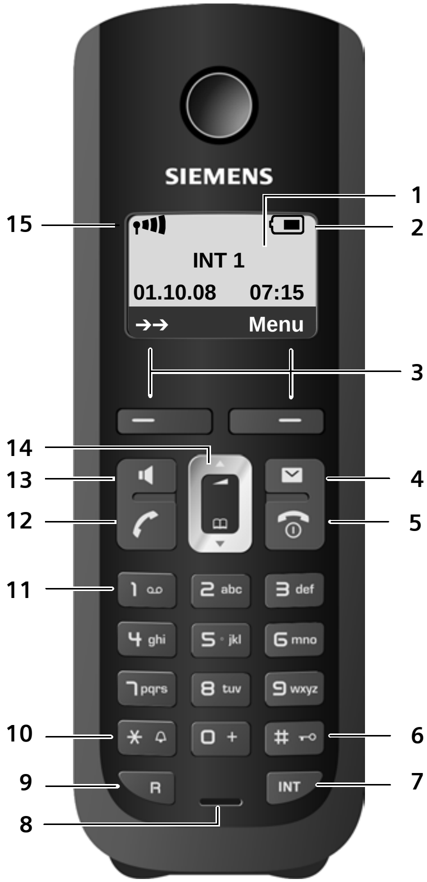
SIEMENS

Gigaset A580 IP

Gigaset

The handset at a glance



Handset keys

1 **Display** in idle status (example)
2 **Charge status of the batteries** (→ Page 28)
3 **Display keys** (→ Page 32)
4 **Message key** (→ Page 60)
Access to calls and message lists;
Flashes: new message, new call or new firmware or new provider profile available
5 **End call key, On/Off key**
End call, cancel function, go back one menu level (press briefly), back to idle status (press and hold), activate/deactivate handset (press and hold in idle status)
6 **Hash key**
Keypad lock on/off (press and hold → Page 34)
Switch between upper/lower case letters and digits for text input (→ Page 165)
7 **Internal key**
Make an internal call (→ Page 86)
8 **Microphone**
9 **Recall key**
Enter recall (press briefly → Page 133)
Fixed line network only:
Insert a dialling pause (press and hold)
10 **Star key**
Idle status:
Ringer melody on/off (press and hold)
Fixed line network: switch between dial pulsing/touch tone dialling
11 **Key 1** (press and hold)
Calling the network mailbox
12 **Talk key**
Accept call, open last number redial list (press briefly in idle status), select connection type and start dialling (press briefly/ press and hold after entering the number → Page 40)
When writing an SMS: send SMS
13 **Handsfree key**
Switch between earpiece and handsfree mode
Lights up: handsfree talking activated
Flashes: incoming call
14 **Control key** (→ Page 32)
15 **Signal strength** (→ Page 28)

Overview of display icons

Charge status of the batteries (flat to full)

(flashes)
Batteries almost empty

(flashes)
Charging

Reception signal strength between the base station and the handset (high to low)

(flashes) No reception signal between the base station and the handset

Keypad lock activated

Ringer deactivated

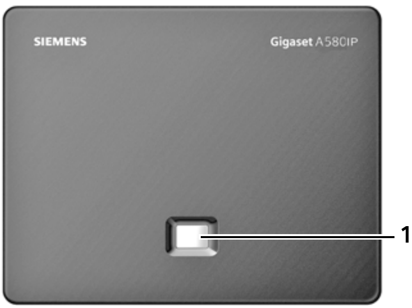
Incoming calls on the fixed line network connection (ringer icon)

Incoming call on a VoIP connection

Alarm clock set

Open last number redial list

The base station at a glance



Base station key

1 **Paging key**
Lights up:
LAN connection active (phone is connected to router)
Flashes:
data transfer to LAN connection
Press **briefly**:
Initiating paging (→ Page 85),
Displaying the IP address on the handset
Press and **hold**:
Set base station to registration mode
(→ Page 83)

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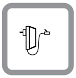
Safety precautions

Warning

Read the safety precautions and the user guide before use.

Explain their content and the potential hazards associated with using the telephone to your children.



Only use the mains adapter supplied, as indicated on the underside of the base station or charging cradle.

Only use the **recommended, rechargeable batteries** (→ Page 163)**,** i.e. never use a conventional (non-rechargeable) battery or other battery types as this could result in significant health risks and personal injury.

The operation of medical appliances may be affected. Be aware of the technical conditions in your particular environment, e.g. doctor's surgery.

Do not hold the rear of the handset to your ear when it is ringing or when the handsfree function is activated. Otherwise you risk serious and permanent damage to your hearing.

The handset may cause an unpleasant humming noise in hearing aids.

Do not install the base station or charging cradle in bathrooms or shower rooms. The handset, base station and charging cradle are not splashproof (→ Page 163).

Do not use the phone in environments with a potential explosion hazard, e.g. paint shops.

If you give your Gigaset to someone else, make sure you also give them the user guide.

Please remove faulty base stations from use or have them repaired by Siemens Service, as they could interfere with other wireless services.

Warning

When the keypad lock is active, you cannot call emergency numbers.

Please note

Not all of the functions described in this user guide are available in all countries.

Gigaset A580 IP – more than just making calls

You can use your phone to make calls and send and receive SMS messages both via the fixed line network and also (cost effectively) via the Internet (VoIP) **without a PC**. - Your phone can do much more besides:

- Make calls with brilliant sound quality (**High Definition Sound Performance HDSP** → Page 9) – for internal calls or calls via VoIP.
- **Press a button** each time you make a call to indicate whether you want to call via the fixed line network or the Internet (→ Page 40).
- Register up to **six** handsets on your base station. With your base station, you can simultaneously conduct two calls via VoIP and one call via the fixed line network.
- **Multiline:** Create up to six VoIP accounts with different VoIP providers. Together with your fixed line number and the Gigaset.net number, your phone can then be reached via up to **eight different phone numbers**.
- Assign each handset its own VoIP number as a send and receive number. If a member of your family is called on their VoIP number, only their handset will ring (→ Page 131).
- You can also use the VoIP accounts with different providers for cost control purposes. When dialling, specify the VoIP connection/the VoIP account you want to use for its lower rates (→ Page 40).
- Setting dialling plans for phone numbers or area codes enables you to automate the selection of the most cost-effective VoIP connection (→ Page 136).
- Use Gigaset.net for VoIP calls. Connect your phone to the mains power supply and the Internet, and enjoy free phone calls on Gigaset.net – without making any further settings (→ Page 48).
- Configure the phone connection for VoIP without a PC. Your phone's connection assistant downloads general data about your VoIP provider from the Internet and guides you through entering your personal data (VoIP/SIP account). This makes it easy for you to start using VoIP (→ Page 21).
- If necessary, make any further VoIP settings on a PC. The phone features a Web interface (**Web configurator**) that can be accessed via your PC's Web browser (→ Page 108).
- Make sure your phone is always up-to-date. Keep yourself informed about **firmware updates** on the Internet and download them onto your phone (→ Page 100).
- You can reduce the transmission power by activating eco mode (→ Page 81).

Your Gigaset A580 IP has a protected operating system that offers **increased security against viruses** from the Internet.

Have fun using your new phone!

VoIP – making calls via the Internet

With VoIP (**V**oice **o**ver **I**nternet **P**rotocol), your calls are not made via a fixed connection as in the telephone network, but rather they are transmitted via the Internet in the form of data packets.

You can take advantage of all the benefits of VoIP with your phone:

- You can make cost-effective calls in high voice quality with callers on the Internet, the fixed line network or the mobile phone network.
- VoIP providers will give you personal numbers, with which you can be reached from the Internet, the fixed line network and any mobile phone network.

To be able to use VoIP, you need the following:

- A broadband Internet connection (e.g. DSL) with flat rate (recommended) or volume-based price.
- Internet access, i.e. you need a router that will connect your phone to the Internet.

  You can find a list of recommended routers on the Internet at:

  www.siemens.com/gigasetcustomercare

  From here, go to the FAQ page and select "Gigaset A580 IP". Search for "Router", for example.
- Access the services of a VoIP provider. You can open up to six accounts with **different** VoIP providers.

Gigaset HDSP – telephony with brilliant sound quality

Your Gigaset IP phone supports the Broadband codec G.722. With your base station and the corresponding handset, you can thus make calls via VoIP with brilliant sound quality (High Definition Sound Performance).

If you register further broadband-capable handsets (e.g. Gigaset S67H, S68H or SL37H) with your base station, internal calls between these handsets will also be conducted via broadband. Preconditions for broadband connections to your base station are:

- **For internal calls**:

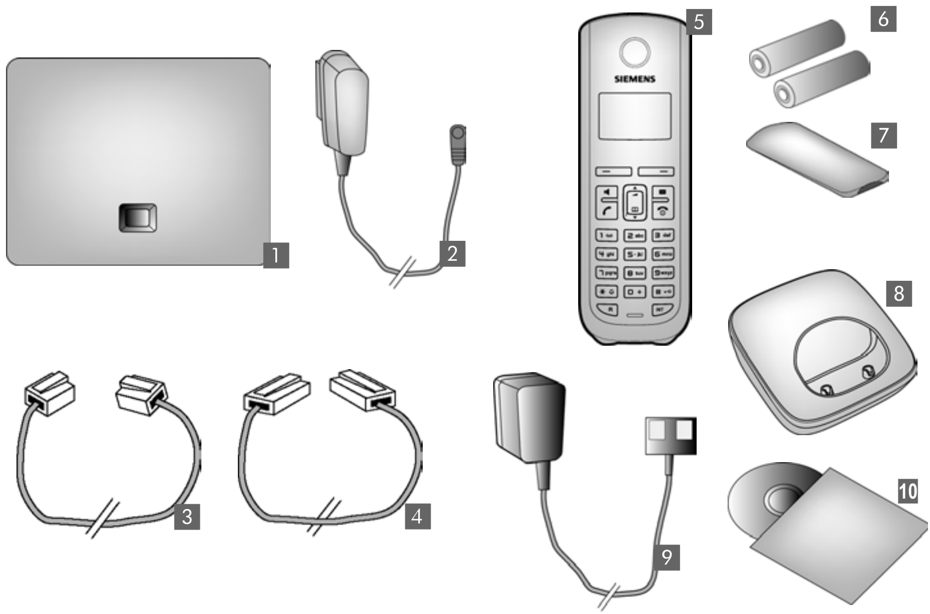
  Both handsets are broadband-capable, i.e. both support the codec G.722.
- **For external calls via VoIP:**
  - You make the call from a broadband-capable handset.
  - You have selected codec G.722 for outgoing calls → Page 127.
  - Your VoIP provider supports broadband connections.
  - The recipient's phone supports codec G.722 and accepts the establishment of a broadband connection.

Please note

The VoIP service **Gigaset.net** (→ Page 48) supports broadband connections.

First steps

Check the pack contents



1 one Gigaset A580 IP base station
2 one mains adapter for connecting the base station to the mains power supply
3 one phone cord for connecting the base station to the fixed line network
4 one Ethernet (LAN) cable for connecting the base station to the router (LAN/Internet)
5 one Gigaset A58H handset
6 two batteries for the handset (uncharged)
7 one battery compartment cover for the handset
8 one handset charging cradle
9 one mains adapter for connecting the charging cradle to the mains power supply
10 one quick guide and a CD containing this user guide

Firmware updates

This user guide describes the basic functions of the firmware version 111.

Whenever there are new or improved functions for your Gigaset A580 IP, base station firmware updates will be made available for you to download to your telephone (→ Page 100). If this results in operational changes to your phone, a new version of the this user guide or the necessary amendments will be published on the Internet at www.siemens.com/gigaset

Select "Gigaset A580 IP" in the product field to open the relevant product page where you will find a link to the user guide.

For information on how to find out the current firmware version of your base station → Page 151 (using the Web configurator) or → Page 160 (during an external call).

Setting up the handset for use

The display is protected by a plastic film.
Please remove the protective film!

Inserting the batteries and closing the battery cover

Warning

Only use rechargeable batteries recommended by Siemens Home and Office Communication Devices GmbH & Co. KG (→ Page 163). Never use a conventional (non-rechargeable) battery or other battery types as this could result in significant health risks and personal injury. For example, the batteries could explode. The phone could also malfunction or be damaged as a result of using batteries that are not of the recommended type.

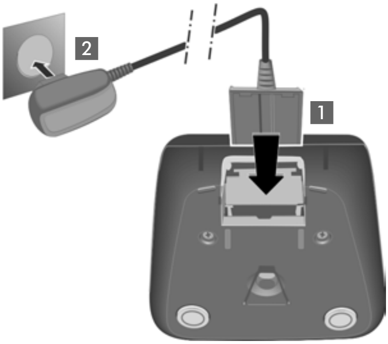
- Insert the batteries the right way round.
  The polarity is indicated in/on the battery compartment.

- First insert the battery cover at the top 1.
- Then press the cover 2 until it clicks into place.

- If you need to open the battery cover, for instance to replace the batteries, place your fingertip in the cavity on the casing and pull the battery cover upwards.

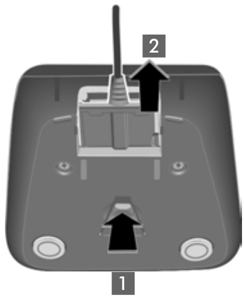
Connecting the charging cradle

The charging cradle is designed to be operated in enclosed, dry areas at temperatures ranging from +5 °C to +45 °C.



- Connect the flat plug from the power supply 1.
- Plug the mains adapter into the plug socket 2.

If you need to disconnect the plug from the charging cradle, press the release button 1 and disconnect the plug 2.



Please note

- Only place the handset in the charging cradle that is intended for it.
- If the handset has turned itself off because the batteries are flat and is then placed in the charging cradle, it will turn itself on automatically.

Should you have any questions or problems → Page 153.

For information on how to attach the charging cradle to the wall → Page 203.

Initial charging and discharging of the batteries

- Place the handset in the charging cradle and wait until the batteries are fully charged (approx. **10** hours).
  Battery charging is indicated in the top right of the display by the flashing battery icon .
- Then remove the handset from the charging cradle and do not replace it until the batteries are fully discharged.
  The charge status is displayed in the idle display.
  (flat to full) flashes, the batteries are almost flat.

After the first battery charge **and** discharge, you may place your handset in the charging cradle after every call.

Warning

- Always repeat the charging and discharging procedure if you remove the batteries from the handset and reinsert them.
- The batteries may warm up during charging.
  This is not dangerous.
- After a while, the charge capacity of the batteries will decrease for technical reasons.

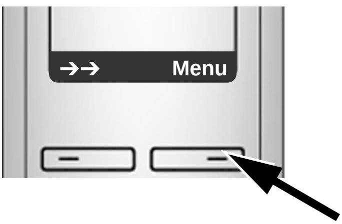
Setting the date and time

The date and time must be set in order to have the correct time for incoming calls and to be able to use the alarm clock.

Please note

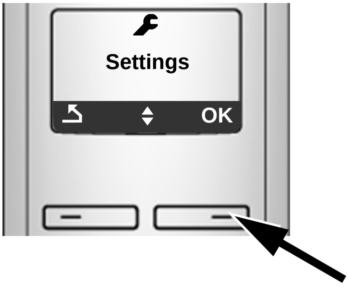
The address of a time server on the Internet is stored on your telephone. The date and time are taken from this time server provided that the base station is connected to the Internet and synchronisation with the time server is activated (→ Page 149). Manual settings are overwritten in this case.

Setting up manually:



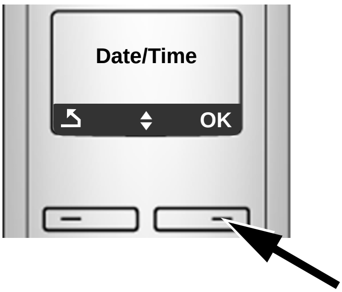
- Press the key below **Menu** on the display screen to open the main menu.

- Press down on the control key repeatedly ...



... until the Settings menu item appears.

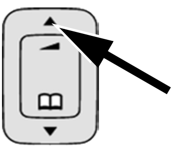
- Press the key below OK on the display screen to confirm your selection.



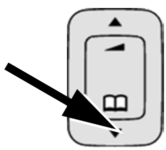
The Date/Time menu item appears on the display.

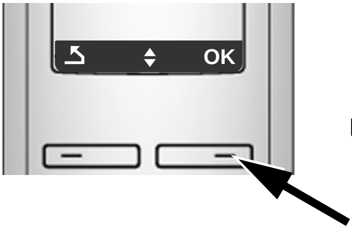
- Press the key below OK on the display screen to open the input field.

- The active line is marked [...].
  Enter day, month and year as an 8-digit number via the handset, e.g. [0 +] [1 ∞] [1 ∞] [0 +] [2 abc] [0 +] [0 +] [8 tuv] for 01/10/2008.



If you want to correct an entry, press up or down on the control key to toggle between fields.





- Press the key below OK on the display screen to confirm your selection.

- Enter the hours and minutes as four digit numbers via the handset e.g. [0 +] [7 pqrs] [1 ꝏ] [5 jkl] for 07:15 am.
  If necessary, use the control key to toggle between fields.
- Press the key below OK on the display screen to confirm your selection.

The display shows Saved.

You will hear a confirmation tone.

- Press and hold the end call key to return to the idle status.

Registering the handset to the base station

Your handset is registered to the base station by default.

For information on how to register additional handsets with the base station and make free internal calls → Page 83.

Installing the base station

The base station is designed for use in closed, dry rooms with a temperature range of +5 °C to +45 °C.

- Set up the base station at a central location on a flat, non-slip surface in your house or apartment.

Please note

Consider the range of the base station.
This is up to 300 m in unobstructed outdoor areas and up to 50 m inside buildings. The range is reduced when eco mode is activated (→ Page 81).

The phone's feet do not usually leave any marks on surfaces. However, due to the multitude of different varnishes and polishes used on today's furnishings, the occurrence of marks on the surfaces cannot be completely ruled out.

For information on how to mount the base station on the wall → Page 203.

Warning

- Never expose the telephone to any of the following: heat sources, direct sunlight or other electrical appliances.
- Protect your Gigaset from moisture, dust, corrosive liquids and vapours.

Connecting the base station

In order to be able to make calls with your phone via the fixed line network and via VoIP, you must connect the base station to the fixed line network and the Internet → **Bild 1**.

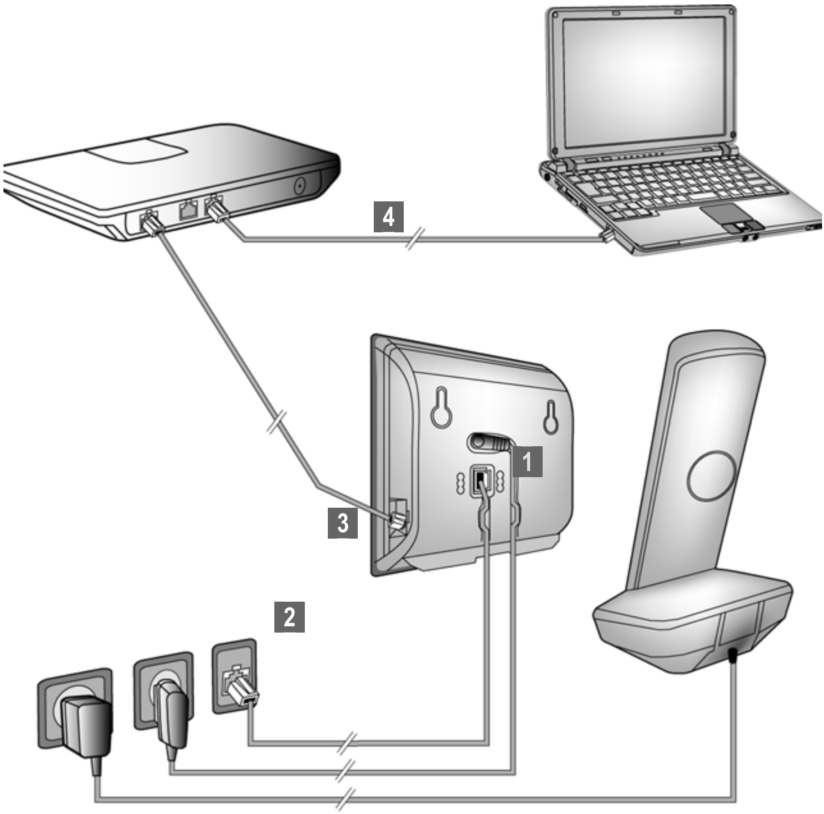


Bild 1 Connecting the phone to the fixed line network and the Internet

Follow the steps in the order given below (→ **Bild 1**):

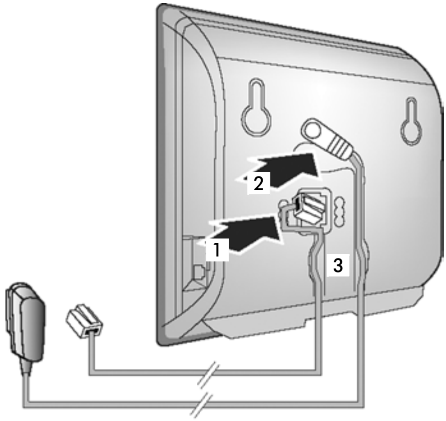
1 Connect the phone cord and power lead to the base station.

