

Configuration

Via provisioning:

↗ Further help in the Auerswald DokuWiki

Process

1. Provide a suitable provisioning file.

2. Activate **Provisioning from PBX search**.

3. Enter the relevant Provisioning server URL.

4. Set up a function key with the extended function provided by provisioning.

SIP

The Session Initiation Protocol (SIP) network protocol sets up a communication session between two or more subscribers. SIP is only used to negotiate the communication conditions. Other protocols such as RTP are responsible for actual data transfer.

Session Timer, Session Timeout and Protocol type settings must be made, for problem-free communication.

You can select protocol type UDP or TCP.

- **UDP:** (the User Datagram Protocol) is used to send data packet over connectionless non-secure communication lines. UDP itself does not verify whether data has been transmitted successfully. When a UDP packet is sent, the sender cannot assume that the packet will definitely arrive at the recipient. This particular protocol only needs a small amount of additional information, which results in a better data throughput rate in a well-functioning network such as a LAN. UDP is used, e.g., for the DNS (Domain Name Server).

- **TCP:** TCP (Transmission Control Protocol) is a transport protocol that segments data into packets up to a specified size and reliably sends these individual data packets to the recipient address, in the correct sequence. In this process, every data packet sent must be resent until it has been confirmed as arrived. To ensure this happens, a large amount of information is sent along with the actual payload data. Most Internet services are implemented with TCP, e.g., HTTP (WWW), SMTP/POPS (e-mail), etc.

Configuration

Web interface:

- **Identities** > **Options for experts**

↗ Details about the settings are in Options for experts

Phone:

- **Settings** > **Identities** > **SIP**

  ↗ Details about the settings are in SIP

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

SIP message

The phone can send and receive SIP messages. A set message can be sent to a recipient with an appropriately configured function key. A received SIP message is displayed as a notification.

Examples

- In a care home, repeated messages that the patient/resident in room X needs help.

- Using sent SIP messages to warn of possible dangers. A silent alarm can also be triggered for this purpose.

↗ Further help under [Silent Alert](#)

Configuration

Web interface:

- **Function keys** > **SIP message**

  ↗ Details about the settings are in [SIP message](#)

Phone:

- **Settings** > **Function keys** > **SIP message**

  ↗ Details about the settings are in [SIP message](#)

Provisioning:

↗ Further help in the [Auerswald DokuWiki](#)

Process

- An SIP message is to be sent and displayed on a different device.

- Depending on the predefined delay, a number of different scenarios are possible:

  - Delay 0 sec: The SIP message is sent immediately.

  - Delay min. 1 sec: A countdown counter appears on the display. The following options are also available

    Dismiss: Cancels the transmission of the SIP message.

    Hide: Displays the Idle screen. The countdown counter continues running in the background.

    Start immediately!: Cancels the countdown counter and the SIP message is sent immediately.

SIPS

SIPS are SIP data encrypted using the TLS protocol. The network protocol sets up a communication session between two or more subscribers.

The provider encrypts external connections to prevent listening in. The processes used to set up and shut down a connection, and signalling, are all encrypted by SIPS.

Requirement for configuration

- The phone's certificate must be known on the PBX.

Configuration

Web interface:

- **Identities** > **Options for experts** > **SIPS**

  ↗ Details about the settings are in Options for experts

Phone:

- **Settings** > **Identities** > **Options for experts** > **Security** > **SIPS**

  ↗ Details about the settings are in Security

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Sound

Sets the sounds for signalling

- inbound calls

- Notifications

- ringtones by call type

Additional ringtones can be added to the device. The number of ringtones is only limited by the amount of memory. Supported file formats:

- .wav

- .mp3

- .ogg.

Maximum music file size: 5 MB. Personalised ringtones will not be backed up.

Priority for defining ringtones

1. A contact/group with a configured ringtone is present locally for a subscriber number. The contact melody defined in the telephone is used first.

2. The alert info in the SIP Invite has a valid URL for a ringtone that can be downloaded externally.

3. The alert info in the SIP Invite has a pseudo URL or an info field that matches one of the predefined designations, e.g. for group calls, internal calls, external calls, alarm calls, door calls or "mute" calls).

4. The identity has a specific ringtone.

5. A defined default ringtone is used.

6. The Android fallback ringtone is used.

Configuration

Web interface:

- **Sound**

  ↗ Description of the operation under [Sound](#)

Phone:

- **Settings** > **Sound**

  ↗ Description of the operation under [Sound](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

Process

1. All available sounds are listed.

2. A plying of the tone is achieved by

   − navigation in the list using the directional pad keys

     and

   − hovering over the selected entry

3. When you confirm your ringtone selection, the menu closes.

   The selected sound is then used as the default for incoming calls.

Speed dial

With the "speed dial" function, you can not only assign internal and external subscriber numbers to a function key, but also PBX programming sequences.

Speed dial keys can be used in the following ways:

To Dial a Subscriber Number

The subscriber number will be dialled immediately when a speed dial button is pressed. If a speed dial button is pressed during a call, a query call is placed and the saved subscriber number is dialled immediately.

Assignment Overview

Speed dial buttons assigned to internal subscriber and group subscriber numbers provide a way for the user to view the LEDs and see which subscribers and groups are active or busy.

Pickup and Call Pickup

Speed dial buttons defined with internal subscriber and group subscriber numbers enable pickup for these subscribers and groups. If the LED next to the button is blinking red because the saved subscriber number is being called, you can take the call on your phone by pressing the speed dial button and then confirming. If a subscriber allows call pick-up (for example on an answering machine), you can still take the call on your phone in the same manner, even after it has already been accepted.

Configuration

Web interface:

- **Function keys** > **Speed dial**

⬈ Details about the settings are in [Speed dial](#)

Phone:

- **Settings** > **Function keys** > **Speed dial**

  ⬈ Details about the settings are in [Speed dial](#)

Provisioning:

⬈ Further help in the [Auerswald DokuWiki](#)

SRTP

SRTP (Secure Real-Time Transport Protocol) is used to encrypt call data.

ℹ️ **Important:** If you are using this on a PBX, the PBX specifies the encryption method. In contrast, you must adjust the encryption settings on the phone.

Configuration

Web interface:

- **Identities** > **Options for experts** > **SRTP**

  ↗ Details about the settings are in Options for experts

Phone:

- **Settings** > **Identities** > **Options for experts** > **Security** > **SRTP**

  ↗ Details about the settings are in Security

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Trace

If the phone malfunctions, a log can be recorded. It can then be sent to the technical departments (Service, Development) so that the cause of the fault can be investigated.

Requirement for use

• Protected data is logged when a trace recording is produced. Before recording starts, you must confirm that you want it to do so, in the dialogue window.

Important: Comply with the legal notes below before recording a network data stream. The recorded data includes, amongst other things, the following elements, which have special legal protection:

• Data streams from other computers in the network

• Connection data sets with the destination, source, and time information for a telecommunication connection.

• The spoken words from both ends of a telecommunication connection

• Phone configuration data, along with the associated PINs and passwords

• Directory data with names and phone numbers

All the phone's users must be informed that this content will be transferred, and who will process the data obtained, before the data is recorded. It is essential to store the recorded data securely and in accordance with legal requirements. If you want to provide your specialised dealer or the manufacturer with the recorded data, for error analysis, you must ensure that the legal requirements are fulfilled.

Configuration

Web interface:

- **Dashboard** > **Status information**

⬈ Details about the settings are in [Status information](#)

Process

Requirements:

- Pop-up blocker has been disabled in the browser or an exception rule has been defined.

1. Start recording the trace by selecting **Dashboard** > **Status information** > **Start trace**.

2. If it is easy to reproduce the faulty behaviour, try to trigger it now.

3. Stop the recording.

4. The file is saved automatically and displayed in the browser's downloads list.

⬈ Contact [Auerswald Support](#)/ [FONtevo Support](#), to arrange for it to be analysed.

Transferring calls

Call transfer types

Transfer the call with an announcement

Transferring a call without an announcement

Calling a 3rd internal subscriber

Call transfer types

A call partner can be transferred (connected) to an internal or external subscriber.

During an existing call, you can perform a call transfer in these ways:

- Transfer the call with an announcement
- Transfer the call without announcement (blind transfer)
- Calling a 3rd internal subscriber and connecting them to an active call partner

Transfer the call with an announcement

Requirement for use

- Initiated query call

Process

To transfer a call, a query call is placed.

The call is announced.

The two call partners are connected with each other.

↗ Description of the operation under <u>Transferring calls</u>

Transferring a call without an announcement

Also called blind transfer.

Requirements for use

- Initiated query call

- To connect two external communication partners, the following is required: Permission granted in the PBX for **Transfer of external calls to extern**

- Internal transfer destination

Process

If an external call partner is to be transferred to another internal subscriber, there is no need to wait until the subscriber called in the query call picks up. The receiver can be put down during calling.

The call to the internal subscriber will be forwarded.

If the subscriber accepts the call, the connection is established.

If the called subscriber does not pick up, the call returns to the caller's own phone (the phone rings) after 2 minutes (or as set on the PBX or at the provider).

↗ Description of the operation under [Transferring calls](#)

Calling a 3rd internal subscriber

Calling a 3rd internal subscriber and connecting them to an active call
partner

Requirements for use

*   Initiated query call

*   To connect two external communication partners, the following is
    required: Permission granted in the PBX for **Transfer of external calls
    to extern**

Process

An additional call is added to a existing call (for example, when a call
waiting call is accepted).

This second subscriber is to be transferred to another subscriber.

To do so, a third call is triggered and the second subscriber is connected
to the third subscriber.

The original, existing call continues.

Update

An update involves updating the firmware To add new features and resolve any problems that might exist. Here, the device sets up a connection with an update server, from which it downloads the latest firmware. (The settings on the device remain unchanged). There are three ways to carry out an update:

- Manual Update

- Automatic update

- Provisioning

  ↗ Further help under Using provisioning

If the Check for updates function is activated, the device searches for updates at these times:

- Update attempt when the telephone is rebooted

- Update attempt after provisioning

- Update attempt according to the schedule.

ℹ **Note:** An update can trigger a data migration. The phone cannot be used while this is happening. A message appears on the display to tell you a data migration is taking place.

! **Important:** If the updater fails to boot the device (e.g. because there is a defective hard disk partition), the system informs you in an error message. Pass the content of this message on to the Service team if requested.

Requirements for configuration

- Address of the update server

Manual Update

During a manual update, the downloading of the update is triggered manually. After the update, the phone restarts automatically.

Configuring a manual update

1. Enter the Update server URL

   In the default settings: https://update.auerswald.de

Web interface:

- **Update**

  ↗ Details about the settings are in Update

Phone:

- **Settings** > **Update**

  ↗ Details about the settings are in Update

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Manual update process

1. **Web interface:** Press **Start update**

   Phone: Press the Install soft key as soon as the message that a new firmware version is available is displayed on the screen.

Automatic update

In the case of an automatic update from the update server, a regular (e.g. weekly) automatic update can be configured. You can set the weekday and time.

Configuring an automatic update

1. Enable Automatically install updates

2. Enter the Update server URL

   In the default settings: https://update.auerswald.de

3. **Web interface:**

   Set the weekdays and time.

Web interface:

- **Update**

  ↗ Details about the settings are in Update

Phone:

- **Settings** > **Update**

  ↗ Details about the settings are in Update

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Automatic update process

It can take up to 60 minutes to perform an automatic update.

When you set the Weekday/Clock Time for an automatic update, the system checks for new firmware during this time period and downloads and installs it if necessary.

Is automatic update is not enabled, the system only checks for a newer version of the firmware during this time period.

If there is a newer version of the firmware, you are informed in a message on the screen.

User roles and permissions

Roles can be assigned to ensure that only particular functions are made available for use by specific users. The configuration can be restricted to just a few functions.

Access is subdivided into two permission levels:

	Admin	**User**
Web interface	Password protection	Password protection
	Unrestricted use.	Restricted use.
		The following options are not possible:
		Reset the device to the factory settings.
		Add or delete users.
		Trace log recording if an error occurs.
		Restrict the configuration for displaying and removing menu options.
	Display all menus.	Displays menu options selected by the administrators.
		Exception: Permanent summary display of:**Dashboard** , **Network**.

User roles and permissions

	Admin	User
phone	Authorisation to add or delete users or administrators.	Only one's own user settings (username, password) can be edited.
	The phone does not distinguish between authorisation levels. The displays are the same for all users.	
	Displays menu options selected by the administrators. Exception: Permanent summary display of:**About phone** , **Network**.	

Configuration

Web interface:

- **User**

  ↗ Details about the settings are in [User](#)

  ⓘ **Note:** By default, one administrator is configured.

  ⚠ **Caution:** Passwords that are unchanged from the factory settings are a security risk. Change the password without delay.

- **Access**

  ↗ Details about the settings are in [Access](#)

Provisioning:

↗ Details about provisioning this function in the [Auerswald DokuWiki](#)

Using provisioning

Provisioning involves automatically configuring one or more phones using a provisioning server URL. Here, the phone sets up a connection with a provisioning server, over which it receives the necessary configuration data. An update is then carried out.

Note: To ensure interference-free operation, some local functions are not available on the terminals after provisioning by a cloud PBX (e.g. COMuniq ONE). Data stored locally on the phone can be deleted or reset.

Provisioning options

- **Provisioning with DHCP data**

  The phone obtains the provisioning server's URL from a DHCP server. Additional data for controlling provisioning can also be transferred along with the URL.

- **Provisioning using a redirect server**

  The phone obtains the provisioning server's URL from the redirect server.

- **Provisioning from PBX search**

  The phone is configured automatically via the PBX.

  Note: Activate **Provisioning from PBX search**, for example, if you want to add PBX functions to the function key assignments.

- **Provisioning using the provisioning server URL**

  The URL is required if provisioning is not performed using DHCP data or a redirect server.

- **SIP provisioning**

The phone is configured using incoming SIP messages.

Requirements for configuration

- The PBX or phone must be connected to a broadband Internet connection.
- Network access with valid settings

  ↗ Further help under Network

- Vaild time

↗ Further help under Date and time

Configuration

Web interface:

- **Using provisioning** > **Basic Settings**
- **Using provisioning** > **Schedule**
- **Using provisioning** > **SIP provisioning**

  ↗ Details about the settings are in Using provisioning

Phone:

- **Settings** > **Using provisioning**

  ↗ Details about the settings are in Using provisioning

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

Automatic provisioning process

1. Enable Automatic provisioning.

2. Select one of these:

   – Provisioning with DHCP data

   – Redirect Server

   – Provisioning from PBX search

   – Provisioning server URL

   You can select combinations of these options.

3. Configure the schedule.

   ℹ Note: You can only configure the schedule in the web interface.

Automatic provisioning is performed as set in the schedule and also each time the device is restarted.

Manual provisioning process (only on the phone)

1. Select one of these:

   – Provisioning with DHCP data

   – Redirect Server

   – Provisioning from PBX search

   – Provisioning server URL

   You can select combinations of these options.

2. Select **Start provisioning**.

   Provisioning is performed.

SIP provisioning process (only in the web interface)

- Switch on SIP provisioning.

↗ You will find more information on configuration via SIP messages and also the provisioning process in the Auerswald DokuWiki.

Voicemail number

A voice mailbox acts as an answering machine. Depending on the setting, it accepts incoming calls and stores recorded messages as audio files (.wav).

Messages and announcements are not stored on the phone, but on the PBX.

The size of the voice mailbox and the authorisation to use it are set on the PBX. A voice mailbox can be assigned to an individual subscriber or to a group. A subscriber can be given the authorisation to use several voice mailboxes.

You can link a configured voice mailbox with the selected identity by entering the associated voicemail number.

Important: The voice mailbox is only available during operation on the following PBXs:

- COMpact 4000
- COMpact 5010 VoIP
- COMpact 5020 VoIP
- COMpact 5000/R
- COMpact 5200/R
- COMpact 5500R
- COMmander 6000/R/RX
- COMtrexx VM
- COMtrexx Advanced
- COMtrexx Business

    In some PBXs, you will need to install another module (available as an optional extra) to run a voice mailbox:

- COMmander 6000/R/RX: COMmander VMF module
- COMpact 5010 VoIP: COMpact 2VoIP module
- COMpact 5020 VoIP: COMpact 2VoIP or 6VoIP module

Requirement for use

- A configured and assigned voice mailbox

Configuration

Web interface:

- **Identities** > **Options for experts** > **Voicemail number**

  ↗ Details about the settings are in Options for experts

Phone:

- **Settings** > **Identities** > **Options for experts** > **Function codes** >
  Voicemail number

  ↗ Details about the settings are in Function codes

Provisioning:

↗ Details about provisioning this function in the Auerswald DokuWiki

XML Minibrowser

The XML Minibrowser is a feature of COMfortel D-series.

The browser enables menus and dialogs that you have created yourself, in XML, to be displayed. You can upload the XML documents from a normal web server, using http, provided they have been stored there.

You can use this function to display menus and dialogs that you have created specially for that purpose.

You can display and modify apps such as phone books created to meet your own special requirements, lists of missed calls, settings on the PBX on the own needs (for example, Do not disturb, Call forwarding etc.).

Requirement for use

- .XML files stored on a web server

Configuration

Web interface:

- **Function keys** > **XML Minibrowser**

  Details about the settings are in XML Minibrowser

Phone:

- **Settings** > **Function keys** > **XML Minibrowser**

  Details about the settings are in XML Minibrowser

Provisioning:

Further help in the Auerswald DokuWiki

Process

- Press the key with the function **XML Minibrowser**.

  After you press that button, the phone sends an HTTP(s) GET command to the server, waits for the reply, and then decodes and displays this reply just like a normal web browser would.

⤢ Further help in the Auerswald DokuWiki

Telephoning

Information about the section

This section describes the different ways that you can use the phone for telephoning.

ⓘ Important: The functions described in these instructions have been tested and in operation on PBXs produced by Auerswald/FONtevo, and then described. If the phone is operated on other PBXs or VoIP providers, functionality may be partially or completely restricted.

ⓘ Note: There are alternatives to the functions below. These alternatives have not always been listed.

To create/end a connection

Pick up the receiver

or

Loudspeaker ⏹

or

Headset ⏹

or

OK ⏹

or

softkey**Accept** / **End**

reject a call

softkey**Reject**

or

Information about the section

Back 🔙

Holding a call/Reconnecting from a call that is on hold

Hold 📞

or

OK 🆗

Dial tones

If the phone is being operated on a PBX in which an exchange line access
number has to be dialled to access the exchange line, provisioning the dial
tone in the Identity area will enable different dial tones to be played for the
user: depending on whether the user is still within the PBX or in the public
network.

⤢ Details about provisioning this function in the Auerswald DokuWiki

Internal dial tone, 3x short

• After the receiver has been picked up within the PBX.

External dial tone, 1x long

• After the receiver has been picked up on the exchange line because:

    − automatic exchange line request is enabled

    − a direct exchange line phone has been selected

      or

    − "0" has been dialled.

Incoming calls

Accepting a call

Pick up the receiver

or

OK ⟦ OK ⟧

or

Headset ⟦ 🎧 ⟧

or

Loudspeaker ⟦ 🔊 ⟧

or

softkey**Accept**

Rejecting a call

softkey**Reject**

Putting an accepted call on hold

Hold ⟦ 📞 ⟧

or

OK ⟦ OK ⟧

Reconnecting an on-hold call

Hold ⟦ 📞 ⟧

or

Incoming calls

OK `OK`

Call-Waiting Calls

Accepting a call

softkey**Accept**

Further help under Call waiting

Rejecting a call

softkey**Reject**

Further help under Call waiting

Forwarding

Transfer [ ]> Enter destination number > Loudspeaker[ ]

or

Transfer [ ]destination number > OK[OK]

Further help under Call waiting

Dialled calls

Dial a Subscriber Number Manually

Pick up the receiver > dial a phone number

↗ Further help under <u>Dial a Subscriber Number Manually</u>

Dial preparation

Dial the phone number > pick up the receiver

↗ Further help under <u>Dial a Subscriber Number Manually</u>

Dialling from within the call history (all calls)

softkey**Calls**> select the required phone number > pick up the receiver

↗ Further help under <u>Overview of call lists</u>

Dialling from within the call history (missed calls)

softkey**Calls**> softkey**Missed**> select the required phone number > pick up the receiver

or

Message key⌷▨⌷ > **You have missed calls.**> OK⌷ᴼᴷ⌷> select the required phone number > pick up the receiver

↗ Further help under <u>Missed calls</u>

Dialling from within the redial list (outgoing)

softkey**Calls**> softkey**Outgoing**> select the required phone number > pick up the receiver

or

Redial list button⬛> select the required phone number > pick up the receiver

Dialling from the contacts list

softkey **Contact**> Select contact > OK⬛> Pick up the receiver

🛈 **Note:** If more than one number is stored for a contact, you can select the number to call before you pick up the receiver.

↗ Further help under Contact

Call

Putting a call on hold

Active call > Hold [☏]

or

OK [OK]

↗ Further help under Call

Reconnecting an on-hold call

To do this, press Hold again [☏]

or

press OK again [OK].

↗ Further help under Call

Accepting a call

During an active call, the LED on the key with the Speed dialling function assigned to it shines red. Press the key with the Speed dialling function assigned to it to pick up the call.

Muting a call

Microphone [🎤]

Switching off muting

To do this, press Microphone again [🎤]

Call

Ending a call

To do this, hang up the receiver

or

softkey**End**

⬈ Further help under Call

Sending a DTMF signal during a call

To do this, enter digits and special characters

⬈ Further help under Call

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Query Calls

Initiating a query call (during a call)

Hold [☎] > enter destination number/contact > OK [OK]

or

OK [OK] > enter destination number/contact > OK [OK]

or

Hold [☎] > softkey**Dial** > enter destination number/contact > OK [OK]

When you connect a COMfortel®D-XT20:

Hold [☎] > Speed dial key (key extension module)

or

press the speed dial button (key extension module)

↗ Further help under Query Calls

Ending the query call and returning to the previous call

softkey**End** > Hold [☎]

or

softkey**End** > OK [OK]

↗ Further help under Query Calls

Splitting a call (during a query call)

Select the call partner to whom you want to change > Hold [☎]

↗ Further help under Query Calls

Ending one of the two calls

Either one of the call partners hangs up the receiver

or

you can select the call partner for whom you want to end the call > soft key**End**

Further help under [Query Calls](#)

Connecting the two call partners to each other

To do this, hang up the receiver

or

softkey**Join**

Further help under [Query Calls](#)

Transferring calls

Transfer the call with an announcement

Requirements:

- Initiated query call

↗ Further help under <u>Query Calls</u>

Proceed as follows::

softkey**Join**

↗ Further help under <u>Transfer the call with an announcement</u>

Transfer the call without announcement (blind transfer)

Proceed as follows:

Hold ☏ > softkey**Dial**> enter destination number > OK ⌗ > softkey**Join**>
To do this, hang up the receiver

↗ Further help under <u>Transferring a call without an announcement</u>

Conference call

All the options listed above are for a conference call that you have initiated yourself.

