V01 03/2022

COMfortel SoftPhone

Operation Manual

Table of Contents

About these instructions

Gender note

For the sake of better readability, the simultaneous use of the language forms masculine, female and diverse (m/f/d) is waived.

All personal designations apply equally to all genders.

Copyright

Trademarks

Microsoft and Windows are registered trademarks of Microsoft Corporation in the USA and/or other countries.

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The latest information

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

You will find the up-to-date instructions on the Internet under Auerswald Support / FONtevo Support.

Setup

Download and Installation

Requirements

- Operating system Windows 10 or later
- Softphone users/subscribers set up in the PBX (see Instructions for the PBX)
- Username and password credentials received via e-mail

Note: The registration information for the COMfortel SoftPhone is emailed to the user when the phone is set up in the PBX.

Note: If you have any questions about the setup, please contact the administration of the PBX.

Configuration

- Softphone is installed

  The installation file for the COMfortel SoftPhone can be found on the Internet at Auerswald Support / FONtevo Support.

- Enter the username and associated password in the Softphone.

Activation errors

Username/Password:

Enter the user name and password again. Ensure you use the correct capitalisation.

Firewall

The firewall blocks access to the software. Allow network access in the Windows settings (**Update and Security** > **Windows Security** > **Firewall & network protection** > **Allow an app through the firewall**).

Go to **COMfortel SoftPhone** and select the required network ranges.

Audio Assistant

Use the Audio Assistant to make microphone and loudspeaker settings.

Requirements:

- Softphone is installed
- Microphone and loudspeaker are connected

Audio Assistant

Follow the Audio Assistant instructions.

Manual audio settings

1. Go to the settings ⚙.
2. Select **Audio**.
3. Test the individual settings under **Device settings** > ▶.

Telephoning

Information about the section

This section contains a list of the different call types. There are often different ways of making those calls. In some places in these Instructions, only one method is described, for the sake of simplicity.

Warning: The COMfortel SoftPhone is not able to make emergency call. Make sure you can make emergency calls elsewhere.

Incoming calls

Accepting a call

Call window >

ringing, on/off

Call window >

The call is then only signalled using the call window.

Note: When the option has been selected, it cannot be cancelled for this call.

Rejecting a call

Call window >

Putting an accepted call on hold

Call menu >

Reconnecting an on-hold call

Call menu >

or

fetch the caller from the waiting position by selecting them in the main window.

Call-Waiting Calls

Requirements for use

- Call waiting must be enabled for the user/subscriber, both in the softphone and on the PBX.
- The called person must currently be on an active call. If the call partner is waiting, the person making the call waiting call hears the busy signal and the call is ended.

Configuration

In the Softphone

- ⚙ > **Audio** > **More functions** > **Call waiting**

on the PBX

- **User** / **Subscriber** > **Phone numbers** > **Reachability**

Further help under More functions

Process

1. A call waiting call is signalled during a conversation.

   First, the call waiting tone is switched on automatically. You hear a ringtone. If you switch the ringtone off, the call is only displayed in the call window. 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.

  The softphone 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.

Silent call waiting

Call window >

The call waiting call continues, but the call is only shown visually on the call window.

Note: When the option has been selected, it cannot be cancelled for this call.

Accepting a call

Call window >

Rejecting a call

Call window >

Forwarding

➞ > enter phone number > Transfer ➞

or

➞ > select contact in contact list > ➞

or

➞ > select contact in contact list. Drag and drop contact into call window transfer area.

Dialled calls

Dial a Subscriber Number Manually

Enter the phone number in the search field >

or

Enter the phone number using the keypad > **Dial**

Dialling from the contacts list

Select a contact in the contact list >

Dialling from the recent calls list

In the main window, select the recent call and select an entry in the call history >

Call

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

During a call, information is displayed about it in the call window (e.g. call duration, account).

Muting a call

Active conversation

Switching off muting

Select again

Sending a DTMF signal during a call

To do this, enter digits and special characters

in the call window, using

or

on the PC keyboard.

Note: No DTMF tones can be entered using the dialpad in the main window.

Hearing the call output on loudspeaker

During the call or when it is ringing, select

Note: The audio settings in the softphone must be configured to enable loudspeaker listening mode to be used. When the function is activated, the microphone (speakerphone) and loudspeaker (speakerphone) are used.

Further help under Device settings

Switching off call output on loudspeaker

Select again

Recording a call

Active conversation

Important: Note the following legal notes prior to recording a call.

The recorded conversation contains the following components:

- The spoken word of both ends of a telecommunication connection

Depending on the configuration, the following metadata can be contained in the file name:

- ID of selected number
- Contact display name
- Subscriber number of the contact
- Name of the account
- Year
- Month:
- Day
- 23 (hour)
- Minute:

- Second:
- Part X of X of the entire recording

Before a conversation is recorded, all call participants must be informed that recording will take place and who will process the data acquired. It is absolutely necessary to store the data safely and in accordance with the 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.

Switch off call recording

Select again

Putting a call on hold

Active conversation

Reconnecting an on-hold call

Call on hold >

Ending a call

Call window >

or

Main window >

Query Calls

A query call works on the same principle as transferring a call with an announcement. It is also similar to a conference call in which there are several users/participants. One or more users are called during an active call, while the other call partners are held in the waiting position and hear the music on hold.

Initiating a query call (during a call)

Call menu >

The active call partner is placed in the waiting position.

Subsequently:

select contact for query call and drag and drop them into the call window conference call/query call area

or

Call menu > + > select contact or enter phone number >

Ending the query call and returning to the previous call

Either one of the call partners hangs up the receiver

or

ending active call with

Then get the previous caller out of the waiting position .

Splitting a call (during a query call)

Put the active call on hold on the call menu or main window .

Then get the previous caller out of the waiting position.

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 >

Connecting the two call partners to each other

Options:

- Initiating a conference: You are taking part in the call.
- Blind transfer: Your call is ended.

Initiating a conference call

Description of the operation under Conference call

Transferring a call without an announcement

Description of the operation under Transferring calls

Transferring calls

A call partner can be transferred, i.e. connected, to an internal or external user.

A call can be transferred as follows:

- Transferring a call without an announcement
- Transfer the call with an announcement

Transferring a call without an announcement

If a call partner is transferred to another user, the call from the person who is transferring the call is ended immediately. If the user accepts the call, the connection is established.

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

Active conversation:

Drag and drop a contact into the call window transfer area.

or

➞ Dial a phone number > ➞

or

➞ > select a contact in the contact list

When the call is ringing:

Drag user from the contact list and drop into the call window transfer area.