2 Connect the base station to the fixed line network and the mains power supply.

3 To connect the base station to the Internet, first connect the base station to the router (connection via router and modem or via router with integrated modem).

4 Connect the PC and router (optional) - for advanced configuration of the base station (→ **Page 108**).

1. Connect the phone cord and power lead to the base station

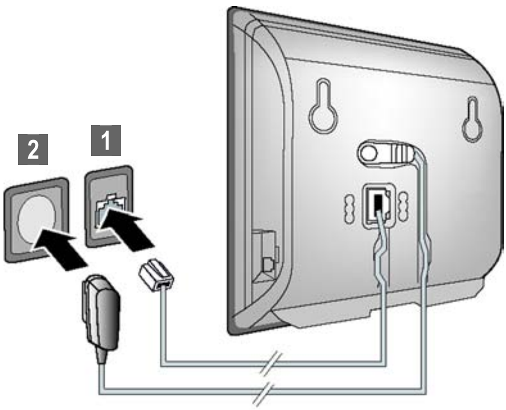


1 Insert the phone cord into the lower connection socket at the rear of the base station.

2 Insert the power lead of the mains adapter into the upper connection socket at the rear of the base station.

3 Push both cables into the appropriate cable channels.

2. Connect the base station to the fixed line network and the mains power supply



1 Insert the phone cord into the fixed line network connection socket.

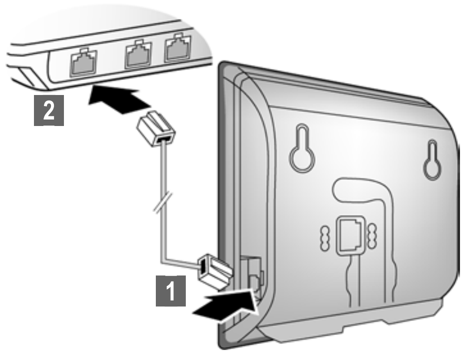
2 **Then** insert the mains adapter into the mains socket.

Warning
- Keep the mains adapter **plugged in at all times** for operation, as the phone does not work without a mains connection.
- Only use the mains adapter and phone cord **supplied**. Pin connections on telephone cables can vary (pin connections → Page 165).

You can now use your phone to make calls via the fixed line network and can be reached on your fixed line number.

3. Connect the base station with the router (Internet)

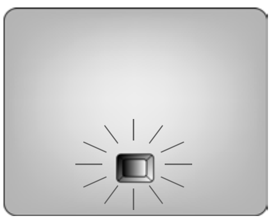
For Internet access you need a router connected to the Internet via a modem (this may be integrated in the router).



1 Connect an Ethernet cable plug into the LAN socket at the side of the base station.

2 Then insert the second Ethernet cable plug into a LAN socket on the router.

As soon as the cable connecting the phone and router is plugged in, the key lights up on the front of the base station (paging key).



You can now establish VoIP connections within Gigaset.net (→ Page 48).

Making settings for VoIP telephony

Before you can use the Internet (VoIP) to phone any other numbers on the Internet, the fixed line network or the mobile phone network, you need the services of a VoIP provider who supports the VoIP SIP standard.

Precondition: You have registered with such a VoIP provider (e.g. via your PC) and set up at least one VoIP account.

To be able to use VoIP, you now need to enter the access data for your VoIP account. You will receive all the necessary data from your VoIP provider. This will include:

Either:

- Your user name (if requested by the VoIP provider), this is the user identification (Caller ID) for your account, which is often identical to your phone number
- Your authentication name or login ID
- The (login) password registered with the VoIP provider
- General settings for your VoIP provider (server addresses etc.)

Or:

- An Auto Configuration Code (Activation Code)

Your Gigaset phone's connection assistant can help you make these entries.

Starting the connection assistant (connection wizard)

Precondition:

The base station is connected to the mains power supply and a router. Your router is connected to the Internet (→ Page 20).

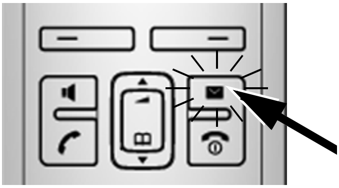
Tip:

Leave IP activated as the default connection for your telephone (default setting → Page 99). The telephone then attempts to establish a connection directly to your VoIP provider's server after the connection assistant is closed. If incorrect/incomplete information means that the connection cannot be established, messages will be displayed (→ Page 29).

Please note

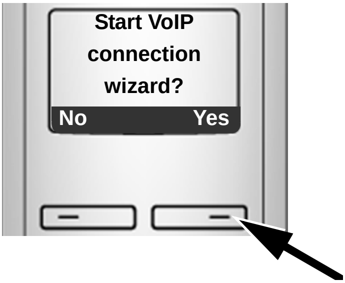
Your phone is preconfigured for dynamic assignment of the IP address. In order for your router to "recognise" the phone, dynamic IP address assignment must also be activated on the router, i.e. the router's DHCP server must be activated.

If the DHCP server cannot or should not be activated, you must first assign a fixed IP address to the phone. For information on how to do this, see Page 103.



As soon as the handset battery is sufficiently charged, the message key ✉ on the handset will flash (approx. 20 minutes after you have put the handset in the charging cradle).

- Press the message key ✉.



You will see the following display.

- Press the key below **Yes** on the display screen.

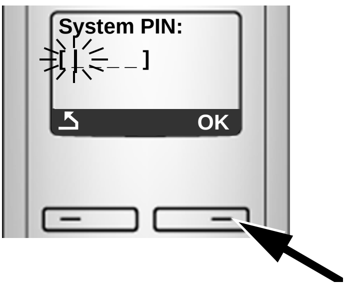
If you press **No**, the procedure that follows is described under "Entering your name in the Gigaset.net directory" on Page 27.

You will be prompted to enter your phone's system PIN.

Please note

To protect your phone and its system settings from unauthorised access, please define a 4-digit number code (system PIN) known only to yourself. These must also be entered before you can register/de-register handsets, or alter the VoIP or LAN settings of your phone.

The default system PIN is 0000 (4 x zero). For how to alter the PIN → Page 97.



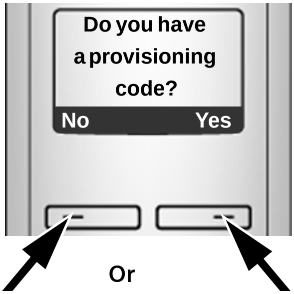
The active line is marked [----].

- Enter your phone's current system PIN using the keypad.
- Press the key below **OK** on the display screen.

The connection assistant is launched.

Please note

The connection assistant will also start automatically if you try to establish a connection via the Internet before you have made the necessary settings. You can also call up the connection assistant at any time via the menu (→ Page 101).



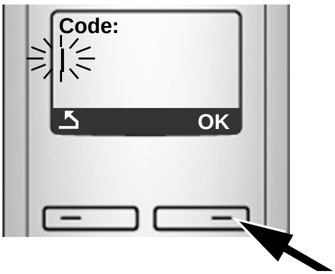
Your VoIP provider has supplied you with an **authentication name/password** and, where applicable, a user name:

- Press the key below No on the display screen.
- For further information, please see "Downloading VoIP provider data" → Page 24

You have received an **Auto Configuration Code** (Activation Code) from your VoIP provider:

- Press the key below Yes on the display screen.
- For further information, please see "Entering an Auto Configuration Code"

Entering an Auto Configuration Code



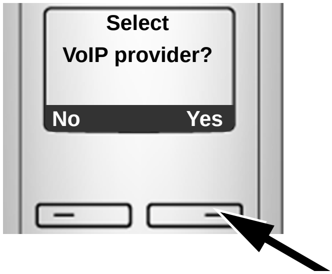
- Enter your Auto Configuration Code using the keypad (max. 32 characters).
- Press the key below OK on the display screen.

All data necessary for VoIP telephony is loaded directly from the Internet to your phone.

When all the data has been successfully loaded onto your phone, Saved appears on the display.

- For further information, please see "Entering your name in the Gigaset.net directory" → Page 27

Downloading VoIP provider data

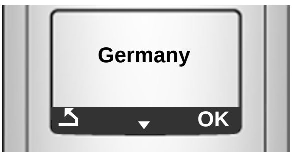


- Press the key below **Yes** on the display screen.

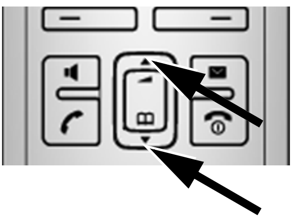
The connection assistant establishes a connection with the Siemens server on the Internet. Various profiles with general access data for different VoIP providers can be downloaded here.

After a brief period you will see the following display:

A list of countries is loaded.



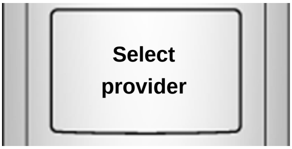
The first country in the list appears in the display.



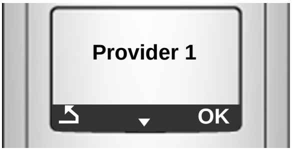
- Press up or down on the control key repeatedly ...

  ... until the country in which you are using the phone appears on the display.

- Press the key below **OK** on the display screen to confirm your selection.



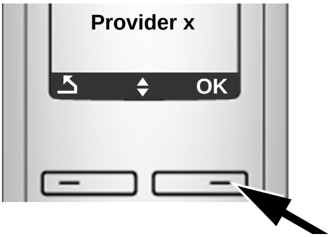
A list of VoIP providers whose data you can download is loaded.



The first VoIP provider in the list appears in the display.

- Press up or down on the control key repeatedly ...

  ... until your VoIP provider appears on the display.



- Press the key below OK on the display screen to confirm your selection.

The general access data for your VoIP provider is downloaded.

When all the data has been successfully loaded onto your phone, Saved appears on the display.

You have not been able to download your provider's data

If the data for your VoIP provider is not available for download, press the display key twice. You can then carry out the following steps with the connection assistant.

You must then make the settings needed for the VoIP provider using the Web configurator (→ Page 118).

Your VoIP provider will supply you with this data.

Entering user data for your first VoIP account

You will now be prompted to enter your personal access data for your VoIP accounts.

The following are provider-dependent:

- Username, Authentication Name, Authentication Password

Or:

- Authentication Name, Authentication Password

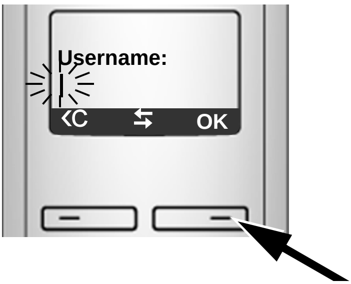
Take care ...

... when entering access data, as it is case sensitive.

To switch between upper and lower case and digits, press the # key (several times if necessary). You can see briefly in the display whether upper case, lower case or digits is selected.

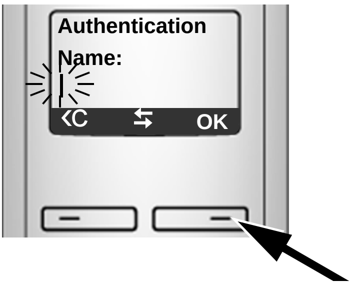
Characters entered incorrectly can be deleted using the left display key below ‹C. The character to the left of the cursor will be deleted.

You can navigate within the input field using the control key (press up/down).

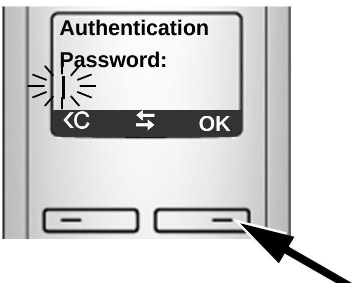


If your VoIP provider does not require a user name, this step can be skipped.

- Using the keypad, enter the user name that you received from your VoIP provider.
- Press the key under OK on the display screen.



- Using the keypad, enter the authentication name that you received from your VoIP provider.
- Press the key under OK on the display screen.



- Use the keypad to enter your password.
- Press the key under OK on the display screen.

If you have made all the required entries, Saved will appear on the display.

Entering your name in the Gigaset.net directory

With Gigaset.net you can call other Gigaset.net users directly over the Internet free of charge, without setting up an account with a VoIP provider and without making any further settings. You can find Gigaset.net subscribers by carrying out a name search in the Gigaset.net directory (→ Page 48).

You will see the following display.

- Press the key below **Yes** on the display screen.

- Using the keypad, enter the name that you would like to be listed under in the Gigaset.net directory. The name may contain a maximum of 25 characters.
- Press the key below **OK**.

If there is already an entry under this name, you will receive a message to this effect and you will be asked to enter a name again: Please choose another name.

If an entry in the Gigaset.net directory is successful, the message Name added to Gigaset.net is displayed.

You have not been able to enter a name ...

If the attempt to create the entry fails (e.g. because the phone is not connected to the Internet), a message to this effect is displayed briefly (→ Page 29). You can then create the entry later via the Gigaset.net directory (→ Page 51).

Completing the VoIP settings

After the entries have been completed, the handset reverts to idle status.

If all the settings are correct and if the phone can establish a connection to the VoIP server, the internal name of the handset will be displayed: (example)

You can now use your phone to make calls via the fixed line network and the Internet! Callers can reach you on your fixed line number and your VoIP number.

Please note

- To ensure that you can always be reached via the Internet, the router must be permanently connected to the Internet.
- If "New firmware available" is shown in the display, an updated firmware is already available on the Internet for your phone. Press the right display key **Yes**, to load the new firmware. The process takes approx. 3 minutes.
- If you try to make a call via a VoIP connection that is not configured correctly, the following VoIP status message will appear in the display: VoIP config. error: xxx (xxx = VoIP status code). The various status codes and their respective meanings can be found in the appendix on → Page 157.

You have set up several VoIP accounts ...

You can enter five additional VoIP accounts (VoIP phone numbers) via the Web configurator at a later stage (→ Page 117). Your phone (together with your fixed line number) can then be reached on up to seven different phone numbers. You can assign the phone numbers to the individual handsets that are registered with the base station as send and receive numbers (→ Page 131).

Icons on the idle display

The following is displayed:

- the internal number, e.g. INT 1.
- the strength of the reception signal between base station and handset:
  - good to poor:
  - no reception: flashes
- battery charge status:
  - (flat to full)
  - flashes: batteries almost flat
  - flashes: charging procedure

No connection to the Internet/VoIP server

If one of the following messages is displayed instead of the internal name after the connection assistant is closed, errors have occurred:

- Server not accessible
- SIP registration failed

Below you will find possible causes and measures you can take.

Server not accessible

The phone has no connection to the Internet.

- Check the cable connection between the base station and the router (the LED on the base station must light up) and the connection between the router and the Internet connection.
- Check whether the phone is connected to the LAN.
  - It may not have been possible to dynamically assign an IP address to the phone

  Or

  - You have assigned a static IP address to the phone that has either already been assigned to another LAN subscriber or does not belong to the router's address block.
  - Press the paging key on the base station. The IP address appears on the handset display.
  - Press the talk key on the handset to end paging call.
  - Start the Web configurator with the IP address.
  - If no connection can be established, change the settings on the router (activate DHCP server) or the phone's IP address.

SIP registration failed

- Your personal data for registering with the VoIP provider may have been entered incompletely or incorrectly.
  - Check your entries for Username, Authentication Name and Authentication Password. Particularly check your use of upper and lower case.
    To do this, open the following menu on your handset:
    Menu → Settings → Base → Telephony → VoIP
    (enter system PIN) (→ Page 102)
- The server address for the VoIP server has not yet been entered, or has been entered incorrectly.
  - Start the Web configurator.
  - Open the following Web page: Settings → Telephony → Connections.
  - Click the Edit button next to the first VoIP connection.
  - Edit the server address where necessary.

Please note

If port forwarding is activated on your router for the ports that have been registered as the SIP port (Standard 5060) and the RTP port (Standard 5004), it is advisable to deactivate DHCP and assign the phone a static IP address (otherwise you may not be able to hear the other party during VoIP calls):

- Assign an IP address via the handset menu:

  Menu → Settings → Base → Local Network

Or

- Assign an IP address via the Web configurator:
  - Open the following Web page: Settings → IP Configuration.
  - Select IP address type.

Please note that the IP address and subnet mask depend on the router's address block.

You must also enter the standard gateway and DNS server. The IP address for the router is generally entered here.

How to proceed

Now you have successfully started your Gigaset you will probably want to adapt it to your personal requirements. Use the following guide to quickly locate the most important subjects.

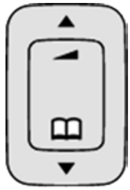
If you are unfamiliar with menu-driven devices such as other Gigaset telephones you should first read the section entitled "**Operating the handset**" → **Page 32**.

Information on ...	... is located here.
Making calls via VoIP or the fixed line network	Page 40
Setting the ringer melody and volume	Page 94
Setting the handset volume	Page 93
Setting Eco mode:	Page 81
Preparing the telephone for SMS reception	Page 71
Operating the telephone on a PABX	Page 106
Registering existing Gigaset handsets to a base station	Page 83
Transferring directory entries from existing Gigaset handsets to the new handset(s)	Page 66
Using online directories	Page 67
Entering additional VoIP accounts	Page 117

If you have any questions about using your phone, please read the tips on troubleshooting (→ Page 153) or contact our Customer Care team (→ Page 152).

Operating the handset

Control key



In the following description, the side of the control key you need to press for each operation is indicated accordingly, e.g. for "press up on the control key".

The control key has a number of different functions:

With the handset in idle status (without a screensaver)

Press briefly to open the handset directory.
Press and hold: to open the list of available online directories.

Call up the menu to set the ringer volume (→ Page 94).

In the main menu, in submenus and lists

/ Scroll up/down line by line.

In input fields

Use the control key to move the cursor to the left or right .

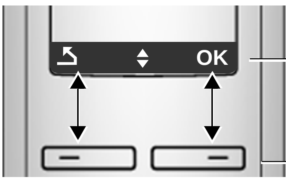
During an external call

Press briefly to open the handset directory.

Adjust the loudspeaker volume for earpiece and handsfree mode.

Display keys

The function of the display keys changes depending on the particular operating situation. Example:



Current display key functions are shown in the bottom display line.

Display keys

Important display keys:

Menu	Open a context-dependent menu.
OK	Confirm selection.
<C	Delete key: deletes one character at a time from right to left.
	Go back one menu level or cancel operation.
→→	Open the last number redial list.

Keys on the keypad

/ / etc.
Press the matching key on the handset.

Enter digits or letters.