Initiating a call

Scenario:

- there is an existing connection between you and at least two other subscribers.

softkey**Conf.**

⤴ Further help under [Conference call](#)

Splitting a call

Select subscriber > OK⌷ OK ⌷

ℹ **Note:** Press again the soft key**Conf.**to restore the conference.

⤴ Further help under [Conference call](#)

Connecting the other two conference call partners to each other

softkey**Join**

⤴ Further help under [Conference call](#)

Splitting a conference call

softkey**Split**

One subscriber is put on hold and there is an active connection with the other subscriber. You can change between holding a call and an active call.

Press Hold to do this🕽.

↗ Further help under Conference call

Holding a conference call and initiating an additional call

Select Conference > Hold🕽> softkey**Dial**> OK ᴼᴷ

The conference call partners are put on hold and hear music on hold.

When the call ends, you can continue the conference call by pressing Hold again🕽.

↗ Further help under Conference call

Ending a call with a conference call partner

Select the subscriber for whom you want to close the connection > soft key**End**

Then, there will only be a connection with the remaining subscriber.

↗ Further help under Conference call

Ending a conference call completely

To do this, hang up the receiver

or

Conference call> softkey**End**

↗ Further help under Conference call

Hands-Free Calling

Accepting a call in hands-free calling mode

Incoming call > Soft key**Accept**

or

Incoming call > Loudspeaker 🔊

or

Incoming call > OK ⌷ OK ⌷

↗ Further help under Hands-free calling mode

Initiating a call in hands-free calling mode

Select phone number/contact > OK ⌷ OK ⌷

or

Select phone number/contact > loudspeaker 🔊

or

Select phone number/contact > wait for a short while

Requirement: Automatic start of callactivated.

↗ Further help under Automatic start of call

↗ Further help under Hands-free calling mode

Ending a call in hands-free calling mode

softkey**End**

or

Loudspeaker ⬜🔊

or

Back ⬜⏪

↗ Further help under Hands-free calling mode

Loudspeaker listening

Switching on during a call through the receiver

Loudspeaker ⬛🔊⬛

↗ Further help under <u>Loudspeaker listening mode</u>

Switching off and returning to a normal call

Loudspeaker ⬛🔊⬛

↗ Further help under <u>Loudspeaker listening mode</u>

Ending a call in loudspeaker listening mode

To do this, hang up the receiver

or

softkey**End**

↗ Further help under <u>Loudspeaker listening mode</u>

Headset Call

Personally initiating a call when headset operation is switched off (headset LED is not shining)

Headset ⌷🎧⌷

or

EHS

↗ Further help under Headset mode

Personally initiating a call when headset operation is switched on (headset LED is shining green)

Headset ⌷🎧⌷

or

softkey**Accept**

or

OK ⌷ OK ⌷

or

EHS

↗ Further help under Headset mode

Transferring an active call to the headset

Headset ⌷🎧⌷

or

EHS

⤴ Further help under Headset mode

Ending an active call in the headset

Headset 🎧

or

softkey**End**

⤴ Further help under Headset mode

Initiating the listen-in function on the receiver during a headset call

Pick up the receiver > press Headset 🎧

⤴ Further help under Headset mode

Cancelling the listen-in function on the receiver during a headset call

To do this, hang up the receiver

⤴ Further help under Headset mode

Caller list

To open the call history

softkey**Calls**

↗ Further help under Caller list

To list missed calls

softkey**Calls**> softkey**Missed**

or

Messages [✉] > **You have missed calls.**> OK[OK]

↗ Further help under Caller list

To list outgoing calls

softkey**Calls**> softkey**Outgoing**

or

Redialling [⬤]

↗ Further help under Caller list

Options for displaying entries in an active list

softkey **Menu**

or

OK [OK](hold down the key for approx. 2 sec. .)

↗ Further help under Caller list

Editing an entry in the call history before making a call

softkey**Calls**> softkey **Menu** > **Edit number before call**> OK⌷ ok

↗ Further help under Caller list

Adding an entry in the call history to the contacts

softkey**Calls**> softkey **Menu** > **Add to contacts**> OK⌷ ok

↗ Further help under Caller list

Deleting entries from the call history

softkey**Calls**> softkey **Menu** > **Clear entry**

or

Delete⌷ ⌫

↗ Further help under Caller list

Deleting an entire call history

Delete⌷ ⌫ (hold down the key for approx. 2 sec. .)

↗ Further help under Caller list

Calling from an opened call history

Pick up the receiver

or

OK ⌷ ok

or

softkey **Menu** > **Call**> OK ⌷ᵒᵏ⌷

or

Loudspeaker ⌷ ◀) ⌷

or

Headset ⌷ ᵒ ⌷

↗ Further help under Caller list

To leave the call history

Back ⌷ ⌫ ⌷

↗ Further help under Caller list

Pickup

Procedure with speed dial configured

The LED on the button with the speed dial function assigned to it flashes red.

When you press the flashing function key, you pick up the call.

↗ Further help under Pickup

Proceed as follows without speed dial configured

Enter pick-up code an phone number of the internal phone > pick up the receiver or confirm with ⌷OK⌷ **OK** .

↗ Further help under Pickup

Using the pickup list

All the calls (incoming or already connected) that can be selected and taken directly are displayed in the pickup list.

ℹ **Note:** The pickup list only displays calls if speed dialling keys have been set up.

Settings for the function key: **Speed dial**:

- disabled:**Do not show in pickup list**
- Optional:**Pickup of connected calls**

If at least one call is present in the pickup list, lift the receiver to take over the call directly.

↗ Further help under Pickup

Device Overview

Front View
Rear View

Front View

Each device has a display and different keys (some of which have LEDs). A device can only be operated using the keys.

Keys and LEDs on the basic unit
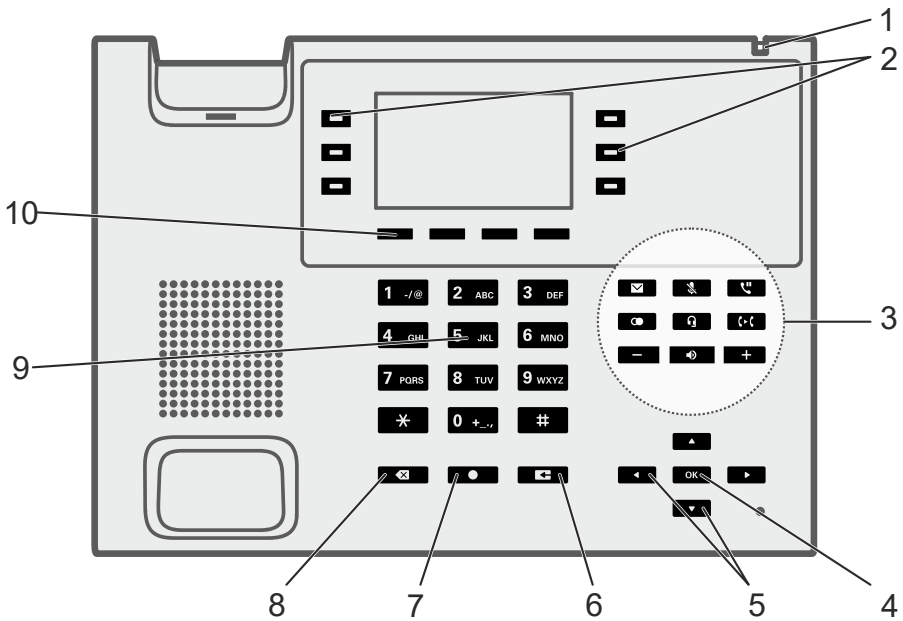
1. Casing LED

2. Function keys (3x on each side)

3. Dedicated function keys

4. OK

5. Directional pad keys (Up, Down, Right, Left)

6. Back

7. Home

8. Delete

9. Keypad

10. Soft keys: situation-dependent function keys below the display (4x)
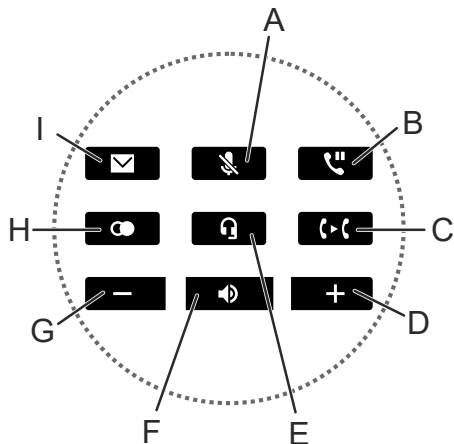
A   Microphone

B   Hold

C   Transfer

D   plus

E   Headset

F   Loudspeaker

G   minus

H   Redialling

I   Messages

Keys and LEDs on the key extension module
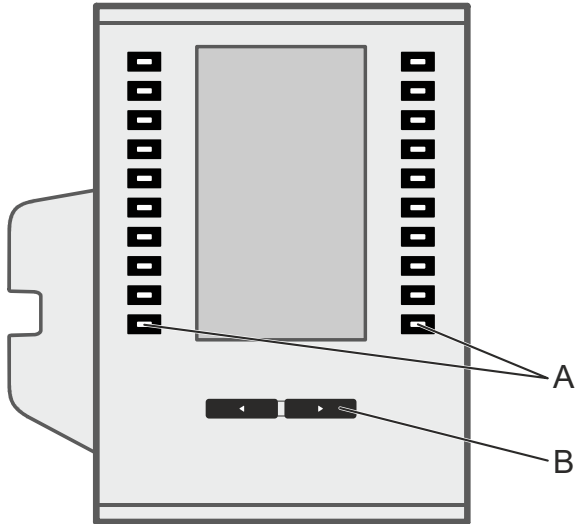
A   Function keys (10x on each side)

B   control rocker

Rear View

The next figures show the connections on the bottom of the different devices.

COMfortel® D-100/D-110
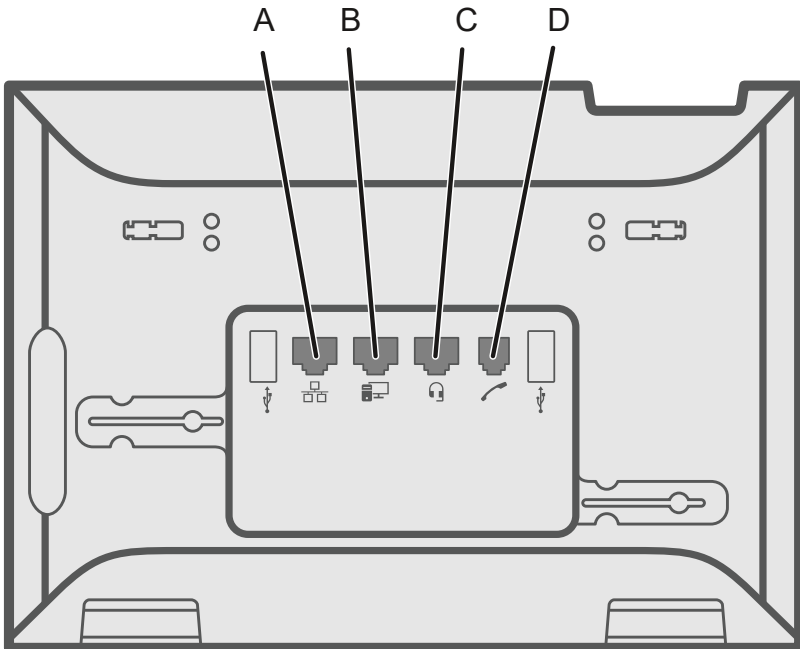
A   Network port

B   Direct PC connection via network cable

C   Headset socket

D   Receiver socket

COMfortel® D-200/D-210
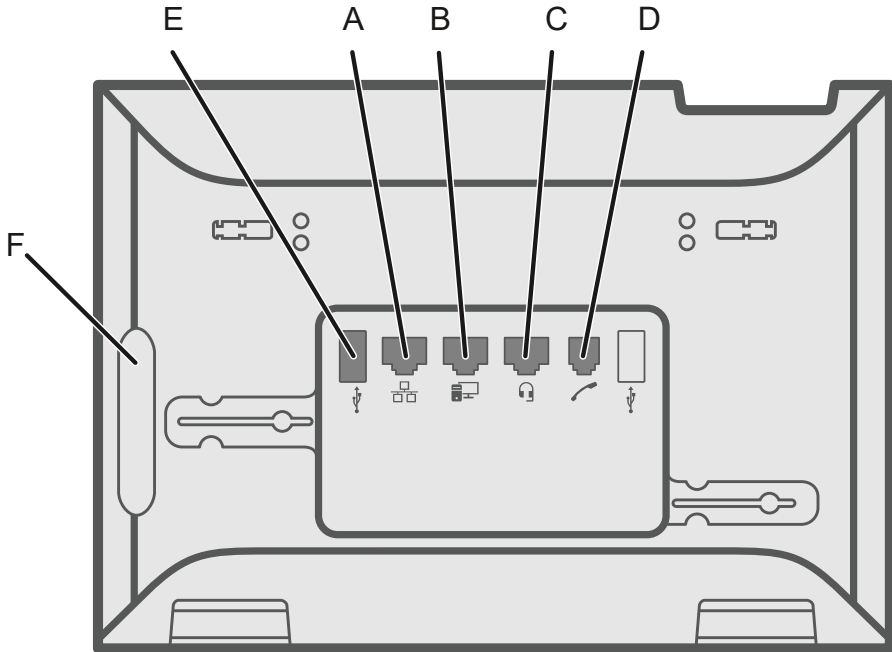
A   Network port

B   Direct PC connection via network cable

C   Headset socket

D   Receiver socket

E   USB port (for COMfortel D-XT20/D-XT20i key extension module, USB headset)

F   Rubber cover: To connect a COMfortel D-XT20 key extension module, remove the cover and attach it to the extension module.

COMfortel® D-XT20/D-XT20i
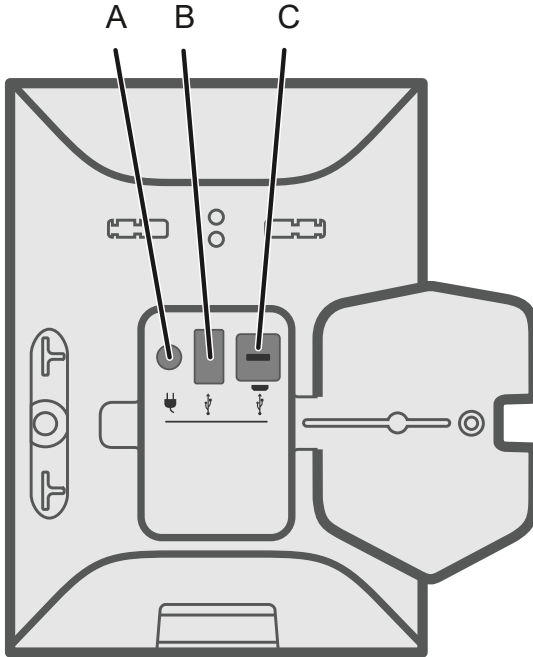
A   Socket for plugging in the COMfortel D-XT-PS wall-mounted power supply (if you connect two or more key extension modules)

B   USB port (exclusively for connecting another COMfortel D-XT20/D-XT20i key extension module)

C   Micro-USB port: Use this to connect the telephone to the key extension module

Using the Web Interface

Basic Knowledge
Dashboard
Display and LED
Function keys
Using provisioning
Identities
Update
Network
LDAP
Telephony
Sound
Date and time
User
Backup and restore
Access

Basic Knowledge

Opening the web interface
Operating Instructions

Opening the web interface

You must log in with the administrator authorisation level the first time you log in or after the device has been reset to its factory settings. When you log in again in the future, the authorisations are linked to the credentials (user/administrator).

Requirement:

• Recommended browsers: Mozilla Firefox, Google Chrome. (current version)

• – You need to know the phone's IP address (you can check it on the phone by selecting **Settings** > **Network**)

• There must be an existing network connection between the PC and phone.

Proceed as follows:

1. Enter the phone's IP address in the browser address line (e.g. http://192.168.178.4).

   The login window opens.

2. Enter your Username and Password access data.

   If you are running the web interface for the first time, enter the following access data:

   – **Username: admin**

   – **Password: admin**

   ⚠ **Caution:** Passwords that are unchanged from the factory settings are a security risk. Change the password in the dialogue without delay.

3. Click on **Login**.

You are now logged into the web interface and can make settings.

Operating Instructions

Overview
Fields and User Elements
Appearance

Overview

The web interface is divided into several areas:
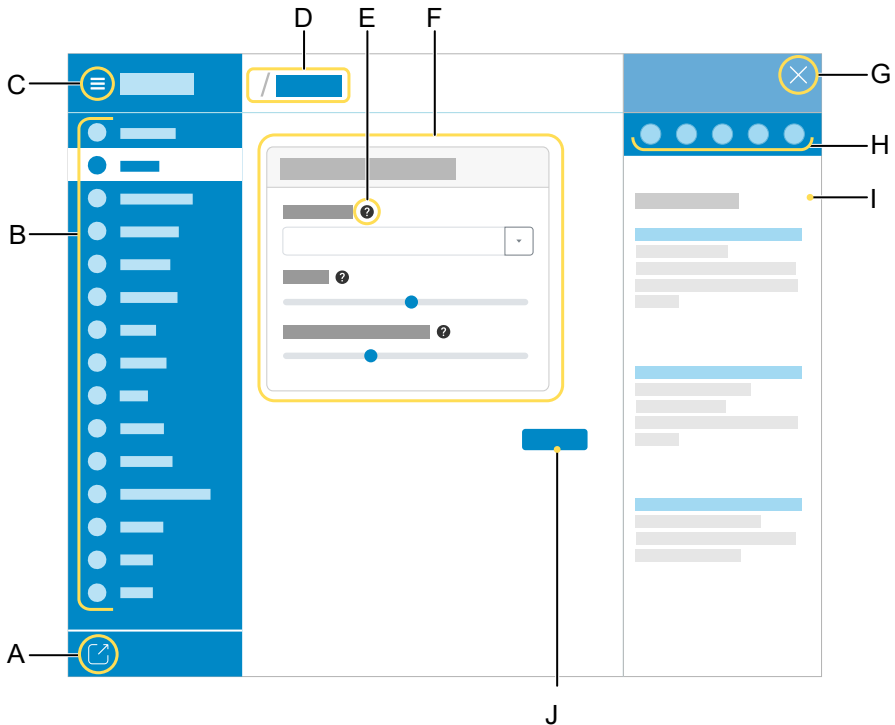
A   Log out user

B   Vertical navigation

Here, click on the required topic to select it.

C   Display or hide vertical navigation.

If the menu structure on the left is being displayed, a vertical navigation is displayed.

D   Breadcrumb navigation

Display the branching within the web interface.

Select the "house" icon to return to the home screen.

E Context-sensitive help

F Content area

The topic selected using the vertical navigation is displayed here.

G Hides the integrated help

H Navigation elements within the integrated help, from left:

- Menu

- Search

- Back

- Forward

- Home

I Display of the contents of the integrated help

J Save

Fields and User Elements

Element	Function
	List field: enter settings in it to configure particular functions. The selected setting is displayed in the list field. Click on the arrow on the right of the list field to display possible options.
	Empty field: enter a name, value or web address in it. Click in the entry field and enter the text.

The following characters must not be used:

" " quotation marks

$ dollar sign

% percent sign

< pointy bracket - less than

> pointy bracket - greater than

\ backslash

^ caret

[ ] Square brackets

{ } curly brackets

Element	Function
	Active text input field.
	Text input field that is already filled.
	Text input field that is locked for processing.
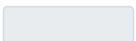
Element	**Function**
	Form field for incrementally increasing/reducing a value.
	You can use a toggle button to enable some functions (move one point to the right) or disable them (move one point to the left). Click on an option to activate it. The previous selection is disabled at the same time.
	Use sliders to make infinitely variable settings (e.g. contrast settings). The position of the slider identifies where the setting is. Drag the slider to the left to reduce the value. Drag the slider to the right to increase the value.
	The intention to delete an item can be confirmed or revoked.
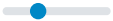
Appearance

The display/legibility of the web interface may be affected by browser settings, e.g.:

- Scale/zoom browsers (e.g.hold down Ctrl + scroll (mouse wheel))

Note: If the web interface is not legible because of scaling/zooming of the browser, you can restore the original view (e.g. Mozilla Firefox under **View** > **Zoom** > **Normal** or Ctrl + 0).

Dashboard

Status information
Language and location
Device name

Status information
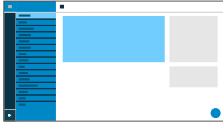
Type

Displays the phone model used.

Firmware version

Displays the firmware version currently in use on your device.

⤴ Further help under Update

Update system version

Displays the update system version in use on your device.

Bootloader version

Displays the bootloader version in use on your device.

Serial number

Displays your device's serial number.

MAC address

Displays your device's physical address on the network.

Network chip

Model of the network chip integrated in the phone.

Logcat

Records all important events, errors and warnings and displays them in the browser. They are only recorded if the developer has implemented logging of errors or events.

Clipboard

Copies status information to the clipboard, so you can access it easily, for example, to make a request for support.

Online help

Displays the online help in your browser.

Start trace

If the phone malfunctions, a log can be recorded. It can then be sent to the technical departments (Service, Development) so that the cause of the fault can be investigated.

Further help under Trace

Legal

Here, you can display the following license information:

- **End User License Agreement for Software (EULA)**

- **Data protection**

- **Open source licenses**

Language and location
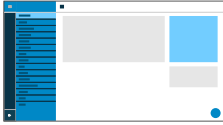
Language web interface

Changes the language used on the web interface.

↗ Further help under Language and location

Phone language

Changes the language used on the phone user interface.

↗ Further help under Language and location

Location

Sets the phone's actual or required location.

Audio frequencies (pitches, pulse/pause characters and/or ringing tones), and also the time and time format, are different for each selected location.

↗ Further help under Language and location

Device name

Device name
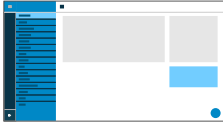
The name you assign to the device.

⬀ Further help under Device Information

Display and LED

Basic Settings
LED notifications

Basic Settings
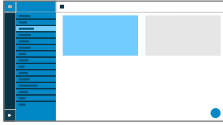
Idle state

After the selected time, the phone is automatically returned to the idle state.

- **Not automatically**

  The phone is not automatically returned to the idle state.

Further help under Idle state

Contrast

Sets the contrast.

Note: Any changes you make to the contrast affect on all key extension modules that are connected.

Brightness

Sets the brightness.