Transfer the call with an announcement

To transfer a call with an announcement, the active call partner is placed in the waiting position, either manually or automatically by dialling a phone number. This enables the person who is transferring the call to announce it.

Requirements:

- Active conversation

Manual call transfer:

1. **Call menu** > ❚❚

   The active call partner is placed in the waiting position.

2. Call menu > ➞
3. Select the phone number or contact to which the call is to be transferred.

Automatic call transfer:

1. **Call menu** > ➞

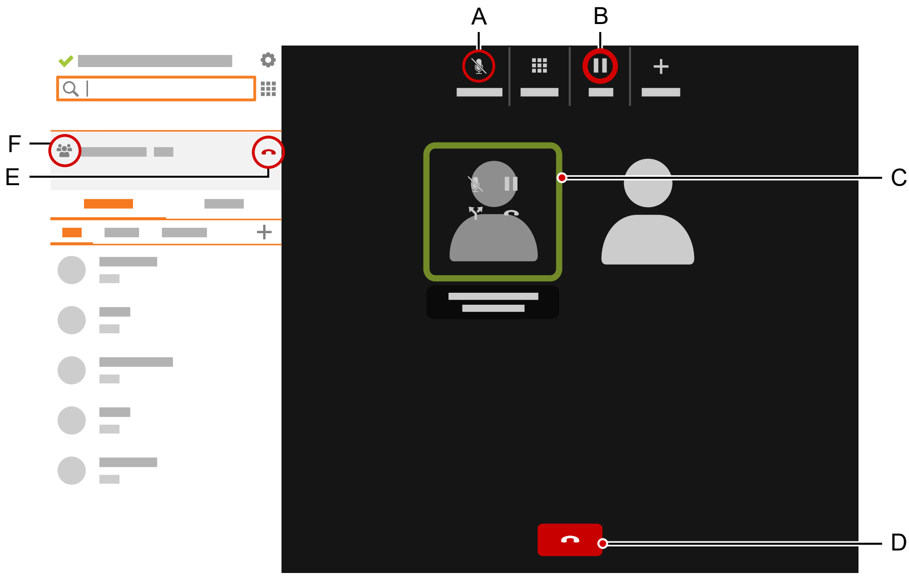
   The active call partner is automatically placed in the waiting position.

2. Select the phone number or contact to which the call is to be transferred.

Conference call

During a conference call, you are talking with two or more participants at once.

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



A. Mute conference call

B. Hold conference call

C. Conference call participant selected. The options in the avatar only apply to the particular user.

   Muting

   Place in waiting position

   Splitting the conference call

End call. Then, there will only be a connection with the remaining conference call participants.

D. end conference

E. end conference

F. An existing conference call

Initiating a conference call

There are two ways of initiating a conference call:

Active conversation:

Drag user from the contact list and drop in the call window conference call/ query call area

or

add a non-configured contact via + > enter phone number in the search field >

When the call is ringing

Drag user from the contact list and drop in the call window conference call area.

The conference call has been successfully created if all participants have taken the call.

Splitting a call

Select the participant you want to put on hold > .

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

Adding conference call participants

Drag user from the contact list and drop in the call window conference call/ query call area

or

add a non-configured contact via + > enter phone number in the search field >

Splitting the conference call

Call window >

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

Connecting call participant(s)

If you would like to leave a conference call, but the conference call participants would like to continue talking with each other, you can connect the two parties with each other. To do so, split the conference call and then transfer them without announcement.

Description of the operation under Transferring calls

Holding a conference call and initiating an additional call

Call menu > II

The conference call participants are put on hold and hear the "music on hold". Another call can be initiated.

Description of the operation under Dialled calls

When that call ends, the conference call can be continued by selecting again .

Ending a call with a conference participant

Select the subscriber for whom you want to close the connection > .

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

Ending a conference call completely

Call window >

or

Main window >

Hands-Free Calling

During hands-free calling, either the loudspeaker and microphone on the PC or the connected USB device (e.g. conference telephone) is used.

: When you click on the loudspeaker in the Call menu, the settings are applied for **Input device (Speaker)** and **Output device (Speaker)**.

Further help under Device settings

Accepting a call in hands-free calling mode

>

Initiating a call in hands-free calling mode

Select a contact from the contact list > >

or

Dial a phone number > >

Ending a call in hands-free calling mode

Call window:

or

Main window:

Speakerphone

In Loudspeaker listening mode, the loudspeaker is switched on. This enables other people in the room to listen in on the call.

Further help under Device settings

Switching on Speakerphone during a Call

Call menu >

Switching off Speakerphone

Call menu >

Ending a Speakerphone call

Call window >

or

Main window >

Caller list

To open the call history

Main window > **History** > **Calls**

Adding an entry in the call history to the contacts

Select list entry > +

Further help under [Main window]

Deleting entries from the call history

Main window > **History** > **Calls** > > **Delete individual items**> select an entry and confirm with Delete.

If necessary, delere more entries > **Done deleting**

Deleting an entire call history

Main window > **History** > **Calls** > > Confirm deleting

Calling from the opened call list

Select list entry >

Instant Messaging

Instant Messaging

Instant Messaging

Configuration

If Instant Messaging is not yet available in the conversation window, proceed as follows:

1. In the conversation window select **Set one here**.
2. Select **Add device type**.
3. Set your account under **Presence account**.

Alternatively, you can set up your Instant Messaging account as follows:

- When adding a contact also via the field **Presence account**
- Via > **Contacts** > **COMfortel SoftPhone Contact Service** > **Use this account for presence**.

Writing a text message

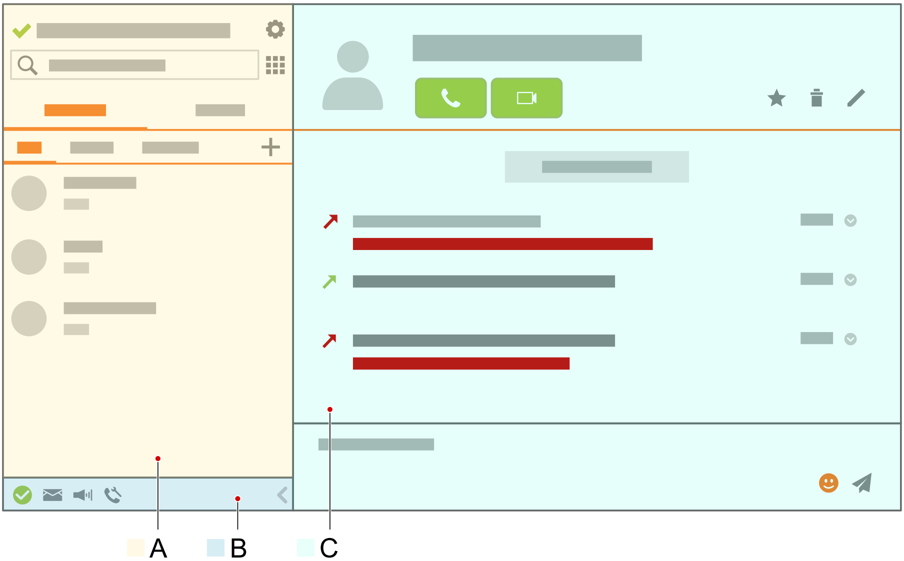
At the bottom of the conversation window, type a text message for the current contact. Send the message with the Enter key or with .