Correcting incorrect entries

You can correct incorrect characters in the text by navigating to the incorrect entry using the control key. You can then:

- Press <C to delete the character to the left of the cursor
- Insert characters to the left of the cursor
- Overwrite the (flashing) character when entering the time and date etc.

Menu guidance

Your telephone's functions are accessed using a menu that has a number of levels.

Main menu (first menu level)

- When the handset is in idle status, press **Menu** to open the main menu.

The main menu functions are shown on the display as a list with name and icon.

To access a function, i.e. to open the corresponding submenu (next menu level):

- Navigate to the function using the control key. Press the display key **OK**.

Submenus

The functions in the submenus are displayed as lists.

To access a function:

- Scroll to the function with the control key and press **OK**.

Or:

- Enter the corresponding digit combination (→ Page 36).

Briefly press the end call key once to return to the previous menu level/ cancel the operation.

Reverting to idle status

You can revert to idle status from anywhere in the menu as follows:

- Press and **hold** the end call key.

Or:

- Do not press any key: after 2 minutes the display will **automatically** revert to idle status.

Any settings you have not confirmed by pressing OK will be discarded.

An example of the display in idle status is shown on (→ Page 28).

Activating/deactivating the handset

With the phone in idle status, press and **hold** the end call key (confirmation tone) to switch off the handset.
Press and **hold** the end call key again to switch the handset on.

Activating/deactivating the keypad lock

The keypad lock prevents any inadvertent use of the phone.

Press and **hold** the hash key to activate or deactivate the keypad lock. You will hear the confirmation tone.

When the keypad lock is activated you will see the icon on the display and a message when you press a key.

The keypad lock is deactivated automatically if someone calls you. It is reactivated when the call is finished.

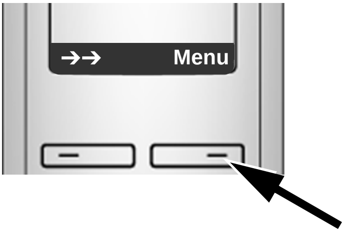
Illustration of operating steps in the user guide

The operating steps are shown in abbreviated form.

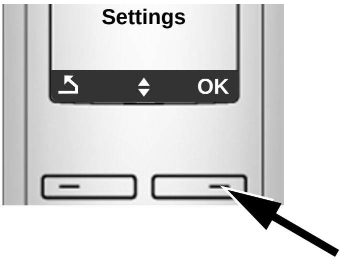
Example:

Menu → Settings → Handset → Auto Answer (✓ = on)

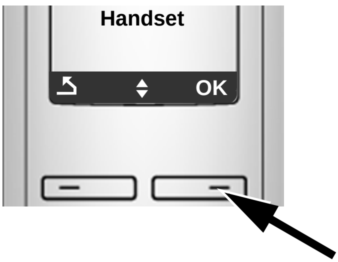
This illustration represents:



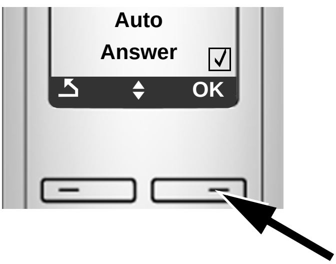
- Press the display key below Menu to open the main menu.



- Press down on the control key repeatedly until the Settings menu item appears on the display.
- Press the display key below OK to confirm your selection.



- Press down on the control key repeatedly until the Handset menu item appears on the display.
- Press the display key below OK to confirm your selection.



The Auto Answer menu item appears on the display.

- Press the key below OK to activate/deactivate the function. If the function is activated, this is indicated by ✓.

Menu trees

Phone menu

Open the main menu on your phone by clicking on the right display key **Menu** when the handset is in idle status. There are two ways to select a function:

Using number combinations ("shortcut")

- Enter the number combination that is in front of the function in the menu tree.
  Example: **Menu** 7 pqrs 3 def 2 abc for "Set handset language".

Scrolling through the menus

- Scroll to the function with the control key (press up or down) and press **OK**.

1 SMS

1-1	Write Message					→ Page 71
1-2	Incoming					→ Page 73
1-3	Outgoing					→ Page 72
1-6	Settings	1-6-1	Service Centres	1-6-1-1	SMS centre # 1	→ Page 75
				:	:	
				1-6-1-4	SMS centre # 4	

3 Select Services

3-1	VoIP	3-1-6	For All Calls	3-1-6-1	Call Diversion	→ Page 55
				3-1-6-2	Call Waiting	→ Page 56
3-2	Phone	3-2-6	For All Calls	3-2-6-1	Call Diversion	→ Page 54
				3-2-6-2	Call Waiting	→ Page 54
3-3	All calls anonymous					→ Page 53
3-4	Next call anony- mous					→ Page 53

4 Alarm Clock

4-1	Activation	→ Page 82
4-2	Wake up time	→ Page 82

5 Voice Mail

5-3	Set Key 1	Net AM:Phone	→ Page 78
		Net AM:IP1	
		: : (dependent on the number of configured VoIP phone numbers and receive numbers on the handset)	
		Net AM:IP6	

7 Settings

7-1	Date/Time					→ Page 14
7-2	Audio Settings	7-2-1	Call Volume	7-2-1-1	Earpiece Volume	→ Page 93
				7-2-1-2	Handsfree Volume	
		7-2-2	Ringer Volume			→ Page 94
		7-2-3	Ringer Melody	7-2-3-1	External Calls	→ Page 94
				7-2-3-2	Internal Calls	
				7-2-3-3	Alarm Clock	
		7-2-4	Advisory Tones			→ Page 95
		7-2-5	Battery Low	7-2-5-1	Off	→ Page 95
				7-2-5-2	On	
				7-2-5-3	During Call	
7-3	Handset	7-3-1	Display	7-3-1-1	Screensaver	→ Page 91
		7-3-2	Language			→ Page 90
		7-3-3	Auto Answer			→ Page 93
		7-3-4	Register Handset			→ Page 83
		7-3-5	Reset Handset			→ Page 96

7-4	Base	7-4-1	Calls List Type	7-4-1-1	Missed Calls	→ Page 61
				7-4-1-2	All Calls	
		7-4-2	Music on hold			→ Page 99
		7-4-3	System PIN			→ Page 97
		7-4-4	Base Reset			→ Page 98
		7-4-5	Additional Features	7-4-5-1	Repeater Mode	→ Page 99
				7-4-5-3	Eco mode	→ Page 81
		7-4-6	Local Network	7-4-6-1	Dynamic IP address	→ Page 103
				7-4-6-2	IP Address	
				7-4-6-3	Subnet Mask	
				7-4-6-4	DNS Server	
				7-4-6-5	Default Gateway	
		7-4-7	Telephony	7-4-7-1	Default Line Type	→ Page 99
				7-4-7-2	Connection Assistant	→ Page 101
				7-4-7-6	Phone	→ Page 106
				7-4-7-7	VoIP	→ Page 101
		7-4-8	Firmware Update			→ Page 100

Web configurator menu

- Home → Page 109
- Settings
  - IP Configuration → Page 115
  - Telephony
    - Connections → Page 117
    - Audio → Page 127
    - Number Assignment → Page 131
    - Call Forwarding → Page 132
    - Dialling Plans → Page 136
    - Network Mailbox → Page 138
    - Advanced Settings → Page 132
  - Messaging
    - Messenger → Page 139
    - E-Mail → Page 141
  - Services → Page 142
  - Handsets → Page 144 and Page 145
  - Miscellaneous → Page 147 to Page 147
- Status
  - Device → Page 151

Making calls with VoIP and the fixed line network

Making an external call

External calls are calls made via the public telephone network (fixed line network) or via the Internet (VoIP). You generally decide which connection type you want to use for a specific call when you dial the number. You have the following options:

- Select the connection type by briefly pressing/pressing and holding the talk key
- Select the connection type via the display key, by assigning VoIP or the fixed line network to your left display key
- Select the connection via your line suffix

Please note

- You can conduct three separate external calls via your base station (using different handsets): two calls via VoIP and one via the fixed line network.
- You can define dialling plans for certain numbers or area codes by determining the connection and therefore the billing method to be used (cost control → Page 136) when these numbers are dialled.
- Dialling with the directory (→ Page 63), shortcut keys (→ Page 65) or last number redial list (→ Page 59) saves repeated keying of phone numbers. You can modify or add to these numbers on a call-to-call basis.
- If you use VoIP to make a call to the fixed line network, you may also have to dial the area code for local calls (depending on the VoIP provider). You can avoid having to dial your own area code by entering it into the configuration (→ Page 135). Your area code is then added automatically when you make local calls.

Use the talk key to select the type of connection and make the call

By briefly pressing or pressing and holding the talk key [talk key], you can determine the type of connection for the call you want to make (fixed line network or VoIP).

Precondition: You enter the number without a line suffix (→ Page 40) and have not defined any dialling plans for this number (→ Page 136).

[keypad] [talk key] Enter number (without suffix) or the IP address and **briefly press/ press and hold** the talk key.

A default connection is established on your phone (fixed line network or VoIP → Page 99/Page 130).

- **Briefly** press the talk key [talk key] if you want to make a call via this default connection.
- Press and **hold** the talk key [talk key] if you want to make the call via the other connection type.

If you have assigned several VoIP numbers to your phone, you can define which VoIP number (VoIP account) is used for external calls from each specific handset (handset send number → Page 131).

Please note

If you are using a GAP compatible handset other than the Gigaset handsets A58H, S67H, S68H, SL37H, S45 and C45, every call will be made via the standard connection, even if you press and hold the talk key. If you want to use the non-default connection to make a call, enter a star (*) at the end of the number.

Use the display key to select the type of connection and make the call

Precondition: Phone or IP is assigned to your handset's left display key (→ Page 92).

Phone / IP — Press the display key to define the connection type.

Enter number or select from the directory.

Press the talk or handsfree key.

Please note

If you have pressed the display key IP before dialling and ...

- ... dialled the number with suffix #1 to #6, your call will be made via the VoIP account assigned to the suffix. The number will not be dialled if the suffix is invalid (e.g. no VoIP connection assigned).
- ... dialled the number without a suffix or with the suffix #0, your call will be made via the handset's VoIP send number.

Do not enter a suffix if you have pressed the display key Phone prior to dialling. Otherwise the suffix will be dialled together with the number via the fixed line network. This may cause errors!

Select and dial a connection via your line suffix

You can configure up to six VoIP numbers on your phone in addition to the fixed line network number and the Gigaset.net number. A (line) suffix is assigned to each number (line) of your phone:

- the fixed line network number has the suffix #0
- VoIP numbers have the suffixes #1 to #6 (→ Page 118)
- and the Gigaset.net number has the suffix #9

When dialling, you can use this line suffix to specify the connection via which you would like to call or be charged.

Enter the number of the party you wish to call.

Add the suffix of the connection (your phone number) from which the call is to be made and charged to.

Press the talk key.

The connection is always made via the line with the assigned suffix, regardless of whether you press the talk key **briefly** or press and **hold**.

Example: If you enter the number 1234567890#1 and press the talk key [talk key], the number 1234567890 will be dialled via the first VoIP connection in the configuration.

Please note

If you specify a suffix for which no VoIP connection is configured in your base station, the VoIP status code 0x33 will be displayed. The number will not be dialled.

Enter the IP address (provider-dependent)

You can also dial an IP address instead of a phone number using VoIP.

- Press the star key [* key] to separate the sections of the IP address (e.g. 149*246*122*28).
- If necessary press the hash [# key] key to attach the SIP port number of the person you are calling to the IP address (e.g. 149*246*122*28#5060).

You **cannot** dial IP addresses using a line suffix.

If your VoIP provider does not support the choice of IP addresses, each part of the address will be interpreted as a normal phone number.

Cancelling the dialling operation

You can cancel the dialling operation with the end call key [end call key].

Dialling emergency numbers – Setting dialling plans

You can use the Web configurator to block certain numbers or to define which of your numbers (fixed line network, VoIP) should be used to call specific numbers (Dialling Plans, → Page 136).

If you enter a number that has a defined dialling plan, the call will be made via the line defined in the dialling plan - regardless of whether the talk key is pressed briefly or pressed and held. Any automatic area code will **not** be prefixed to the number.

Emergency numbers

Dialling plans for emergency numbers (e.g. the **local** emergency service number) are preconfigured for certain countries. Emergency calls are then always made via the fixed line network.

You cannot delete or deactivate these dialling plans. However, you can change the connection through which each emergency number should be called (e.g. if the phone is not connected to the fixed line network). You must make sure that the VoIP provider for the selected connection supports emergency calls.

If your phone does not have default dialling plans for emergency calls, you should define the rules yourself (→ Page 136). Assign them to a connection that you know supports emergency calls. Calls to emergency numbers are always supported by fixed line networks.

Please note: If no rules are defined for emergency numbers and you have programmed an automatic local area code (→ Page 135), the code will be prefixed to emergency numbers as soon as they are dialled via a VoIP connection.

Please remember

Emergency numbers cannot be dialled if the keypad lock is activated. Before dialling, press **and hold** the hash key [#], to release the keypad lock.

Ending a call

[End call key] Press the end call key.

Accepting a call

The handset indicates an incoming call in three ways: by ringing, by a display on the screen and by the flashing handsfree key [Handsfree key].

Please note

Only calls to receive numbers assigned to your handset will be signalled (→ Page 131).

Calls made to a number that is not assigned to a handset as a receive number will not be signalled on any handset.

You can accept the call by:

- Pressing the talk key [Talk key].
- Pressing the handsfree key [Handsfree key].

If the handset is in the charging cradle and the Auto Answer function is activated (→ Page 93), the handset will take a call automatically when you lift it out of the cradle.

To deactivate the ringer, press the **Menu** display key and select Silent. You can accept the call so long as it is displayed on the screen.

Please note

You can reject VoIP calls by pressing the end call key [End call key]. The caller receives an appropriate message (provider-dependent).

Calling Line Identification

When you receive a call from the Internet, the caller's number or the name they have specified is displayed on the screen.

When you receive a call from the fixed line network, the caller's number and/or name is displayed on the screen if the following conditions are met:

- Your fixed line network provider supports CLIP, CLI and CNIP:
  - CLI (Calling Line Identification): the caller's number is transmitted
  - CLIP (Calling Line Identification Presentation): the caller's number is displayed
  - CNIP (Calling Name Identification Presentation): the caller's name is displayed
- You have requested CLIP or CNIP from your network provider.
- The caller has requested CLI from the network provider.

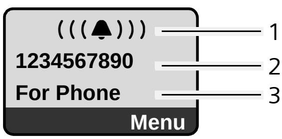
If the phone number is identified and the caller's number is saved in your handset's local directory, the name will be displayed from the directory.

If the caller's number is not saved in the local directory the caller's surname and first name will be displayed from the currently installed online directory.
Precondition: You have activated this option (see "Selecting and registering online directories for access", Page 143).

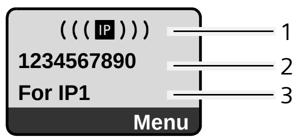
Call display

You can use the display to determine whether the call is for your fixed line network number or one of your VoIP numbers.

Calls to your fixed line number



Calls to one of your VoIP numbers



1 Ringer icon
2 Number or name of the caller if available
3 Receive number: indicates which of your phone numbers the caller has dialled. Assign the name when you are configuring the phone with the Web configurator (→ Page 119/Page 125). For calls from Gigaset.net, For Gigaset.net is displayed.

Adopting the name from the online directory

You can display the name under which the caller is saved in the online directory.

Preconditions:

- The provider of the online directory you have set for your telephone (→ Page 143) supports this function.
- You have activated the "display caller name" function via the Web configurator (→ Page 143).

- If applicable, the phone number is transferred and not withheld.
- Your telephone is connected to the Internet.
- The caller's number is not saved in the handset's local directory.

Withholding calling line display

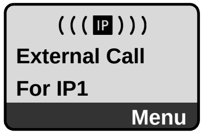
The number or name of the caller is not displayed in the following cases:

- The caller has activated the "anonymous calling" function.
- A caller from the fixed line network has not requested Calling Line Identification from the fixed line network provider.

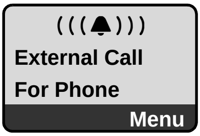
The following is displayed in place of the number:

- With a call to one of your VoIP phone numbers, the display is provider-dependent (examples):

or

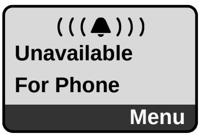


- For a call to your fixed line network number:
  – If no number is transmitted:



  – If the caller has withheld Calling Line Identification.

  – If the caller has not arranged Calling Line Identification.



Screen display with CNIP (fixed line network)

Precondition: Call is made from a fixed line network connection to your fixed line network number.

If you have CNIP, then the name (town/city) that is registered with your network provider for the caller's number will **also** be displayed. If the caller's number is stored in your directory then the directory entry will be displayed.

The display shows:

- External Call, if no number is transmitted.
- Withheld, if the caller has suppressed Calling Line Identification.
- Unavailable, if the caller has not arranged Calling Line Identification.

VoIP: Displaying the called party's phone number (COLP)

Preconditions:

- Your VoIP provider supports COLP (**Co**nnected **L**ine Identification **P**resentation). You may have to ask your provider to activate COLP (contact your VoIP provider for more information).
- The called party has not activated COLR (**Co**nnected **L**ine Identification **R**estriction).

For outgoing VoIP calls, the phone number of the connection on which the call is received is displayed on the handset.

The displayed number may differ from the number you have dialled. Examples:

- The called party has activated call forwarding.
- The call is answered by another connection within a PABX system.

If there is an entry in the directory for this phone number, the corresponding name will be displayed.

Please note

- The number of the connection you have reached (or the assigned name) will also be displayed instead of consultation calls.
- When the phone number is copied to the directory (Menu → Copy to Directory) and the last number redial list, the dialled number (not the displayed number) is copied.

Handsfree talking

In handsfree mode, instead of holding the handset to your ear you can put it down, for example on the table in front of you. This allows others to participate in the call.

Activating/deactivating handsfree mode

Activating while dialling

Enter the number.

Briefly press/press and hold the talk key, to select the connection type (→ Page 40).

- You should inform your caller before you use the handsfree function so that they know someone else is listening.

Switching between earpiece and handsfree mode

▶ Press the handsfree key [handsfree] to activate or deactivate handsfree talking during a call.

If you wish to place the handset in the charging cradle during a call:

▶ Press and hold the handsfree key [handsfree] while placing the handset in the charging cradle.

▶ If the handsfree key [handsfree] does not light up, press the key again.

For instructions on how to adjust the loudspeaker volume → Page 93.

Muting the handset

You can deactivate the microphone in your handset during an external call. Your caller will hear hold music, if activated (→ Page 99).

INT	Press the display key to mute the handset.
↰ / [end call]	Press the display or the end call key, to reactivate the microphone.

Making cost-effective calls

You can use your phone's cost control function for calls to fixed line or mobile phone networks. Open accounts with various VoIP providers who offer favourable rates for calls to other networks. In the phone configuration, define the most cost-effective VoIP connections (account), e.g. for specific regional, national and mobile network prefixes, to be used when calls are made (Dialling Plans, → Page 136). Or define the VoIP connection to be used when dialling the number (→ dialling with line suffix, Page 40).

If you are using your fixed line network to make a call, choose a network provider that offers particularly favourable rates (call-by-call).

Displaying the call duration

The duration of each call appears in the display for calls made via a fixed line network and VoIP

- during the conversation,
- until about three seconds after the call has ended if you do not replace the handset in the charging cradle.

Please note

The actual duration of the call can vary from that shown by a few seconds.

VoIP telephony via Gigaset.net

You can use **Gigaset.net** to make free phone calls via the Internet **directly** to other Gigaset.net users, without having to set up an account with a VoIP provider or make any further settings. You simply have to connect your phone to the power supply and the Internet connection and, if necessary, enter yourself in the Gigaset.net online directory under a name of your choice (→ Page 27/ Page 51).

Gigaset net is a VoIP service provided by Siemens Home and Office Communication Devices GmbH und Co KG, which is available to all users with a Gigaset VoIP device.