Note: Any changes you make to the brightness affect on all key extension modules that are connected.

Further help under Display and LED

LED notifications

Seen missed calls

Sets the message key LED's reaction to seen missed calls.

- **LED lights up**

  The message key LED shines and a text message is displayed in the Notifications menu.

- **LED is flashing**

  The message key LED flashes and a text message is displayed in the Notifications menu.

- **LED off**

  The message key LED does not flash and a text message is displayed in the Notifications menu.

- **Disabled**

  The message key LED does not flash and no text message is displayed in the Notifications menu.

Further help under Display and LED

Heard voice messages

Sets the message key LED's reaction to heard voice messages.

- **LED lights up**

The message key LED shines and a text message is displayed in the Notifications menu.

- **LED is flashing**

  The message key LED flashes and a text message is displayed in the Notifications menu.

- **LED off**

  The message key LED does not flash and a text message is displayed in the Notifications menu.

- **Disabled**

  The message key LED does not flash and no text message is displayed in the Notifications menu.

Further help under Display and LED

Message key LED

Controls the behaviour of the message key LED. If this option is deselected, the message key LED are switched off permanently.

Further help under Display and LED

Function keys

Overview
Edit function keys
Edit parameters
Edit Icon

Overview

Function keys

You can see the interface in which you configure the function keys for the phone in the Function keys tab.

Phone user interface
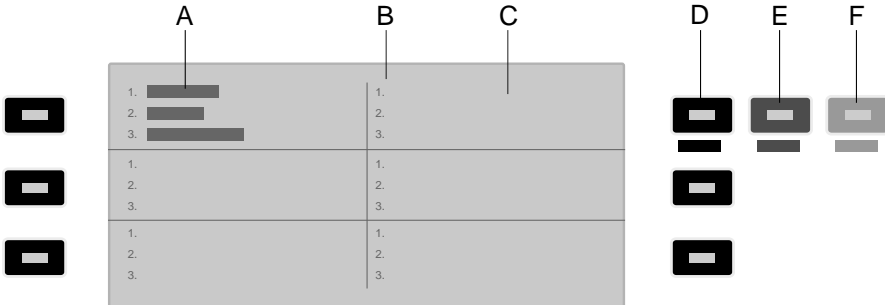
A   Shows that the phone's top left function key has had a function assigned to it on all 3 levels.

B   Lists all 3 possible levels to which a function can be assigned

C   Display

D   **Level 1** (always visible)

Click on the selected key to open the Edit function keys window

E   **Level 2** (opens when you pass over it with the mouse)

Click on the selected key to open the Edit function keys window

F **Level 3** (opens when you pass over it with the mouse)

Click on the selected key to open the Edit function keys window

Note: If a key already has a function assigned to it, the LED shines orange. You can assign a different function.

Key extension module (1 – 3)

You can see the interface in which you configure the function keys for the selected module in Key extension module tab 1 – 3. Up to 3 key extension modules can be connected.

Phone user interface key extension module
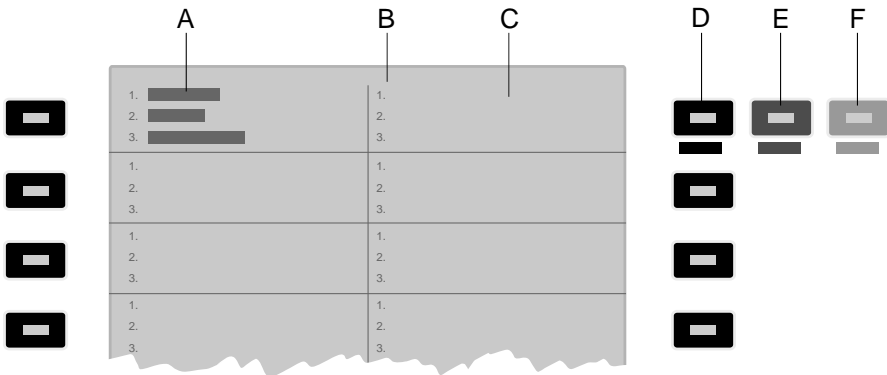
A   Shows that the phone's top left function key has had a function assigned to it on all 3 levels.

B   Lists all 3 possible levels to which a function can be assigned

C   Display

D **Level 1** (always visible)

Click on the selected key to open the Edit function keys window

E **Level 2** (opens when you pass over it with the mouse)

Click on the selected key to open the Edit function keys window

F **Level 3** (opens when you pass over it with the mouse)

Click on the selected key to open the Edit function keys window

Note: If a key already has a function assigned to it, the LED shines orange. You can assign a different function.

Edit function keys
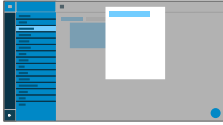
Level (1 – 3)

Click on one of the Level tabs (1 – 3) to jump between the 3 different key levels.

Function

Selects the function and displays the necessary parameters.

ℹ **Note:** To use additional function templates, activate the Auerswald/ Fontevo templates.

↗ Further help under Function templates

↗ Details about the settings are in Edit parameters

✎ Edit Icon

Displays the menu in which you select and adjust the required image file for use as an icon for the function key.

↗ Details about the settings are in Edit Icon

Key label

Displayed beside each function key on the screen.

Reset level

Removes the function assigned to the key on the current level.

Reset all levels

Deletes the functions assigned to the key on all levels.

Cancel

Closes the menu without adopting changes.

Save

Saves the entered parameters and close the window.

Edit parameters

Speed dial

Identity

Activate config

Anonymous call

Automatic configuration switching

Call forwarding immediately

Call forwarding on busy

Call forwarding on no reply

Call waiting

CF ext. calls: Immediately

CF ext. calls: On busy

CF ext. calls: On no reply

Do not disturb

DTMF-Relay

Group

Group (incoming)

Group (outgoing)

Park position

Relays

Silent Alert

SIP message

VPN

XML Minibrowser

Speed dial

Number/URI

Enter a phone number or address such as 198@192.168.XX.YY or 198@company.com, here.

↗ Further help under Speed dial

Identity

Identity for which this function is set up.

↗ Further help under Identities

Subscription

Activates the sending of BLF subscriptions to the PBX.

ℹ **Note:** It may be a good idea to deactivate subscriptions in situations when it is necessary to prevent the PBX from being overloaded by high volumes of transmitted data.

Pickup

Activates the Pickup function.

ℹ **Note:** Pickup only works if the Subscription function is activated.

ℹ **Note:** Pickup should not be used if the key is only used to switch functions in the PBX using a programming code. Deactivate Pickup in that case.

↗ Further help under Pickup

InterCom

Enables the InterCom function. The connection with the destination number is established directly. No ringing or active call acceptance.

Further help under InterCom

Signalise missed call

If this function is enabled, missed calls associated with the speed dial number are signalled with flashing green LEDs.

Further help under Speed dial

Pickup-choice on key-press

If this function is switched on, a list of all the calls relevant for the function key, that can be taken via pickup, is displayed.

Note: If this function is switched on, a call cannot be taken by pressing a key. Instead, a list of all the calls that can be taken via pickup is displayed, even if the list only contains one call.

Further help under Pickup

Do not show in pickup list

If this function is switched on, calls that can be taken by pickup are not forwarded to the global pickup list.

Further help under Pickup

Pickup of connected calls

If this function is switched on, calls that have already been connected can be picked up. This option is useful, for example, if an answering machine or mailbox number has been assigned to the speed dialling key. The calls are included in the global pickup list.

Further help under Pickup

Signal presence

If this function is switched on, the following states are transmitted (if they are supported by the PBX):

- Successful registration of the monitored phone number on the PBX

- Do not disturb

- Call forwarding

Default: disabled.

Identity

Identity

Identity used to trigger subsequent calls.

⬈ Further help under [Identities](#)

Activate config

Identity

Identity for which this function is set up.

Further help under Identities

Configuration id

The PBX assigns the configuration ID automatically.

Further help under Function templates

Anonymous call

No parameters necessary.

⬈ Further help under Anonymous call

Automatic configuration switching

Identity

Identity for which this function is set up.

⤴ Further help under Identities

ID

Identification number for system relay with Automatic configuration switchover operating mode (900, in the factory settings).

⤴ Further help under Function templates

Call forwarding immediately

Identity

Identity for which this function is set up.

↗ Further help under Identities

ℹ **Note:** Only assign the Identity parameter if PBX function templates are used.

↗ Further help under Function templates

Destination

Call forwarding destination

↗ Further help under Call forwarding

Call forwarding on busy

Identity

Identity for which this function is set up.

↗ Further help under Identities

ℹ **Note:** Only assign the Identity parameter if PBX function templates are used.

↗ Further help under Function templates

Destination

Call forwarding destination

↗ Further help under Call forwarding

Call forwarding on no reply

Identity

Identity for which this function is set up.

↗ Further help under [Identities](#)

ℹ **Note:** Only assign the Identity parameter if PBX function templates are used.

↗ Further help under [Function templates](#)

Destination

Call forwarding destination

↗ Further help under [Call forwarding](#)

Call waiting

No parameters necessary.

↗ Further help under Call waiting

CF ext. calls: Immediately

Identity

Identity for which this function is set up.

⤴ Further help under [Identities](#)

Call forwarding for external calls

Point-to-multipoint connection: External phone numbers (MSN, see PBX configuration), for which call forwarding is to be performed.

PTP connection: Header plus extension for which call forwarding is to be performed.

Destination

Call forwarding destination

⤴ Further help under [Function templates](#)

CF ext. calls: On busy

Identity

Identity for which this function is set up.

↗ Further help under [Identities](#)

Call forwarding for external calls

Point-to-multipoint connection: External phone numbers (MSN, see PBX configuration), for which call forwarding is to be performed.

PTP connection: Header plus extension for which call forwarding is to be performed.

Destination

Call forwarding destination

↗ Further help under [Function templates](#)

CF ext. calls: On no reply

Identity

Identity for which this function is set up.

⬈ Further help under Identities

Call forwarding for external calls

Point-to-multipoint connection: External phone numbers (MSN, see PBX configuration), for which call forwarding is to be performed.

PTP connection: Header plus extension for which call forwarding is to be performed.

Destination

Call forwarding destination

⬈ Further help under Function templates

Do not disturb

Identity

Identity for which this function is set up.

⬈ Further help under Identities

⬈ Further help under Do not disturb

DTMF-Relay

DTMF sequence

Numbers needed to perform the required function.

⬈ Further help under Function templates

Group

Identity

Identity for which this function is set up.

⬈ Further help under [Identities](#)

Group

Group telephone number

⬈ Further help under [Function templates](#)

Group (incoming)

Identity

Identity for which this function is set up.

⤴ Further help under Identities

Group

Group telephone number

⤴ Further help under Function templates

Group (outgoing)

Identity

Identity for which this function is set up.

⤴ Further help under Identities

Group

Group telephone number

⤴ Further help under Function templates

Park position

Number/URI

Phone number of the park position.

Identity

Identity for which this function is set up.

⤴ Further help under Park position

Relays

Identity

Identity for which this function is set up.

⬀ Further help under [Identities](#)

ID

Relay ID number according to PBX.

⬀ **Note:** The Relay function key is for relays with light, door opener or calculator operating mode. This function key is not suitable for manual relays. For additional Information, refer to the Advanced Information about the PBX.

⬀ Further help under [Function templates](#)

Silent Alert

URL

Required for forwarding the alarm message.

⚠ **Caution:** The function can only be executed if the appropriate URL has been entered.

Delay

Countdown counter. An action is triggered after the countdown reaches zero.

↗ Further help under [Silent Alert](#)

SIP message

Number/URI

Enter a phone number or address such as 198@192.168.XX.YY or 198@company.com, here.

Identity

Identity for which this function is set up.

Message

This is where you enter the text for sending an SIP message.

Delay

Countdown counter. An action is triggered after the countdown reaches zero.

⬈ Further help under SIP message

VPN

No parameters necessary.

⬈ Further help under Network

XML Minibrowser

Number/URI

Web server address

⬈ **Further help under** [XML Minibrowser](#)

Edit Icon

Browse

Displays a dialogue in which you can select a previously saved image file in the appropriate folder on the data storage device.

Further help under Function keys

Close

Closes the menu without adopting changes.

Further help under Function keys

Delete

Deletes the user-defined icon. The default icon for the selected function is displayed.

Further help under Function keys

Download

Saves the icon locally with a size of 64 x 64 px as `icon.png`.

Further help under Function keys

Set

Applies the icon for these views:

Edit Icon

- Edit function keys menu, web interface
- Function keys list, web interface
- Home screen function keys, telephone
- Edit function keys menu, telephone

↗ Further help under Function keys

Adjustment tool

Tool for adjusting the icon's image size and screen window.

↗ Further help under Function keys

Icon Preview

Displays a preview of the final icon.

↗ Further help under Function keys

Using provisioning

Basic Settings
Schedule
SIP provisioning
Additional function templates

Basic Settings

Automatic provisioning

Automatic provisioning is performed according to the schedule and each time the phone is switched on.

📷 Further help under Using provisioning

Provisioning with DHCP data

The phone obtains the provisioning server's URL from a DHCP server. Additional data for controlling provisioning can also be transferred along with the URL.

📷 Further help under Using provisioning

Redirect Server

The phone obtains the provisioning server's URL from the redirect server.

📷 Further help under Using provisioning

Provisioning from PBX search

Automatic provisioning of the phone is performed via the PBX.

📷 Further help under Using provisioning

Provisioning server URL

This URL is required if it is not obtained via DHCP or a redirect server.

Further help under Using provisioning

Schedule

Weekdays

The weekdays on which automatic provisioning is carried out.

↗ Further help under Using provisioning

Time

Time at which automatic provisioning is carried out. The time window is 60 minutes.

ℹ **Note:** After a reboot, a random time between 00:00 and 05:00 is entered for provisioning. This prevents load spikes on the server.

↗ Further help under Using provisioning
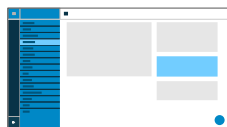
SIP provisioning

SIP provisioning

Starts remote configuration using SIP messages.

Further help under [Using provisioning](#)
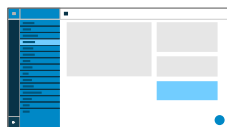
Additional function templates

Vendor templates

Activates templates and scripts that are specially designed to run on Auerswald/FONtevo PBXs.

Further help under Function templates

Identities

Managing Identities
Basic Settings
Options for experts
Codecs
Registrar
NAT
Outbound proxy

Managing Identities
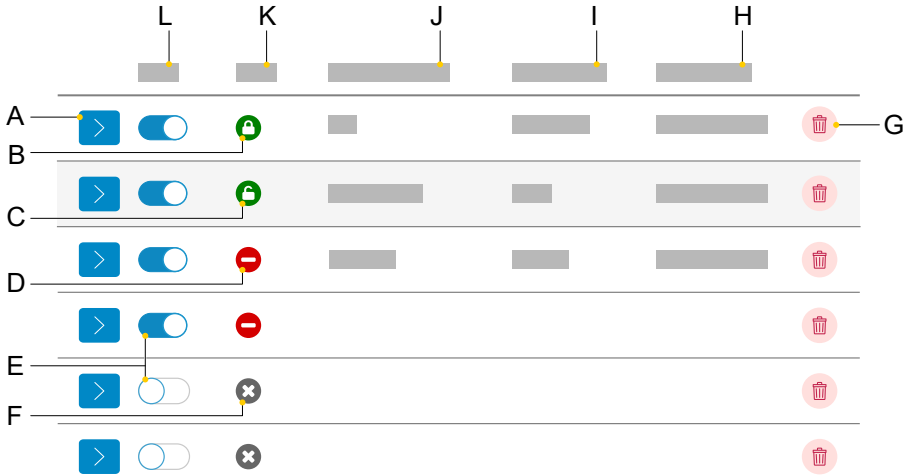
A   Displays the settings for an identity

B   Identity correctly configured, registered and encrypted

C   Identity correctly configured, registered and unencrypted

D   Identity incorrectly configured, unregistered and unencrypted

E   Activate/deactivate an identity

F   Identity deactivated

G   Resets the selected identity

H   **SIP Registrar**

I   **Display name**

J   **SIP Username**

K   **Status**

L   **Active**

Basic Settings

Active

Press this to switch the selected identity on.

⬈ Further help under Identities

SIP Username

Used to identify the device to the VoIP provider or in the PBX. Identical to the user name assigned by the VoIP provider or the internal phone number registered in the PBX.

⬈ Further help under Identities

Password

Used to verify the user name. Identical to the password assigned by the VoIP provider when the identity was being created or the password entered in the PBX.

⬈ Further help under Identities

Display name

Freely selectable name.

The display name is displayed both in the web interface, under the display name, and on the phone, below the status bar.

If you do not set a display name, the display on the phone contains these elements: SIP user name@SIP registrar.

↗ Further help under Identities

Authentication name

Enter the authentication username provided by the VoIP provider.

You need the authentication username when you register with a VoIP provider. If you connect to an Auerswald PBX, you do not need to enter the authentication ID.

↗ Further help under Identities

SIP Registrar

This tells the phone where it can register itself. The information given here is the URL set by the VoIP provider on the IP address on which the registering PBX can be accessed.

↗ Further help under Identities

Ringer Tone

Assigns a ringtone to the identity.

↗ Further help under Sound

Options for experts
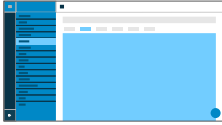
Fallback for

No fallback

No fallback is possible for the current identity.

Configured/Unconfigured identity

Sets the main identity for which the fallback identity applies.

Configured identity: displays **Display name**

Configured identity: displays **Identity** + seq. no.

Notices: In the case of a main identity, the field is greyed out and displays the entered fallback identity. A fallback identity can also be created for identities that have not yet been configured.

Further help under Fallback identity

CLIR type

(Number suppression)

Area in the From header in which the VoIP provider expects to receive number presentation suppression. Selection as set in the PBX/at the provider.

Anonymous

The sent display text in the From header is "anonymous".

User anonymous

Both the display text and the User name section in the From header is "anonymous".

↗ Further help under Anonymous call

Voicemail number

Enter the voicemail number assigned by the VoIP provider or the voicemail number entered in the PBX.

↗ Further help under Voicemail number

Pickup code

This is required to perform a call pickup. Enter the character string stored on the PBX/at the provider, e.g.**##06** for Auerswald PBXs.

Music on hold

If a connection or call is on hold, the "music on hold" is played.

↗ Further help under Music on hold

IP version

IPv4

Sets IPv4 for the registrar.

IPv6

Sets IPv6 for the registrar.

Auto

Automatically sets the protocol used by the registrar.

↗ Further help under Identities

SRTP

Transport protocol for encrypted connections.

Mandatory

This setting forces voice encryption to be on. If the voice partner (VoIP provider, PBX, external VoIP subscriber) does not support SRTP, the connection is not established.

Preferred

Switches on negotiation for the encryption of call data via SRTP. When a call is made, the call partner will be asked if encryption is possible. If selected, voice data is transmitted in encrypted form. If not selected, it is not encrypted.

Disabled

This setting forces voice encryption to be off. If the voice partner (VoIP provider, PBX, external VoIP subscriber) does not support encryption, the connection is not established.

↗ Further help under SRTP

SIPS

Activates the transmission of encrypted SIP messages over TLS for connections with this identity.

The destination in the invite package header is contacted with an encrypted transmission.

ℹ️ **Note:** To create a successful, secure connection, a certificate must be provided for the provided host, if necessary.

↗️ Further help under [SIPS](#)

↗️ Further help under [Certificate](#)

Peer-to-peer TLS

Forces the encryption of SIP messages over the entire route to the destination.

ℹ️ **Note:** The call is not made if TLS is not available on the entire connection.

↗️ Further help under [Identities](#)

Certificate

If the SIPS function is activated, the system checks whether the certificate belongs to the domain/IP.

↗️ Further help under [Certificate](#)

Session timer

Switches on the check after a connection for a call that is still in existence.

ℹ️ **Note:** When the SIP session timer is switched on, this may result in the call being interrupted more frequently after the specified interval, if

a VoIP provider has not implemented session renewal properly. In this case, set a different session timeout or disabled the session timer.

↗ Further help under SIP

Session timeout (in min.)

2 … 255 minutes, default: 15 minutes

Specifies the number of minutes after which the SIP session timer is to check a call's connection.

↗ Further help under SIP

Protocol type

UDP

(User Datagram Protocol). Used to send data packets over connectionless non-secure communication lines.

ℹ **Note:** If very large data packets are present, TCP is used instead of UDP. The maximum size of a data packet can vary according to the network. (RFC 3261 > TCP)

TCP

(Transmission Control Protocol). Segments data into packets, from a specified size, and sends these individual data packets to the recipient address until receipt has been confirmed.

↗ Further help under SIP

❗ **Important:** If encryption using SIPS is enabled, the TCP transport protocol is used. Manual settings are overwritten.

Subscription interval (min)

This sets the frequency at which the status of potential changes are queried on the PBX. Default: 45 minutes

The value you enter here should be a compromise between a short interval (which generates traffic) and rapid updates.

↗ Further help under SIP

Retry subscriptions

Sets the interval at which attempts are made to configure a subscription on the PBX/provider, if an error occurs.

- **1x**

  The device sends exactly one subscription to the PBX/provider. No other subscriptions are sent.

- **Fixed interval**

  Attempts to subscribe on the PBX/provider are made at the specified interval.

- **Redouble interval**

  The set number of seconds doubles after each attempt to subscribe on the PBX/provider.

↗ Further help under SIP

Retry subscriptions: Interval in seconds

Sets the time gap between two subscription attempts.

Minimum: 10 sec

Default: 180 sec

Further help under SIP

Retry register

Sets the interval at which attempts are made to register on the PBX/ provider, if an error occurs.

- **1x**

  The device makes exactly one attempt to register on the PBX/provider. There are no more registration attempts.

- **Fixed interval**

  Attempts to register on the PBX/provider are made at the specified interval.

- **Redouble interval**

  The set number of seconds doubles after each attempt to register on the PBX/provider.

Further help under SIP

Retry register: Interval in seconds

Sets the time gap between two subscription attempts.

Minimum: 1 sec

Default: 10 sec

Further help under SIP

Name sources

Active

The sequence in which the name sources are displayed in the list sets the sequence in which they are applied. The first source that contains a name is used for the display.

Inactive

You can select inactive name sources and drag and drop them into the Active list to arrange them, or remove them from the list.

Reset to default

Returns the list to its default state.

↗ Further help under [Name sources](#)

DTMF method

Specifies the DTMF method used to transmit signals.

- **RTP event**

  Transmission of event packets in the RTP stream

- **Inband**

  Transmission of coded sound signals, directly in the RTP stream

- **SIP info**

  Transmission of SIP info messages

↗ Further help under [DTMF](#)

Codecs
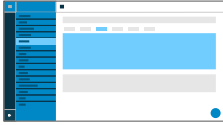
Codecs

Active

The sequence in which the active codecs are displayed in the list sets the sequence in which they are applied.