User Interface

Basic Knowledge

Overview

The User Interface is divided into several areas:



A. **Main window**

- Account state
- Keypad
- Search field
- Contacts

Further help under Main window

B. **Status bar**

- Presence change
- Voice messages

- Error messages
- Audio
- Call settings

Further help under Status bar

C. **Call window**

- Telephoning
- Contact settings
- Call information
- Instant Messaging

Further help under Call window

Main window

A. Account/Account state

   Registered

   Is registered

   Not registered

B. Search field

C. Contacts and recent calls display selection options

D. Contact display selection options

E. Avatar of the contact you have created

F. Subscriber number of the contact

G. Contact display name

H. Add/Import contacts

I. Keypad for entering phone numbers directly

Note: You cannot enter DTMF sounds here. Use the keypad in the Call menu.

J. Settings

Add Contact

The +

Each contact must have at least one phone number and at least one associated entry in the Surname, Display name or Company entry fields. Any number of phone numbers can be added per contact.

Entering a device type with associated phone number:**Add device type** > **Device type**

The Display name is used for the display in the main and call window. If you have not entered a name in the Display name entry field, the name is automatically generated from the entries in the First Name and Last Name fields. If no data is present either in the First name or Last Name entry field, the display name is generated from the entry in the Company entry field.

Import contacts

Main window > +> Import contacts

Select the required CSV file and set the encoding and separator (usually UTF-8 and semicolon, i.e. ;).

Analyze file: Assign the columns in the CSV file to the Contact fields and generate the contacts.

Note: If they cannot be generated, adjust the encoding and separator, if necessary, and/or modify the CSV file.

Status bar

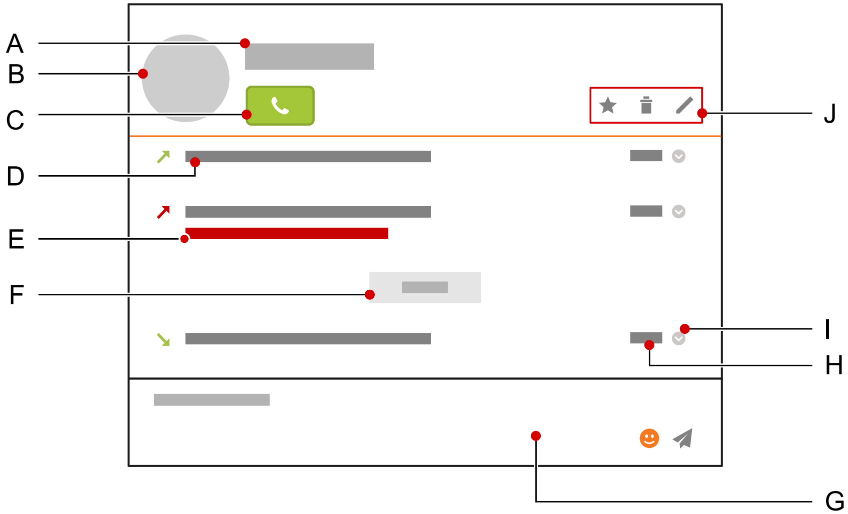
The status bar is on the bottom edge of the main window.

Presence change	Online Invisible Away Busy Offline Lunch break On the phone Briefly absent

Note: The presence status can be selected, but it currently cannot be output at the destination.

Voicemail	Calls the voice mailbox whose extension has been specified.
Error messages	Is displayed if there is an error. If the icon is selected, the Extended Settings are displayed, with additional details about the error that has occurred.
Audio	Opens the volume settings for the default loudspeaker and default microphone.
Call settings	Opens the call settings in the Settings menu.

Call window



A. Destination display name

B. Destination avatar

C. Call button. A selection list is displayed if more than one phone number has been entered for a contact.

D. Call type/call information

Successful outgoing call

Unsuccessful outgoing call

Accepted call

Rejected incoming call

Rejected outgoing call

Missed call

Successful transferred call

E. Error code for an unsuccessful call
F. Date
G. Instant Messaging
H. Time
I. Other call details (account, phone number, duration)
J. Setting options

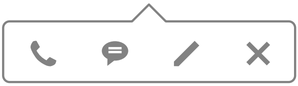
Adds the selected contact to the Favourites.

Deletes the history or the recent call and contact.

The selected contact can be edited.

Compact mode

A menu bar is displayed when compact mode is activated, and you select a contact:



Initiates a voice call. If more than one phone number has been entered for a contact, a selection list is displayed.

Note: Only a limited number of phone numbers can be displayed in compact mode.

Opens the call window for Instant Messaging.

Opens the window in which you edit a saved contact.

Closes the menu bar

Note: When entering a text message or making a call, the conversation window opens. Then to return to compact mode, select in the status bar .

Call menu

A menu with additional options (orange: activated option; grey: deactivated option) is displayed during an active call.

Muting

Switch on speaker

Display keypad. DTMF sounds can be entered using it.

Displays the call or audio quality statistics

Records the call

Stops the call

Provides options for call forwarding

Starts a conference call

Settings

Information about the section

There are two ways of opening the Settings menu in the softphone:

- On the right of the main window, next to the account name: select
- On the bottom edge of the main window: select

Note: You can search for individual parameters in the Settings area using the key combination **Ctrl + F**.

Provisioned Settings

Provisioned Settings: Accounts

The following parameters are provisioned and overwritten if they are changed manually.

SIP Credentials

- **Domain**
- **User name**
- **Password**

Further help under SIP Credentials

Optional SIP credentials

- **Outbound proxy**

Further help under Optional SIP credentials

Functions

- **MWI for incoming voice messages**
- **Register on startup**
- **Subscribe presence**
- **Publish presence**
- **Use BLF**

Further help under Functions

Phone number modification

- **Remove dial chars**
- **Use country code**
- **Default country for numbers without country code**

Further help under Phone number modification

Compatibility modes

- **DTMF mode**

Further help under Compatibility modes

Network

- **Transport**
- **Keep alive time-out**
- **RPort**
- **RPort media**
- **STUN**

Further help under Network

Audio codecs

- **Audio codecs**

Further help under Audio codecs

Encryption

- **SRTP key negotiation**

Further help under Encryption

Provisioned Settings: Contacts

The following parameters are provisioned and overwritten if they are changed manually.