You can call other subscribers to Gigaset.net **free of charge**, i.e. there are no telephone charges other than the costs for your Internet connection. Connections to/from other networks are not possible.

Every Gigaset VoIP device is assigned a Gigaset.net phone number by default (→ Page 160).

All registered subscribers are included in the Gigaset.net directory, which you are able to access.

You can use an echo service provided by the phone number **12341#9** on Gigaset.net to test your VoIP connection.

After an announcement, the echo service sends back the voice data you have received immediately in the form of an echo.

Exclusion of liability

Gigaset.net is a voluntary service provided by Siemens Home and Office Communication GmbH & Co KG with no liability or guarantee for the availability of the network. This service can be terminated at any time with a notice period of three months.

Please note

If you do not use your Gigaset.net connection for six weeks, it is automatically deactivated. You cannot be reached for calls from Gigaset.net.

The connection is reactivated:

- as soon as you start a search in the Gigaset.net directory or
- make a call via Gigaset.net (dial a number with #9 at the end) or
- activate the connection via the Web configurator (→ Page 125).

Search for subscribers in the Gigaset.net directory

Your handset is in idle status.

▼	Press and **hold**.
▲▼	If necessary, select Gigaset.net directory from the list of available online directories and press **OK**.

Or:

- Open the directory with the button.
- Select Gigaset.net from the directory and press the talk key . A connection to the Gigaset.net directory is established.

Please note

- ◆ The Gigaset.net directory entry is transferred to a handset when it is registered with the base station. Provided the handset can send and receive directory entries.
- ◆ Calls to the Gigaset.net directory are always **free of charge**.
- ◆ You can also open the Gigaset.net directory by dialling **1188#9** (phone number of the Gigaset.net directory) and pressing the talk key .

If no connection can be made to the Gigaset.net directory, an error message will be sent and the handset will go into idle status.

Once the connection has been established, you will be asked to enter a name that you want to search for.

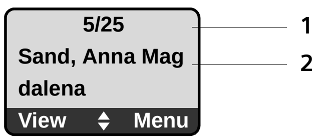
Nickname:
Enter the name or part of a name (max. 25 characters).

Menu — Press the display key.

Start search
Select and press OK.

If the search has been successful, a hit list will be displayed of all the names that begin with the specified character string.

Example:



1. 5/25: Entry number/number of hits
2. Name of the entry, the name is displayed in full, if necessary over several lines

You can scroll through the hit list with .

If it has **not** been possible to find a **matching** entry, a corresponding message is displayed. You have the following options:

- Press the display key New to start a new search.

Or

- Press the display key Change to change the search criteria. The previously entered name is copied and can be edited.

If there are **too many matching** entries in the Gigaset.net directory, the message Too many entries found is displayed instead of a hit list.

- Press the display key Refine to start a refined search. The previously entered name is copied and can be edited/expanded.

Calling participants

	Select the subscriber from the hit list.
	Press the talk key.

Viewing the subscriber's number

	Select the subscriber from the hit list.
View	Press the display key.

The display shows the Gigaset.net number and the subscriber's name, whereby the name may appear over a couple of lines.

Example:

Please note

- Connections to Gigaset.net are always established via the Internet irrespective of which default connection is set on your phone. Pressing and holding the talk key or pressing it briefly and a "*" at the end of the number have no effect.
- You can open the Gigaset.net directory and establish connections, even if you have not entered yourself in the Gigaset.net directory.

Using other functions

Precondition: The hit list is displayed.

(select entry) → **Menu**

The following functions can be selected with :

Copy to Directory

Copy the entry to the handset directory. The number and name (abbreviated if necessary, max. 16 characters) are copied to the directory.

- Edit and save entry where appropriate (→ Page 64).

The hit list is displayed again.

Show number

Display the number of the entry.

- Press the display key **OK**, to return to the hit list.

New search

Start a search with a new name (→ Page 49).

Refine search

Start detailed search. The previously entered Nickname can be edited/expanded and the search can be restarted (→ Page 49).

Your name in Gigaset.net

→ "Entering, editing and deleting own entry" Page 51.

Please note

If you select a Gigaset.net number from the local directory, the connection is automatically established via Gigaset.net (Internet).

Entering, editing and deleting own entry

You have the following options:

- Edit the name of your entry in the Gigaset.net directory
- Delete your entry from the Gigaset.net directory
- If you did not enter a name when using the phone for the first time (→ Page 27), specify a name and enter yourself in the directory.

Viewing own entry

- ▶ You are connected to the Gigaset.net directory:
  Select Menu → Your name in Gigaset.net and press OK.

Or:

- ▶ You are in a Gigaset.net hit list:
  Select Menu → Own information and press OK.

Your Gigaset.net number and, where applicable, your currently entered name are displayed.

Entering/editing a name

Edit	Press the display key.
	Edit name or enter new name (max. 25 characters) and press OK. You can delete the name with ‹C.

If there is no existing entry with this name in the Gigaset.net directory, the name is saved. A message to this effect is displayed. The handset switches to idle status.

If there is an existing entry with this name, or the entered name contains impermissible characters, you will be requested to enter a different name.

Delete your own entry from the Gigaset.net directory

Precondition: You are connected to the Gigaset.net directory:

Menu → Your name in Gigaset.net / Own information
Select and press OK.

Edit	Press the display key.
‹C	Delete name and press OK.

Your entry is deleted from the directory. You are no longer "visible" to other Gigaset.net subscribers. However, you can still be reached via your Gigaset.net number.

Please note

For how to display your Gigaset.net number → Page 160.

Calling a Gigaset.net subscriber

You can call a Gigaset.net subscriber directly via the Gigaset.net directory (see above) or via their Gigaset.net number:

/ Enter the Gigaset.net number (including the #9) or select from the handset directory.

Press the talk key.

Every number ending with #9 is dialled via Gigaset.net.

Network services

Network services are functions made available by your fixed line network or VoIP provider.

Anonymous calling - withholding caller ID

Phone number identification can be withheld (CLIR = Calling Line Identification Restriction). Your phone number will not be displayed when making outgoing calls. You are calling anonymously.

Preconditions:

- For anonymous calls via your fixed line network connection you need to have requested the relevant service (feature) from your fixed line network provider.
- Anonymous calls are only possible via VoIP connections through providers that support the "anonymous calling" function. You may have to ask your VoIP provider to activate this function.

Activating/deactivating "anonymous calling" for all calls

Withholding caller ID can be activated/deactivated permanently for all your phone's connections (fixed line network and VoIP).

When this function is activated, the phone number will be withheld both for fixed line network calls and for calls via a VoIP connection. Withholding caller ID is activated for all registered handsets.

Menu → Select Services

All calls anonymous
Select and press **OK** (✓ = on).

If the All calls anonymous function is activated, Withhold Number is shown in the handset's idle display.

Activating/deactivating "anonymous calling" for the next call

You can activate/deactivate withholding caller ID for the next call.

Menu → Select Services → Next call anonymous

Yes / No	Select and press **OK**.
⌨	If necessary, enter the phone number with line suffix.
Send	Press the display key. The phone number is dialled. If you have not specified a line suffix, the number will be dialled via the default connection.

Further network services for fixed line networks

The following network services can only be used for making calls via the fixed line network. You will need to request them from your fixed line network provider.

▶ **If you require assistance, please contact your network provider.**

Settings for all calls

If you have completed one of the following procedures, a code is sent.

▶ After confirmation from the telephone network, press the end call key.

You can set the following features:

General call forwarding (= Call Diversion)

Menu → Select Services → Phone → For All Calls → Call Diversion

All Calls
Select and press OK.

On — Select and press OK.

If necessary, enter the number to which the call is to be forwarded. You can enter a fixed line, VoIP or mobile number.

OK — Press the display key.

▶ Press the end call key after the announcement from the telephone network.

Call forwarding is only activated for your fixed line network number. Calls to your VoIP number are not forwarded. For information on how to forward calls to your VoIP numbers, → Page 55.

Deactivating call forwarding

Menu → Select Services → Phone → For All Calls → Call Diversion

All Calls
Select and press OK (✓ = on).

Off — Select and press OK.

Press and **hold** (idle status).

Call forwarding for fixed line network numbers is deactivated.

Activating/deactivating call waiting

If call waiting is activated, a caller on the fixed line network will hear the ringing tone if you are already conducting a phone conversation using your fixed line connection. This call is announced acoustically and visually on your handset screen.

Calls on the VoIP connection are not shown as call waiting. They are signalled on other registered handsets. If no other handset is available, the caller will hear the busy tone.

Menu → Select Services → Phone → For All Calls → Call Waiting

On / Off — Select and press OK.

▶ Press the end call key after the announcement from the telephone network.

Please note

The setting does not affect the procedure for call waiting on the VoIP connection. For how to activate/deactivate call waiting for the VoIP connection
→ Page 56.

Functions during a call

Consultation call

During a call:

Menu → External Call

Select and press OK.

Enter a number or copy it from the directory and press OK.

The number will be dialled via the fixed line network.

Please note

After a few seconds, the number selected for a consultation call is saved in the last number redial list.

- Conference call:
  - Talk to both participants: Menu → Conference Call.
  - End call with both participants: press the end call key.

Further network services for VoIP

You can use the following network services to make calls via the VoIP connection.

Functions for the next call

You can withhold your fixed line network number for the next call (CLIR) provided your network provider supports the feature.

After the call, the setting is reset and your number is transmitted again.

Menu → Select Services → VoIP → Withhold Number

Enter the number of the other caller.

OK Press the display key.

The number is always dialled via VoIP. If no suffix is entered, it is selected via the VoIP send number.

The setting is reset after this call, even if you call the dialled number again from the last number redial list.

Settings for all calls

General call forwarding (= Call Diversion)

Precondition: The VoIP provider supports call forwarding.

Menu → Select Services → VoIP → For All Calls → Call Diversion

The display shows a list of configured and activated VoIP phone numbers and the Gigaset.net number of your telephone. Numbers for which call forwarding is activated are marked with ✓.

⇕	Select the phone number for which you want to activate or change call forwarding and press **OK**.

All Calls / No Answer / When Busy

	Select and press **OK** (✓ = on).
On	Select and press **OK**.
⌨	If necessary, enter the number to which the call is to be forwarded. You can state a fixed line network, VoIP or mobile number for forwarding calls from a VoIP number. You must state another Gigaset.net number for call forwarding from your Gigaset.net number.
OK	Press the display key.
☎	Press and **hold** (idle status).

Call forwarding is activated for the selected phone number (receive number). This does not result in calls for the other VoIP numbers or your fixed network numbers being forwarded. For information on how to forward calls to your fixed network number → Page 54.

Deactivating call forwarding

⇕	Select the VoIP phone number for which you want to deactivate call forwarding and press **OK**.

All Calls / No Answer / When Busy

	Select and press **OK** (✓ = on).
Off	Select and press **OK**.
☎	Press and **hold** (idle status).

Call forwarding is deactivated.

Please note

Forwarding VoIP phone numbers can result in additional costs. Please consult your VoIP provider.

Activating/deactivating call waiting

Precondition: Your phone will permit two parallel VoIP connections (see Allow 1 VoIP call only → Page 127).

If call waiting is activated, a caller on one of your VoIP connections will hear the ringing tone if you are already on a call using this VoIP connection. This call is announced acoustically and visually on your handset screen.

Calls on the fixed line connection are not signalled as call waiting. They are signalled on other registered handsets to which the fixed line number has been allocated as a receive number. If no other handset is available, the caller will hear the busy tone.

Accepting/rejecting a waiting call → Page 58.

Menu → Select Services → VoIP → For All Calls → Call Waiting

On / Off — Select and press **OK**.

Press and **hold** (idle status).

Please note

The setting applies to all VoIP phone numbers. It does not affect the procedure for call waiting on the fixed line network. For how to activate/deactivate call waiting for the fixed line connection → Page 54.

Functions during a call

Precondition: Your phone will permit two parallel VoIP connections (see Allow 1 VoIP call only → Page 127).

Consultation call

During an external call:

Menu → External Call

Select and press **OK**.

Enter the number (if necessary the suffix) or copy it from the directory.

Press the talk key.

The number is always dialled via VoIP. If no suffix is entered, it is selected via the VoIP send number.

Please note

After a few seconds, the number selected for a consultation call is saved in the last number redial list.

If the participant does not answer:

- Press the display key **End**, to return to the waiting call.

If the participant answers, you have the following options:

- Toggling:
  - Use the control key to toggle between the participants.
  - End call with active participant: **Menu** → End Active Call.
    Press the end call key, to end the current call. The participant who was previously waiting will automatically call you back.
- Conference call:
  - Talk to both participants: **Menu** → Conference Call.
  - End conference call (toggle): **Menu** → End Conference.
  - End call with both participants: press the end call key.
- Call forwarding (provider-dependent):
  You can connect the two external participants.
  Preconditions:
  – You are toggling calls and you phoned the currently active participant yourself.
  – You have activated call forwarding via the Web configurator (→ Page 134).

▶ Press the end call key ☎.

If call forwarding was successful, a message will appear to this effect. The handset will then switch to idle.

If call forwarding was not successful, the participant who was previously waiting will call you back.

Accepting a waiting call

Precondition: Call waiting is activated (→ Page 56).

Menu → Accept Call Waiting

You have the option of toggling or holding a conference call.

Please note

- If the first call was an internal call, the internal connection is ended.
- An internal, waiting call is shown on the display. You can neither accept the internal call nor reject it.
- When receiving an SMS, you will hear a call waiting tone (without screen display).

Using lists

The options are:
- Last number redial list
- SMS list
- Calls list
- Network mailbox

Last number redial list

The last number redial list contains the ten numbers last dialled with the handset (max. 32 numbers). If one of the numbers is in the directory, the corresponding name will be displayed.

Dialling from the last number redial list

[talk key]	Press the key **briefly**.
[control key down]	Select entry.
[talk key]	Briefly press/press and hold the talk key. The number is dialled using the selected connection type (→ Page 40).

Managing entries in the last number redial list

[talk key]	Press the key **briefly**.
[control key down]	Select entry.
Menu	Open menu.

The following functions can be selected with [control key up/down]:

Use Number
: Save or modify a saved number and then dial with [talk key] **or** save as a new entry; to do so, press Menu → Copy to Directory → OK after the number appears on the display.

Copy to Directory
: Copy an entry to the local directory (→ Page 66).

Delete Entry
: Delete selected entry.

Delete List
: Delete complete list.

Opening lists with the message key

You can use the message key ✉ to open the following list selection:

- Calls list
- SMS list → Page 73
- Network mailbox → Page 79

  A separate list is displayed for each network mailbox.

  Precondition: Its number is saved in the base station, it is switched on (→ Page 138) and the corresponding VoIP/fixed line network number is assigned to the handset as a receive number.

An advisory tone sounds as soon as a **new message** arrives in a list. The ✉ key flashes (it goes off when the key is pressed). The message You have new messages appears in the display in **idle status**.

List selection

The lists displayed after pressing the message key ✉ depend on whether there are any new messages.

✉ **key does not flash (no new messages):** All lists are displayed. Select a list with ⬍. To open, press OK.

✉ **key flashes (new messages received):** All the lists that contain new messages are displayed as well as the network mailbox lists, whose connection is assigned to the handset as a receive number (Netz-AB Phone, Netz-AB IP1,...).

Incoming SMS message list

All received SMS messages are saved in the incoming message list → Page 73.

Network mailbox lists

If you select a network mailbox list and press OK, you are connected directly to the network mailbox. For information on the network mailbox → Page 79.

Calls list

The calls list contains the last 20 numbers, depending on the type of list set

- all calls
  - answered calls
  - unanswered calls
- missed calls
  - unanswered calls

Multiple calls from the same number will be stored once in the list of missed calls (the latest call). The number of calls from this number is shown in brackets after the entry.

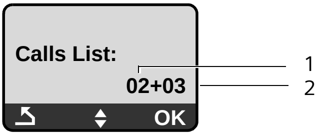
Multiple calls from the same number are stored several times in the list of answered calls.

Please note

Only calls to the receive numbers assigned to your handset are stored in the calls list (→ Page 131).

If no receive numbers are assigned, all calls will be stored in the calls list for all handsets.

The calls list is displayed as follows:



1 Number of new entries
2 Number of old, read entries

Setting the call history type

Menu → Settings → Base → Calls List Type

Missed Calls / All Calls
Select and press OK (✓ = on).

Press and **hold** (idle status).

The calls list entries are retained when you change the list type.

Open the calls list

→ Calls List 01+02
Select and press OK.

Select entry.

The last incoming call is displayed in the **calls list**.

List entry

Example of a list entry:

- Status of entry
  New Call: new missed call.
  Old Call: entry already read.
  Call recv.: call answered (in list type All Calls).
- Entry number
  01/02 means e.g.: first new entry of a total of two new entries.
- Number or name of caller
- Call date and time (if set → Page 90).

Managing entries in the calls list

Precondition:

You have opened the calls list and selected an entry.

Menu Press the display key.

The following functions can be selected with ◆:

Delete Entry
    Delete selected entry.

Copy to Directory
    Copy a displayed number to the directory.

Show Name
    VoIP:
    If a URI was received and stored for a VoIP call, this will be shown. The URI is dialled if you press the talk key. The URI is not entered on the last number redial list.

    Fixed line network: → "Displaying CNIP information" Page 62.

Show Connection
    Display the name of the connection (fixed line network/VoIP account), for which the call was received.

Delete List
    Delete complete list.

Displaying CNIP information

If you have CNIP, you can display the name and town/city that is registered with your network provider for this number.

You have selected a list entry.

Menu → Show Name

If the name and town are not shown, it means that the caller has not requested Calling Line Identification or that Calling Line Identification has been withheld.

Press **OK** to return to the list.

Using directories

The options are:

- the local handset directory
- where applicable, public online directories (country and provider-specific) → Page 67

Local handset directory

You can save up to 150 entries in your handset's local directory (number dependent on the number of individual entries).

You can create a personalised directory for your own individual handset. However, you can send the list or individual entries to other handsets → Page 66.

Please note

For quick access to a number from the directory (shortcut), you can assign the number to a key (→ Page 65).

In the **directory** you can save numbers and corresponding names.

- With the handset in idle status, open the directory by pressing the key.

Length of the entries

Number:	max. 32 digits
Name:	max. 16 characters

Please note

Some VoIP providers do not support local calls for calls to the fixed line network. In this case, always enter the fixed line number with the area code in your directory. Alternatively, you can also use the Web configurator to define an area code, which is automatically prefixed to all numbers that are dialled via VoIP without an area code (→ Page 135).

Saving the first number in the directory

Open the directory.

The display shows Dir. empty New Entry?.

OK Press the display key.

Enter the number and press OK.

Enter the name and press OK.

Storing a number in the directory

→ **Menu** → New Entry

Enter the number and press **OK**.

Enter the name and press **OK**.

Please note

- For information on how to enter IP addresses → Page 42.
- If you want to dial a number using a certain line connection each time, you can add the relevant line suffix to the number in question (Page 40).
- If you add a star (*) at the end of the number, the number will be dialled via the non-standard connection (→ Page 99), even if you briefly press the talk key. Provided that no dialling plan has been defined for the number (→ Page 136).
- You can use the Web configurator to save the directory to a file on your PC, where it can be edited and then sent back to the handset (→ Page 145). Or you can transfer Outlook contacts from the PC to the handset's directory.

Order of directory entries

The directory entries are usually sorted in alphabetical order. Spaces and digits take first priority. The sort order is as follows:

1 Space

2 Digits (0–9)

3 Letters (alphabetical)

4 Other characters

To work round the alphabetical order of the entries, insert a space or a digit in front of the name. These entries will then move to the beginning of the directory. Names that you have prefixed with a star will move to the end of the directory.

Selecting a directory entry

Open the directory.

You have the following options:

- Use to scroll through the entries until the required name is selected.
- Enter the first character of the name and scroll to the entry using if required.