To achieve the best possible balance between bandwidth and speech quality, we recommend you do not change the sequence of the codecs.

Inactive

You can select inactive codecs and drag and drop them in the »Active« list to arrange them, or remove them from the list.

Further help under Codecs

Registrar

SIP domain

Name space that contains all the entries within one directory, specified by the VoIP provider/PBX operator.

Further help under [Identities](#)

Port

The registrar's port. The (recommended) default value is 5060.

⚠ **Caution:** Different entries can cause malfunctions if the provider makes changes. An SRV query is only possible with the default value.

⚠ **Caution:** Each time a port is opened on the router, this creates a potential security risk. It is essential you put additional protective measures in place.

Further help under [Identities](#)

Time lapse registration (min.)

5 … 60 minutes

The registration time is the number of minutes that must elapse before the telephone reregisters itself with the VoIP provider.

The value you enter here should be a compromise between a short timespan (which generates traffic with the provider) and a long re-registration period.

↗ Further help under [Identities](#)

Early re-register (sec)

Shortens a re-registration by the specified number of seconds. Early re-registration prevents the registration from expiring prematurely.

↗ Further help under [Identities](#)

Refresh register at 1/2

Turns on re-registration after half of the negotiated maximum registration period.

↗ Further help under [Identities](#)

NAT

SIP NAT traversal

Requirements:

- Configured DNS server

- If the option "active" is selected: **Registrar** enabled for the phone

Active

NAT traversal is performed by the telephone. To do this, you must also specify a STUN server.

⚠ **Caution:** Each time a port is opened on the NAT router, this creates a potential security risk. Implement appropriate security measures.

Inactive

NAT traversal is not performed by the telephone. The SIP request also contains an "rport" part in which the IP addresses used here (local, public) are transported. The router which connects the local area network with the Internet should be a properly functioning SIP-aware router which performs NAT traversal.

↗ Further help under NAT

RTP NAT traversal

Requirement:

- Configured DNS server

Active

NAT traversal is performed by the telephone. To do this, you must also specify a STUN server.

⚠ **Caution:** Each time a port is opened on the NAT router, this creates a potential security risk. Implement appropriate security measures.

ℹ **Note:** If there are problems with unilateral call connections, a STUN server should be used.

Inactive

NAT traversal is not performed by the telephone. The router which connects the local area network with the Internet should be a properly functioning SIP-aware router which performs NAT traversal.

↗ Further help under [NAT](#)

NAT Keep-Alive Interval (sec.)

After the time specified here has passed, NAT Keep-Alive packets are sent to the firewall to maintain the NAT mapping.

ℹ **Note:** Some providers lock accounts if the NAT Keep-Alive Interval is too short. As a rule, this is reported with error message 503 during SIP registration. If this problem occurs, it is recommended to raise the value (e.g. to 180).

↗ Further help under [Identities](#)

Use ICE

Enable the best possible transmission route to be found between the phone and PBX/provider.

Note: ICE must only be activated if the other end supports ICE.

Further help under [Identities](#)

Outbound proxy

Mode

Inactive

Deactivate this function if

- the registrar can be accessed without outbound proxy
- no outbound proxy is available

Automatically

The PBX automatically determines an outbound proxy.

Manually

The outbound proxy is a fixed default setting. Enter this data as an IP address or a URL.

Further help under Outbound proxy

URL

Server address for the outbound proxy.

Note: The **URL** must be filled out if mode **Manually** is selected.

Further help under Outbound proxy

Port

Port for the outbound proxy.

ℹ **Note:** The **Port** must be filled out if mode **Manually** is selected.

↗ Further help under Outbound proxy

Alternative URL

Alternative server address for the outbound proxy.

↗ Further help under Outbound proxy

Alternative port

Alternative port for the outbound proxy.

↗ Further help under Outbound proxy

Update

Basic Settings
Restart and factory reset
Schedule

Basic Settings
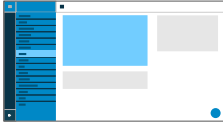
Check for updates

If selected, the phone automatically checks for current updates.

Further help under Update

Automatically install updates

If selected, available updates are installed automatically.

Further help under Update

Update server URL

In the default settings: https://update.auerswald.de

Further help under Update

Start update

Starts installation of a new firmware version.

Further help under Update

Restart and factory reset
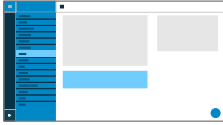
Restart phone

Restarts the telephone.

ℹ️ **Caution:** This interrupts all connections, all calls and data services. Unsaved data is lost!

↗️ Further help under Update

Reset to default

Resets the telephone to the factory settings. Any saved data and settings are deleted. The phone is shut down and then restarted.

↗️ Further help under Reset to default

Schedule
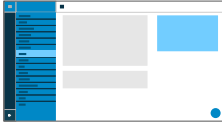
Weekdays

The weekdays on which the phone checks for available updates and installs them.

⬈ Further help under Update

Time

Time at which the phone checks for available updates and installs them.

⬈ Further help under Update

Network

Current Settings
IPv4
IPv6
VLAN
VPN
Topology

Current Settings
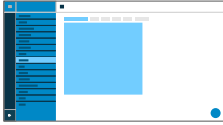
Lists all the network settings made on the phone.

DNS

IP address of the DNS server used here.

Further help under [IPv4](#)

IP addresses

Protocol

Shows which network protocol is in use.

IP address

Displays the phone's IP address.

Prefix

Different subnet mask notation.

Example: 24 corresponds to 255.255.255.0

Forwarding routes

Protocol

Shows which network protocol is being used by the forwarding route.

Gateway

Displays the IP address of the forwarding route.

If only one router is in use, the DNS's IP address is the same as that of the gateway. If more than one server is being used to handle these tasks, the IP addresses will be different.

IPv4

IPv4 mode

DHCP

The phone obtains its IP address automatically from a DHCP server.

Static

The telephone uses a fixed IP address (supplied by the system administrator).

↗ Further help under [IPv4](#)

IP address

If you select **DHCP:** the phone's IP address is displayed

If you select **Static:** you can enter/change the phone's IP address

In the factory settings: 0.0.0.0.

↗ Further help under [IPv4](#)

Subnet mask

Example: 225.255.255.0

↗ Further help under [IPv4](#)

Gateway

Example: 192.168.178.1

The gateway provides interoperability between networks, so data can pass from one to another. The provider or the network system administrator provide the gateway.

Further help under IPv4

DNS 1

The DNS server is needed for Internet access. The provider or network system administrator sets the DNS server. The provider or network system administrator sets the DNS server.

Further help under IPv4

DNS 2

Optional

Further help under IPv4

IPv6

IPv6 mode

Auto

The phone obtains its IP address automatically from a DHCP server.

Disabled

Disables IPv6 mode. IPv4 over DHCP is used.

Further help under [IPv6](#)

Privacy Extensions

If this function is activated, IPv6 addresses that do not allow the user to be identified are created. The host part of the IPv6 address is anonymised.

Further help under [IPv6](#)

DNS 1

The DNS server is needed for Internet access. The provider or network system administrator sets the DNS server. The provider or network system administrator sets the DNS server.

Further help under [IPv6](#)

IPv6

DNS 2

Optional

⬀ Further help under IPv6

VLAN

Phone VLAN

Switches on VLAN for the phone.

⤴ Further help under VLAN

Phone VLAN ID

Enter the phone's VLAN ID: 1 … 4094

The associated data packets are tagged with this ID.

⤴ Further help under VLAN

Phone VLAN priority

0 (lowest priority) to 7 (highest priority)

⤴ Further help under VLAN

PC port VLAN

Switches on VLAN for the PC port.

⤴ Further help under VLAN

PC port VLAN ID

Enter the PC port's VLAN ID: 1 … 4094

The associated data packets are tagged with this ID.

Further help under VLAN

PC port VLAN priority

0 (lowest priority) to 7 (highest priority)

Further help under VLAN

VPN

VPN

Switches on VPN for the phone.

Further help under VPN

VPN type

IPsec Xauth PSK

Authentication is performed using a shared secret and a personal secret.

IPsec Xauth RSA

Authentication is performed using a key pair: one private key and one public key.

IPsec Hybrid RSA

Mixed form **IPsec Xauth PSK** and **IPsec Xauth RSA**

Open VPN

OpenVPN libraries are used to set up a VPN over an encrypted TLS/SSL connection.

L2TP IPsec PSK

IPsec variant performed using the Layer 2 Tunneling Protocol. Authentication is performed using a shared secret and a personal secret.

L2TP IPsec RSA

IPsec variant performed using the Layer 2 Tunneling Protocol.
Authentication is performed using a key pair: one private key and one
public key.

↗ Further help under [VPN](#)

Name

Freely selectable name for the VPN connection.

↗ Further help under [VPN](#)

Server

Server address for the VPN connection.

Your network administrator will give you the server address.

↗ Further help under [VPN](#)

User name

User name for the VPN connection.

Your network administrator will give you your user name.

↗ Further help under [VPN](#)

Password

Password for the VPN connection.

VPN

Your network administrator will give you the password.

⬈ Further help under VPN

DNS server

Address of the DNS server used for the VPN connection (e.g. 8.8.8.8).

Your network administrator will give you the DNS server address.

⬈ Further help under VPN

Search domain

Search domain addresses for the VPN connection.

Your network administrator will give you the search domain addresses.

⬈ Further help under VPN

Forwarding routes

Settings for the network, IP address range (e.g. 10.0.0.0/8).

Your network administrator will give you the IP address range for the forwarding routes.

⬈ Further help under VPN

IPsecIdentifier

⚠ **Caution:** IPsecIdentifier settings can vary greatly, depending on which server is in use. The network administrator must predefine the configuration.

VPN

⤢ Further help under [VPN](#)

---

preinstalled IPsecSecret

Authentication key.

⤢ Further help under [VPN](#)

IPsecUserCert

This is generated by the system administrator or by the user.

The IPsecUserCert must fulfil these criteria:

- X.509 standard
- .pem format

⤢ Further help under [VPN](#)

IPsecUserPrivateKey

This is generated by the system administrator or by the user.

The private key must fulfil these criteria:

- X.509 standard
- .pem format

⤢ Further help under [VPN](#)

IPsecCaCert

Acts as a "trust anchor" for the connection.

The IPsec CA certificate (IPsecCaCert) must fulfil these criteria:

- X.509 standard

- .pem format

Further help under [VPN](VPN)

IPsecServerCert

Server-side certificate.

The IPsec CA certificate (IPsecCaCert) must fulfil these criteria:

- X.509 standard

- .pem format

Further help under [VPN](VPN)

OpenVPN Config

Contents of a configuration file (openvpn.cfg). Component part of setting up an open VPN.

Further help under [VPN](VPN)

Topology

Basic Settings
Jitter buffer
STUN server

Basic Settings

Check hostname

Checks whether the certificate belongs to the domain/IP.

Further help under Certificate

FrameSize

Sets the RTP packet size (msec Audio/RTP package).

Local SIP port

1024 … 65535

Default: 5060

Sets the outgoing port for SIP messages.

Further help under SIP

RTP off when silent

Ends the sending of RTP packets if no active conversation is taking place.

Note: If this function is switched on, parts of the call can seem like breaks in the connection because background noises are cut off sharply.

⬈ Further help under [Topology](#)

LLDP

If this function is enabled, LLDP packages are sent regularly to the directly connected LLDP-enabled switch.

⬈ Further help under [LLDP](#)

LLDP: Interval in seconds

LLDP package sending frequency

30 to 3600 sec

Minimum: 30 sec

Default: 30 sec

⬈ Further help under [LLDP](#)

Jitter buffer

Initial value

Initial prefetch delay value (msec).

Note: The value must lie between Min. prefetch and Max. prefetch. If the Initial value = 0, prefetch is deactivated.

Further help under Jitter buffer

Min. prefetch

Minimum prefetch delay (msec).

Default: 60 msec

Further help under Jitter buffer

Max. prefetch

Maximum prefetch delay (msec).

Default: 240 msec

Further help under Jitter buffer

Maximum

Delay that can be set for the jitter buffer (msec).

Default: 360 msec

⬈ Further help under Jitter buffer

STUN server

STUN server management
Settings

STUN server management
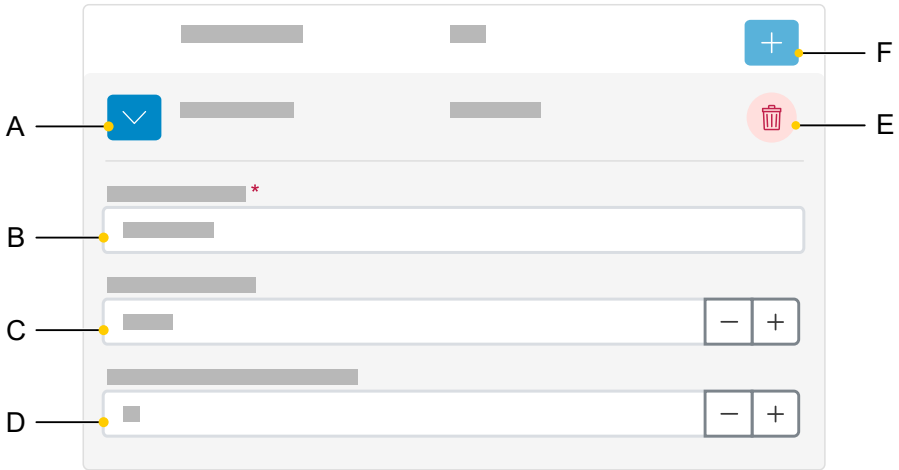
A   Displays/hides all settings for a STUN server.

B   **STUN server URL** field, entry mandatory

C   **STUN server port** field, 1 … 65535.

D   **Registration time lapse (min.)**, 5 … 60 minutes.

E   Deletes the selected STUN server.

F   Adds another STUN server to the list. The list is dynamic. Settings can be made for up to eight STUN servers.

Settings

STUN server URL

Requirement:

- Configured DNS server

Enter the STUN server URL.

Further help under [Topology](#)

STUN server port

1 … 65535

Important: Every time port forwarding is performed, there is a security risk. For this reason, we recommend you use port forwarding as little as possible.

Note: If you encounter problems with half-open (dropped) call connections, enabling the RTP port might help.

Further help under [Topology](#)

Time lapse registration (min.)

5 … 60 minutes

The registration time is the number of minutes that must elapse before the telephone reregisters itself with the VoIP provider.

The value you enter here should be a compromise between a short timespan (which generates traffic with the provider) and a long re-registration period.

Further help under Topology

LDAP

Basic Settings
LDAP filter

Basic Settings
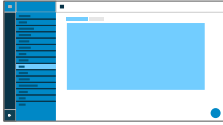
LDAP-Protocol

Protocol used on the LDAP server

LDAP

By default, unencrypted data traffic, Port 389.

LDAP StartTLS

Data traffic encrypted using Transport Layer Security, Port 389.

LDAPS

LDAP via SSL technology, Port 636.

Active

Switches on the directory service request.

⤴ Further help under LDAP

Server

The LDAP server's domain name or IP address

⤴ Further help under LDAP

Port

The LDAP server's port. On PBXs, use the preset port.

LDAPS (recommended): Port 636

LDAP StartTLS: Port 389

LDAP: Port 389

Further help under LDAP

User name

User name assigned when the LDAP server was being configured. Used for identification to the LDAP server.

Note: You can find the user name in the web interface of your PBX under **User data** > **Address Books** > **Settings** in the row **Simplified user name** next to **User name**(e.g. cn=auerswaldschandelah,dc=auerswald).

Further help under LDAP

Password

This is the password assigned when the LDAP server was being configured. It is used for identification to the LDAP server.

Note: You can find the password in the web interface of your PBX under**User data** > **Address Books** > **Settings** in the row **Simplified user name** next to **Password**.

Further help under LDAP

Search area

Point in the folder from which the LDAP search starts.

Example: BaseDN dc=auerswald

⤢ Further help under LDAP

LDAP filter
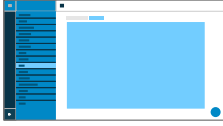
Only experts should make the settings available in the **LDAP** > **LDAP filter** area. LDAP filter syntax is used. RFC 2254 and RFC 4515 are used.

To enable contact data to be transferred from the LDAP server, enter the following values, listed below, in the end device.

Name filter

Sets the name filter used for the LDAP search.

Further help under [LDAP]

Number filter

Filter for assigning numbers to names for incoming and outbound calls.

Further help under [LDAP]

Number filter dial preparation

Filter for displaying the suggestions list in dial preparation.

Further help under [LDAP]

Max. LDAP contacts

Sets the max. number of contacts that the LDAP server is to output.

⬈ Further help under LDAP

Max. matches

Sets the max. number of search results that the LDAP server is to output.

⬈ Further help under LDAP

Name attributes

Relevant name attributes.

⬈ Further help under LDAP

Phone number attributes

Relevant phone number attributes.

⬈ Further help under LDAP

Sort by

Sorting algorithm (by First name, by Last name)

⬈ Further help under LDAP

Name format

Defines the format in which names are displayed (First name first, Last name first).

↗ Further help under LDAP

Number search incoming

If "Number search incoming" is enabled, the display name is searched for via LDAP, for incoming calls.

↗ Further help under LDAP

Number search outgoing

If "Number search outgoing" is enabled, the display name is searched for via LDAP, for outgoing calls.

↗ Further help under LDAP

Telephony

Call signalling
Direct call
Dialled calls
Call forwarding
Call history
Automatic call acceptance

Call signalling

Call waiting

Indicates an incoming call during a call.

Audible indication

This acoustic signal indicates that there is an incoming call during a call.

If this option is disabled, incoming calls are only signalled by a change in the display content during a call.

Do not disturb

Rejects calls to the phone.

Call Completion

The callback is triggered automatically by the PBX.

Further help under Call Completion

Direct call
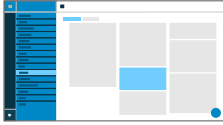
Destination

Enter the phone number that is to be called after the specified waiting time has elapsed.

ℹ️ **Notices:**The function is only active if a destination has been entered. Comply with the phone number format set for the PBX when entering external phone numbers.

↗ Further help under Direct call

Waiting time before call starts in s

Shows the number of seconds the device will wait before starting the call to the specified destination.

↗ Further help under Direct call

Dialled calls
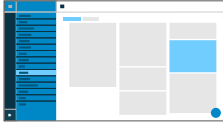
Anonymous call

Stops your phone number from being displayed to the person you are calling

Automatic start of call

The call starts automatically after the predefined time.

Seconds until start of call

You can use the sliding adjuster to set the number of the seconds that the phone waits before making the automatic call.

Call forwarding

Immediately

Forwards incoming calls to the set destination number immediately.

↗ Further help under Call forwarding

On busy

Forwards incoming calls to the set destination number if the subscriber is busy.

↗ Further help under Call forwarding

On no reply

Forwards incoming calls to the set destination number if these calls have not been taken within a set time.

↗ Further help under Call forwarding

Call forwarding destination

Enter the destination number here.

↗ Further help under Call forwarding

Waiting time before call forwarding

Sets the number of the seconds that the phone waits before activating call forwarding.

Further help under [Call forwarding](#)

Call history

Log calls

Activates the logging of incoming and outgoing calls.

Note: If the function is switched off, calls cannot be started by redialling the number.

Further help under Log calls

Automatic call acceptance

On SIP request

Received calls use an Alert Info header containing `info=alert-autoanswer` to request immediate call acceptance.

↗ Further help under Automatic call acceptance

After timeout

Sets the number of seconds that the phone waits before automatically taking an incoming call. Automatic call acceptance is prevented by selecting the setting **Not automatically**.

↗ Further help under Automatic call acceptance

Beep when connecting

Uses a tone to signal incoming calls.

↗ Further help under Automatic call acceptance

Interrupt active calls

Active calls are put on hold when InterCom calls are received.

Note: For active calls to be interrupted, the Call waiting function must be enabled.

Further help under Automatic call acceptance

Sound

Ringtones by call type
Ringtone management

Ringtones by call type
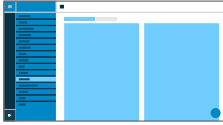
#Numbering

Displays the sequential number within the list.

⬈ Further help under Sound

Search term

Assigns the selected ringtone to the alert info from the PBX/provider's SIP Invite. Six names are predefined by default.

⬈ Further help under Sound

Ringtones by call type

List of possible ringtones. Assign a ringtone to a call type here.

⬈ Further help under Sound

Ringtone management
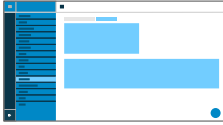
Import

Uploads the selected audio file to the device.

There are two steps in the procedure:

- **Browse:** Displays a dialogue in which you select the required file.
- **Import:** Uploads the selected audio file to the phone.

Note: Max. permitted file size per file: 5 MB.

Further help under Sound

Memory utilisation

Displays how much memory is occupied by imported ringtones.

Further help under Sound

Date and time
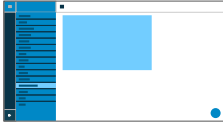
Time Phone

Display of the time set in the phone.

Update via C .

Date Phone

Display of the date set in the phone.

Update via C .

Automatic time (NTP)

The time is automatically transferred from the NTP server.

If NTP is not enabled, there is no synchronisation between the phone and the time server. It is possible that the date and time may be displayed incorrectly.

Further help under Date and time

NTP server

The time server that supplies the time stamp. Use the address of a time server that is geographically close by, to achieve maximum accuracy.

Example: pool.ntp.org

↗ Further help under <u>Date and time</u>

Time zone

Data on the phone that has a time stamp (such as call history entries, calendar entries and e-mails) is displayed with the correct time.

↗ Further help under <u>Date and time</u>

Time format

12 h: time display in 12-hour format. In addition, the time range AM/PM is displayed.

24 h: time display in 24-hour format.