Windows Contact Service

Access data

- **Enable**

Further help under Access data

Provisioned Settings: Settings

The following parameters are provisioned and overwritten if they are changed manually.

Behaviour

Behaviour

- **Minimise to tray**
- **Minimise on close**

Further help under Behaviour

Automation

General

- **Start COMfortel SoftPhone with the operating system**

Further help under General

Call events

- **Enable call recording for all calls**

Further help under Call events

Advanced

Global STUN

- **STUN**
- **STUN server**
- **STUN port**

Further help under Global STUN

TLS

- **Override domain name**
- **Load domain certificate**
- **Use only strong ciphers**
- **Protocol suite**

Further help under TLS

Accounts

Information about the section

The existing accounts are listed in the main window (left).

A symbol before an account shows its current status.

Further help under Main window

If you select an account, its settings are displayed in the right-hand window.

SIP accounts

SIP Credentials

Domain

The SIP registrar server hostname or IP Address. The domain can also be called a proxy or registrar.

Examples:

- `sip.example.com`
- `192.168.0.240`

A port can be added. Add it at the end of the domain:`port`

Examples:

- `sip.example.com:5060`
- `192.168.0.240:5060`

Further help under Provisioned Settings: Accounts

User name

Freely selectable name. For authentication against the SIP registrar. This can be a name or the phone number, for example.

Further help under Provisioned Settings: Accounts

Password

Freely selectable password. For authentication against the SIP registrar.

Note: Password that consists of at least 8 characters containing digits, letters (upper and lower case, but no German umlauts and ß) and special characters - _ . ! ~ * ' ( ) & = + $,.

Further help under Provisioned Settings: Accounts

Optional SIP credentials

Authorised username

Only fill this field if necessary for configuration on the PBX.

Outbound proxy

Enter use of the outbound proxy as shown in the entry field below.

Only enter the proxy if the PBX requires it.

Further help under Provisioned Settings: Accounts

Functions

Caller ID

Caller ID to be displayed for outgoing calls. Note that the PBX generally ignores or overwrites this setting.

MWI for incoming voice messages

Controls whether and when the softphone logs in to the server using MWI.

If the PBX supports it, the user is informed about new voicemail messages.

Further help under Provisioned Settings: Accounts

Register on startup

The softphone is automatically registered on startup.

Registration is usually necessary so that incoming calls and, in some configurations, also outgoing calls, can be made.

Note: You can also register the smartphone manually, on this page, if required. Select **Deregister** / **Login**.

Further help under Provisioned Settings: Accounts

Subscribe presence

Displays the presence status of your contacts (e.g. **Online** , **Busy**).

Publish presence

Sends the presence status to the server.

Use BLF

Enables the Busy Lamp Field if the PBX supports this function.

Further help under Provisioned Settings: Accounts

Use custom ringtone

Selects a specific ringtone for calls received on this account.

Do not play ringback tones

Disables the destination ringing tone for outgoing calls, except if the ringing tone is sent via EarlyMedia.

Enable typing notification

If the function is enabled, you can see when your chat partner is currently writing a text message.

Phone number modification

Remove dial chars

Characters entered here are removed from the phone number before dialling.

Further help under Provisioned Settings: Accounts

Use country code

Use a default country code for phone numbers that have been saved without a prefix.

A prefix can also be used, e.g. for international phone numbers.

Further help under Provisioned Settings: Accounts

Default country for numbers without country code

Select the required country for the default country code.

Further help under Provisioned Settings: Accounts

Prefix

Enter a specific number (if necessary), to set a prefix before the phone number, as a country code, e.g. `00` for most European countries.

This field can also be used for a particular prefix, e.g. `9`for outgoing calls or an outgoing extension.

Preconfigured Extensions

Check for voicemail

Voice mailbox phone number.

The softphone uses this extension, assigned on the PBX, to listen to voicemail messages.

This extension is dialled in the status bar when selecting the account after selecting .

Transfer to voicemail

Voice mailbox phone number that the softphone uses to leave voicemail messages.

This extension is used to forward incoming calls to the voice mailbox. To do this, select first in the Call menu when there is an incoming call → and then in the status bar .

Compatibility modes

DTMF mode

Setting controls how the softphone sends DTMF signals to the PBX. These are used to control IVRs (Interactive Voice Responses).

Example of use of an IVR at start of a phone call: For Sales, press 1, for Service, press 2 etc.

This setting must match the configuration on the PBX.

- COMtrexx: **Exchange lines** > **Providers and Accounts** > **Provider** > **RTP** > **DTMF signalling**

Further help under Provisioned Settings: Accounts

RTCP Feedback

The RTCP protocol is used to negotiate and store quality of service (QoS) parameters.

Send KPML (Cisco Unified Communications Manager)

Uses Keypad Markup Language (KPML). This is mainly used in combination with the Cisco Call Manager and replaces the other DTMF functions.

Force RFC 3264 hold (Cisco Unified Communications Manager)

Forces a change in the Hold and Unhold packets and fixes some incorrect implementations that do not output the supported Hold methods correctly.

Use this setting if you have problems putting a call on hold and are certain that NAT is not the cause.

Cisco Call forwarding (Cisco Unified Communications Manager)

Enables Cisco's call forwarding on the server.

This option only works if you configure the telephone type as a Cisco softphone, instead of the standard third party SIP softphone, in the Cisco Call Manager.

Network

Transport

The protocol for sending and receiving SIP packages.

Note: This setting is provisioned. No adjustment usually necessary here.

UDP

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

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.

TLS

(Transport Layer Security). Encrypts data and transports it securely. TLS is primarily used to protect http connections e.g. for commercial transactions (https).

Further help under Provisioned Settings: Accounts

Registration expiry mode

The maximum time between sequential registrations on the server. Here, the softphone registers 10% sooner than configured, to take into account network delays and new transmissions.

Note: The registration process is used to inform the server of the PBX's location, to ensure that incoming calls are received correctly. Registration might be required for outgoing calls from the server.

Subscription expiry

Enter the maximum time between sequential registrations.

Subscription expiry mode

After the set time, the softphone automatically sends a new log-off request to the server.

Subscription expiry

Sets the time after which the softphone sends a new log-off request to the server.