Dialling with the directory

→ (Select entry).

Briefly press/press and hold the talk key. The number is dialled using the selected connection type (→ Page 40).

Please note

You can only dial IP addresses via VoIP.

Managing directory entries

→ (Select entry).

Editing entries

Menu → Edit Entry

Edit the number if required, and press OK.

Edit the name if required, and press OK.

Using other functions

Menu Press the display key.

The following functions can be selected with :

Use Number
Save or modify a saved number and then dial with **or** save as a new entry; to do so, press Menu → Copy to Directory → OK after the number appears on the display.

Delete Entry
Delete selected entry.

Send Entry
Send a single entry to a handset (→ Page 66).

Delete List
Delete **all** directory entries.

Send List
Send complete list to a handset (→ Page 66).

Shortcut
Assign the phone number of the current entry to a number key as a shortcut (shortcut key).

Using shortcut keys

You can assign phone numbers from the local directory to number keys on your handset (→ Shortcut, Page 65). Number keys to which phone numbers are assigned are known as shortcut keys.

- Press and **hold** the required shortcut key.

If there is a valid line suffix at the end of the number in the directory (e.g.: #1), the number will be dialled via the line belonging to the suffix (→ Page 118).

A number with a hash key (#) at the end will be dialled via the default connection.

A number with a star (*) at the end will be dialled via the non-default connection (→ Page 99).

If no suffix is entered, the number will be dialled via the default connection. Exception: A dialling plan has been defined for the number (→ Page 136).

The number is dialled via the default connection. Exception: A dialling plan has been defined for the number (→ Page 136).

Transferring the directory to another handset

Preconditions:

- The sending and receiving handsets must both be registered to the same base station.
- The other handset and the base station can send and receive directory entries.

→ (Select entry) → Menu (Open menu) → Send Entry / Send List

Select the internal number of the receiving handset and press OK.

Successful transmission is acknowledged with the message Entry copied. . You can transfer several individual entries one after the other by responding OK to the Next entry? prompt.

Please note:

- Entries with identical numbers are not overwritten on the receiving handset.
- The transfer is cancelled if the phone rings or if the memory of the receiving handset is full.

Copying a displayed number to the directory

You can copy numbers displayed in a list, e.g. the calls list or last number redial list, to the directory.

If you have CNIP, the first 16 characters of the transmitted name are copied to the Enter name line.

A number is displayed:

Menu → Copy to Directory

- Complete the entry → Page 64.

Copying a number from the directory

There are many operating situations in which you can open the directory, e.g. to copy a number. Your handset need not be in idle status.

Open the directory.

Select entry.

Menu Press the display key.

Select function with .

Using the public online directory

You can use public online directories (= online directories and classified directories, e.g. "Yellow Pages") depending on your provider.

You can define which public online directories you wish to use via the Web configurator (→ Page 143).

Exclusion of liability

Siemens Home and Office Communication GmbH & Co KG assumes no guarantee or liability for the availability of this service. The service may be discontinued at any time.

Opening an online/classified directory

Precondition: The handset is in idle status.

You will find the entries for online directories (e.g. Online Directory) in the local directories of the registered handsets. You can use these entries to access the online directories assigned to your handset (→ Page 143). These entries appear at the top of the directory.

- Open the directory with the button.
- Select an entry from the online directory/classified directory and press the talk key.

A connection to the online directory is established.

Please note

You can establish a connection to your provider's online directory by dialling 1#91:

- When the handset is idle, dial 1#91 and then press the talk key.
- To establish a connection to the classified directory, dial 2#91.
- To establish a connection to the Gigaset.net directory, dial 1188#9.

Searching for an entry

Precondition: You have opened the online directory/classified directory.

There are two types of online directories:

- Online directories that only allow you to search by name (e.g. online classified directories).

  Once the connection is established, you will be prompted to enter a name immediately. The display shows Surname:.

  - Enter the name/trade sector (see below.).
- Online directories that allow you to search for names and numbers.
  - Select Search by Name/Search by Number with and press OK.
  - Enter the name or number (see below).

Enter the name/trade sector you are searching for

Surname: (online directory) / Categor./Name: (classified directory)

Enter the name or part of a name (max. 32 characters) and press OK.

City: Enter the name of the town/city in which the party you are searching for lives and press OK.

The search is launched.

You complete the Surname:, Categor./Name: and City: fields. Entering text → Page 165.

If several towns/cities are listed with the same name, More than one city found. Select city? is displayed:

OK Press the display key.

Select the town. View allows you to view detailed information about the entry.

OK Press the display key to continue the search.

If the town/city entered is not found or if no corresponding name is listed for the town/city, a message to this effect is displayed. You have the following options:

- Press the display key New to start a new search.

Or

- Press the display key Change to change the search criteria. The stated name and town are adopted and can be changed.

If the hit list is too large (more than 99 hits), no hits are displayed. A message detailing the number of hits is displayed. You have the following options:

- Press the display key Refine to start a refined search (→ Page 70).

Or

- Press the display key View. The hit list is displayed (→ Search result (hit list)).

Enter the number you are searching for

Number: Enter the number (max. 32 characters) and press OK.

OK Launch the search.

If the number is not found:

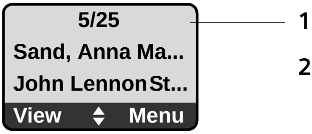
- Press the display key New to start a new search.

Or

- Press the display key Change, to correct the entered number.

Search result (hit list)

The search result is shown as a list on the display. Example:



1. 5/25: Entry number/number of hits
2. Two lines containing the name, industry sector or address of the participant (shortened if necessary)

You have the following options:

- You can scroll through the hit list with .
- Press the display key **View**. Displays the unabridged details of the entry (name, trade sector where applicable, address, telephone numbers). You can scroll through the hit list with .

You have the following additional options via **Menu**:

New search Start a new search.

Refine search

Refine search criteria and restrict list (→ Page 70).

Copy to Directory

Copy the number and name of the entry to the handset's local directory (→ Page 66). If the entry contains several numbers, a directory entry is created for each number. The surname and first name of the entry are copied to the directory name field (shortened if necessary, a maximum of 16 characters are transferred).

Calling participants

Precondition: A hit list is displayed or you have opened the detailed view of an entry (**View** display key).

- Press the talk key , to call the participant.

If the entry only contains one phone number, this is the one that is dialled.

If the entry contains several phone numbers (e.g. fixed line network and mobile numbers), a list of numbers is displayed.

- Select the number with and press the talk key again.

Start detailed search

You can use the search options available in the detailed search (first name and/ or street) to limit the number of hits returned by a previous search.

Precondition: A search result is displayed (hit list with multiple entries or a message indicating too many hits).

Refine — Press the display key.

Or

Menu → Refine search
Select and press OK.

The search criteria from the previous search are adopted and entered in the corresponding fields.

Surname: (online directory) / Categor./Name: (classified directory)
If necessary, change the name/trade sector or extend the partial name and press OK.

Street: If necessary, enter the street name (max. 32 digits) and press OK.

City: If necessary, change the name of the town and press OK.

First name: (only in the online directory)
If necessary, enter the first name (max. 32 characters).

OK — Start detailed search.

Please note

The order in which the search criteria are displayed can be changed independently of the directory.

SMS (text messages)

You can only send and receive SMS messages via the fixed line network.

When an SMS is sent, the base station automatically establishes a connection via the fixed line network.

Your device is delivered ready to send SMS messages as soon as the phone is connected to the fixed line network.

Preconditions:

- Calling Line Identification is enabled for your fixed line network connection.
- Your fixed line network provider supports the SMS service on the fixed line network (information on this is available from your network provider).
- You are registered with your SMS service provider to send and receive SMS messages.

SMS messages are exchanged between SMS centres operated by service providers. You must enter the SMS centre through which you wish to send and receive SMS messages into your phone. You can receive SMS messages from **every** SMS centre that is entered (maximum of four), provided you have registered with your service provider.

Your SMS messages are sent via the **SMS centre** that is entered as the active **send service centre**. However, you can activate any other SMS centre as the active send service centre to send a current message (→ Page 75).

If no SMS service centre is entered, the SMS menu only contains the entry Settings. Enter an SMS Service Centre (→ Page 75).

Information on writing an SMS can be found in the appendix (→ Page 165). An SMS may contain up to 160 characters.

Please note:

- Each incoming SMS is signalled by a single ring (ringer as for external calls). If you accept such a "call", the SMS will be lost. To prevent this ring, suppress the first ring for all external calls (→ Page 76).
- If your phone is connected to a PABX → Page 75.
- To receive SMS messages you must be registered with your SMS service provider.

Writing/sending an SMS

Menu → SMS → Write Message

Write an SMS. For how to enter text, → Page 165.

Menu → Send Text

Select and press OK.

Enter the number with area code (including your local area code) from the directory or key it in manually, and press OK.
The SMS is sent.

Please note

- If you are interrupted by an external call while writing an SMS, the text is automatically saved in the draft message list.
- If the memory is full, or if the SMS function on the base station is being used by another handset, the operation is cancelled. An appropriate message appears in the display. Delete SMS messages you no longer require or send the SMS later.

Draft message list

You can save an SMS in the draft message list, and edit and send it later.

Saving an SMS in the draft message list

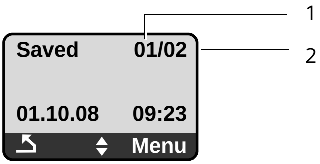
Writing an SMS (→ Page 71).

Menu → Save Text

Opening the draft message list

Menu → SMS → Outgoing

The first entry in the list is displayed, e.g.:



1 Current Number
2 Total number of SMS in the draft message list

Reading or deleting individual SMS

- Open the draft message list.

[▲▼] Select SMS.

Menu → Read SMS

Select and press **OK** to read the SMS. Scroll through the SMS using [▲▼].

Or:

Menu → Delete Entry

Select and press **OK** to delete the SMS.

Writing/changing an SMS

You are reading an SMS in the draft message list.

Menu Press the display key.

You have the following options:

Write Message
Write and then send a new SMS or save.

Use text
Change the text of the stored SMS and then send it.

Deleting draft message list

▶ Opening draft message list.

Menu → Delete List

Select and press OK.

OK — Press the display key to confirm the delete. The list is cleared.

Press and **hold** (idle status).

Receiving an SMS

All received SMS messages are saved in the incoming message list. Since an SMS remains in the list even after it has been read, you should **regularly delete SMS messages from the list** (→ Page 73).

Incoming message list

The incoming message list contains:

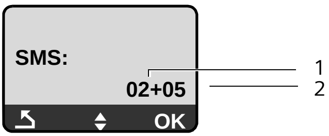
- All received SMS messages, starting with the latest.
- SMS messages that could not be sent due to an error.

New SMS messages are signalled on all Gigaset A58H handsets by a message in the display, the flashing message key and an advisory tone.

Opening the incoming message list with the key

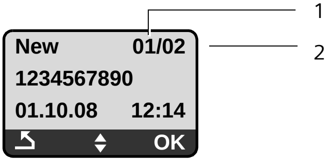
Press.

The incoming message list is displayed as follows (example):



1 Number of new entries
2 Number of old, read entries

An entry in the list is displayed as follows:



1 Current number of SMS currently displayed
2 Total number of new SMS

Opening the incoming message list via the SMS menu

Menu → SMS → Incoming

Reading or deleting individual SMS

▶ Open the incoming message list.

Select SMS.

Menu → Read SMS
Select and press OK to read the SMS. Scroll through the SMS using ⬍.
After you have read a new SMS, its status turns to "old".

Or:

Menu → Delete Entry
Select and press OK to delete the SMS.

Deleting the incoming message list

All **new and old** SMS messages in the list are deleted.

- Opening the incoming message list.

Menu → Delete List
Select and press OK.

OK Press the display key to confirm the delete. The list is cleared.

Replying to or forwarding SMS

You are reading an SMS.

Menu Press the display key.

You have the following options:

Reply
Write and send a reply SMS directly (→ Page 71).

Use text
Edit the text of the SMS and then send it.

Send Text
Forward the text of an SMS to another recipient.

Changing the character set

You are reading an SMS.

The SMS contains symbols and has probably been created with a foreign character set.

Menu Press the display key.

Character Set
Text is shown in the selected character set.

After closing the SMS, the settings are reset.

Adding the number of the message sender to the directory

You are reading an SMS in the incoming message list.

- Pressing the display key Menu.
- Complete the entry, → Page 64.

Setting SMS centres

Entering/changing SMS centre number

- You should find out about the services and special functions offered by your service provider **before you make a new application** and before you delete preconfigured phone numbers.

Menu → SMS → Settings

Service Centres
Select and press OK

Select an SMS centre (e.g. SMS centre #3) and press OK.

SMS Select and press OK.

Enter and change the SMS centre number and press OK.

Press and **hold** (idle status).

To activate an SMS centre as the active send service centre

Normally an SMS is sent via SMS centre # 1.

Menu → SMS → Settings

Service Centres
Select and press OK.

Select an SMS centre (e.g. SMS centre # 3) via which the SMS is to be sent, and press OK.

Active Send Srvc. Cent.
Select and press OK, to activate the SMS centre (✓ = on).

If a different SMS centre was active previously, then this will be deactivated. For SMS centres 2, 3 and 4, the settings only apply to the next SMS. After this, SMS centre # 1 is reset.

SMS on a PABX

- You can only receive an SMS when the **Calling Line Identification** is **forwarded** to the extension of the PABX **(CLIP)**. The CLIP evaluation of the phone number for the SMS centre takes place in your **Gigaset**.
- If required, you must prefix the number for the SMS centre with the access code (depending on your PABX).

  If in doubt, test your PABX, e.g. by sending an SMS to your own number: once with and once without the access code.
- When you send SMS messages, your sender number may be sent without your extension number. In this case the recipient cannot reply to you directly.

Sending and receiving SMS messages **on ISDN PABXs** is only possible via the MSN number assigned to your base station.

Activating/deactivating first ring muting

Menu	Press the display key.
7pqrs 4ghi 9wxyz 1 9wxyz	Press keys.
0+ **OK**	Make the first ring audible.
Or:	
1 **OK**	Mute the first ring.

Activating/deactivating the SMS function

If you deactivate the SMS function, you cannot send or receive any SMS messages with your phone.

The settings you have made for sending and receiving SMS messages (e.g. the numbers of the SMS centres) and the entries in the incoming and draft message lists are saved even after you turn off your phone.

Menu	Press the display key.
7pqrs 4ghi 9wxyz 2abc 6mno	Enter the digits.
0+ **OK**	Deactivate SMS function.
Or:	
1 **OK**	Activate SMS function (default setting).

SMS troubleshooting

Error codes when sending

E0	Calling Line Identification (CLIR) permanently withheld or Calling Line Identification is not enabled for the fixed line network.
FE	Error occurred during SMS transfer.
FD	Connection to SMS centre failed; see self-help.

Self-help with errors

The following table lists error situations and possible causes and provides notes on troubleshooting.

You cannot send messages. 1. SMS transmission has been interrupted (e.g. by a call). ▸ Re-send the SMS. 2. The network provider does **not** support this feature. 3. No number or an invalid number is entered for the SMS centre set as the active send service centre. ▸ Enter the number (→ Page 75). 4. Fixed line network: You have not requested the Calling Line Identification Presentation (CLIP) service. ▸ Ask your fixed line network provider to enable this service.
You receive an incomplete SMS. 1. Your phone's memory is full. ▸ Delete old SMS messages (→ Page 73). 2. The SMS provider has not yet sent the rest of the message.
The SMS is played back. 1. The "display call number" service is not activated. ▸ Ask your fixed line network provider to enable this feature (there is a charge for this). 2. Your mobile phone operator and your fixed line network SMS service provider have not agreed on a co-operation. ▸ Obtain information from your fixed line network SMS service provider. 3. Your terminal has been recorded by your SMS provider as having no fixed line network SMS functionality, i.e. you are no longer registered with the provider. ▸ Have the device (re-)registered to receive SMS messages.
Messages are only received during the day. The terminal is recorded in your SMS provider's database as having no fixed network SMS functionality, i.e. you are no longer registered with the provider. ▸ Obtain information from your fixed line network SMS service provider. ▸ Have the device (re-)registered to receive SMS messages.

Using the network mailbox

Some fixed network providers and VoIP providers offer answer machines on the network – these are known as network mailboxes.

Each network mailbox accepts incoming calls made via the corresponding line (fixed line network or corresponding VoIP phone number). To record all calls, you should therefore set up network mailboxes for both the fixed line network and for each of your VoIP connections.

You can activate/deactivate the network mailboxes for your VoIP connections using the Web configurator. To do this, you only require the network mailbox phone number.

For information on how to activate/deactivate network mailboxes via the Web configurator and to change their assigned numbers, → Page 138.

You need to have **requested** the network mailbox for your fixed line network connection from your fixed line network provider. You can store the phone number for the fixed line network mailbox in the Web configurator on the base station (→ Page 138).

You cannot activate/deactivate the network mailbox for the fixed line network connection via the Web configurator. For how to activate/deactivate the network mailbox for the fixed line network connection please refer to the fixed line network provider's information.

Please note

- For many VoIP network mailboxes, the phone number is automatically saved on the base station when the general VoIP provider data is downloaded.
- If you have registered a Gigaset C47H, S67H or S68H handset to your base station, you can also enter and activate the network mailbox via this handset. For further information on this, read the user guide for the Gigaset C470 IP or S675 IP on the Internet.

Configuring the network mailbox for fast access

With fast access you can dial a network mailbox directly.

Assigning key 1 of the handset, changing assignments

The setting for fast access is handset-specific. You can assign a different mailbox to key [1] on each registered handset.

No mailbox is preconfigured for fast access in the default settings.

Preconditions:

- At least one receive number is assigned to the handset.
- The corresponding network mailbox has been entered and activated for at least one of the receive numbers on the handset.

[1] Fast access has not yet been set on the handset:
Press and hold key [1].

Or:

Menu → Voice Mail → Set Key 1

Select and press OK.

Select network mailbox and press OK (✓ = on).

The selection includes the network mailboxes that belong to a VoIP/fixed line network connection and whose phone number is assigned to the handset as a receive number. Net AM: xxx where xxx is replaced by the standard name in the connection (IP1 to IP6, Phone).

If a number is already saved to the base station for the selected network mailbox, fast access is activated.

Press and **hold** (idle status).

If no number has been saved for the network mailbox the message Not possible! is displayed. You then have to first enter the mailbox number using the Web configurator (→ Page 138).

Please note

You can only assign fast access to one mailbox.

However, you can also call the network mailbox assigned to a handset's receive numbers directly via the message key (→ Page 79).

Calling the network mailbox and listening to messages

Press and **hold**.

If you have set a network mailbox for fast access you will be connected directly to this network mailbox (external call).

If necessary, press the handsfree key.

You will hear the answer machine announcement.

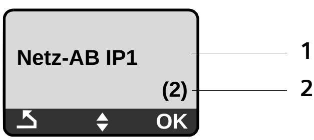
Listening to messages on the network mailbox

Under the message key you will find a list for each network mailbox that fulfils the following requirements:

- The corresponding connection is allocated to the handset as a receive number.
- The network mailbox phone number is saved on the base station.
- The network mailbox is activated (activated → Page 138).

You can use the list to call the network mailbox directly and listen to the messages.

When you press the message key, the following is displayed (example):



1 Network mailbox name. Netz-AB IP1, ..., Netz-AB IP6 or Netz-AB Phone is displayed. IP1 ... are the standard names of the corresponding connections. The default names are always displayed irrespective of which connection name you specified when configuring via the Web configurator.

2 The number of new messages is displayed (2 = two new messages). If there are no new messages, (0) is displayed. The number of messages stored in the network mailbox is not displayed.

Displaying new messages on the handset

If a new message is present on one of the network mailboxes that is assigned to the handset via its receive number, the message key ✉ flashes.

Calling the network mailbox and checking messages

✉ Press the message key.