↗ Further help under <u>Date and time</u>

User

User administration
Settings

User administration
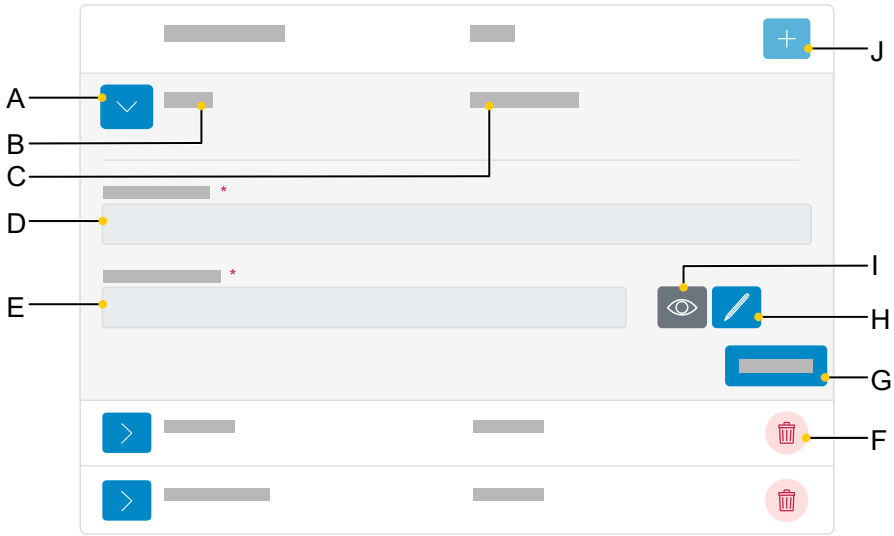
A Displays/hides the settings for a user

B Displays username as assigned in D.

C Displays the role as assigned when the new user was created.

> ⓘ **Note:** Administrator rights can only be assigned when a new
> user is created.

D **Username** field

E **Password** field

F Deletes a user.

G Stores all settings for a user.

H Click on the Edit icon to assign a new password.

I Displays/hides the currently set password (only in Edit mode).

J   Adds a new user to the list.

Settings
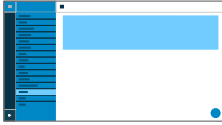
User name

Press this to display the user name.

ℹ️ **Note:** Do not enter colons "**:** In the user name:

↗ Further help under <u>User roles and permissions</u>

Password

Enter/change the password here.

When you set a new password, comply with the conventions below. The password must:

- be different from the previous one used
- contain at least one number
- contain at least one capital letter
- contain at least one lower case letter
- contain at least one special character
- be at least 9 characters long
- be different from the user name

↗ Further help under <u>Passwords</u>

Admin

Assigns administrator rights to the user.

Note: Administrator rights can only be assigned when a new user is created.

Further help under [User roles and permissions](#)

Backup and restore
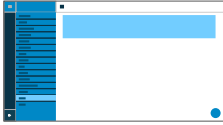
Create backup

Exports all phone settings to an .xml file.

A copy of the file will be saved on the PC.

Note: Depending on the setting for your browser, when you export the file, you will either be prompted to specify where the file will be stored or the file will be saved to your PC's Downloads folder. Example, Mozilla Firefox: **Settings** > **Files and applications** > **Downloads**.

Further help under

Restore backup

Uploads all settings in the saved .xml file to the phone.

There are two steps in the procedure:

- **Browse:** Displays a dialogue in which you can select a previously saved copy of the phone settings in the appropriate folder on the data storage device.

- **Import:** Uploads the settings from the selected .xml file to the phone.

Further help under

Access

General settings
Telephone menu

General settings

Developer options
Action URL
Logging
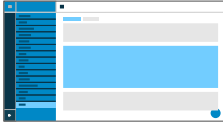
Developer options

Android Debug Bridge (adb)

Enables the PC to access the phone via the software interface. You can then use ADB to execute commands on the Android system.

Action URL

Evaluate action URLs

Triggers running of action URLs. The device receives HTTP requests from the PC and executes them.

Note: Activating **Evaluate action URLs** is not possible if the function **Web administration (https)** is inactive.

Further help under Action URL

Action URL server

Required to send events. No entry required if only commands are received.

Further help under Action URL

Generate new token

Provides a static password for the phone interface.

Further help under Action URL

Logging
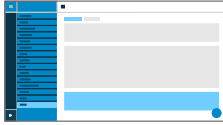
Extended Logging

Enables debug info recording.

Note: In some situations, extended logging can have a negative impact on system performance.

Further help under Logging

Telephone menu
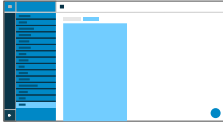
Filter settings

Limits the number of menu options are displayed. For administrators, the reduced display is applied to the phone. For users, the reduced display is applied both to the phone and the web interface.

Note: Only administrators can see this page.

Further help under User roles and permissions

Using the Phone

Basic Knowledge
Calls
Contacts
Pickup
Settings

Basic Knowledge

Display
Notifications
Keys and LEDs

Display

Display

Display

Display overview
System icons

COMfortel D-100 / D-110 / D-200 / D-210 - Firmware V2.6 - Advanced Information V10 03/2023
- 333 -

Display overview

Home

The next figure shows the telephone display with the Home screen open.
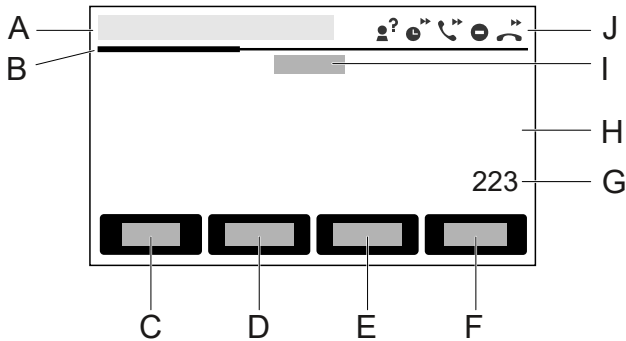
A   Status bar

B   Horizontal scroll bar. Displays the current level.

C   softkey **Menu**

D   softkey**Calls**

E   softkey **Contact**

F   softkey**Level**

G   Automatically generated label for a function key that has a function assigned to it (in this case: internal number 223)

H   Empty area, used to display other function keys

I   Displays the subscriber name, as set in**Settings** > **Identities** > **Display name**

J   System icons display: In the area**Settings** > **Telephony**functions have been enabled.

Varying display names

Different display names can be sent. The display names will be used both on your own device and when SIP messages are sent.

Display names on the home screen
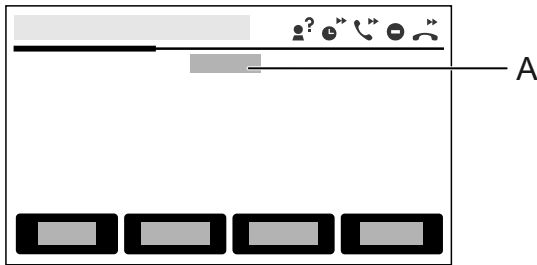
A Possible displays (sequence indicates priority):

  – Primary username (only provisionable)

  – Display name as assigned in Identities

  – Device name

  – SIP username@SIP registrar (xxxx@xxx.xxx.xxx.xxx)

  – No WebUI (if web administration (https) is deactivated and no identity has been configured)

  – IP address (if device is connected but not registered)

  – No display

list

Appearance in lists

Elements on the display are shown in different ways to indicate their state:
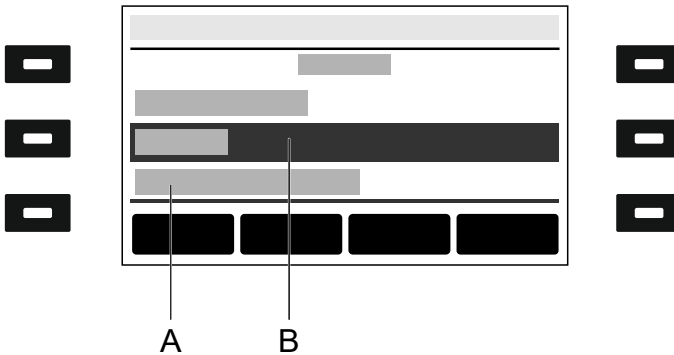
A   B

A   **underlined:**The selected element is locked and cannot be selected by pressing OK⌷ok⌷.

B   **negative:**An element is displayed as a negative image in a dark bar if you activate it using the direction keys and you can display a sub-menu by pressing OK⌷ok⌷.

All call lists

Overview of call lists
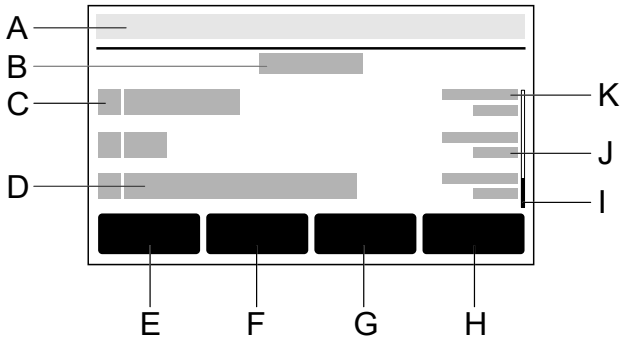
A   Status bar: Displays the date, time and, if required, other system icons

B   Title of the current list:**All** / **Missed** / **Outgoing**

The number in brackets is the number of calls in the corresponding list.

C   Call-type icon

↗   Details about the settings are in Display

D   Subscriber number display or name display for a saved contact.

E   softkey**All:** Lists all calls by how recent they are.

F   softkey**Missed:** Lists all calls, sorted chronologically.

G   softkey**Outgoing:** Lists all outgoing calls, sorted chronologically.

H   softkey**Menu:**Press this to access the following options for using the selected subscriber number:

- **Call**

- **Edit number before call**

- **Add to contacts**

- **Clear entry**

I Scroll bar

J Date

K Call date

Note: If you pick up the receiver while the caller list is open, the most recently dialled subscriber number will be dialled immediately. To remove entries from a call list, press Delete⬛. To delete the entire list, press Delete ⬛ for approximately 2 seconds. If you want to exit the caller list without dialling a phone number, press Back⬛.

Details about the settings are in Display

Incoming calls

The following information is displayed when incoming calls are received by a COMfortel D-series telephone:

- the destination that has been called

- the caller

- the person being called

- the call path

🛈 **Important:** The call paths are only displayed if the telephone has Feature level 2 or above (device is operating on a COMtrexx) or after the D-series convenience package has been activated (device is operating on an Auerswald/FONtevo PBX).

Internal

A **Ringing**
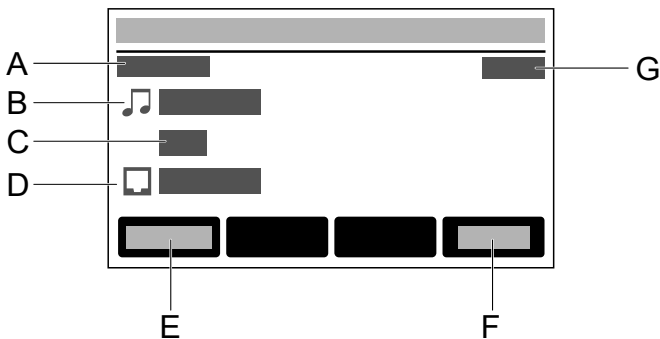
B Caller name

C Telephone number of the caller

D   Own called identity

E   **Accept**(Accept)

F   **Reject**(Rejecting)

G   Call duration

External
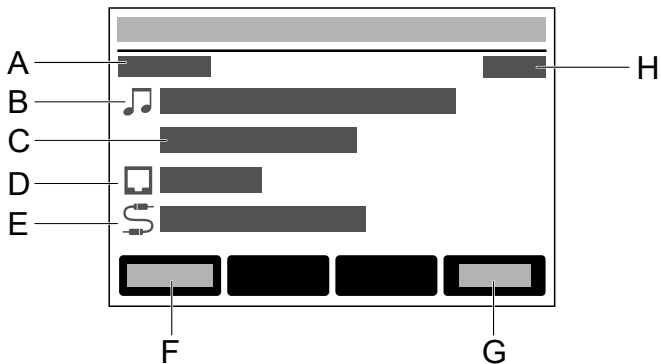
A   **Ringing**

B   Caller name

C   Telephone number of the caller

D   Own called identity

E   Name of the called exchange line number

Ist kein Name vergeben: Anzeige der gerufenen Nummer.

F   **Accept**(Accept)

G   **Reject**(Rejecting)

H   Call duration

group calling
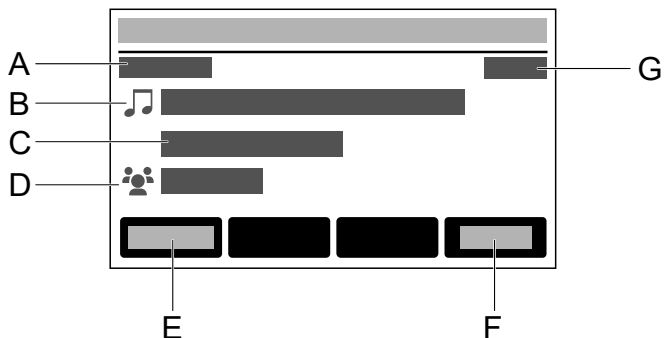
A **Ringing**

B Caller name

C Telephone number of the caller

D Name of the called group

E **Accept**(Accept)

F **Reject**(Rejecting)

G Call duration

Forwarded call
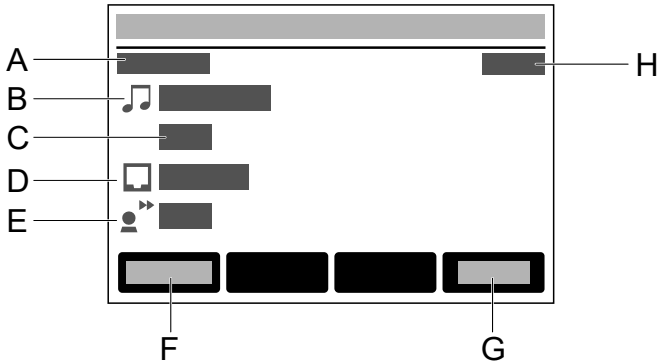
A **Ringing**

B Caller name

C Telephone number of the caller

D Own identity

E Phone number of the person who is forwarding the call

F **Accept**(Accept)

G **Reject**(Rejecting)

H Call duration

Call waiting

Use the Call waiting function to signal incoming calls during an on-going conversation.

Further help under Call waiting

Do not disturb

Use the Do not disturb function to switch off incoming calls if necessary.

⬈ Further help under Do not disturb

Pickup

Use the pickup function to take a call, originally sent to another internal telephone, or your own voice mailbox, on your own phone.

⬈ Description of the operation under Pickup

Call forwarding

Use the Call forwarding function to forward calls independently of the current situation (immediately, on busy or on no reply).

⬈ Further help under Call forwarding

Active conversation

Active conversation between 2 SUBs
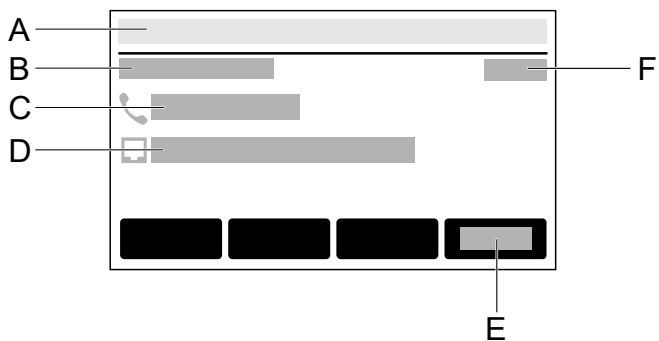
A   Status bar

B   **Connected.**

C   Identity of called SUB

D   Own identity

E   **End**

F   Conversation duration

Putting a conversation on hold

Put subscriber 1 on hold
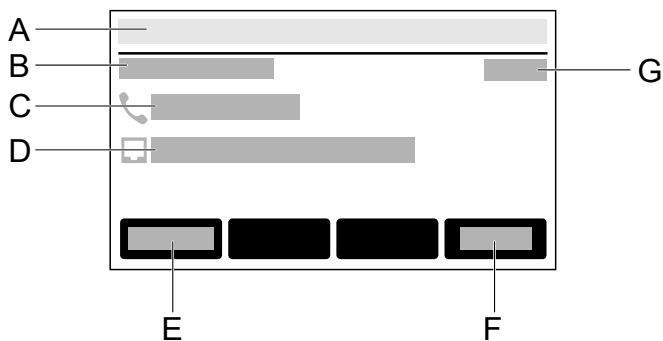
A   Status bar

B   **Holding**

C   Identity of SUB put on hold

D   Own identity

E   **Dial**

F   **End**

G   Conversation duration

Put 1 subscriber on hold, 1 subscriber in call
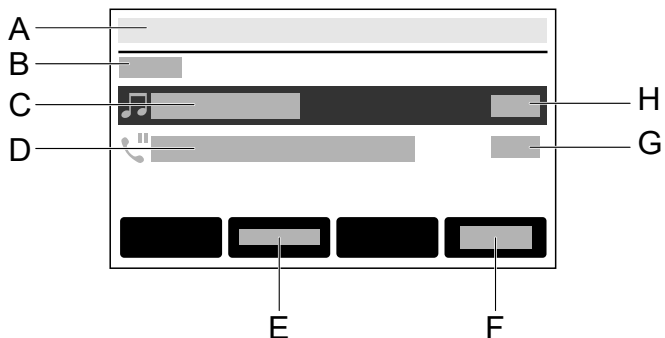
A   Status bar

B   **Call**

C   SUB in call

D   Held SUB

E   **Join**Connects SUB in call with held SUB

F   **End**

G   Duration of call on hold

H   Duration of incoming call

Put 1 subscriber on hold, 1 subscriber in conversation
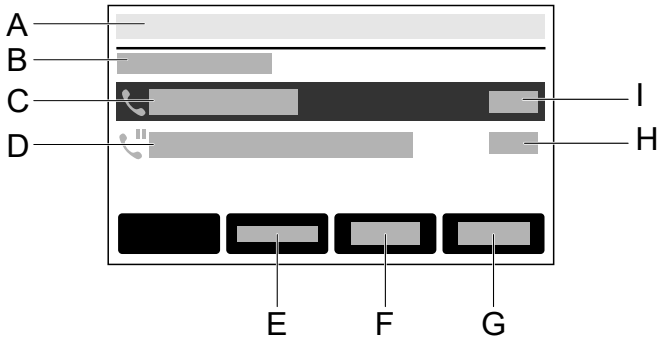
A   Status bar

B   **Connected.**

C   SUB in conversation

D   Held SUB

E   **Join**Connects SUB in call with held SUB

F   **Conf.**

G   **End**

H   Duration of call on hold

I   Duration of active conversation

Conference call

Preliminary conference call setup stage
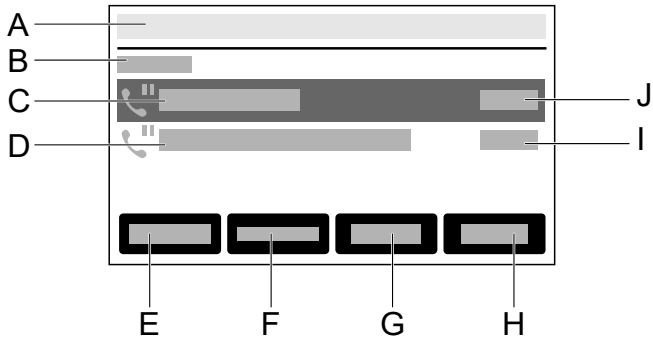
A   Status bar

B   **Holding**

C   SUB 1 on hold

D   SUB 1 on hold

E   **Dial**

F   **Join**

G   **Conf.**

H   **End**

I   Call duration SUB 2

J   Call duration SUB 1

Active conference call
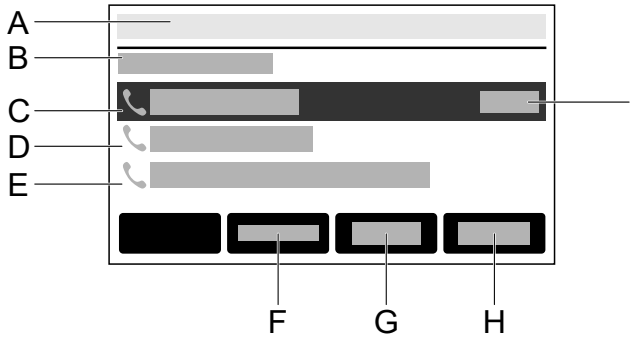
A   Status bar

B   **Connected.**

C   **Conference call**(active)

D   Conference call participant 1

E   Conference call participant 2

F   **Join**

G   **Split**

H   **End**

I   Conference call duration

Transfer

Call forwarding using transfer
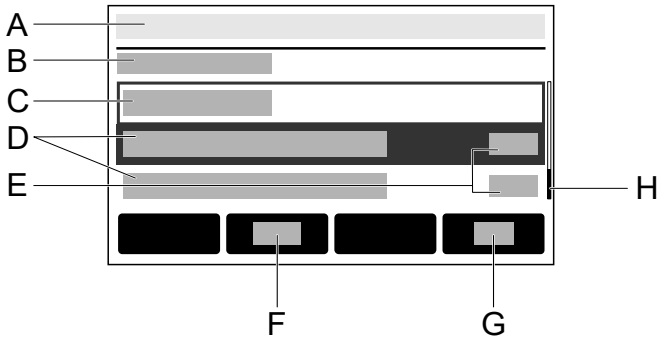
A Status bar

B **Transfer target**

C Enter the transfer target

Suggested contacts, each consisting of:

D Contact name

E Subscriber number of the contact

F Toggle between alphanumeric and numerical entry

G **OK**

H Scroll bar for navigating the list

System icons

⤳ Call forwarding immediately

📞 Call forwarding on busy

📞 Call forwarding by time

👤? Anonymous call

⊖ Do not disturb

📞 Incoming call

📞 Active call

📞 Call on hold

⟩⟨ Forwarded call

👤 Call Completion

🎵 Ringing

▯ Identity

⤳ Line

👥 Group

↙ Accepted call

↙ₓ Missed call

↗ Successful outgoing call

↗ₓ Unsuccessful outgoing call

⚇ Active conference call

⁝⚇ Conference call participant

⚇" Own device is on hold

⚇" Own device is being forwarded

✓ Identity active

✕ Identity inactive

− Identity not configured

Notifications

ℹ **Note:** The options and displays provided in the Notifications menu option are identical to those for the message key ▨.

↗ Further help under Dedicated function keys

One missed call

Select to open the list **Calls** > **Missed**.

A new voice message

The mailbox is called and the dialler is displayed.

Follow the spoken instructions to find out what to do next.

▨ Display/hide messages

Pressing the message key displays the messages. If no interaction with the message is intended, pressing the message key again makes the view disappear.

Keys and LEDs

Keypad

Dedicated function keys

Function Keys near the Screen

Soft Keys below the Display

Delete, Home and Back

Directional pad and OK

LED signalling on function keys

Casing LED

Selecting a Key Level

Changing the input mode

Keypad

Use the keypad to dial subscriber numbers and to enter numbers and
letters. Press the digit keys one or more times until the required character
appears. The assignment is then listed in the table below. The characters
available vary according to the language configured for the device.