Keep alive time-out

After the set time, the softphone automatically sends a new log-off request to the server, to maintain the connection with the server.

Note: If the server does not receive a request, it is in an idling state and maintains the connection until the specified time has passed. Specify whether and for how long the server is to wait for a new request, here.

Note: Ensure that the NAT (firewall, router) has the correct port assignment for TCP and UDP connections.

Subscription expiry

Sets the time after which the softphone sends a new request to the server.

Further help under Provisioned Settings: Accounts

RPort

NAT-processing on the basis of RFC 3881. Recommended for TCP and TLS configurations.

Further help under Provisioned Settings: Accounts

RPort media

For NAT-related missing audio signals, for some defective implementations (e.g. if the client is behind a symmetrical NAT combined with a CUCM server).

Note: Only select this option if you really have to, for fault resolution.

Further help under Provisioned Settings: Accounts

STUN

Custom STUN

Activates the use of the values entered in the following fields for the STUN server.

Global STUN

These global settings are used:**Settings** > **Functions** > **Advanced** > **Global STUN**. Consequently, the values entered here, in the account, are not applied.

Further help under Provisioned Settings: Accounts

STUN server

The STUN server that is to be used.

The following address should be used, as far as possible:
`stun.auerproxy.de`.

Further help under Provisioned Settings: Accounts

STUN port

The STUN port that is to be used.

This port should be used, as far as possible:`3478`.

Further help under Provisioned Settings: Accounts

STUN refresh period

The interval at which the connection to the STUN server is refreshed.

Further help under Provisioned Settings: Accounts

Audio codecs

Select one or more codecs for use with this account.

Note: Only codecs that are supported on the PBX and the VoIP operating mode set on it can be selected. First check what codecs are present on the PBX.

- COMtrexx: **Administration** > **VoIP** > **VoIP configuration** > **Presetting of the VoIP channels**

Only codecs listed in the relevant operating mode can be used which are listed under **codecs internal**.

Available codecs

Shows unselected codecs.

To select a codec: select it and then press . The codec is added to the list:**Selected codecs**.

Selected codecs

Changing a codec's priority (the higher in the list, the higher the priority):. You can also drag and drop the entries to move them.

To remove a codec from this list: select the relevant codec and then press . The codec is added to the list:**Available codecs**.

Further help under [Provisioned Settings: Accounts]

Encryption

Certificate file

Certificate file exported from PBX. Always use the **PBX certificate**, not the **Root certificate**.

You will find the certificate on the PBX's as follows: **Administration** > **Certificate** > **SIPS/SRTP internal**.

Use certificate as

Do not use

The certificate set here will not be used.

Default

The certificate is used that may be set under **Settings** > **Functions** > **Advanced** > **TLS**.

Note: Do not use this option here. Do not store a certificate on the specified path.

Use certificate

The certificate set here will be used.

Generate self-signed certificate

A self-signed certificate will be created and used.

SRTP key negotiation

Select if encryption is to be performed using the protocol **SDES**.

Further help under [Provisioned Settings: Accounts]

Enable ZRTP

Note: This transport protocol is not supported by the PBXs.

Contacts

Information about the section

The main window (left) contains a list of the existing contact services for integrating contacts.

When you select a contact service, its settings are displayed in the right-hand window.

To add contact services, select the button **Add**.

Overview

Add

Adds a contact service.

Rename

For renaming individual contact services.

To do so, click in the title and enter the new title.

Note: This function is not available for the contact services **Auerswald** and **Windows**.

Delete

Deletes the contact service.

Note: This function is not available for the contact services **Auerswald** and **Windows**.

COMfortel SoftPhone

Access data

Filter

Access data

Use this account for outgoing calls

Specifies the account used for outgoing calls.

Use this account for presence

Note: The presence status can be selected, but it currently cannot be output at the destination.

Filter

Hide contacts without name

Contacts without a name are not displayed.

Hide contacts without phone

Contacts without a phone number are not displayed.

Windows

Access data

Enable

Enable the use of contacts saved in Windows.

Further help under Provisioned Settings: Contacts

Use this account for outgoing calls

Specifies the account used for outgoing calls.

Use this account for presence

Note: The presence status can be selected, but it currently cannot be output at the destination.

Filter

Hide contacts without name

Contacts without a name are not displayed.

Hide contacts without phone

Contacts without a phone number are not displayed.

LDAP

Access data

Enable

Enable the use of LDAP contacts.

Authentication type

Specifies how LDAP contacts can be accessed.

Note: Auerswald/FONtevo PBXs currently only support the option **Simple**.

Use TLS

The TLS certificate is used for secure data transfer.

Ignore TLS certificate

Disables the request for a TLS certificate.

Data is still encrypted before transfer. However, no TLS certificates are requested.

HOST

IP address of the LDAP server used here. This is usually the PBX's IP address.

User name

Name for logging in to the LDAP server on the PBX. It contains your phone number:**telephoneNumber=user phone number, dc=auerswald**.

You will find the user name on the PBX as follows:

- COMtrexx: **User** > **Phone numbers** > **Configure**

If contact groups have been created on the PBX and one needs to be transferred into the softphone, the User name then contains the name of the relevant contact group.

You will find the contact group name on the PBX as follows:

- COMtrexx: **User data** > **Contacts and assignment** > **Settings**

Password

Password for logging in to the LDAP Server on the PBX.

You will find the password on the PBX as follows:

- COMtrexx: **User** > **Phone numbers** > **Configure**

You will find the password for contact groups on the PBX as follows:

- COMtrexx: **User data** > **Contacts and assignment** > **Settings**

DC

LDAP server domain components.

Note: The setting for Auerswald/FONtevo PBXs is always **dc=auerswald**. Do not change it.

Limit the number of search results to

Specifies the maximum number of search results to be displayed.

Note: The maximum number of search results to be displayed is restricted to 50 on the PBX. If the number of search results exceeds that maximum value, the last entries are not displayed. Restrict the search if that happens.

Use this account for outgoing calls

Specifies the account used for outgoing calls.

Use this account for presence

Note: The presence status can be selected, but it currently cannot be output at the destination.

Filter

Hide contacts without name

Contacts without a name are not displayed.

Hide contacts without phone

Contacts without a phone number are not displayed.

Attribute

Attribute

Select the attributes displayed for contacts.

Select the attributes that are to be displayed in your contact list.

Note: The right-hand column contains the designations of the individual attributes in the LDAP server. Do not make any changes there.

Note: The Instant Messaging with LDAP contacts function is supported with the following setting:**Attribute** > **Work IP Phone**: **telephoneNumber**.

Outlook

Access data

Enable

Enables the use of Outlook Contacts.