Netz-AB Phone / Netz-AB IP1 / ... / Netz-AB IP6

Select the network mailbox and press **OK**.

You are connected directly to the network mailbox (external call) and hear its announcement.

Please note

- The network mailbox is automatically called via the corresponding connection. An area code predefined for your phone is **not** prefixed.
- After the call, the number of new messages on the handset returns to (0), even if not all or no new messages have been listened to.

Messages can generally be played back using your handset keypad (digit codes). Listen to the announcement.

You need to define how the digit codes for VoIP should be converted to DTMF signals and transmitted. This setting should be made via the Web configurator → Page 132.

Ask your VoIP provider which type of DTMF transmission it supports.

ECO DECT: Reducing the power consumption and transmission power of the base station

The base station of your phone is an ECO DECT base station, this means that:

- The base station and the charging cradle use less power because they are equipped with a power-saving mains adaptor.
- The reduction of the handset's transmission power is dependent on the handset's proximity to the base station.
- The base station can also be switched to Eco mode. Eco mode reduces the transmission power and the power consumption of the base station further.

  Eco mode means:
  80% reduction of the transmission power in standby operation and when making calls. Eco mode is available when the handset(s) and the base station are close together, e.g. when the phone is being used in an office.

  The setting can be made on the handset.

Activating/deactivating Eco mode

Menu → Settings → Base → Additional Features

Eco mode Select and press OK (✓ = on).

Reception display

Screen icon	Reception strength:
(signal icons, 4 levels) (flashes)	– good to low – no reception

Please note

- Activating **Eco mode** reduces the range of the base station.
- **Eco mode** and any repeater support (→ Page 99) cancel each other out, i.e. both functions cannot be used at the same time.

Setting the alarm clock

Precondition: The date and time have already been set (→ Page 14).

Activating/deactivating the alarm clock

Menu → Alarm Clock

Activation Select and press OK (✓ = on).

After you activate the alarm clock, the menu for setting the wake-up time opens automatically.

If necessary, enter the wake-up time in 4 digits (hours and minutes) and press OK.

Press and **hold** (idle status).

If the alarm clock is set, the wake-up time appears with the ⏰ icon in the display instead of the date.

Changing the wake-up time

Menu → Alarm Clock → Wake up time

Enter the 4-digit wake-up time (in hours and minutes) and then press OK.

Press and **hold** (idle status).

When the alarm clock rings...

A wake-up call with the selected ringer tone is signalled on the handset (→ Page 94). The wake-up call lasts for a maximum of 5 minutes. If no key is pressed, the wake-up call is repeated twice at five minute intervals and then turned off for 24 hours.

During a call, the wake-up call is only signalled by a short tone.

Alarm repeated after 5 minutes (snooze mode)

Snooze Press the display key or any other key (apart from the left display key).

If you press Snooze three times, the alarm clock switches off for 24 hours.

Turning off the alarm clock for 24 hours

Off Press the display key.

Using several handsets

Registering handsets

You can register up to six handsets to your base station.

When you register a new Gigaset handset, the base station transfers the corresponding entries to its local directory to enable you to use online directories on your new handset.

- Online Directory (→ Page 67) for the public online directory.
- Yellow Pages (→ Page 67) for the classified directory.
- Gigaset.net for the Gigaset.net directory (→ Page 48)

Precondition: The handset can send and receive directory entries (see handset user guide).

Successful registration is acknowledged with the message Data Transfer x entries rec for this reason.

Please note

- If several handsets are registered on your base station, you can simultaneously make two calls via the Internet and one via the fixed line network. Up to two additional internal connections are also possible.
- Selecting the connection via the talk key (→ Page 40) is not supported on GAP handsets. If you enter a phone number without a line suffix and without defining a dialling plan for the phone number, it will be dialled via the default connection (→ Page 99). If you enter a "*" (star) at the end of the phone number, it will be dialled via the non-default connection.
- After registration, all the phone numbers for the phone will be assigned to the handset as receive numbers. It will use the fixed line network number and the first VoIP number in the configuration as send numbers.
  For how to change the assignments → Page 131.

Registering another Gigaset A58H handset on the Gigaset A580 IP

Before you can use your handset, you must register it to the base station.

You must initiate handset registration on the handset and on the base station.

The handset will return to idle status if registration was successful. The handset's internal name is shown in the display, e.g. INT 1. If it does not appear, repeat the procedure. Registration can take up to one minute.

- **On the handset: Select** Menu → Settings → Handset → Register Handset and press OK.
- Enter the system PIN of the base station (the default is 0000) and press OK. Handset is registering flashes on the display.
- Within the next 60 seconds press and **hold** (for approx. 3 seconds) the registration/paging key (→ Page 2) **on the base station**.

The handset is assigned the lowest available internal number (1-6). The internal number appears in the display after registration, e.g. INT 2. This means that the internal number 2 has been assigned to the handset.

Please note

If six handsets are already registered to the base station, there are two options:

- The handset with the internal number 6 is in idle status: the handset you wish to register is assigned the number 6. The handset that was previously number 6 is de-registered.
- The handset with the internal number 6 is being used: the handset you wish to register cannot be registered.

Registering other handsets on the Gigaset A580 IP

You can register other Gigaset handsets and handsets for other devices with GAP functionality as follows.

- Start the registration procedure **on the handset** in accordance with the handset's operating instructions.
- Press and hold (for approx. 3 seconds) the registration/paging key (→ Page 2) **on the base station**.

De-registering handsets

You can de-register any registered handset from any registered Gigaset A58H handset.

Open list of internal participants.
The handset which you are using is marked with **<**.

Select the handset to be de-registered.

Menu — Press the display key.

De-register Handset

Select and press OK.

Enter the system PIN of the base station (the default is 0000) and press OK.

De-register handset?

Press OK, to confirm the prompt.

Press and **hold** (idle status).

The handset is de-registered immediately, even if it is not in idle status.

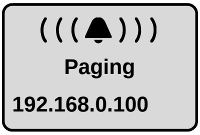
Locating a handset ("paging")

You can locate your handset using the base station.

- **Briefly** press the registration/paging key on the base station (→ Page 2).
  All handsets will ring at the same time ("paging"), even if ringer melodies are switched off.

The current (local) IP address for the base station appears in the handset displays.

Example:



Ending paging

- **Briefly** press the registration/paging key on the base station (→ Page 2).

Or

- Press the talk key or end call key on any handset.

Or

- Do not press any key on the base station or handset:
  After approx. 30 seconds, the paging call will end **automatically**.

Please note

- An incoming external call will not interrupt the paging process.
- If there is an internal connection between the two handsets, paging is not possible.

Changing a handset's internal number

A handset is **automatically** assigned the lowest free number when it is registered. In the list of internal participants, the handsets are sorted according to their internal number.

You can change the internal number of all registered handsets (1–6). The numbers 1-6 can only be assigned once each.

INT — Open the list of registered handsets.

Menu — Press the display key.

Edit Handset Number

Select and press OK.

Select handset.

Enter the new internal number (1–6). The handset's old number is overwritten.

If necessary, select another handset and change its number.

After all the changes are completed:

OK — Press the display key to save the input.

Press and **hold** (idle status).

You will hear the error tone if an internal number has been allocated twice.

- Repeat the procedure with a free number.

Changing the name of a handset

The names "INT 1", "INT 2" etc. are assigned automatically on registration. You can change these names. The changed name is displayed in every handset's list.

INT — Open the list of registered handsets.

Select handset.

Menu — Press the display key.

Change Handset Name

Select and press OK. The handset's current name is displayed.

<C — Delete the old name if necessary.

Enter new name (max. 10 characters) and press OK.

Press and **hold** (idle status).

Please note

If you delete the current handset name with <C and then press OK without entering a new name, the handset will automatically be allocated the standard name "INT x" (x= internal number).

Making internal calls

Internal calls to other handsets registered on the same base station are free.

Calling a specific handset

INT — Open the list of registered handsets.

Select handset.

Press the talk key.

Or:

INT — Open the list of registered handsets.

Enter the internal handset number (1–6).

Calling all handsets ("group call")

INT Open the list of registered handsets.

* Press the star key. All handsets are called.

Or:

INT Open the list of registered handsets.

Call All Select.

Press the talk key. All handsets are called.

Ending a call

Press the end call key.

Please note

- You can reject an internal call by pressing the end call key. Other handsets will continue to signal an internal "group call".
- If the called handset is not answered, the busy tone sounds after approx. 3 minutes.

Transferring a call to another handset

You can forward (connect) an external call, made via the fixed line network or VoIP, to another handset.

Precondition: You are conducting an external call.

INT / INT Open the list of registered handsets.

Select handset or All.

OK / Press the display key or the talk key.

Or:

INT / INT Open the list of registered handsets.

Enter the internal number of the handset.

The external participant hears hold music if activated (→ Page 99).

When the internal participant answers:

- If necessary announce the external call.

Press the end call key.

The external call is transferred to the other handset.

If the internal participant does **not** respond or is busy:

Menu → Back

Select and press OK.

You are reconnected with the external participant.

You can also press the end call key when forwarding a call before the internal party picks up the call.

Then, if the internal participant does not answer or the line is busy, the call will automatically return to you (the display will show Recall).

Initiating internal consultation call, conference call

You are talking to an **external** participant (via fixed line network or VoIP) and can call an **internal** participant at the same time to hold a consultation call.

Precondition: You are conducting an external call.

INT / INT — Open the list of registered handsets.

Select handset or All.

OK / — Press the display key or the talk key.

Or:

INT / INT — Open the list of registered handsets.

Enter the internal number of the handset.

The external participant hears hold music if activated (→ Page 99).

When an internal participant answers you can speak to them.

You have the following options:

Ending a consultation call

Menu → Back

Select and press OK.

You are reconnected with the external participant.

Initiating a conference call

Menu → Conference Call

Select and press OK.

You are in a three-way conference call with the external participant and the internal participant.

During an internal consultation/conference call

If the internal participant who has been called ends the call (press end call key), you will be reconnected with the external participant.

If you press the end call key , the external call will be transferred to the internal participant (→ "Transferring a call to another handset" Page 87).

Accepting/rejecting call waiting during an internal call

If you receive an **external** call during an **internal** call, you will hear the call waiting tone (short tone). With Calling Line Identification, the caller's number or name will appear in the display.

Ending an internal call, accepting an external call

Menu → Accept Call Waiting
Select and press OK.

The internal call is **ended**. You are connected to the external caller.

Rejecting an external call (only possible for calls to your fixed line network number)

Menu → Reject Call Waiting
Select and press OK.

The call waiting tone is turned off. You remain connected with the internal participant. The ringer melody can still be heard on the other registered handsets.

Handset settings

Your handset is preconfigured, but you can change the settings to suit your individual requirements.

Changing the date and time

Please note

The address of a time server on the Internet is stored on your telephone. The date and time are taken from this time server provided that the base station is connected to the Internet and synchronisation with the time server is activated (→ Page 149).
Manual settings are overwritten in this case.

To manually change the time, open the input field with:

Menu → Settings → Date/Time
Select and press **OK**.

Enter Date: Enter the day, month and year in 8-digit format, e.g. [0] [1] [1] [0] [2] [0] [0] [8] for 01/10/2008.

Enter Time: Enter the hours and minutes in 4-digit format, e.g. [0] [7] [1] [5] for 07:15 AM.

OK Press the display key

Changing the display language

You can view the display texts in different languages.

Menu → Settings → Handset → Language
Select and press **OK**.

The current language is indicated by a ✓.

Select a language and press **OK**.

Press and **hold** (idle status).

If you accidentally choose a language you do not understand:

Menu [7] [3] [2]
Press keys in sequence.

Select the correct language and press **OK**.

Activating/deactivating the screensaver

You can have a screensaver displayed on the handset. The screensaver replaces the display screen when the handset is in idle status. It hides the date, time and internal name.

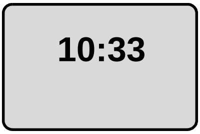
You have the following options:

No Screensaver

The screensaver is deactivated. The idle display status (→ Page 1) is displayed.

Digital Clock

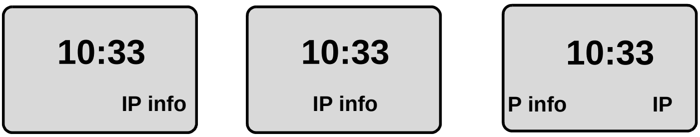
Approx. 10 seconds after the handset returns to idle status, a digital clock appears on the display.



Info Services

Precondition: You have activated the display of info services via the Web configurator (→ Page 142).

Approx. 10 sec after the handset returns to idle status a digital clock and (if available) text information from the Internet appears in the display (e.g. weather reports, RSS feeds). The text information is displayed below the clock as a scrolling message.



The text information is initially displayed once. Then only the digital clock is displayed.

The text information is displayed again when:

- New information is received
- You remove the handset from the charging cradle or place it in the charging cradle
- You press any key on the handset

The display backlight switches itself on.

The text information can be made up of separate pieces of information (→ Page 142). The weather report is preset.

Please note

- If you have set the screensaver Info Services, and you want to make a call or change settings on the handset, you may, if necessary, need to repeat the first key press (e.g. press the Menu key twice to open the main menu). The first key press activates the display of text information.
- If you have not activated the display of info services (→ Page 142), only the digital clock is displayed on the screensaver Info Services.
- The screensaver is not displayed in certain situations, e.g. during a call or if the handset is de-registered.
- If the screensaver conceals the display, press the end call key **briefly** to show the idle display with time and date.

Setting the screensaver

Menu → Settings → Handset → Display → Screensaver

The current setting is displayed.

No Screensaver / Digital Clock / Info Services

Select and press OK. A brief preview of the selected screensaver is displayed. The selection is marked with ✓.

Press and **hold** (idle status).

The selected screensaver is displayed after approx. 10 seconds.

Quick access to functions

The →→ function (open the last number redial list) is preset on the left display key of your handset. You can change the configuration, i.e. assign a different function to the display key.

To start the function, you then simply need to press the button.

Changing the assignment of the display key

- When the handset is in idle status, press and **hold** the left display key.
- Select a function with the control key and press OK.

The following features are available:

- INT (INT)
  Open the list of internal participants with the INT key.
- SMS (SMS)
  Opens the SMS submenu for writing, sending and reading SMS messages (→ Page 71): Menu → SMS
- Fixed Line call (Phone)
  Opens the pre-dialling option for making a call via the fixed line network.
- IP call (IP)
  Opens the pre-dialling option for making a call via VoIP.

Activating/deactivating auto answer

If you activate this function, when you receive a call you can simply lift the handset out of the charging cradle without having to press the talk key.

Menu → Settings → Handset

Auto Answer

Select and press OK (✓ = on).

Press and **hold** (idle status).

Changing the handsfree/earpiece volume

You can adjust the volume for handsfree talking to one of five settings (1-5, e.g. volume 3 =) and the earpiece volume to one of three (1-3, e.g. volume 2 =).

Setting the volume during a call

The setting applies to the current mode (earpiece or handsfree).

You are conducting an external call.

Press up on the control key.

Set the volume.

The setting will automatically be saved after approx. 3 seconds, if not then press the display key OK.

If is assigned with another function, e.g. toggling (→ Page 57):

Menu Open menu.

Volume Select and press OK.

Set the volume.

Adjusting the volume via the menu

Briefly press up on the control key.

Call Volume

Select and press OK.

Earpiece Volume / Handsfree Volume

Select and press OK.

Adjust the volume and press OK.

Press and **hold** (idle status).

Please note

You can also adjust the volume via Menu → Settings → Audio Settings → Call Volume.

Setting ringers

- Volume:
  You can choose between five volumes (1–5; e.g. volume 3 =) and the "crescendo" ring (6; volume increases with each ring =).
- Ringer melodies:
  You can select various ringer form a list of pre-loaded ringer melodies. The first three melodies are the "classical" ring melodies.

You can set different ringer melodies for the following functions:

- External Calls: for external calls
- Internal Calls: for internal calls
- Alarm Clock: for the alarm clock

Setting the ringer volume

The ringer volume is the same for all types of ring.

When the handset is in idle status:

Briefly press up on the control key.

Ringer Volume
Select and press OK.

Adjust the volume and press OK.

Please note

You can also adjust the ringer volume via Menu → Settings → Audio Settings → Ringer Volume.

Setting the ringer melody

Set different ringer melodies for external calls, internal calls and the alarm clock.

Briefly press up on the control key.

Ringer Melody
Select and press OK.

External Calls / Internal Calls / Alarm Clock
Select and press OK.

Select melody (✓ = on) and press OK.

Press and **hold** (idle status).

Please note

You can also adjust the ringer melody via Menu → Settings → Audio Settings → Ringer Melody.

Activating/deactivating the ringer

You can deactivate the ringer on your handset before you answer a call or when the handset is in idle status; the ringer can be deactivated permanently or just for the current call. The ringer cannot be re-activated while an external call is in progress.

Deactivating the ringer permanently

[* ♫] Press **and hold** the star key until the icon appears in the display.

Reactivating the ringer

[* ♫] Press and **hold** the star key in idle status.

Deactivating the ringer for the current call

Menu Open menu.

Silent Select and press OK.

Activating/deactivating advisory tones

Your handset uses various advisory tones to tell you about different activities and statuses.

- **Key click**: every key press is confirmed.
- **Confirmation tone** (ascending tone sequence): at the end of an entry/setting and when an SMS or a new entry arrives in the calls list
- **Error tone** (descending tone sequence): when you make an incorrect entry
- **Menu end tone**: when scrolling to the end of a menu

You cannot deactivate the confirmation tone for placing the handset in the base station.

Briefly press up on the control key.

Advisory Tones

Select and press OK (✓ = on).

Press and **hold** (idle status).

Setting the battery low tone

The **battery warning tone** advises that the batteries need to be charged. You can activate it, deactivate it or decide whether or not it should sound during a call.

Briefly press up on the control key.

Battery Low

Select and press OK.

On / Off / During Call

Select and press OK (✓ = on).

Press and **hold** (idle status).

Restoring the handset default settings

Each individual handset setting is reset, in particular the language, display, volume, ringer melodies and alarm clock settings (→ Page 90). The last number redial list is cleared.

This will not affect entries in the directory, calls list or SMS lists, or the handset's registration to the base station.

Menu → Settings → Handset → Reset Handset

OK Press the display key.
You can cancel the factory reset by pressing the end call key or the display key (back).

End call key: Press and **hold** (idle status).

Setting the base station via the handset

The base station settings are carried out using a registered Gigaset A58H handset.

Some settings can also be carried out via the base station Web configurator. Look out for the 💻 icon.

Protecting against unauthorised access

Protect the system settings of the base station with a PIN known only to you. The system PIN must be entered, for example, when activating and deactivating the handset, when changing the VoIP settings, with firmware updates, when resetting the base station default settings and for launching the Web configurator.

Changing the system PIN

You can change the 4-digit system PIN set on the base station (default setting: 0000) to a 4-digit PIN known only by you.

Menu → Settings → Base → System PIN

⌨	Enter the current system PIN and press OK.
⌨	Enter your new system PIN and press OK.
⌨	Now re-enter the new system PIN and press OK.

For security reasons, "****" is displayed instead of the system PIN.

☎	Press and **hold** (idle status).

Resetting the system PIN

If you have forgotten your system PIN, you can reset the base station to the original PIN **0000**: To do this, you must reset your base station using the key on the base station (→ Resetting the base station using a key on the base station on Page 98).

Please note that this will restore all other base station settings too (→ Page 98).

Restoring the base station to the factory settings

Resetting the base station via the menu

Each individual setting is reset, in particular:

- VoIP settings such as VoIP provider and account data and DTMF settings (→ Page 101, Page 117, Page 132)
- Settings for the local network (→ Page 103, Page 115)
- Default connection (→ Page 99)
- The names of the handsets (→ Page 86)
- SMS settings (e.g. SMS centres → Page 71)
- Eco mode is deactivated
- PABX connection settings (→ Page 106)

SMS lists, calls list are deleted.