Key	Assignment
1	1 - / @
2	a b c 2 A B C ä Ä
3	d e f 3 D E F
4	g h i 4 G H I
5	j k l 5 J K L
6	m n o 6 M N O ö Ö
7	p q r s 7 P Q R S ß
8	t u v 8 T U V ü Ü
9	w x y z 9 W X Y Z
0	Space . , 0 + : ' \ " ? ¿ ! ¡ _
*	* ( ) [ ] < = > « » ^ ´ ` ~ { }
#	# % & § £ $ €

Dedicated function keys

✉ Messages with LED

- Displays these lists:
    - Voice messages
    - Missed calls

- Pressing the message key displays the messages. Depending on the device variant, you can press the message key again to clear the display if you do not want to respond to the message.

ℹ **Note:** The behaviour of the LED message key can be configured via **Settings** > **Display and LED** > **Seen missed calls/Heard voice messages**

🎤 Microphone with LED

- Switches the microphone off (mute) or on again.
- LED is shining red: microphone is disabled

⏸ Hold with LED

- You can put an active call on hold or reconnect it again.

⏺ Redialling

- Displays the outgoing calls list.

🎧 Headset with LED

- Press the Headset key to enable or disable the headset.
- LED is shining green: headset operation is active
- LED is flashing green: incoming call
- LED is shining red: active hedaset call

☎ Transfer without LED

• You can transfer calls with or without prior notification.

⊟ minus

• Reduces the volume of the ringer, receiver, or loudspeaker, depending on operational state

◄ Loudspeaker with LED

• LED is flashing green: incoming call

• LED flashing red: loudspeaker listening is enabled

• LED is shining red: hands-free calling is enabled

⊞ plus

• Increases the volume of the ringer, receiver, or loudspeaker, depending on operational state.

Function Keys near the Screen

The basic version of the device has six function keys which you can program to suit your needs. Each function key has a multi-colour LED that indicates its status, based on its assignment.
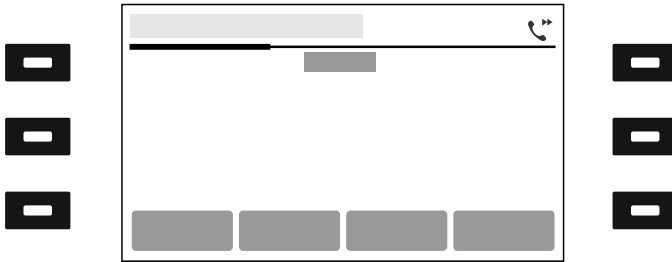
Using the function keys you have programmed from the idle state

Run functions using the programmed function keys to the right or left of the screen:

Function Keys near the Screen
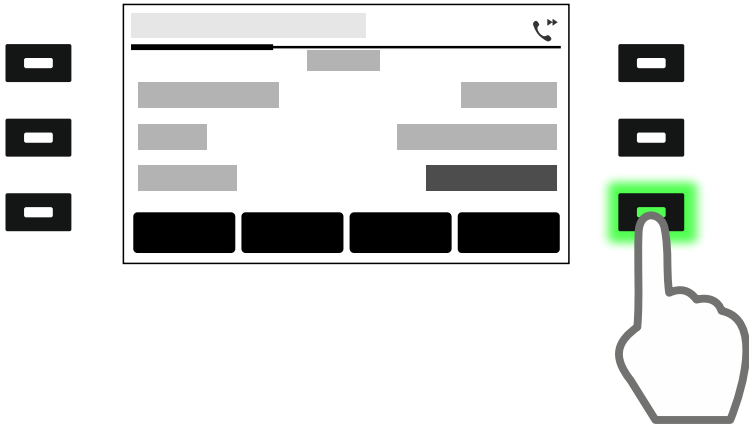
Press the button at the bottom, on the right, enables a function. The LED on the button shines green.

Further help under Call forwarding

Navigation in menus and lists

Press the top or bottom function key on the right of the screen: you jump up or down one screen or scroll through a list/menu quickly.
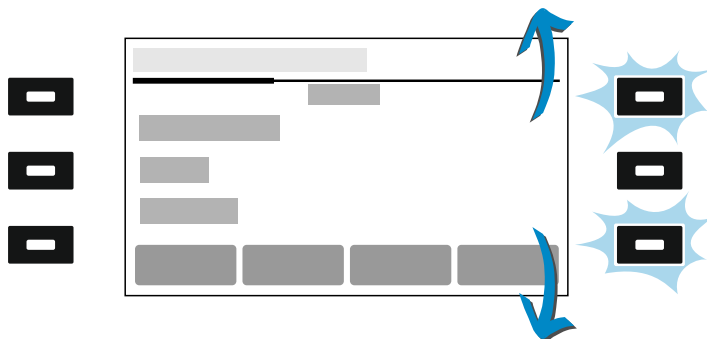
Key extension module COMfortel® D-XT20/D-XT20i

The number of programmable function keys on the COMfortel D-200/
D-210 can be extended by adding up to three COMfortel D-XT20/D-XT20i
key extension modules (not included in the scope of delivery). Modules
are attached to the right-hand side of the phone. When the maximum
number of modules has been installed, 105 freely programmable function
keys are available.

Function Keys near the Screen
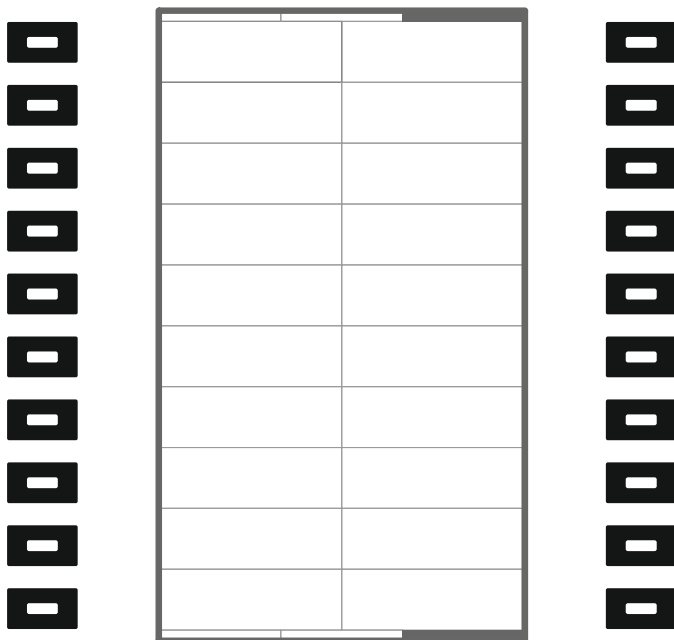
The programmable function keys can each be defined three times (on three levels).

Soft Keys below the Display

The soft keys have different functions, depending on the operational state or which menu has been selected. Each function is displayed directly above the key in the display.
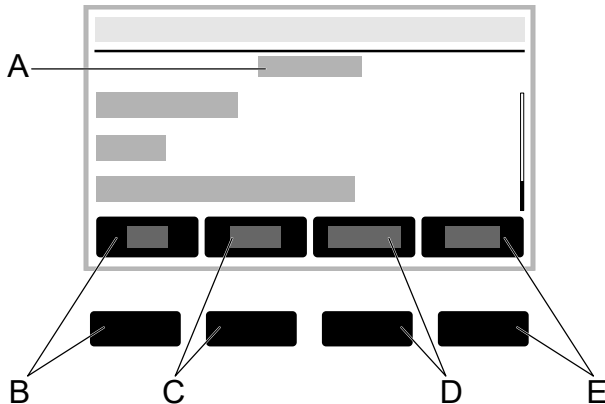
A   Menu name, here **Contact**

B   softkey**Fav.** / **All:**Switches between the lists **Contact** / **Favourites**.

C   softkey**Delete**Deletes a saved contact.

D   softkey**Details:**Press this to display more saved phone numbers for a contact.

E   →**Fav.** / →**Fav.:**Turns a contact into a favourite or favourite into a normal contact.
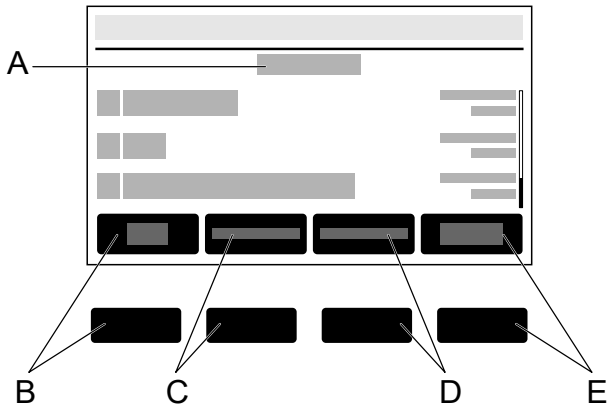
A   Menu name, here**Calls**

B   softkey**All:** Lists all calls by how recent they are.

C   softkey**Missed:**Press this to list all missed calls by how recent they are.

D   softkey**Outgoing:**Press this to list all outgoing calls by how recent they are.

E   softkey**Menu:**Press this to access more options for the selected phone number.

Delete, Home and Back

⌫ Delete

- Press this to delete the last character before the cursor.

● Home

- Press this to display the home screen. Previously started actions will continue in the background.

⌫ Back

- Press this to return to the most recently opened menu and end the current action.

Directional pad and OK

Use the directional pad to navigate in the display.

Use the directional pad direction keys to start operation and activate required elements in lists and menus. By pressing OK[ok] the activated element is selected and the element's function is executed.

Elements on the display are shown in different ways to indicate their state:
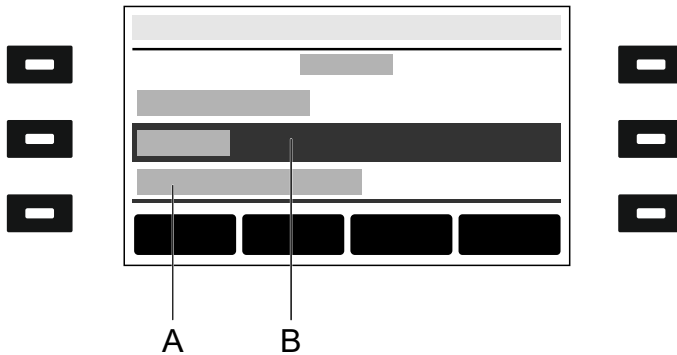
A   B

A   **underlined:**The selected element is locked and cannot be selected by pressing OK[ok].

B   **negative:**An element is displayed as a negative image in a dark bar if you activate it using the direction keys and you can display a sub-menu by pressing OK[ok]

LED signalling on function keys

Each LED indicates the status depending on the key definition.

*: The function key or the described behaviour of the LED is only visible if the function key has been provisioned by a PBX or activated by manufacturer templates.

Function	LED display	Status
Speed dial	Shining red	Outgoing call from speed dial number or currently active conversation on speed dial number
	Flashing red	Inbound call to the speed dial number
	Flashing green	Missed call from speed dial number
	Shining green	Registered target subscriber
	Off	No active calls or conversations
	flashing yellow	Sending subscription to PBX
	shining yellow	Subscription failed
		unregistered target subscriber with enabled function **Signal presence**

Function	LED display	Status
Identity	Shining green	Active, registered identity, not a standard identity
	Off	Active, registered identity, not a default identity
	flashing yellow	Current registration attempt
	shining yellow	Incorrect identity
Call waiting	Shining green	Function enabled
	Off	Function key configured, function deactivated
Do not disturb	Shining red	Function enabled
	Flashing red *	Time gap between pressing the function key and acknowledgement by the PBX/provider
	Flashing yellow *	Current attempt to register the function key on the PBX/provider, using SIP Subscription
	Shining yellow *	Error

Function	LED display	Status
Automatic configuration switching	Shining green	Function enabled
	Off	Function key configured, function deactivated
Anonymous call	Shining green	Function enabled
	Off	Function key configured, function deactivated
Call forwarding on busy	Shining green	Function enabled
	Off	Function key configured, function deactivated
	Flashing green *	Time gap between pressing the function key and acknowledgement by the PBX/ provider
	Flashing yellow *	Current attempt to register the function key on the PBX/provider, using SIP Subscription
	shining yellow	Error, function key perhaps not completely configured

LED signalling on function keys

Function	LED display	Status
Call forwarding on no reply	Shining green	Function enabled
	Off	Function key configured, function deactivated
	Flashing green *	Time gap between pressing the function key and acknowledgement by the PBX/provider
	Flashing yellow *	Current attempt to register the function key on the PBX/provider, using SIP Subscription
	shining yellow	Error, function key perhaps not completely configured
Call forwarding immediately	Shining green	Function enabled
	Off	Function key configured, function deactivated
	Flashing green *	Time gap between pressing the function key and acknowledgement by the PBX/provider

Function	LED display	Status
	Flashing yellow *	Current attempt to register the function key on the PBX/provider, using SIP Subscription
	shining yellow	Error, function key perhaps not completely configured
VPN	Shining green	Function enabled
	flashing yellow	Establishing VPN connection
	Off	Function key configured, function deactivated
XML Minibrowser	None	Performs the stored functions
Switch relay	Flashing green (approx. 2 sec.)	Relay has been switched
Group, **Group (incoming)**, **Group (outgoing)***	Shining green	Logged into the group
	Off	Logged out of the group
	Flashing green	Login process

Function	LED display	Status
	flashing yellow	Sending subscription to PBX
	shining yellow	Subscription failed
Call forwarding ext. calls: On busy*	Shining green	Function enabled
	Off	Function key configured, function deactivated
	Flashing green	Time gap between pressing the function key and acknowledgement by the PBX/provider
	flashing yellow	Current attempt to register the function key on the PBX/provider, using SIP Subscription
	shining yellow	Error, function key perhaps not completely configured
Call forwarding ext. calls: On no reply*	Shining green	Function enabled
	Off	Function key configured, function deactivated

Function	LED display	Status
	Flashing green	Time gap between pressing the function key and acknowledgement by the PBX/provider
	flashing yellow	Current attempt to register the function key on the PBX/provider, using SIP Subscription
	shining yellow	Error, function key perhaps not completely configured
Call forwarding ext. calls: Immediately*	Shining green	Function enabled
	Off	Function key configured, function deactivated
	Flashing green	Time gap between pressing the function key and acknowledgement by the PBX/provider
	flashing yellow	Current attempt to register the function key on the PBX/provider, using SIP Subscription
	shining yellow	Error, function key perhaps not completely configured

LED signalling on function keys

Function	LED display	Status
Activate config*	Shining green	Active configuration
	Off	Configuration inactive
Park position	Shining red	Call in the internal park zone
	Off	Free park position
From device version COMfortel® D-400		
Browser	None	Calls the set page in the browser
Bluetooth	Off	Bluetooth not active
	Shining green	Bluetooth active
	shining yellow	Error: maybe no Bluetooth dongle inserted

Casing LED

The Casing LED indicates the following statuses:

- Flashing red (quickly): incoming call
- Lit in red: active call

Selecting a Key Level

Change between key levels by pressing the following keys:

On the home screen

- directional pad Left and Right keys
- softkey**Level**

On the key extension module

- control rocker

On the basic device, you can see the current key level on the horizontal scroll bar under the date display.

On the key extension module, you can see the current key level on the horizontal scroll bar on the top and bottom edge of the display.

The display content changes according to the selected key level.

Changing the input mode

If you want to use the keypad to enter data in fields, press and hold (for approx. 2 seconds) to toggle the input mode (abc < > 123) ✳ .

Calls

Navigation
Menu

Navigation

⤢ Further help under [Display overview](#)

All

Lists all calls, sorted chronologically.

Missed

Press this to list all missed calls by how recent they are.

Outgoing

Press this to list all outgoing calls by how recent they are.

Menu

Call

Calls the previously selected phone number.

Edit number before call

Displays the previously selected phone number. You can then modify the number by selecting:

* Delete
* Complete

Add contact

Jumps to the menu **Add Contact**.

↗ Details about the settings are in [Edit Contact](#)

Clear entry

Deletes the selected entry in the Call history.

Note: Alternatively, you can delete entries in the caller list by pressing Delete ⬚. To delete the entire list, press Delete ⬚ for approximately 2 seconds.

Using the Call History in Different Ways

1. Select an entry in the call history.
2. By pressing OK ⬚ for approximately 2 seconds.

The options as described are listed.

3. Select any required options.

Contacts

Navigation
Edit Contact

Navigation

Fav.

Jumps from the list of all contacts to the Favourites list.

All

Jumps from the Favourites list to the list of all contacts.

Details

Displays all the details for the selected contact.

Delete

Deletes the selected contact.

→Fav./≠ Fav.

→Fav. Marks the selected contact as a Favourite.

→Fav.turns the selected contact, which is marked as a favourite, back into a normal contact.

Add Contact

Creates a new contact.

This is then added to the appropriate source list (Favourites or Contacts), from which a new contact is added.

↗ Details about the settings are in Edit Contact

Navigation

⬈ Further help under Contact

Edit Contact

Enter the name of the contact.

Select this so that you can enter a name using the keypad.

Via the soft key**More**you can display and edit the following name details:

- Displayed name

- Last name

- First name

- Middle name

Enter a phone number

Select this so that you can enter a phone number using the keypad.

Via the soft key**Type**you can display and assign the following phone number details:

- Private

- Mobile

- Work

- Other

  Note: Via the soft key**Default**you can set the default phone number to be used if several phone numbers have been assigned to a contact.

No.+/Del Nr

Adds a phone number to a contact, or removes it from the contact.

Save

Saves the contact settings.

Pickup

Pickup

All the calls that can be picked up from the pickup list are displayed and sorted by time in the app **Pickup**.

The pickup list for the display of calls of a single key that can be picked up appears with the enabled function **Pickup-choice on key-press**.

Further help under Pickup

Settings

Network
Identities
Sound
Language and location
Display and LED
Telephony
Function keys
Update
Using provisioning
Access
Date and time
Contacts/LDAP
About phone

Network

Current Settings
IPv4 settings
IPv6 Settings
VLAN
VPN
Topology

Current Settings

The current settings will be shown.

IPv4 settings

IPv4 mode

DHCP

The phone obtains its IP address automatically from a DHCP server.

Static

The telephone uses a fixed IP address (supplied by the system administrator).

↗ Further help under IPv4

IP address

If you select **DHCP:** the phone's IP address is displayed

If you select **Static:** you can enter/change the phone's IP address

In the factory settings: 0.0.0.0.

↗ Further help under IPv4

Subnet mask

Example: 225.255.255.0

↗ Further help under IPv4

Gateway

Example: 192.168.178.1

The gateway provides interoperability between networks, so data can pass from one to another. The provider or the network system administrator provide the gateway.

↗ Further help under [IPv4](IPv4)

DNS 1

The DNS server is needed for Internet access. The provider or network system administrator sets the DNS server. The provider or network system administrator sets the DNS server.

↗ Further help under [IPv4](IPv4)

DNS 2

Optional

↗ Further help under [IPv4](IPv4)

IPv6 Settings

IPv6 mode

Auto

The phone obtains its IP address automatically from a DHCP server.

Disabled

Disables IPv6 mode. IPv4 over DHCP is used.

Further help under IPv6

Privacy Extensions

If this function is activated, IPv6 addresses that do not allow the user to be identified are created. The host part of the IPv6 address is anonymised.

Further help under IPv6

DNS 1

The DNS server is needed for Internet access. The provider or network system administrator sets the DNS server. The provider or network system administrator sets the DNS server.

Further help under IPv6

DNS 2

Optional

Further help under IPv6

VLAN

Phone VLAN

Switches on VLAN for the phone.

↗ Further help under VLAN

Phone VLAN ID

Enter the phone's VLAN ID: 1 … 4094

The associated data packets are tagged with this ID.

↗ Further help under VLAN

Phone VLAN priority

0 (lowest priority) to 7 (highest priority)

↗ Further help under VLAN

PC port VLAN

Switches on VLAN for the PC port.

↗ Further help under VLAN

PC port VLAN ID

Enter the PC port's VLAN ID: 1 … 4094

The associated data packets are tagged with this ID.

↗ Further help under VLAN

PC port VLAN priority

0 (lowest priority) to 7 (highest priority)

⤴ Further help under VLAN

VPN

VPN

Switches on VPN for the phone.

↗ Further help under VPN

User name

User name for the VPN connection.

Your network administrator will give you your user name.

↗ Further help under VPN

Password

Password for the VPN connection.

Your network administrator will give you the password.

↗ Further help under VPN

Topology

Basic Settings
STUN server

Basic Settings

Check hostname

Checks whether the certificate belongs to the domain/IP.

↗ Further help under Certificate

FrameSize

Sets the RTP packet size (msec Audio/RTP package).

Local SIP port

1024 … 65535

Default: 5060

Sets the outgoing port for SIP messages.

↗ Further help under SIP

Jitter buffer

Initial value: Initial prefetch delay value (msec).

ⓘ **Note:** The value must lie between Min. prefetch and Max. prefetch. If the Initial value = 0, prefetch is deactivated.

Min. prefetch: Minimum prefetch delay (msec).

Default: 60 msec

Max. prefetch: Maximum prefetch delay (msec).

Default: 240 msec

Maximum: Delay that can be set for the jitter buffer (msec).

Default: 360 msec

↗ Further help under Topology

RTP off when silent

Ends the sending of RTP packets if no active conversation is taking place.

ⓘ **Note:** If this function is switched on, parts of the call can seem like breaks in the connection because background noises are cut off sharply.

↗ Further help under Topology

LLDP

If this function is enabled, LLDP packages are sent regularly to the directly connected LLDP-enabled switch.

↗ Further help under LLDP

LLDP: Interval in seconds

LLDP package sending frequency

30 to 3600 sec

Minimum: 30 sec

Default: 30 sec

↗ Further help under LLDP

STUN server

⊕ / Add STUN server

Adds another STUN server to the list. The list is dynamic. Settings can be made for up to eight STUN servers.

STUN server URL

Requirement:

• Configured DNS server

Enter the STUN server URL.

↗ Further help under Topology

STUN server port

1 … 65535

ⓘ **Important:** Every time port forwarding is performed, there is a security risk. For this reason, we recommend you use port forwarding as little as possible.

ⓘ **Note:** If you encounter problems with half-open (dropped) call connections, enabling the RTP port might help.

↗ Further help under Topology

Time lapse registration (min.)

5 … 60 minutes

STUN server

The registration time is the number of minutes that must elapse before the telephone reregisters itself with the VoIP provider.

The value you enter here should be a compromise between a short timespan (which generates traffic with the provider) and a long re-registration period.

↗ Further help under Topology

Identities

Managing Identities
Basic Settings
Options for experts

Managing Identities

Display settings

1. Select an identity in the list.

2. Press OK to confirm your selection ⌷ OK ⌷.

You can see all settings in the in the individual menu options.

ⓘ **Note:** You can only display settings for set-up identities.

↗ Further help under Identities

Creating a new identity

1. Select an unconfigured identity (**---**) from the list.

2. Confirm your selection with OK ⌷ OK ⌷.

3. Make the settings as described under **Identities**.

A new identity is added to the list.