Outlook Profile

Name of the profile created in Outlook, whose Contacts are to be transferred.

Outlook Profile Password

Password assigned in Outlook for the profile whose Contacts are to be transferred.

Note: The password is not the same as your e-mail password.

Note: Only enter the password if the system administrator has asked you to do so.

Use this account for outgoing calls

Specifies the account used for outgoing calls.

Use this account for presence

Note: The presence status can be selected, but it currently cannot be output at the destination.

Filter

Hide contacts without name

Contacts without a name are not displayed.

Hide contacts without phone

Contacts without a phone number are not displayed.

Attribute

Attribute

Select the attributes displayed in the contact list.

In the right-hand column, select the appropriate designation for the attributes present in the Outlook Address Book.

Google

Access data

Enable

Enables the use of Google Contacts.

Generate token

Generate new token via **Generate**.

The token is entered in Google and in the softphone once. Follow the instructions in the dialogue window.

Note: If access to the Google Contacts is denied, check that the token entered in Google and the softphone match each other.

Use this account for outgoing calls

Specifies the account used for outgoing calls.

Use this account for presence

Note: The presence status can be selected, but it currently cannot be output at the destination.

Filter

Hide contacts without name

Contacts without a name are not displayed.

Hide contacts without phone

Contacts without a phone number are not displayed.

XML

Access data

Filter

Access data

Enable

Enables the use of XML contacts.

Local path/URL

Local path or URL of the XML file that contains the contacts.

Authentication type

Specifies how XML contacts can be accessed.

User name

Name for identification to the XML contact source.

The user name is specified when the contact source is entered, e.g. when the local XML file is created.

Note: The user name need not necessarily be assigned when the XML source is created. However, it does protect against unauthorised third parties accessing the data.

Password

Password for identification to the XML contact source.

The password is specified when the contact source is entered, e.g. when the local XML file is created.

Note: The password need not necessarily be assigned when the XML source is created. However, it does protect against unauthorised third parties accessing the data.

Use this account for outgoing calls

Specifies the account used for outgoing calls.

Use this account for presence

Note: The presence status can be selected, but it currently cannot be output at the destination.

Filter

Hide contacts without name

Contacts without a name are not displayed.

Hide contacts without phone

Contacts without a phone number are not displayed.

Media

Audio

Audio

Device settings

Input device (default)

Select a microphone in the list, to be the default device.

Output device (default)

Select a loudspeaker in the list, to be the default device.

Input device (Speaker)

Select a microphone in the list for use if "hands-free calling" mode is selected.

Output device (Speaker)

Select a loudspeaker in the list for use if "hands-free calling" mode is selected.

Ringing Device

Select a loudspeaker from which the softphone rings.

Automatic microphone selection

The softphone selects the microphone automatically.

Select this option if, for example, you are using a portable device for a conference call and would like to connect a USB headset during this time, to continue the conference call.

Echo cancellation

The echo that the other calling party might hear is reduced or removed.

We recommend you always leave this function activated unless you are using a desk telephone with integrated echo cancellation.

Note: It is technically impossible to cancel the echo coming from the other end. If you hear an echo when using the softphone, the other calling party must enable echo cancellation, if necessary.

Automatic gain control

Automatically regulates the microphone level, to ensure your voice is always at the correct volume, no matter which microphone you use or how far you are from the microphone.

We recommend you always leave this function activated.

Noise suppression

Removes disruptive background noise during the phone call.

More functions

EarlyMedia Call mute (outgoing calls)

Mutes the EarlyMedia function.

During the call phase, audio, e.g. the call tone, signal tone, an announcement or voice, can be transferred (EarlyMedia). This includes transferred ringing tone sounds or announcement texts such as "This number is not in use."

Whether EarlyMedia is supported depends on the provider. It is possible that the provider allows the negotiation but suppresses the audio transmission.

Call waiting

Call waiting signals incoming calls during a call.

A call is signalled by

- ringtone
- an incoming call in the call window

If the option is disabled, the person making the call hears the busy signal.

Ring also through PC speaker

Ringtone can be heard through the PC loudspeaker as well as the headset.

This is useful for hearing the ringtone for an incoming call if a headset is normally used but is currently not being worn.

Disable DTMF sounds

The softphone outputs no sounds when the keypad is used.

This has no effect on the DTMF sounds that are sent to the server. Instead, it only prevents the local playback of the same sounds on the softphone.

Sounds

New Instant Message

A sound is output when a new message arrives. Use the default file (.*wav) or, to import one, select [...].

GUI

Appearance

Language

Select the language.

Appearance

Balloons in Instant Messaging

If a text message is received, it will be displayed in the main window with speech bubbles.

Deactivate avatars

The user avatars are hidden.

Animations

Animations are used, e.g. rapid screen change.

Compact mode

In compact mode, only the left-hand main window is displayed. The right-hand call window remains closed if a contact or an entry in the recent calls list is selected.

If this option is deselected, the call window remains open. ‹ (status bar) closes the call window temporarily.

Theme

Select a theme for the user interface.

Behaviour

Behaviour

Start minimised

The softphone is started in minimised mode.

Always on top

The softphone window is permanently displayed in the foreground, on top of all other windows.

Minimise to tray

Minimise: The softphone is moved to the notification area, as an icon, when this button is selected on the title bar.

Further help under Provisioned Settings: Settings

Minimise on close

Close: The softphone is minimised when this button is selected on the title bar.

Further help under Provisioned Settings: Settings

Collapse on hangup

If the right-hand call window is open and the call is ended, the softphone changes to compact mode. Consequently, the call window is closed and the left-hand main window also remains open.

Focus window on incoming Instant Message

The softphone window is automatically activated when an Instant Message is received.

Window flashes on incoming Instant Message

The softphone window flashes when an Instant Message is received.

Activate window on incoming call

The softphone window is automatically activated when a call is received.

Window flashes on incoming call

The softphone window flashes when a call is received.

Pop-up notifications

when a contact goes online

Notification if a contact in the contact list goes online.

for Instant Messages

Notification if a new Instant Message arrives.

for incoming calls

Notification of an incoming call.

for new voicemails

Notification if a new voicemail message arrives.

when an audio device is (dis)connected

Notification if an audio device is connected or disconnected.

when the network status has changed

Notification when the network status changes (LAN, WiFi, etc.)

Functions

Calls

Call settings

Manage incoming calls

Enables call forwarding or automatic call acceptance.

Call forwarding

Instant forwarding

The call is immediately forwarded to the destination number specified below.

Forward after (1 - 600 secs)

The call is forwarded to the destination number specified below after the specified wait time.