The following are **not** reset:

- Date and time
- System PIN

The handsets are still registered.

Menu → Settings → Base → Base Reset

(keypad)	Enter the system PIN and press OK.
OK	Press the display key to confirm.

Resetting the base station using a key on the base station

As with resetting the base station via the menu, all individual settings are reset. The **system PIN will also be reset to "0000"** and **all handsets registered above and beyond the scope of delivery will be de-registered**.

Please note

For how to reregister the handsets after resetting, where applicable → Page 83.

- Remove the cable connections from the base station to the router (→ Page 20) and fixed line network (→ Page 19).
- Remove the base station mains adapter from the socket (→ Page 19).
- **Press and hold** the registration/paging key (→ Page 2).
- Plug the mains adapter back into the power socket.
- Press and hold the registration/paging key (at least 10 sec.).
- Release the registration/paging key. The base station has now been reset.

You will then need to "prepare to use" the base station again, i.e. re-establish the cable connections to the fixed line network and the router and make the settings for VoIP telephony (→ Page 21).

Activating/deactivating music on hold

Menu → Settings → Base

Music on hold

Select and press OK to activate or deactivate the hold music (✓ = on).

Activating/deactivating repeater mode

With a repeater you can increase the range and reception strength of your base station. You will need to activate repeater mode. This will terminate any calls being made via the base station at that time.

Precondition: A repeater is registered with the base station.

Menu → Settings → Base → Additional Features

Repeater Mode

Select and press OK (✓ = on).

OK Press the display key to confirm the security prompt.

Please note

Repeater support and Eco mode (→ Page 81) cancel each other out, i.e. both functions cannot be used at the same time.

Setting default connection

You can make settings according to whether you want to make calls via VoIP or the fixed line network as standard.

Please note

The default connection is only relevant when dialling numbers that are not subject to dialling plans (→ Page 136) and are entered without a line suffix (→ Page 40).

Menu → Settings → Base → Telephony → Default Line Type

IP / Phone

Select and press OK (✓ = on).

When making calls:

- **Briefly** press the talk key if you want to make a call via this default connection.
- Press and **hold** the talk key if you want to make the call via the other connection type.

For how to make the setting on the Web configurator → Page 130.

Updating the base station firmware

If necessary, you can update your base station firmware.

The firmware update is downloaded directly from the Internet by default. The relevant Web page is preconfigured in your phone.

Precondition:

The base station is in idle status, i.e.:

- No calls are being made via the fixed line network or VoIP.
- There is no internal connection between registered handsets or to GHC devices.
- No other handset has opened the base station menu.

Starting the firmware update manually

Menu → Settings → Base

Firmware Update
Select and press OK.

Enter the system PIN and press OK.

The base station is connected to the Internet.

Yes Press the display key to start the firmware update.

Please note

- The firmware update can last up to 3 minutes.
- When updating from the Internet, checks are made to ensure that no newer version of the firmware exists. If this is not the case, the operation is terminated and a message is issued to that effect.

Automatic firmware update

Your phone will check daily whether a newer firmware update is available via the Internet on the Siemens configuration server. If this is the case, the message New firmware available will be displayed when the handset is in idle status, and the message key flashes.

Press the message key.

Yes Press the display key to confirm the prompt.

The firmware will be loaded onto your phone.

Please note

If you respond to the prompt with No, the display will not be repeated. The message New firmware available will only be shown again if a newer version of the firmware than the one rejected is available.

You can deactivate the automatic version check via the Web configurator (→ Page 149).

Making VoIP settings on the handset

In order to be able to use VoIP, you must set a few parameters for your phone.

You can make the following settings using your handset.

- ◆ Download the general access data for your VoIP provider from the Siemens configuration server and store them on your phone.
- ◆ Enter your personal access data for your first VoIP account (first VoIP phone number). You can configure the access data for five further VoIP accounts via the phone's Web configurator.
- ◆ Set the phone's IP address in the LAN.

The connection assistant on your phone can help you make the settings.

You can set these and other parameters conveniently via the Web configurator on a PC connected to your local network (→ Page 108).

Using the connection assistant

The connection assistant starts automatically when you set the handset and base station up for the first time, or when you try to connect to the Internet before making the necessary settings.

You can also start the connection assistant via the menu:

Menu → Settings → Base → Telephony

Connection Assistant

Select and press OK.

Enter the system PIN and press OK.

For how to enter VoIP settings using the connection assistant → Page 21.

Changing settings without the connection assistant

You can change your provider's VoIP settings and the VoIP user data via the menu without starting the connection assistant.

Downloading your VoIP provider's settings

The general settings for various VoIP providers are available to download on the Internet. The relevant Web page is preconfigured in your phone.

Precondition: Your phone is connected to the Internet.

Menu → Settings → Base → Telephony → VoIP

Enter the system PIN and press OK.

Select VoIP Provider

Select and press OK.

The phone establishes a connection to the Internet.

Select country and press OK.

Select your VoIP provider and press OK.

Your VoIP provider data is downloaded and saved in your phone.

If your VoIP provider is not included in the list, you need to enter or adjust the general VoIP settings manually via your phone's Web configurator → Page 118.

Please note

If an error occurs during the download, an error message will be displayed. You can find possible messages and measures in the table on Page 153.

Automatic update for the VoIP provider settings

After the first download of the VoIP provider settings, your phone will check daily whether a newer version of the file for your VoIP provider is available via the Internet on the Siemens configuration server. If this is the case, the message New profile available will be displayed when the handset is in idle status, and the message key flashes.

[message key]	Press the message key.
Yes	Press the display key to confirm the prompt.
[keypad]	Enter the system PIN and press OK.

The new data for your VoIP provider will be downloaded and saved on the phone.

If you reply to the prompt with No, the display will not be repeated. The message New profile available will only be shown again if a newer version of the VoIP settings than the one rejected is available.

You can deactivate the automatic version check via the Web configurator (→ Page 149).

Entering/changing VoIP user data

You must complete the VoIP settings with your personal data. You will receive all necessary data from your VoIP provider.

Menu → Settings → Base → Telephony → VoIP

[keypad]	Enter the system PIN and press OK.

Select Username / Authentication Name / Authentication Password one after another and press OK.

[keypad]	Enter/change the user data and press OK. When making these entries, please remember the VoIP user data is case sensitive. To enter text → Page 165.

Enter the caller ID for your VoIP provider account as the Username. The Username is usually identical to your Internet phone number (the first part of your SIP address → Page 121).

For Authentication Name and Authentication Password enter the provider-dependent access data that has to be transferred by the phone to the SIP service at registration.

Please note

A previously set password is not displayed.

Setting the phone's IP address in LAN

The base station requires an IP address in order to be "recognised" by the LAN (the router).

The IP address can be assigned to the base station automatically (by the router) or manually.

- If performed **dynamically,** the router's DHCP server automatically assigns the base station an IP address. The base station's IP address can be changed according to router settings.
- If performed manually/**statically**, you assign the base station a static IP address. This may be necessary depending on your network configuration.

For information on how to make the local network settings on the Web configurator → Page 115.

Activating/deactivating dynamic assignment

Menu → Settings → Base → Local Network

Enter the system PIN and press OK.

Dynamic IP address (✓ = on)

Select and press OK to change the current settings.

If you deactivate dynamic assignment, you must set the IP address and subnet mask of the base station, the standard gateway and DNS server manually. A corresponding message is displayed.

Please note

To assign the IP address dynamically, the DHCP server on the router must be activated. Please also read the user guide for your router.

Viewing/changing the base station IP address

You can only change the IP address if you have deactivated dynamic assignment.

192.168.2.2 has been preset by default.

Menu → Settings → Base → Local Network

Enter the system PIN and press OK.

IP Address

Select and press OK.

The current IP address is displayed.

If necessary, enter the IP address and press OK.

For information on the IP address, please see Page 115 and the glossary on Page 185.

Viewing/changing the subnet mask

You can only change the subnet mask if you have deactivated dynamic assignment.

255.255.255.0 has been preset by default.

Menu → Settings → Base → Local Network

Enter the system PIN and press OK.

Subnet Mask

Select and press OK.

The current subnet mask is displayed.

If necessary, enter the subnet mask and press OK.

For information on the subnet mask, please see Page 115 and the glossary on Page 190.

Viewing/changing the DNS server

Enter the IP address for the preferred DNS server. The DNS server (Domain Name System) converts the symbolic name of a server (DNS name) into the public IP address for the server when the connection is made.

You can specify your router's IP address here. The router forwards phone address requests to its DNS server.

192.168.2.1 has been preset.

Menu → Settings → Base → Local Network

Enter the system PIN and press OK.

DNS Server

Select and press OK.

If necessary, enter the IP address of your preferred DNS server and press OK.

Viewing/changing the default gateway to the Internet

Enter the IP address for the standard gateway, by means of which the local network is connected to the Internet. This is generally the local (private) IP address for your router (e.g. 192.168.2.1). Your phone requires this information to be able to access the Internet.

192.168.2.1 has been preset.

Menu → Settings → Base → Local Network

Enter the system PIN and press OK.

Default Gateway

Select and press OK.

If necessary, enter the IP address of the standard gateway and press OK.

Activating/deactivating display of VoIP status messages

If the function is activated, a VoIP status code for your service provider is displayed.

Activate the function if, for example, you have problems with VoIP connections. You will receive a provider-specific status code, which supports the service when the problem is analysed. You will find a table with the possible status screens in the appendix (→ Page 157).

Menu → Settings → Base → Telephony → VoIP

Enter the system PIN and press OK.

Status on HS

Select and press OK (✓ = on).

Press and **hold** (idle status).

For how to make the setting on the Web configurator → Page 151.

Checking the base station MAC address

Depending on your network configuration, you may have to enter your base station MAC address, for example, into your router's access control list. You can check your base station MAC address:

Menu 7pqrs 4ghi 9wxyz 7pqrs 5jkl

The base station MAC address is displayed.

Press and **hold** (idle status).

For information on how to check your MAC address on the Web configurator → Page 151.

Operating the base station on the PABX

The following settings are only necessary when your PABX requires them; see the PABX user guide. The settings only affect fixed line network connections.

You cannot send or receive SMS messages on PABXs that do not support Calling Line Identification.

Changing the dialling mode

You can set the dialling mode.

Menu → Settings → Base → Telephony → Phone → Dialling Mode

Tone / Pulse

Select and press OK (✓ = on).

Press and **hold** (idle status).

Please remember

◆ Suffix dialling (for selecting the connection) is not possible in conjunction with pulse dialling: a hash "#" is displayed during dialling but ignored when pulse dialling is used.

◆ Enter an asterisk "*" to switch temporarily to touch tone dialling. The asterisk is not displayed.

Setting recall

Your phone is preset at the factory for operation on the main connection (recall 250 ms). For operation on a PABX, you may have to change this value. Please refer to the user guide for your PABX.

Menu → Settings → Base → Telephony → Phone → Recall

Select recall and press OK.

The current setting is indicated by ✓.

Press and **hold** (idle status).

Setting pauses

Changing pause after line seizure

You can set the length of the pause inserted between pressing the talk key and sending the phone number.

Menu 7pqrs 4ghi 9wxyz 1 6mno

Enter a number for the length of the pause (**1** = 1 sec.; **2** = 3 sec.; **3** = 7 sec.) and press OK.

Changing the pause after the recall key

You can change the length of the pause if your PABX requires this (refer to the user guide for your PABX).

Menu 7 pqrs 4 ghi 9 wxyz 1 1

Enter a number for the length of the pause (**1** = 1 sec.; **2** = 2 sec.; **3** = 3 sec.; **4** = 6 sec.) and press **OK**.

Switching temporarily to touch tone dialling (DTMF)

If your PABX still operates with dial pulsing (DP), but you need touch tone dialling for a connection (e.g. to listen to the network mailbox for your fixed line network connection), you must switch to touch tone dialling for the call.

Precondition: You are currently conducting an external call via the fixed line network or you have dialled an external fixed line network number or an external call is signalled.

Menu Open menu.

Tone dialling

Select and press **OK**.

Touch tone dialling is now activated **for this call only**.

Web configurator – setting the phone using a PC

The Web configurator is the Web interface for your phone. It allows you to select the settings for your phone's base station via your PC's Web browser.

The Web configurator on your phone provides you with the following options:

- Configure your phone access to the local network (IP address, gateway to the Internet).
- Configure your phone for VoIP. Assign up to six VoIP phone numbers to your telephone.
- Load new firmware onto the phone if necessary.
- Use Internet services: Enable access to an online directory, display text information on the handset (info services) and synchronise the telephone's date/time with a time server on the Internet.
- Copy contacts from the Outlook address book on your PC into the handset directories or back up your handsets' directories on your PC.
- Manage the names and internal numbers of registered handsets and your local directories
- Obtain information about your phone's status (firmware version, MAC address etc.).

Preconditions:

- A standard Web browser is installed on the PC, e.g. Internet Explorer version 6.0 or higher, or Firefox version 1.0.4 or higher.
- The phone and PC are connected with each other via a router.

Please note

- Depending on your VoIP provider, it is possible that you will be unable to change individual settings in the Web configurator.
- The phone is **not** blocked while you select your settings in the Web configurator. You can also use your phone to make calls or change base station or handset settings on your handset at the same time.
- While you are connected to the Web configurator, it is blocked to other users. It cannot be accessed by more than one user at any time.

Connect the PC with the telephone's Web configurator

Precondition: The settings of an available firewall installed on your PC allow the PC and phone to communicate with each other.

There are two ways of connecting your PC to the base station Web configurator:

◆ via the (local) IP address of the base station

◆ via Gigaset config

Establishing a connection via the IP address:

▶ Establish the telephone's current IP address on the handset:

You can see the phone's current IP address in the handset display by **briefly** pressing the registration/paging key on the base station.

Your phone's IP address can change if you have activated dynamic IP address assignment (→ Page 115).

Warning

If one of the four parts of the IP address contains leading zeros (e.g. 002), these zeros must not be entered in the Web browser address field. Otherwise the Web browser will not be able to establish a connection to the Web configurator.

Example: The IP address 192.168.002.002 is displayed on the handset. 192.168.2.2 should be entered in the address field.

▶ Launch the Web browser on your PC.

▶ Enter **http://** and the telephone's current IP address (for example: http://192.168.2.2) into the address field of the Web browser.

▶ Press the return key.

A connection is established to the phone's Web configurator.

Establish a connection via Gigaset config:

Precondition: The router is connected to the Internet and your PC can access the Internet via the router.

▶ Launch the Web browser on your PC.

▶ Enter the following URL into the Web browser's address field:

http://www.Gigaset-config.com.

▶ Press the return key.

You will receive a message stating that the connection will be forwarded to your base station.

If several Gigaset VoIP phones can be reached via your Internet connection, you will be asked to which one of these phones you would like to be connected.

After successfully forwarding the connection, the Login Web page of the Web configurator will be displayed in the Web browser.

Please note

The connection between the PC and the Web configurator is a local connection (LAN connection). The Internet is only accessed to establish the connection.

Logging in, setting the Web configurator language

Once you have successfully established the connection, the Login Web page will be displayed in the Web browser.

You can select the language you want the menus and Web configurator dialogues to be displayed in. The language that is currently selected is displayed in the top field of the Web page.

- If necessary, click ▾ to open the list of available languages.
- Select the language.
- Enter your phone's system PIN (default setting: 0000) in the bottom field of the Web page, to access the Web configurator functions.
- Select OK.

Once you have successfully logged in, a Home screen opens with general information on the Web configurator.

Please note

- If you have forgotten your system PIN, you must restore your device's factory settings. Please note that this will restore all other settings too (→ Page 98).
- If you do not make any entries for a lengthy period (approx. 10 minutes), you will be automatically logged off. The next time you try to make an entry or open a Web page, the Login Web page will be displayed. Enter the system PIN again to log in again.
- Any entries that you did not save on the phone before automatic log-off will be lost.

Logging off

In the menu bar (→ Page 111) at the top right of every Web page in the Web configurator, you will see the Log Off command. Select Log Off to log off from the Web configurator.

Warning

Always use the Log Off command to end the connection to the Web configurator. If, for example, you close the Web browser without logging off beforehand, it is possible that access to the Web configurator will be blocked for a few minutes.

Structure of the Web pages

The Web pages contain the UI elements displayed in Bild 1.

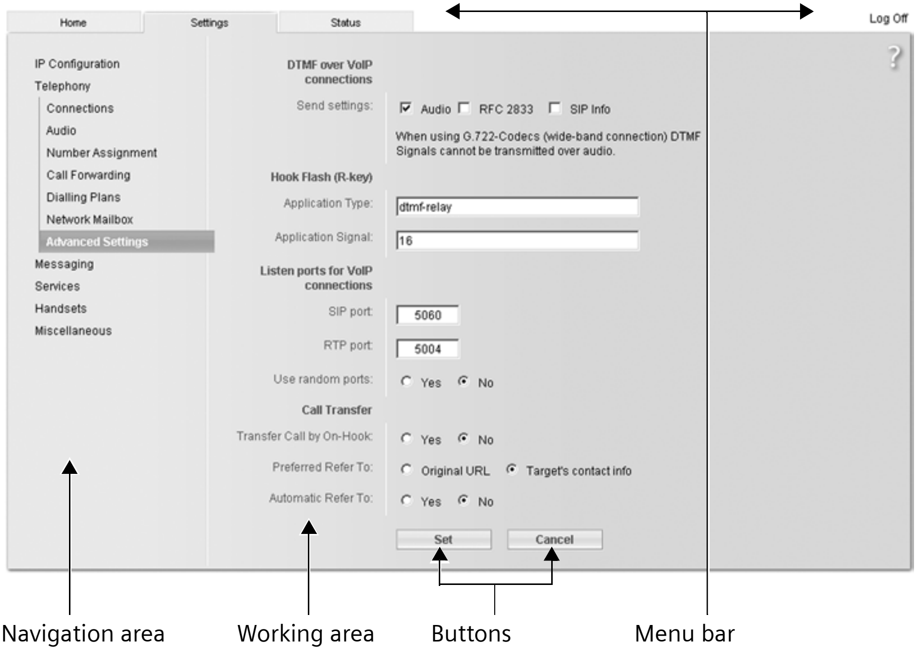


Bild 1 Example of the structure of a Web page

Menu bar

In the menu bar, the Web configurator menus are displayed in the form of tab pages.

The following menus are available:

Home
The start screen is opened once you have registered with the Web configurator. It contains information on the Web configurator functions.

Settings (→ Page 114)
This menu allows you to make settings on your phone.
If you select the Settings menu, a list with this menu's functions is displayed in the navigation area (→ Page 112).

Status (→ Page 151)
This menu provides you with information about your phone.

Log Off
You will find the Log Off function to the right of the menu bar on every Web page (→ Page 110).

Please note

An overview of the Web configurator menu → Page 39.

Navigation area

In the navigation area, the functions of the menu selected in the menu bar (→ Page 111) are listed.

If you select a function, the associated page opens in the working area with information and/or fields for your inputs.

If a function is assigned subfunctions, these are displayed with the function as soon as you select the function. The relevant page for the first subfunction is displayed in the working area.

Working area

Depending on the function selected, information or dialogue boxes are displayed in the working area, which allow you to make or change your phone settings.

Making changes

Make settings for entry fields, lists or options.

- There may be restrictions regarding the possible values for a field, e.g. the maximum number of characters, entering special characters or certain value ranges.
- To open a list, select . You can choose between default values.
- There are two kinds of options:
  - Options in a list, from which you can activate one or several options. Active, i.e. selected options are highlighted with , non-active options with . You can activate an option by selecting . The status of the other options in the list does not change. You can deactivate an option by selecting .
  - Alternative options. The active option in the list is highlighted with , and the non-active with . You can activate an option by selecting . The previously activated option is deactivated. You can only deactivate an option by activating another option.

Entering Cyrillic and Arabic characters

In the following, the specified maximum number of characters permitted in a field refers to Latin characters and digits (1 character = 1 byte), i.e. 1 character means 1 byte.