↗ Further help under Identities

Deleting an identity

1. Select the identity you want to delete in the list.

2. softkey**Clear**Press

   ↗ Further help under Identities

ⓘ **Important:** All settings made for the selected identity are deleted without asking you to confirm. If you back up the settings for an identity to a data storage device first, you can restore them if required.

Basic Settings

Active

Press this to switch the selected identity on.

↗ Further help under Identities

SIP Username

Used to identify the device to the VoIP provider or in the PBX. Identical to the user name assigned by the VoIP provider or the internal phone number registered in the PBX.

↗ Further help under Identities

SIP Registrar

This tells the phone where it can register itself. The information given here is the URL set by the VoIP provider on the IP address on which the registering PBX can be accessed.

↗ Further help under Identities

Password

Used to verify the user name. Identical to the password assigned by the VoIP provider when the identity was being created or the password entered in the PBX.

↗ Further help under Identities

Display name

Freely selectable name.

The display name is displayed both in the web interface, under the display name, and on the phone, below the status bar.

If you do not set a display name, the display on the phone contains these elements: SIP user name@SIP registrar.

Further help under Identities

Authentication name

Enter the authentication username provided by the VoIP provider.

You need the authentication username when you register with a VoIP provider. If you connect to an Auerswald PBX, you do not need to enter the authentication ID.

Further help under Identities

Options for experts

Registrar
SIP
NAT
Function codes
Audio
Security
Outbound proxy
Fallback for
Ringer Tone
Name sources

Registrar

SIP domain

Name space that contains all the entries within one directory, specified by the VoIP provider/PBX operator.

⤴ Further help under [Identities](#)

Port

The registrar's port. The (recommended) default value is 5060.

⚠ **Caution:** Different entries can cause malfunctions if the provider makes changes. An SRV query is only possible with the default value.

⚠ **Caution:** Each time a port is opened on the router, this creates a potential security risk. It is essential you put additional protective measures in place.

⤴ Further help under [Identities](#)

Time lapse registration (min.)

5 … 60 minutes

The registration time is the number of minutes that must elapse before the telephone reregisters itself with the VoIP provider.

The value you enter here should be a compromise between a short timespan (which generates traffic with the provider) and a long re-registration period.

⤴ Further help under [Identities](#)

Early re-register (sec)

Shortens a re-registration by the specified number of seconds. Early re-registration prevents the registration from expiring prematurely.

Further help under Identities

Refresh register at 1/2

Turns on re-registration after half of the negotiated maximum registration period.

Further help under Identities

SIP

Session timer

Switches on the check after a connection for a call that is still in existence.

ℹ Note: When the SIP session timer is switched on, this may result in the call being interrupted more frequently after the specified interval, if a VoIP provider has not implemented session renewal properly. In this case, set a different session timeout or disabled the session timer.

↗ Further help under SIP

Session timeout (in min.)

2 … 255 minutes, default: 15 minutes

Specifies the number of minutes after which the SIP session timer is to check a call's connection.

↗ Further help under SIP

Protocol type

UDP

(User Datagram Protocol). Used to send data packets over connectionless non-secure communication lines.

ℹ Note: If very large data packets are present, TCP is used instead of UDP. The maximum size of a data packet can vary according to the network. (RFC 3261 > TCP)

TCP

(Transmission Control Protocol). Segments data into packets, from a specified size, and sends these individual data packets to the recipient address until receipt has been confirmed.

↗ Further help under [SIP](SIP)

ⓘ **Important:** If encryption using SIPS is enabled, the TCP transport protocol is used. Manual settings are overwritten.

↗ Further help under [SIP](SIP)

Subscription interval (min)

This sets the frequency at which the status of potential changes are queried on the PBX. Default: 45 minutes

The value you enter here should be a compromise between a short interval (which generates traffic) and rapid updates.

↗ Further help under [SIP](SIP)

Retry subscriptions

Type of repetition

Sets the interval at which attempts are made to configure a subscription on the PBX/provider, if an error occurs.

- **Never**

  The device sends exactly one subscription to the PBX/provider. No other subscriptions are sent.

- **Fixed interval**

  Attempts to subscribe on the PBX/provider are made at the specified interval.

- **Redouble interval**

    The set number of seconds doubles after each attempt to subscribe on the PBX/provider.

Interval in seconds

Sets the time gap between two subscription attempts.

Minimum: 10 sec

Default: 180 sec

Further help under [SIP](#)

Retry register

Type of repetition

Sets the interval at which attempts are made to register on the PBX/provider, if an error occurs.

- **Never**

    The device makes exactly one attempt to register on the PBX/provider. There are no more registration attempts.

- **Fixed interval**

    Attempts to register on the PBX/provider are made at the specified interval.

- **Redouble interval**

    The set number of seconds doubles after each attempt to register on the PBX/provider.

Interval in seconds

Sets the time gap between two subscription attempts.

Minimum: 1 sec

Default: 10 sec

Further help under SIP

DTMF method

Specifies the DTMF method used to transmit signals.

- **RTP event**

  Transmission of event packets in the RTP stream

- **Inband**

  Transmission of coded sound signals, directly in the RTP stream

- **SIP info**

  Transmission of SIP info messages

Further help under DTMF

NAT

SIP NAT traversal

Requirements:

- Configured DNS server
- If the option "active" is selected: **Registrar** enabled for the phone

Active

NAT traversal is performed by the telephone. To do this, you must also specify a STUN server.

⚠ **Caution:** Each time a port is opened on the NAT router, this creates a potential security risk. Implement appropriate security measures.

Inactive

NAT traversal is not performed by the telephone. The SIP request also contains an "rport" part in which the IP addresses used here (local, public) are transported. The router which connects the local area network with the Internet should be a properly functioning SIP-aware router which performs NAT traversal.

↗ Further help under [Identities](#)

RTP NAT traversal

Requirement:

- Configured DNS server

Active

NAT traversal is performed by the telephone. To do this, you must also specify a STUN server.

NAT

⚠️ **Caution:** Each time a port is opened on the NAT router, this creates a potential security risk. Implement appropriate security measures.

ℹ️ **Note:** If there are problems with unilateral call connections, a STUN server should be used.

Inactive

NAT traversal is not performed by the telephone. The router which connects the local area network with the Internet should be a properly functioning SIP-aware router which performs NAT traversal.

↗ Further help under Identities

NAT Keep-Alive Interval (sec.)

After the time specified here has passed, NAT Keep-Alive packets are sent to the firewall to maintain the NAT mapping.

ℹ️ **Note:** Some providers lock accounts if the NAT Keep-Alive Interval is too short. As a rule, this is reported with error message 503 during SIP registration. If this problem occurs, it is recommended to raise the value (e.g. to 180).

↗ Further help under Identities

Use ICE

Enable the best possible transmission route to be found between the phone and PBX/provider.

ℹ️ **Note:** ICE must only be activated if the other end supports ICE.

↗ Further help under Identities

Function codes

Pickup code

This is required to perform a call pickup. Enter the character string stored on the PBX/at the provider, e.g.**##06** for Auerswald PBXs.

Voicemail number

Enter the voicemail number assigned by the VoIP provider or the voicemail number entered in the PBX.

Further help under <u>Voicemail number</u>

CLIR type

(Number suppression)

Area in the From header in which the VoIP provider expects to receive number presentation suppression. Selection as set in the PBX/at the provider.

Anonymous

The sent display text in the From header is "anonymous".

User anonymous

Both the display text and the User name section in the From header is "anonymous".

Further help under <u>Anonymous call</u>

Audio

Music on hold

If a connection or call is on hold, the "music on hold" is played.

↗ Further help under Music on hold

Play popup notification sound

Enables audible signalling of new popup messages.

ℹ **Note:** No tone is output when a new popup message is displayed during an active call.

↗ Further help under Notifications

Codecs

Press this to display the codecs in the sequence in which they are used.

If you select a codec, you can replace it with a different codec, or none at all.

To achieve the best possible balance between bandwidth and speech quality, we recommend you do not change the sequence of the codecs.

ℹ **Note:** Codecs that are assigned multiple times are automatically removed from the list and replaced with**---**.

↗ Further help under Codecs

Security

SIPS active

Enables the encrypted transfer of signalling data.

↗ Further help under [SIPS](#)

↗ Further help under [Certificate](#)

↗ Further help under [Identities](#)

Peer-to-peer TLS

Forces the encryption of SIP messages over the entire route to the destination.

ℹ **Note:** The call is not made if TLS is not available on the entire connection.

↗ Further help under [Identities](#)

SRTP

Transport protocol for encrypted connections.

Mandatory

This setting forces voice encryption to be on. If the voice partner (VoIP provider, PBX, external VoIP subscriber) does not support SRTP, the connection is not established.

Preferred

Switches on negotiation for the encryption of call data via SRTP. When a call is made, the call partner will be asked if encryption is possible. If

selected, voice data is transmitted in encrypted form. If not selected, it is not encrypted.

Disabled

This setting forces voice encryption to be off. If the voice partner (VoIP provider, PBX, external VoIP subscriber) does not support encryption, the connection is not established.

↗ Further help under [SRTP](#)

↗ Further help under [Identities](#)

Certificate

If the SIPS function is activated, the system checks whether the certificate belongs to the domain/IP.

ℹ **Note:** For each device variant, simply shows whether a certificate is installed and where processing can take place, if relevant.

↗ Further help under [Identities](#)

↗ Further help under [Certificate](#)

Outbound proxy

Mode

Inactive

Deactivate this function if

- the registrar can be accessed without outbound proxy

- no outbound proxy is available

Automatically

The PBX automatically determines an outbound proxy.

Manually

The outbound proxy is a fixed default setting. Enter this data as an IP address or a URL.

↗ Further help under Outbound proxy

URL

Server address for the outbound proxy.

ℹ **Note:** The **URL** must be filled out if mode **Manually** is selected.

↗ Further help under Outbound proxy

Port

Port for the outbound proxy.

ℹ **Note:** The **Port** must be filled out if mode **Manually** is selected.

↗ Further help under Outbound proxy

Alternative URL

Alternative server address for the outbound proxy.

Further help under [Outbound proxy]

Alternative port

Alternative port for the outbound proxy.

Further help under [Outbound proxy]

Fallback for

No fallback

No fallback is possible for the current identity.

⤢ Further help under Fallback identity

Configured/Unconfigured identity

Sets the main identity for which the fallback identity applies.

Configured identity: displays **Display name**

Configured identity: displays **Identity** + seq. no.

ℹ **Note:** A fallback identity can also be created for identities that have not yet been configured.

⤢ Further help under Fallback identity

Ringer Tone

Ringer Tone

Assigns a ringtone to the identity.

↗ **Further help under** [Sound](#)

Name sources

The sequence in which the name sources are displayed in the list sets the sequence in which they are applied. The first source that contains a name is used for the display.

Name sources that are not to be displayed can be removed from the list. Depending on the device variant, press

- 🗑

- **Delete**

- Selecting a list entry opens the selection list, select **---**.

Depending on the device variant, to add a name source, press

- ➕

  ℹ **Note:** Only displayed if there is a free place in the list.

- Selecting an empty list entry (**---**) opens the selection list of all the available name sources.

Depending on the device variant, the default state of the name sources is created by

- **Reset to default**

- **Default**

↗ Further help under Name sources

Sound

Procedure for selecting a ringtone:

- If you hover over a ringtone, the system plays it.

- If you confirm a selected ringtone, this enables it.

Default ringtone

Assigns a ringtone to inbound calls. This ringtone can be set individually so the user can identify different call types → **Ringtones by call type**.

Further help under Sound

Default notification

Assigns a ringtone to notifications (e.g. callback on busy).

Further help under Sound

Ringtones by call type

Opens a list with six predefined default names.

Select an entry in the list to open another menu in which you can assign a ringtone to the call type.

Search term

Alert info from the PBX/provider's SIP Invite.

Ringer Tone

Assigns a ringtone to the alert info from the PBX/provider's SIP Invite.

Further help under Sound

Language and location

Phone language

Changes the language used on the phone user interface.

↗ Further help under Language and location

Location

Sets the phone's actual or required location.

Audio frequencies (pitches, pulse/pause characters and/or ringing tones), and also the time and time format, are different for each selected location.

↗ Further help under Language and location

Language web interface

Changes the language used on the web interface

↗ Further help under Language and location

Display and LED

Idle state

After the selected time, the phone is automatically returned to the idle state.

- **Not automatically**

  The phone is not automatically returned to the idle state.

↗ Further help under [Idle state](#)

Brightness

Sets the brightness.

ℹ **Note:** Any changes you make to the brightness affect on all key extension modules that are connected.

↗ Further help under [Display and LED](#)

Contrast

Sets the contrast.

ℹ **Note:** Any changes you make to the contrast affect on all key extension modules that are connected.

↗ Further help under [Display and LED](#)

Seen missed calls

Sets the message key LED's reaction to seen missed calls.

- **LED lights up**

The message key LED shines and a text message is displayed in the Notifications menu.

- **LED is flashing**

The message key LED flashes and a text message is displayed in the Notifications menu.

- **LED off**

The message key LED does not flash and a text message is displayed in the Notifications menu.

- **Disabled**

The message key LED does not flash and no text message is displayed in the Notifications menu.

Further help under Display and LED

Heard voice messages

Sets the message key LED's reaction to heard voice messages.

- **LED lights up**

The message key LED shines and a text message is displayed in the Notifications menu.

- **LED is flashing**

The message key LED flashes and a text message is displayed in the Notifications menu.

- **LED off**

The message key LED does not flash and a text message is displayed in the Notifications menu.

- **Disabled**

The message key LED does not flash and no text message is displayed in the Notifications menu.

Further help under Display and LED

Telephony

Call waiting

Audible indication

Do not disturb

Call forwarding

Anonymous call

Automatic start of call

Direct call

Automatic call acceptance

Call Completion

Log calls

Call waiting

Call waiting

Indicates an incoming call during a call.

Audible indication

Audible indication

This acoustic signal indicates that there is an incoming call during a call.

If this option is disabled, incoming calls are only signalled by a change in the display content during a call.

↗ Further help under Call waiting

Do not disturb

Do not disturb

Rejects calls to the phone.

Call forwarding

Immediately

Forwards incoming calls to the set destination number immediately.

↗ Further help under [Call forwarding](#)

Call forwarding destination

Enter the destination number here.

↗ Further help under [Call forwarding](#)

On busy

Forwards incoming calls to the set destination number if the subscriber is busy.

↗ Further help under [Call forwarding](#)

Call forwarding destination

Enter the destination number here.

↗ Further help under [Call forwarding](#)

On no reply

Forwards incoming calls to the set destination number if these calls have not been taken within a set time.

↗ Further help under [Call forwarding](#)

Call forwarding destination

Enter the destination number here.

⬀ Further help under Call forwarding

Waiting time before call forwarding

Sets the number of the seconds that the phone waits before activating call forwarding.

⬀ Further help under Call forwarding

Anonymous call

Anonymous call

Stops your phone number from being displayed to the person you are calling

Automatic start of call

Automatic start of call

The call starts automatically after the predefined time.

Seconds until start of call

You can use the sliding adjuster to set the number of the seconds that the phone waits before making the automatic call.

Direct call

Destination

Enter the phone number that is to be called after the specified waiting time has elapsed.

Notices: The function is only active if a destination has been entered. Comply with the phone number format set for the PBX when entering external phone numbers.

Further help under Direct call

Waiting time before call starts in s

Shows the number of seconds the device will wait before starting the call to the specified destination.

Further help under Direct call

Automatic call acceptance

On SIP request

Received calls use an Alert Info header containing `info=alert-autoanswer` to request immediate call acceptance.

⬈ Further help under [Automatic call acceptance](#)

After timeout

Sets the number of seconds that the phone waits before automatically taking an incoming call. Automatic call acceptance is prevented by selecting the setting **Not automatically**.

⬈ Further help under [Automatic call acceptance](#)

Beep when connecting

Uses a tone to signal incoming calls.

⬈ Further help under [Automatic call acceptance](#)

Interrupt active calls

Active calls are put on hold when InterCom calls are received.

ℹ **Note:** For active calls to be interrupted, the Call waiting function must be enabled.

⬈ Further help under [Automatic call acceptance](#)

Call Completion

Call Completion

The callback is triggered automatically by the PBX.

↗ Further help under Call Completion

Log calls

Log calls

Activates the logging of incoming and outgoing calls.

Note: If the function is switched off, calls cannot be started by redialling the number.

Further help under Log calls

Function keys

Managing function keys
Edit function keys
Edit parameters

Managing function keys

Choose a key

Selects a key for adjusting settings.

⬈ Further help under Function keys

Delete

Deletes the marked function key in the list.

⬈ Further help under Function keys

Insert

Adds an entry to the list, above the marked position.

⬈ Further help under Function keys

Edit

Opens a dialogue in which parameters can be configured for the selected key.

⬈ Further help under Function keys

Edit function keys

Function

Selects the function and displays the necessary parameters.

Note: To use additional function templates, activate the Auerswald/ Fontevo templates.

Further help under Function templates

Details about the settings are in Edit parameters

Key label

Displayed beside each function key on the screen.

Delete

Deletes the marked function key in the list.

Cancel

Closes the menu without adopting changes.

Save

Saves the entered parameters and close the window.

Edit parameters

Speed dial

Identity

Activate config

Anonymous call

Automatic configuration switching

Call forwarding immediately

Call forwarding on busy

Call forwarding on no reply

Call waiting

CF ext. calls: Immediately

CF ext. calls: On busy

CF ext. calls: On no reply

Do not disturb

DTMF-Relay

Group

Group (incoming)

Group (outgoing)

Park position

Relays

Silent Alert

SIP message

VPN

XML Minibrowser

Speed dial

Number/URI

Enter a phone number or address such as 198@192.168.XX.YY or 198@company.com, here.

↗ Further help under Speed dial

Identity

Identity for which this function is set up.

↗ Further help under Identities

Subscription

Activates the sending of BLF subscriptions to the PBX.

ℹ **Note:** It may be a good idea to deactivate subscriptions in situations when it is necessary to prevent the PBX from being overloaded by high volumes of transmitted data.

Pickup

Activates the Pickup function.

ℹ **Note:** Pickup only works if the Subscription function is activated.

ℹ **Note:** Pickup should not be used if the key is only used to switch functions in the PBX using a programming code. Deactivate Pickup in that case.

↗ Further help under Pickup

InterCom

Enables the InterCom function. The connection with the destination
number is established directly. No ringing or active call acceptance.

Further help under InterCom

Signalise missed call

If this function is enabled, missed calls associated with the speed dial
number are signalled with flashing green LEDs.

Further help under Speed dial

Pickup-choice on key-press

If this function is switched on, a list of all the calls relevant for the function
key, that can be taken via pickup, is displayed.

Note: If this function is switched on, a call cannot be taken by pressing
a key. Instead, a list of all the calls that can be taken via pickup is
displayed, even if the list only contains one call.

Further help under Pickup

Do not show in pickup list

If this function is switched on, calls that can be taken by pickup are not
forwarded to the global pickup list.

Further help under Pickup

Pickup of connected calls

If this function is switched on, calls that have already been connected can be picked up. This option is useful, for example, if an answering machine or mailbox number has been assigned to the speed dialling key. The calls are included in the global pickup list.

Further help under Pickup

Signal presence

If this function is switched on, the following states are transmitted (if they are supported by the PBX):

- Successful registration of the monitored phone number on the PBX

- Do not disturb

- Call forwarding

Default: disabled.

Identity

Identity

Identity used to trigger subsequent calls.

⬈ Further help under Identities

Activate config

Identity

Identity for which this function is set up.

⬆ Further help under Identities

Configuration id

The PBX assigns the configuration ID automatically.

⬆ Further help under Function templates

Anonymous call

No parameters necessary.

⬈ Further help under Anonymous call

Automatic configuration switching

Identity

Identity for which this function is set up.

⬈ Further help under [Identities](#)

ID

Identification number for system relay with Automatic configuration switchover operating mode (900, in the factory settings).

⬈ Further help under [Function templates](#)

Call forwarding immediately

Identity

Identity for which this function is set up.

⤴ Further help under Identities

ℹ **Note:** Only assign the Identity parameter if PBX function templates are used.

⤴ Further help under Function templates

Destination

Call forwarding destination

⤴ Further help under Call forwarding

Call forwarding on busy

Identity

Identity for which this function is set up.

↗ Further help under Identities

ℹ **Note:** Only assign the Identity parameter if PBX function templates are used.

↗ Further help under [Function templates](Function%20templates)

Destination

Call forwarding destination

↗ Further help under [Call forwarding](Call%20forwarding)

Call forwarding on no reply

Identity

Identity for which this function is set up.

⬈ Further help under [Identities](#)

ℹ **Note:** Only assign the Identity parameter if PBX function templates are used.

⬈ Further help under [Function templates](#)

Destination

Call forwarding destination

⬈ Further help under [Call forwarding](#)

Call waiting

No parameters necessary.

↗ Further help under [Call waiting](#)

CF ext. calls: Immediately

Identity

Identity for which this function is set up.

⬈ Further help under [Identities](#)

Call forwarding for external calls

Point-to-multipoint connection: External phone numbers (MSN, see PBX configuration), for which call forwarding is to be performed.

PTP connection: Header plus extension for which call forwarding is to be performed.

Destination

Call forwarding destination

⬈ Further help under [Function templates](#)

CF ext. calls: On busy

Identity

Identity for which this function is set up.

⤴ Further help under Identities

Call forwarding for external calls

Point-to-multipoint connection: External phone numbers (MSN, see PBX configuration), for which call forwarding is to be performed.

PTP connection: Header plus extension for which call forwarding is to be performed.

Destination

Call forwarding destination

⤴ Further help under Function templates

CF ext. calls: On no reply

Identity

Identity for which this function is set up.

↗ Further help under Identities

Call forwarding for external calls

Point-to-multipoint connection: External phone numbers (MSN, see PBX configuration), for which call forwarding is to be performed.

PTP connection: Header plus extension for which call forwarding is to be performed.

Destination

Call forwarding destination

↗ Further help under Function templates

Do not disturb

Identity

Identity for which this function is set up.

Further help under Identities

Further help under Do not disturb

DTMF-Relay

DTMF sequence

Numbers needed to perform the required function.

⬈ Further help under [Function templates](#)

Group

Identity

Identity for which this function is set up.

⬀ Further help under Identities

Group

Group telephone number

⬀ Further help under Function templates

Group (incoming)

Identity

Identity for which this function is set up.

⤴ Further help under Identities

Group

Group telephone number

⤴ Further help under Function templates

Group (outgoing)

Identity

Identity for which this function is set up.

⤢ Further help under Identities

Group

Group telephone number

⤢ Further help under Function templates

Park position

Number/URI

Phone number of the park position.