Transfer Destination

Destination number for call transfer.

Automatic call acceptance

Instant auto answer

The call is immediately accepted, automatically.

Call Acceptance after (1 -600 sec.)

The call is accepted automatically after the specified wait time.

Play sound on auto answer

A sound is played during automatic call acceptance.

Keep settings after restart

All call settings on this page are retained when the softphone is closed.

Headset

External Headset / Device

External Headset / Device

Sonstige Geräte anschließen

For using any standard USB device.

Jabra device

For using a device made by Jabra.

Plantronics device (32-bit only)

For using a 32-bit device made by Plantronics.

Sennheiser device

For using a device made by Sennheiser.

Automation

General

Start COMfortel SoftPhone with the operating system

The softphone is automatically started when the operating system boots.

Further help under Provisioned Settings: Settings

Check for updates

The softphone automatically searches for updates.

Integration

Register Callto, SIP and tel URIs in the operating system

If this option is selected, a setting is made in the operating system that URIs with these prefixes are executed with the softphone.

Integrate COMfortel SoftPhone in Microsoft Outlook

A phone number selected in Microsoft Outlook is dialled with the softphone.

Call events

Enable call recording for all calls

All incoming calls are automatically recorded.

Important: Note the following legal notes prior to recording a call.

The recorded conversation contains the following components:

- The spoken word of both ends of a telecommunication connection

Depending on the configuration, the following metadata can be contained in the file name:

- ID of selected number
- Contact display name
- Subscriber number of the contact
- Name of the account
- Year
- Month:
- Day
- 23 (hour)
- Minute:
- Second:
- Part X of X of the entire recording

Before a conversation is recorded, all call participants must be informed that recording will take place and who will process the data acquired. It is absolutely necessary to store the data safely and in accordance with the 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.

Further help under Provisioned Settings: Settings

Directory to store the recorded conversations

Select a local storage location for recorded conversations.

A recording can be started in the active call window: select ◉.

Important: Note the following legal notes prior to recording a call.

The recorded conversation contains the following components:

- The spoken word of both ends of a telecommunication connection

Depending on the configuration, the following metadata can be contained in the file name:

- ID of selected number
- Contact display name
- Subscriber number of the contact
- Name of the account
- Year
- Month:
- Day
- 23 (hour)
- Minute:
- Second:
- Part X of X of the entire recording

Before a conversation is recorded, all call participants must be informed that recording will take place and who will process the data acquired. It is absolutely necessary to store the data safely and in accordance with the 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.

File name for the recorded conversations

Sets the structure of the automatically generated file name.

The parameters below can be used in whatever way required: the sequence is freely definable.

Parameter	Meaning
DNID	ID of selected number
name	Contact display name
phone	Subscriber number of the contact
number	Subscriber number of the contact
Account	Name of the account
YYYY	Year
MM	Month:
DD	Day
hh	23 (hour)
NN	Minute:

Parameter	Meaning
ss	Second:
recording_part	Part X of X of the entire recording

Example:`recorded_conversation_{YYYY}-{MM}-{DD}-{HH}_{NN}_{SS}_{account}_{number}_part{recording_part}`

On transfer request mode

Always accept

While you are being called, or during an active call, the call can be taken by a different internal telephone, provided that the call pick-up has been permitted for the user/subscriber involved, on the PBX.

Setting in the PBX under

- COMtrexx: **User** > **Configure** > **Settings** > **Call pick-up**

Always reject

In general, incoming calls or active calls cannot be taken over by a different internal telephone.

InterCom

If this function is activated, a bi-directional connection is created between the InterCom source and InterCom destination.

An incoming call is accepted automatically. No ringing or active call acceptance.

To initiate an outgoing InterCom call, a prefix for international calls is set before the phone number. You can find this in the documentation of the PBX under **Short reference** > **InterCom**.

Prerequisite: InterCom permission must have been granted on the PBX:

- COMtrexx: **User** > **Configure** > **Settings** > **Intercom permission**

Auto reject calls if status is set to

Incoming calls are automatically rejected when the selected presence status occurs.

Event rules

Add rule

You can define any number of event rules here to suit your requirements and application scenarios.

Edit Open URL Rule

Event

Select, to set the following options for this.

Call state changes to

Select a status. Different options are available depending on the previously selected event.

Call direction

Select the applicable call direction for the selected event **Call status change**.

Do action

Open URL

Opens the URL entered in the field below.

Open/Execute Application

Runs the executable program file entered in the field below (e.g. *.exe) and opens the relevant program.

REST API

Opens the URL for a REST API, entered in the field below, and so runs that service.

Address

Depending on which action has been selected in the previous field, you must enter a URL or an executable program here.

The following parameters can also be used.

Parameter	Meaning
DNID	ID of selected number
name	Contact display name
phone	Subscriber number of the contact
number	Subscriber number of the contact
Account	Name of the account

Example: `www.auerswald.uk_name.`.

This Internet page is called when there is an incoming or outgoing call from this saved contact.

Filter

As far as possible, enter using a regular expression (regex).

Provisioning

During provisioning, the phone is automatically configured via a provisioning server. 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.

User name

User name assigned when the SIP account was configured. Used for identification to the provisioning server.

Note: You can find the username under **Settings** > **Accounts** > **[Own accounts]** > **SIP Credentials**.

Password

Password assigned when the SIP account was configured. Used for identification to the provisioning server.

Note: You can find the password under **Settings** > **Accounts** > **[Own accounts]** > **SIP Credentials**.

Remember me

The softphone remains logged on for provisioning with the user name and password registered here.

Registering automatically

Each time the program starts, the softphone is registered for this function and provisioning is carried out.

Note: If, for example, the user name or other SIP credentials change while the softphone is in use, the softphone must be restarted. Provisioning is run again.

Advanced

Global STUN

STUN

Activates the use of the values entered in the following fields for the STUN server.

Note: You can also enable these settings in the Account settings under **Settings** > **Account** > **Advanced**. Then, the global values entered here are used instead of the account-specific values.

Further help under Provisioned Settings: Settings

STUN server

The STUN server that is to be used.

The following address should be used, as far as possible: `stun.auerproxy.de`.

Further help under Provisioned Settings: Settings

STUN port

The STUN port that is to be used.

This port should be used, as far as possible:`3478`.

Further help under Provisioned Settings: Settings

STUN refresh period

The interval at which the connection to the STUN server is refreshed.

TLS

Override domain name

A domain name entered here overwrites the domain name in the certificate specified in the field below.

Caution: A TLS initialisation error is triggered if a domain name is entered in this field, but the field below does not contain a certificate.