Identity

Identity for which this function is set up.

⬈ Further help under Park position

Relays

Identity

Identity for which this function is set up.

Further help under Identities

ID

Relay ID number according to PBX.

Note: The Relay function key is for relays with light, door opener or calculator operating mode. This function key is not suitable for manual relays. For additional Information, refer to the Advanced Information about the PBX.

Further help under Function templates

Silent Alert

URL

Required for forwarding the alarm message.

⚠️ **Caution:** The function can only be executed if the appropriate URL has been entered.

Delay

Countdown counter. An action is triggered after the countdown reaches zero.

↗ Further help under Silent Alert

SIP message

Number/URI

Enter a phone number or address such as 198@192.168.XX.YY or 198@company.com, here.

Identity

Identity for which this function is set up.

Message

This is where you enter the text for sending an SIP message.

Delay

Countdown counter. An action is triggered after the countdown reaches zero.

Further help under SIP message

VPN

No parameters necessary.

↗ Further help under Network

XML Minibrowser

Number/URI

Web server address

⬀ Further help under [XML Minibrowser](#)

Update

Start update

Starts the update.

↗ Further help under Update

Check for updates

If selected, the phone automatically checks for current updates.

↗ Further help under Update

Automatically install updates

If selected, available updates are installed automatically.

↗ Further help under Update

Update server URL

In the default settings: https://update.auerswald.de

↗ Further help under Update

Restart phone

Restarts the telephone.

ℹ **Caution:** This interrupts all connections, all calls and data services. Unsaved data is lost!

↗ Further help under Update

Reset to default

Resets the telephone to the factory settings. Any saved data and settings are deleted. The phone is shut down and then restarted.

↗ Further help under [Reset to default](#)

Using provisioning

Start provisioning

Starts manual provisioning.

Automatic provisioning

Automatic provisioning is performed according to the schedule and each time the phone is switched on.

⬈ Further help under Using provisioning

Provisioning with DHCP data

The phone obtains the provisioning server's URL from a DHCP server. Additional data for controlling provisioning can also be transferred along with the URL.

⬈ Further help under Using provisioning

Redirect Server

The phone obtains the provisioning server's URL from the redirect server.

⬈ Further help under Using provisioning

Provisioning from PBX search

Automatic provisioning of the phone is performed via the PBX.

⬈ Further help under Using provisioning

Provisioning server URL

This URL is required if it is not obtained via DHCP or a redirect server.

↗ Further help under Using provisioning

Vendor templates

Activates templates and scripts that are specially designed to run on Auerswald/FONtevo PBXs.

↗ Further help under Function templates

Access

Extended Logging

Enables debug info recording.

Note: In some situations, extended logging can have a negative impact on system performance.

Further help under Logging

Web administration (https)

Activates access to the web interface.

Note: You cannot call web interface if this function is inactive.

Android Debug Bridge (adb)

Enables the PC to access the phone via the software interface. You can then use ADB to execute commands on the Android system.

Password web interface

Displays the password entry/modification dialogue for the web interface.

Note: When you define the password, comply with the conventions for secure passwords.

Further help under Passwords

Evaluate action URLs

Triggers running of action URLs. The device receives HTTP requests from the PC and executes them.

Note: Activating **Evaluate action URLs** is not possible if the function **Web administration (https)** is inactive.

Further help under Action URL

Action URL server

Required to send events. No entry required if only commands are received.

Further help under Action URL

Date and time

Automatic time (NTP)

The time is automatically transferred from the NTP server.

If NTP is not enabled, there is no synchronisation between the phone and the time server. It is possible that the date and time may be displayed incorrectly.

↗ Further help under Date and time

NTP server

The time server that supplies the time stamp. Use the address of a time server that is geographically close by, to achieve maximum accuracy.

Example: pool.ntp.org

↗ Further help under Date and time

Time format

12 h: time display in 12-hour format. In addition, the time range AM/PM is displayed.

24 h: time display in 24-hour format.

↗ Further help under Date and time

Time zone

Data on the phone that has a time stamp (such as call history entries, calendar entries and e-mails) is displayed with the correct time.

Date and time

⬀ Further help under [Date and time]

Contacts/LDAP

Active

Switches on the directory service request.

↗ Further help under LDAP

Max. LDAP contacts

Sets the max. number of contacts that the LDAP server is to output.

↗ Further help under LDAP

Max. matches

Sets the max. number of search results that the LDAP server is to output.

↗ Further help under LDAP

Sort by

Sorting algorithm (by First name, by Last name)

↗ Further help under LDAP

Name format

Defines the format in which names are displayed (First name first, Last name first).

↗ Further help under LDAP

Number search incoming

If "Number search incoming" is enabled, the display name is searched for via LDAP, for incoming calls.

Further help under LDAP

Number search outgoing

If "Number search outgoing" is enabled, the display name is searched for via LDAP, for outgoing calls.

Further help under LDAP

About phone

Device name

The name you assign to the device.

⬈ Further help under Device Information

Model

Displays your device's model details.

Serial number

Displays your device's serial number.

Keypad serial number

Displays the serial number of the keyboard PCB installed on your device.

Build number

Displays the build number of the software version installed on the device.

MAC address

Displays your device's physical address on the network.

Android version

Displays the Android version installed on your device.

Kernel version

Displays the kernel version in use on your device.

Update system version

Displays the update system version in use on your device.

Bootloader version

Displays the bootloader version in use on your device.

Network chip

Model of the network chip integrated in the phone.

SPL version

Displays the SPL version.

Licenses

Here, you can display the following license information:

- **End User License Agreement**
- **Data protection**
- **Open source licenses**
- **Android licenses**
- **Webclient licenses**

Further help under Device Information

Glossary

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VoIP account

VoIP address

VoIP provider

VPN

XML

Account

Authorisation to access a VoIP provider or a VoIP PBX. Users must identify themselves by logging in with a username and a password.

Browser

Special software to display websites from the Internet.

Call deflection

Call Deflection – refers to call forwarding during the ringing period. If Call Deflection has been activated, the called subscriber can deflect the call on a case-to-case basis while the telephone is still ringing.

Certificate

Digital key issued by a Certification Authority (CA) to ensure the authentication and secure transmission of e-mails, web sites, etc.

Client

Piece of computer hardware or software that accesses a service made
available by a server.

Codec

A codec is a method that encodes (digitizes) analogue voice data
for transmission and again decodes again, meaning converts into
back into voice. There are various codecs that feature different voice
data compression rates thereby require different band widths for data
transmission. The equality of VoIP calls is dependent on the codec used.

DHCP

Dynamic Host Configuration Protocol – client/server protocol for dynamically allocating IP addresses and network parameters. The IP addresses are requested by the DHCP clients (PCs in the network) on the DHCP server (for example, a router or the Internet service provider). The DHCP server takes these IP addresses from a set address pool and sends them to the client. In addition, the client receives additional information (for example, the addresses for the standard gateway and DNS server).

The IP address is temporarily allocated for a certain amount of time. If the address is no longer required by the client, the server has access to it again, and can allocate it to another client.

Dial preparation

Dialling of a number before a call is initiated by lifting the receiver.

DNS

Domain Name Service – needed to translate Internet addresses. The name of a computer on the Internet (for example**www.auerswald.de** is assigned to the corresponding IP address. This service is provided by DNS servers at the various Internet service providers or by upper domain servers.

Domain

Globally unique name of a website, consisting of third-level domain (e. g. the service name "www"), second-level domain (e. b. "auerswald") and top-level domain (e. g. the country code "uk"). The domain is part of the URL.

Downdate

Reset of the telephone to the previous firmware version.

DTMF

Dual Tone Multi Frequency dialling – dialling method that transmits the
dialled number to the exchange as two superimposed sinusoidal signals.

EHS

Electronic Hook switch: a device that connects a wireless (Bluetooth or DECT standard) headset with the phone, electronically, making particular functions available on the headset. Phone calls can be controlled entirely from the headset, via EHS: conversations can be accepted or ended directly on the headset, conversations can be split between the headset and phone and the phone can be muted. Additional features can be implemented, depending on product.

End device

Device that can be operated on a communication network or on a PBX, e. g. telephone, fax machine, answering machine, ISDN PC controller, etc.

Ethernet

Network system with a speed of 10/100/1000 Mbit/s developed by the companies INTEL, DEC and Xerox.

Firewall

Network security component that uses a set of security rules to protect a computer network or an individual computer against unauthorised access to or from the network.

Firmware

Operating software on the telephone or PBX that is stored in the device itself. The device cannot function without firmware.

Frame size

Sets the RTP packet size (msec Audio/RTP package).

Gateway

PC or router that acts as an intermediary between two networks. The Internet service provider is the gateway for direct Internet dial-up connections. If you use a router, this is the gateway in a local network.

HOST

Component of a data processing system that manages larger application programs and data volumes and makes them available to lower-order servers and clients.

HTTP

Hypertext Transfer Protocol Secure – communication protocol for tap-proof data transfer with encryption within the World Wide Web.

ICE

Interactive Connectivity Establishment is used when two nodes must communicate as directly as possible, over the Internet, but the presence of NATs and firewalls makes it difficult for them to do so. ICE uses the STUN (Session Traversal Utilities for NAT)- and TURN (Traversal Using Relays Around NAT) protocols to establish a connection.

IMAP

Internet Message Access Protocol – protocol to download e-mails from a server, with extended functions as compared to POP3 (e-mails are stored on a server, not on the client PC, so that they cannot be lost due to a failure of the client PC).

IP address

Unique numerical address within a TCP/IP network that is assigned to one device and ensures that data packages reach the correct recipient.

Ipsec

The telephone uses the IPsec protocol to transport IP packages cryptographically secured via public networks.

IPv4

IPv4 addresses consist of four numbers between 0 and 255 that are separated by a full stop, for example, 192.168.0.14. The network addresses are divided up into different classes (Class A to C) depending on how many computers can be located in a network.

Certain ranges are reserved for operating local networks:

- Class A: 10.x.x.x (for networks with up to 16.5 million PCs)

- Class B: 172.16.x.x to 172.31.x.x (for networks with up to 65534 PCs)

- Class C: 192.168.0.x to 192.168.255.x (for networks with up to 254 PCs)

These addresses have no validity on the Internet. This means that data packets with this kind of sender or recipient address cannot be sent over the Internet. However, they can be used in local networks without restriction. The benefit of this is that if data from a local network configured in this way comes in contact with the Internet, none of the data on the computers in the local network can leak out or be accessed externally.

IPv6

IPv6 addresses consist of eight blocks, each containing four numbers or letters, and each separated by a colon, e.g. 3001:00FF:ABC0:0EAC:0001:0000:0000:000F or, in short form, 3001:FF:ABC0:EAC:1::F.

Thanks to the longer address format, IPv6 offers a greater number of possible IP addresses than the Internet Protocol Version 4 (IPv4).

jitter buffer

Buffers RTP packages, to compensate irregular transfer rates of the Internet.

A high jitter buffer reduces short interruptions during the call. In return the delay of the RTP packages increases.

LDAP

Lightweight Directory Access Protocol – protocol that enables you to query the data in a directory service (a hierarchical database) over the network.

LED

Light emitting diode - electronic device similar to a lamp and used for displays (also multi-coloured).

LLDP

Link Layer Discovery Protocol – the protocol IP phones use to collect and query device-specific information from devices directly connected in the network.

LLDP-MED – LLDP software upgrade that supports special VoIP apps.

MAC

Media Access Control address – specific, unique hardware address specified for each individual single network card used in the whole world. This means that the network card unique and can be controlled without any problems and detected by the network The IP addresses on the TCP/IP network are always mapped to the respective MAC address.

NAT

Network Address Translation – permits the mapping of a specific IP address used within a network to another IP address used by another network. This function is e.g. performed by a router which connects a local area network to the Internet.

NAT traversal

Method to establish and hold communication connections that allows IPsec-encrypted data packets to pass trough NAT transition points.

NTP

Network Time Protocol – protocol that synchronises the time on a computer or on other devices in the network with other computers or with an external clock via the Internet. An NTP server is selected to perform the time synchronisation. The time provided by the NTP server is synchronised as exactly as possible to Coordinated Universal Time (UTC).

If a local NTP server is available in a local network, it can be selected, or a public NTP server from a public list on the Internet can be used.

Open source licenses

Software license acknowledged by the Open Source Initiative (OSI).
Open-source licenses are the basis of free-of-charge applications whose
programming code may be modified without restrictions.

PoE

Power over Ethernet (PoE) is a method by which small devices in a network, for example, VoIP telephones, receive both power and data via an Ethernet cable. Benefit: Fewer obtrusive power cables are required.

PoE is described in the IEEE 802.3af standard.

POP3

Post Office Protocol, version 3 – ASCII protocol to download e-mails from a server.

Port

A single PC can simultaneously establish multiple connections and provide multiple services for other PCs. Ports are used to differentiate these connections from parallel connections. Due to a common agreement, PCs usually provide their services on ports 1 to 1023. Outgoing connections are usually established starting at Port 1024. Most services that use the standard port numbers (for example, web browsers use port 80).

Prefetch

The loading of memory contents before a requirement for them has become obvious. When the requirement actually occurs, this enables a faster access speed to be achieved.

Using provisioning

Automatded configuration of one or more telephone(s) from a provisioning server.

Redirect Server

Supports the setup of a connection by identifying alternative addresses of a recipient in the database of the registrar and returning them to the originator of the request. The originator can then send a new request to the alternative address.

Registrar

A domain name registrar is an organisation or company that registers Internet domains.

A SIP registrar tells the phone where it can register itself. The information given here is the URL set by the VoIP provider on the IP address on which the registering PBX can be accessed.

router

Routers connect two separate networks. This means, for example, that you can connect the local network to another LAN or a WAN. When a PC wants to send a data packet, this packet must travel over a router. The router uses the IP address to detect the network it must send the data to. In addition to connecting to networks, routers can also execute certain control functions, such as maintaining a simple firewall. In a home network, routers are usually used to connect the local network to the Internet and to enable simultaneous Internet access for multiple users.

RTP

Transport Protocol – a transport protocol that guarantees complete data transport in real time, especially of audio/video data, for which a packet loss of 1 to 20% is acceptable, depending on the codec in use. RTP does not guarantee the service quality of the data transfer.

Rubber cover

Rubber cover which must be removed before the key extension module can be mounted.

SPL

Secondary Program Loader: a simplified version of the Bootloader for initial hardware configuration.

Server

Computer or software that performs different tasks as part of a network and, for example, provides other users (clients) of this network with specific information, data or services.

SIP

The Session Initiation Protocol (SIP) network protocol sets up a communication session between two or more subscribers. SIP is only used to negotiate the communication conditions. Other protocols such as RTP are responsible for actual data transfer.

SIP port

Port of the local system used as the communication port for the SIP transfer.

SIPS

Session Initiation Protocol Secure – SIPS are SIP data encrypted using the TLS protocol. The network protocol sets up a communication session between two or more subscribers. The provider encrypts external connections to prevent listening in. The processes used to set up and shut down a connection, and signalling, are all encrypted by SIPS.

SLAAC

Stateless Address Autoconfiguration – process for automatically and statelessly configuring IPv6 addresses on a network interface. "Stateless" means that the IPv6 address is not assigned and stored centrally. The host generates its own IPv6 address, using additional information as an aid. SLAAC is the further development of "classic" processes for IP auto configuration in IPv4. IPv6 routers have an active role, unlike those in IPv4. IPv6 offers the same convenience as operating a very simple DHCP server.

SMTP server

Simple Mail Transfer Protocol – network protocol used for the secure transfer of e-mails between the PBX and external mail servers. The PBX needs an external SMTP server (outgoing mail server) to send system e-mails or forward voice mail messages and fax messages.

STUN

Simple Transversal of UDP over NATs – used to determine the public IP
address of an Internet connection. To do this, a STUN request is sent to a
STUN server, which then shares its own IP address allocated by the VoIP
provider with the Internet connection.

Subnet mask

Configuring subnetworks makes it possible to connect many totally different networks located in totally different areas because each subnetwork receives its own address and can therefore be addressed by the router. A subnetwork is created when the IP address is linked with a subnet mask:

If a bit in the subnet mask is set, the corresponding bit in the IP address is considered a part of the network address. If a bit is not set in the subnet mask, the corresponding bit in the IP address is used as a part of the PC address.

The value of the subnet mask, as with IP addresses, is often given in decimal form (for example, 255.255.255.0 for IPv4 addresses).

Switch

Active network distributor that distributes data packages to a destination defined by an IP address among the different segments of a network.

TCP

TCP (Transmission Control Protocol) is a transport protocol that segments data into packets up to a specified size and reliably sends these individual data packets to the recipient address, in the correct sequence. In this process, every data packet sent must be resent until it has been confirmed as arrived. To ensure this happens, a large amount of information is sent along with the actual payload data. Most Internet services are implemented with TCP, e.g., HTTP (WWW), SMTP/POPS (e-mail), etc.

TLS

ransport Layer Security – protocol for the encryption and secure transmission of data on the Internet. TLS is often used to protect http connections, e. g. for commercial transactions (https).

UDP

User Datagram Protocol) – protocoll used to send data packets over non-secure communication lines without a connection. This means that successful transmission is dependent on the application and is therefore not always guaranteed. UDP itself does not verify whether data has been transmitted successfully. When a UDP packet is sent, the sender cannot assume that the packet will indeed arrive at the recipient. This particular protocol needs only a small amount of additional information, and results in a better data throughput rate in a well-functioning network, e.g., on a LAN. UDP is used, e.g., for the DNS (Domain Name Server).

Update

Installation of a new firmware version to integrate new features and to solve possible problems. Previous settings of the device remain unchanged.

update server

Server that provides up-to-date data, e. g. firmware files, provider profiles, etc., for downloading to a telephone.

URI

Uniform Resource Identifier. Used to uniquely identify every single item in the World Wide Web, irrespective of whether it is a page with text, a video, a sound file, a moving image, a static image or a programme.

A URI usually describes:

- the mechanism used to access a resource
- the specific computer on which the resource is present
- the specific name of the resource (or the file name) on the computer

URL

Uniform Source Locator – complete address of a resource (e. g. a website), consisting of a scheme (e. g. „http://") and a scheme-specific part (e. g. domain „www.auerswald.de/" and path „en/products/pbx/home-office.html...").

VLAN

Virtual Local Area Network. This is a virtual network within a local network.

VLAN-enabled switches ensure that data packets within the subnet are not forwarded to a different VLAN, although the subnets might be connected to shared switches.

Among other things, a VLAN can be used to keep IP telephony separate from normal network traffic, at network level.

VoIP

Voice over Internet – Internet telephony. An Internet connection (for example, DSL) is used for telephoning instead of an analogue or ISDN line. In this case, digital voice data is sent as IP packets from from one telephone to another. This functions the same way as the transmission of a webpage over the Internet.

The transmission quality and the reliability of Voice over IP depends to a great extent on the quality of the Internet connection used.

VoIP account

Account configured with a VoIP provider who provides the necessary access data required for VoIP calls. To do this, use your name and address data to register on the provider's website. After that, you will be assigned one or more telephone numbers that can be reached from the land line and the Internet, and also an account with a username and password. The registered connection usually becomes active within a few minutes and can be used very soon after that.

VoIP address

VoIP telephone number plus domain, separated by the @ character: <subscriber>@domain.

VoIP provider

Internet service provider offering internet telephony (VoIP, Voice over Internet Protocol).

VPN

A VPN (Virtual Private Network) integrates distant or neighbouring networks (LAN) into your own network via a VPN tunnel. The tunnel serves as a shell in which the data is embedded and transported from one network to the other.

XML

Extensible Markup Language – a markup language used to display hierarchically structured data, in a text file.

XML data can be opened and edited in an editor. Computers can also read and write XML format. XML consists of tags which are enclosed by pointed brackets: "<" ">". Unlike XML, HTML has only one fixed set of tags, such as, for example <h1>, or <li>. You can define your own custom tags in XML. The standard only defines what a tag must look like, not what it means. Tags can be nested, enabling a hierarchy to be created.

A Document Type Definition (DTD) describes the structure and grammar used in documents. It is an XML system component and is enabled by default.

Abbreviations

AM		answering machine
ARA	Auerswald Remote Access	
CLIR	Calling Line Identification Restriction	Phone number is not displayed if that is appropriate
CTI	Computer Telephony Integration	Computer-aided telephony
DDI	Direct Dial-In	Extension
DECT	Digital Enhanced Cordless Telecommunications	Digital radio telephone network
DHCP	Dynamic Host Configuration Protocol	Protocol for assigning the network configuration to network clients
DNS	Domain Name Service	
DSL	Digital Subscriber Line	Digital subscriber connection
DTMF	Dual Tone Multi Frequency	Dual TMrequency
ESMTP	Simple Mail Transfer Protocol	An integral parrt of e-mail, extended SMTP
HTML	Hypertext Markup Language	Defines the structure of a website

HTTP	Hypertext Transfer Protocol	Hypertext transmission protocol
IMAP	Internet Message Access Protocol	Enables online access to an e-mail inbox (live access)
IP	Internet Protocol	Internet Protocol
Ipsec	Internet Protocol Security	Encryption extension od the Internet Protocol (IP)
IPv4	Internet Protocol verion 4	
IPv6	Internet Protocol verion 6	
ISDN	Integrated Services Digital Network	Digital network with integrated services
LAN	Local Area Network	Local network
LDAP	Lightweight Directory Access Protocol	Protocol for easy access to directory server
LED	Light-emitting diode	Light emitting diode
MAC	Media Access Control	Physical addree of a network interface
NAT	Network Address Translation	
NTP	Network Time Protocol	
PBX	Private Branch Exchange	Private PBX
PoE	Power over Ethernet	

POP3	Post Office Protocol, Version 3	
RTP	Real-Time Transport Protocol	
SIP	Session Initiation Protocol	
SIPS	Session Initiation Protocol Secure	
SLAAC	Stateless Address Autoconfiguration	Process for automatically and statelessly configuring IPv6 addresses on a network interface.
SMTP	Simple Mail Transfer Protocol	Protocol that is used to exchange e-mails in computer networks
SRTP	Secure Real-Time Transport Protocol	
SSL	Secure Sockets Layer	Most widely used security protocol, provides a secure channel between two computers working over the Internet or an internal network
STUN	Simple Transversal of UDP over NATs	
TCP	Transmission Control Protocol	

Abbreviations

TLS	Transport Layer Security	
UDP	User Datagram Protocol	
URI	Uniform Resource Identifier	
URL	Uniform Resource Locator	Uniform Resource Pointer
USB	Universal Serial Bus	
VLAN	Virtual Local Area Network	
VMB	Voice Mailbox	voice mailbox
VoIP	Voice over Internet Protocol	
VPN	Virtual Private Network	
XML	Extensible Markup Language	