Further help under Provisioned Settings: Settings

Load domain certificate

Note: Do not import a certificate here. The PBX certificate is set under **Settings** > **Accounts** > **SIP** > **Advanced** > **Encryption**.

Further help under Provisioned Settings: Settings

Use only strong ciphers

Only the secure encryption algorithms within each TLS suite will be used for the encryption and secure transmission of data.

Further help under Provisioned Settings: Settings

Disable certificate verification

Disables the verification of the certificate specified above.

Caution: Selecting this option creates a high security risk for encrypting and securely transmitting data and must be avoided if at all possible.

Protocol suite

Select a protocol suite. The correct setting for integrating the Softphone into the Auerswald COMtrexx PBX is**TLS v1.2**.

Caution: SSL v2/3 (insecure) presents a security risk and must be avoided if at all possible.

Further help under Provisioned Settings: Settings

Network

SIP options:

Port

Local system port for SIP transfer.

Open random port above 32000

Select this option if there is a problem with the router.

RTP options:

Port

Local system port for RTP transfer. Specially used for transferring audio data.

Open random port above 32000

Select this option if there is a problem with the router.

Glossary

Technical Terms and Functions

Account

Authorisation to access a VoIP provider or a VoIP PBX. Users must identify themselves by logging in with a username and a password.

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

A (digital) certificate is similar to a digital passport that serves to identify the owner of the certificate. Its authenticity can be verified by cryptographic procedures. The certificate thus confirms the identity of a person or a company by means of an asymmetric key, thus ensuring that the person or company really is the person/company it claims to be. Certificates are issued by an officicial institution, the Certification Authority (CA).

Certificate authorities

A certificate authority is an organisation that issues digital certificates. A digital certificate serves to assign a specific public key to a person or organisation. The certificate authority certifies this assignment by adding its own digital signature.

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 quality of VoIP calls depends 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.

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.g. “auerswald“) and top-level domain (e.g. the country code “uk“). The domain is part of the URL.

Driver

Programme or software module that controls the interaction between an operating system or programme and the interfaces to connected devices (keyboard, printer, monitor, etc.) or virtual devices.

End device

Device that can be operated on a communication network or on a PBX, e.g. phone, fax machine, answering machine etc.

Ethernet

Network system with a speed of 10/100/1000 Mbit/s 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.

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 proxy

Proxy for the HTTP protocol, over which Internet sites are called or files downloaded.

HTTP

Hypertext Transfer Protocol Secure – communication protocol for tap-proof data transfer with encryption within the World Wide Web.

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 are bi-directional, so that a person who is close by can use the intercom created to talk to the caller.

IP address

Unique numerical address within a TCP/IP network that is assigned to one device and ensures that data packages reach the correct recipient.

IPv4

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 greater coding, IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

LAN

Local Area Network - connects computers over company or home networks.

LDAP

Lightweight Directory Access Protocol) - enables you to query the data in a directory service (a hierarchical database) over the network.

MWI

Message Waiting Indication - a SIP function that signals newly received voicemail messages on internal VoIP end devices. The notification can be made for all voice mailboxes for which the relevant user has been entered.

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.

Network

System of more than one computer and other communication devices This enables multiple users to access common resources such as files, printers, etc.

Outbound proxy

Intermediate server that processes all VoIP requests and connections going to the provider (except registration).

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 use the standard port numbers (for example, web browsers use port 80).

Provisioning

Automatded configuration of one or more telephone(s) from a provisioning server.

Proxy server

Acts as the interface to the Internet or between two networks. A proxy server receives queries from a computer in an Intranet, for example. It then connects to the Internet (on behalf of the computer) to forward the queries and return the (if necessary, filtered) responses. To achieve this, the HTTP proxy uses the Internet's HTTP protocol.

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.

REST API

Representational State Transfer (REST)/Application Programming Interface (API) - a programming interface that uses HTTP queries to, for example, access information in an XML file.

This service is executed when the user enters a URL. A particular web site is then read and an XML file is opened.

REST is used for connection to cloud services and also enables interaction with them.

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

Real Time Transport Protocol – protocol for ensuring complete data transportation 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.

RTCP

Real Time Transport Protocol – protocol for negotiating and complying with quality of service (QoS) parameters.

The quality of service ensures consistently high voice quality by prioritising voice data (RTP data packets).

Server

Computer or a software that fulfils different tasks as part of the network and provides, for example, other users (clients) in this network with certain information, data and services.

SIP

Session Initiation Protocol - network protocol which provides establishing a communication session between two or more users. 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.

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.

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) - protocol used to send data packets over non-secure communication lines without a connection. This means that successful transmission is dependent on the application and is therefore not always guaranteed. UDP itself does not verify whether data has been transmitted successfully. When a UDP packet is sent, the sender cannot assume that the packet will indeed arrive at the recipient. This particular protocol needs only a small amount of additional information, and results in a better data throughput rate in a well-functioning network, e.g. on a LAN. UDP is used by DNS servers, for example.

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 Resource 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...“).

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).

Abbreviations

CD	Call deflection	call forwarding via the called person
DHCP	Dynamic Host Configuration Protocol	
DNS	Domain Name Service	
DSL	Digital Subscriber Line	Digital subscriber connection
DTMF	Dual Tone Multi Frequency	Dual Tone Multi Frequency
HTML	Hypertext Markup Language	
HTTP	Hypertext Transfer Protocol	Hypertext transmission protocol
HTTP	Hypertext Transfer Protocol Secure	Hypertext Transfer Protocol Secure
IP	Internet Protocol	Internet Protocol
IPv4	Internet Protocol verion 4	
IPv6	Internet Protocol verion 6	
LAN	Local Area Network	Local network
LDAP	Lightweight Directory Access Protocol	Protocol for easy access to directory server
MWI	Message Waiting Indication	
NAT	Network Address Translation	
PBX	Private Branch Exchange	Private PBX

REST API	Representational State Transfer/ Application Programming Interface	
RTP	Real-Time Transport Protocol	
RTCP	Real Time Control Protocol	
SIP	Session Initiation Protocol	
SIPS	Session Initiation Protocol Secure	
SMTP	Simple Mail Transfer Protocol	
SRTP	Secure Real-Time Transport Protocol	
STUN	Simple Transversal of UDP over NATs	
TCP	Transmission Control Protocol	
UDP	User Datagram Protocol	
URI	Uniform Resource Identifier	Uniform Identifier for Resources
URL	Uniform Resource Locator	Uniform Resource Pointer
USB	Universal Serial Bus	
UTC	Coordinated Universal Time	
VoIP	Voice over Internet Protocol	
XML	Extensible Markup Language	