Cyrillic and Arabic characters require 2 bytes each, i.e. with a field length of 16 characters, for example, you can enter a maximum of 8 Cyrillic or Arabic characters.

If you enter too many characters into a field, the entry will be rejected (not saved on the base station). The "old" field content (e.g. the standard settings) will remain in place and will be displayed again when the web page is updated. No warning/confirmation is given.

Applying changes

As soon as you have made your change on a page, activate the new setting on the phone by selecting Set.

If your input in a field does not comply with the rules for this field, an appropriate error message will be displayed. You can then repeat the input.

Warning

Changes that have not been saved on your phone are lost if you move to another Web page or if the Web configurator is logged off, e.g. due to exceeding the time limit (→ Page 110).

Buttons

Buttons are displayed in the bottom section of the working area.

Set — Save entries on the phone.

Cancel — Reject changes made on the Web page and reload the settings that are currently saved in your phone to the Web page.

Opening Web pages

A brief outline of the navigation to the individual Web configurator functions is given below.

Example

Setting DTMF signalling

Settings → Telephony → Advanced Settings

To open the Web page, carry out the following steps after registration:

- Select the Settings menu in the menu bar.
- Select the Telephony function in the navigation area.
  The Telephony subfunctions are displayed in the navigation tree.
- Select the Advanced Settings subfunction.

The Web page from Bild 1 will be shown in the Web browser.

Setting the phone with the Web configurator

You can make the following settings using the Web configurator:

- Connect your phone to the local network (→ Page 115)
- Configuration for telephony
  - Enter settings for the VoIP provider and configure or activate/deactivate VoIP accounts (→ Page 118)
  - Specify the name of the fixed line network (→ Page 125)
  - Activate/deactivate the Gigaset.net connection (→ Page 125)
  - Activate/deactivate call forwarding for calls to your VoIP numbers or to your Gigaset.net number (→ Page 132)
  - Configure settings to improve voice quality for the VoIP connections (→ Page 127)
  - Define the standard connection for your telephone (fixed line network or VoIP) (→ Page 130)
  - Assign VoIP phone numbers to the individual handsets (→ Page 131)
  - Define user-specific dialling plans for emergency numbers and for cost control purposes (→ Page 136)
  - Enter and activate/deactivate the network mailbox for each number assigned to the telephone (→ Page 138)
  - Define the type of DTMF signalling (e.g. for remote operation of a VoIP network mailbox) and the recall key function for VoIP (→ Page 132).
  - Enter settings for call forwarding via VoIP (call placing, i.e. connecting two external callers to each other → Page 134)
- Output of information from an IP info service on the handset (→ Page 142)
- Select an online phone directory, activate/deactivate the caller name display from the online directory (→ Page 143)
- Synchronise date and time on the base station with a time server on the Internet (→ Page 149)
- Start firmware updates (→ Page 147)
- Manage registered handsets
  - Change names and internal numbers of the registered handsets (→ Page 144)
  - Copy contacts from your PC's Outlook address book to the handset directories or save handset directories to your PC (→ Page 145)
  - Activate/deactivate the display of VoIP status messages on your handset (→ Page 147)

IP Configuration

Assigning the IP address

Select the necessary settings for operating your phone in your local network and to connect it to the Internet. For more detailed explanations of the individual components/terms, see the glossary (→ Page 179).

- Open the following Web page: Settings → IP Configuration.
- In the Address Assignment area, select the IP address type.

Select Obtained automatically if you want your phone to be assigned a dynamic IP address by a DHCP server in your local network. No further settings are necessary for the local network.

Select Static if you would like to set up a static local IP address for your phone. A static IP address is, for example, useful if port forwarding or a DMZ is set up on the router for the phone.

The following fields are displayed when you select IP address type = Static:

IP address

Enter an IP address for your phone. This IP address allows your phone to be reached by other subscribers in your local network (e.g. PC).

192.168.2.2 has been preset.

Please note the following:

- The IP address must be from the address block for private use that is used in the router. This is generally in the range 192.168.0.1 – 192.168.255.254 with Subnet mask 255.255.255.0. The subnet mask determines that the first three parts of the IP address must be identical for all subscribers in your LAN.
- The static IP address must not belong to the address block (IP pool range) that is reserved for the DHCP server of the router. It must also not be used by another device on the router.

If necessary, check the settings on the router.

Subnet mask

Enter the subnet mask for your device's IP address. For addresses from the address block 192.168.0.1 – 192.168.255.254, the subnet mask 255.255.255.0 is generally used. This is preconfigured when the phone is supplied.

Default gateway

Enter the IP address for the standard gateway, by means of which the local network is connected to the Internet. This is generally the local (private) IP address for your router (e.g. 192.168.2.1). Your phone requires this information to be able to access the Internet.

192.168.2.1 has been preset.

Preferred DNS server

Enter the IP address for the preferred DNS server. DNS (Domain Name System) allows you to assign public IP addresses to symbolic names. The DNS server is required to convert the DNS name into the IP address when a connection is being established to a server.

You can specify your router's IP address here. The router forwards phone address requests to its DNS server.

192.168.2.1 has been preset.

Alternate DNS server (optional)

Enter the IP address for the alternative DNS server that should be used in situations where the preferred DNS server cannot be reached.

- Select Set to save the changes.

Or

- Select Cancel to reject the changes.

Allowing access from other networks

The default setting for your phone is set so that you can only access your phone's Web configurator via a PC that is in the same local network as your phone. The subnet mask of the PC must match that of the phone.

You can also allow access from PCs in other networks.

Warning

Expansion of access entitlement to other networks increases the risk of unauthorised access. It is therefore recommended that you deactivate remote access again if you no longer require it.

- Open the following Web page: Settings → IP Configuration.
- In the Remote Management area, activate the Yes option to permit access from other networks.

  To deactivate remote access, activate the No option Access is then limited to PCs in your own local network.

Access to the Web configurator services from other networks is only possible if your router is set accordingly. The router must pass on the service requests from "outside" to Port 80 (default port) of the phone. Please also read the user guide for your router.

To establish a connection, the public IP address or the DNS name of the router and, where applicable, the port number on the router must be indicated in the Web browser of the remote PC.

Configuring telephone connections

You can configure up to eight numbers on your phone: your fixed line network number, your Gigaset.net number and six VoIP numbers.

You need to set up a VoIP account with a VoIP provider for each VoIP phone number. You must save the access data for each account and for the relevant VoIP provider in the phone. You can assign a name to each connection (VoIP and fixed line network).

To configure the connections:

▶ Open the following Web page: Settings → Telephony → Connections.

A list (→ Bild 2) will be shown containing all the possible connections that you can configure, or have already configured, on your phone.

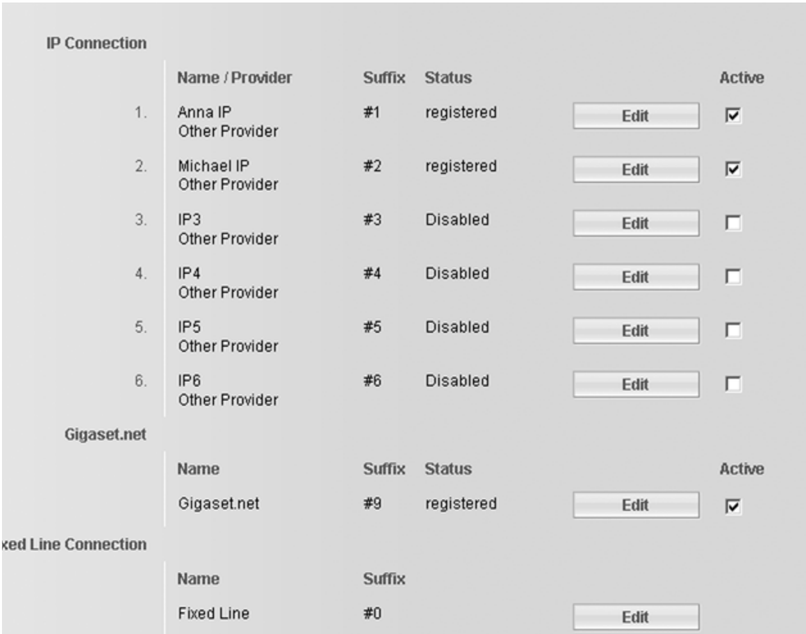


Bild 2 List of possible connections

The list will show the following:

Name / Provider

Name of the connection. This will show the name that you have defined for the connection (→ Page 119, Page 125) or the default name (IP1 to IP6 for VoIP connections, Fixed Line for the fixed line network connection and Gigaset.net).

VoIP connections also display the name of the VoIP provider with which you have opened the account. If the name is unknown the display will show Other Provider.

Suffix

Line suffix that you have to add to the phone number of an outgoing call to allow the account assigned to the suffix to be used as the sending account.

Example

If you dial 123456765**#1**, the connection will be made and billed through the first VoIP account, regardless of the VoIP number you have assigned to your handset as the send number and whether you briefly press/press and hold the talk key.

If you dial 123456765**#0**, the connection will be made via the fixed line network.

Status

The status of the connection will be shown for VoIP connections:

Registered

The connection is activated. The phone has been successfully registered. You can use the connection to make calls.

Disabled

The connection is deactivated. The phone is not registering with the corresponding account with the VoIP service. You cannot use the connection to make or receive calls.

Registration failed / Server not accessible

Your phone was unable to register with the VoIP service, e.g. because the VoIP access data is incorrect or incomplete or the phone is not connected to the Internet. For further information on this in the section, please also refer to "Questions and answers" → Page 153.

Active

You can use the option in the Active column to activate (☑) and deactivate (☐) VoIP connections. If a connection is deactivated, the phone will not register for this connection. The connection can be activated/deactivated by clicking directly on the option. The change does not need to be saved.

To configure a connection or to change the configuration of a connection:

- Select Edit button next to the connection.

This will open a Web page where you can make the settings needed. More information is available

- in the section "Configuring the VoIP connection" → Page 118 or
- in the section "Configuring the fixed line connection" → Page 125

Configuring the VoIP connection

- Open the following Web page: Settings → Telephony → Connections.
- Select the Edit button next to the VoIP connection that you want to configure or the configuration you wish to change.

This will open a Web page where you can make the settings that your phone needs to access your provider's VoIP server.

The Web page always displays the following areas:

- IP Connection (→ Page 119)
- Auto Configuration (→ Page 119)
- Personal Provider Data (→ Page 121).

The areas

- General Provider Data (→ Page 122) and
- Network (→ Page 123)

can be shown and hidden by selecting the Show Advanced Settings and Hide Advanced Settings buttons.

You must enter the VoIP provider's general access data in these areas. You can download this data for many VoIP providers from the Internet (→ "Area: Auto Configuration" Page 119).

- Make the settings on the Web page.
- Save them in the phone → Page 125.
- Activate the connection if necessary → Page 125.

Area: IP Connection

Connection Name or Number

Enter a name for the VoIP connection or the VoIP phone number (max. 16 characters). This name is used to display the connection on the handset and the Web configurator interface, e.g. during allocation of send and receive numbers (→ Page 131), for the call display (→ Page 44).

Area: Auto Configuration

The entire configuration process or a large part of the configuration for a VoIP connection is automated for many VoIP providers. You can download the necessary VoIP access data to your phone from the Internet.

You have the following options:

- Fully automated configuration

  Preconditions:

  – You have received an **Auto Configuration Code** from your VoIP provider.

  – The general access data for your VoIP provider is available for downloading.

  You can download all the data required for VoIP access from the Internet:

  - Enter the Auto Configuration Code you received from your VoIP provider in the Auto Configuration area in the Auto Configuration Code field (maximum 32 characters).
  - Select the Start Auto Configuration button.

The telephone establishes a connection to the Internet and downloads all data required for the VoIP connection, i.e. the general provider information and your personal provider data (account data) are saved to your base station.

If you have already entered details on the Web page, this is deleted as soon as Start Auto Configuration is selected. The fields in the Personal Provider Data and General Provider Data areas and the server addresses in the Network area are overwritten by the downloaded data.

Generally, you should not have to enter any additional data on this Web page.

Please note

If the message Download of settings not possible! File is corrupt! appears, no data will be loaded onto the phone. Possible causes of this are:

- The incorrect code has been entered (e.g. upper/lower case rules have not been followed). If necessary, enter the code again.
- The file that has been downloaded is invalid. Please consult your VoIP provider.

When the download is complete, the Connections list will be displayed.

- Activate the connection as described on Page 125.

You can then be reached on the corresponding VoIP phone number.

- Automatic configuration of general VoIP provider data

  Precondition: You have received your account details from your VoIP provider (e.g. Authentication Name, Authentication password).

  Profile files of the most important VoIP providers are available to download on the Siemens Internet server. The address for the server is stored in your phone (→ Page 147).

  To load the data onto your telephone, proceed as follows:

  - Select Select VoIP Provider in the Auto Configuration area. This will display information on the download procedure.

Please note

If you select the Select VoIP Provider button, any changes that have been made to the Web page will be saved and checked. Values may need to be corrected before the Select VoIP Provider operation is started.

The download procedure consists of several steps:

- Select the Next button.
- From the list, select the country for which the list of VoIP providers is to be loaded.
- Select the Next button.
- Select your VoIP provider from the list.
  If your provider is not included in the list, select Other Provider. In this case you will have to enter the general provider data by hand (see "Area: General Provider Data" and "Area: Network" below).
- Select the Finish button.

The details of the selected provider are loaded to your phone and saved under General Provider Data (→ Page 122) and Network (→ Page 123). You cannot make any further entries in these areas.

The Provider field shows the name of the selected provider or Other Provider. A link to the provider's homepage is displayed where available.

To complete configuration of your VoIP connection, enter your account data in the Personal Provider Data area.

Area: Personal Provider Data

Enter the configuration data that is necessary for accessing your VoIP provider's SIP service. You will receive this data from the VoIP provider.

The field names in this area (Authentication Name etc.) listed in the following are standard names and may change. If you have already downloaded the provider's general details ("Select VoIP Provider" button, see above), field entries will be replaced by provider-specific names to facilitate orientation (e.g. SIP-ID instead of Authentication Name).

Authentication Name

Specify the registration or authentication ID agreed with your VoIP provider (maximum 32 digits). The registration ID serves as the access ID that your phone must specify when registering with the SIP proxy/registrar server. The Authentication Name is mainly identical to the Username, i.e. to your Internet phone number.

Authentication password

Enter the password that you have agreed with your VoIP provider in the Authentication password field (maximum 32 characters). The phone needs the password when registering with the SIP proxy/registrar server.

Username

Enter the caller ID for your VoIP provider account (maximum 32 characters). This ID is usually identical to the first part of your SIP address (URI, your Internet phone number).

Example

Example: If your SIP address is "987654321@provider.com", enter "987654321" as the Username.

Display name (optional)

Enter any name that should be shown in the other caller's display when you call them via the Internet (example: Anna Sand). All characters in the UTF8 character set (Unicode) are permitted. The name must not exceed 32 characters.

If you do not enter a name, your Username or your VoIP phone number will be displayed.

Ask your VoIP provider if this feature is supported.

Area: General Provider Data

If you have downloaded the general settings for the VoIP provider from the Siemens configuration server (→ Page 119), then the fields in this area will be preset with the data from the download. Generally speaking, you do not need to configure any settings in this area.

Domain
Specify the last part of your SIP address (URI) here (maximum 74 characters).

Example

For the SIP address "987654321@provider.com", enter "provider.com" in Domain.

Proxy server address

The SIP proxy is your VoIP provider's gateway server. Enter the IP address or the (fully-qualified) DNS name of your SIP proxy server (maximum 74 characters). **Example:** myprovider.com.

Proxy server port

Enter the number of the communication port that the SIP proxy uses to send and receive signalling data (SIP port).

Port 5060 is used by most VoIP providers.

Registrar server

Enter the (fully-qualified) DNS name or the IP address of the registrar server (maximum 74 characters).

The registrar is needed when the phone is registered. It assigns the public IP address/port number to your SIP address (Username@Domain) that were used by the phone at registration. With most VoIP providers, the registrar server is identical to the SIP server. **Example:** reg.myprovider.com.

Registrar server port

Enter the communication port used in the registrar. It is mainly port 5060 that is used.

Registration refresh time

Enter the time intervals at which the phone should repeat the registration with the VoIP server (SIP proxy) (a request will be sent to establish a session). The repeat is required so that the entry of the phone in the tables of the SIP proxy is retained and the phone can therefore be reached. The repeat will be carried out for all activated VoIP phone numbers.

The default is 180 seconds.

If you enter 0 seconds, the registration will not be repeated periodically.

Area: Network

Please note

If you have downloaded the general settings for your VoIP provider from the Siemens configuration server (→ Page 120), then some fields in this area will be preset with the data from the download (e.g. the settings for the STUN server and outbound proxy).

If your phone is connected to a router with NAT (Network Address Translation) and/or a firewall, you must select some settings in this area so that your phone can be reached from the Internet (i.e. can be addressed).

Through NAT, the IP addresses of subscribers in the LAN are concealed behind the public IP address of the router.

For incoming calls

If port forwarding is activated or a DMZ is set up for the phone on the router, no special settings are required for incoming calls.

If this is not the case, an entry in the NAT routing table (in the router) is necessary in order for the phone to be reached. This entry is created when the phone is registered with the SIP service. In the interest of security, this entry is automatically deleted at certain intervals (session timeout). The phone must therefore confirm its registration at certain intervals (see NAT refresh time, → Page 124), so that the entry stays in the routing table.

For outgoing calls

The phone needs its public address in order to receive caller voice data.

There are two possibilities:

- The phone requests the public address from a STUN server on the Internet (Simple Transversal of UDP over NAT). STUN can only be used with asymmetric NATs and non-blocking firewalls.
- The phone does not direct the connection request to the SIP proxy but to an outbound proxy on the Internet that supplies the data packets along with the public address.

The STUN server and outbound proxy are used alternately to work around the NAT/firewall in the router.

STUN enabled

Select Yes if you want your phone to use STUN as soon as it is used on a router with asymmetric NAT.

STUN server

Enter the (fully-qualified) DNS name or the IP address of the STUN server on the Internet (maximum 74 characters).

If you selected Yes in the STUN enabled field, you must enter a STUN server here.

STUN port

Enter the number of the communication port on the STUN server. The default port is 3478.

STUN refresh time

Enter the time intervals at which the phone should repeat the registration with the STUN server. The repeat is required so that the entry of the phone in the tables of the STUN server is retained. The repeat will be carried out for all activated VoIP phone numbers.

Ask your VoIP provider for the STUN refresh time.

The default is 240 seconds.

If you enter 0 seconds, the registration will not be repeated periodically.

NAT refresh time

Specify the intervals at which you want the phone to update its entry in the NAT routing table. Specify an interval in seconds that is a little shorter than the NAT session timeout.

As a rule you should not need to change the preconfigured value for the NAT refresh time.

Outbound proxy mode

Specify when the outbound proxy should be used.

Always

All signalling and voice data sent by the phone is sent to the outbound proxy.

Auto

Data sent by the phone is only sent to the outbound proxy when the phone is connected to a router with symmetric NAT or blocking firewall. If the phone is behind an asymmetric NAT, the STUN server is used.

If you have set STUN enabled = No or have not entered a STUN server, the outbound proxy is always used.

Never

The outbound proxy is not used.

If you do not make an entry in the Outbound proxy field, the phone behaves independently of the selected mode, as with Never.

Outbound proxy

Enter the (fully qualified) DNS name or the IP address of your provider's outbound proxy (maximum 74 characters).

Please note

With many providers, the outbound proxy is identical to the SIP proxy.

Outbound proxy port

Enter the number of the communication port used by the outbound proxy.

The default port is 5060.

Saving settings on the phone

- Select Set to save the changes.
  The Connections list will be shown after saving (→ Bild 2 on Page 117).

If you want to discard the changes:

- Select the Cancel button.

If all fields are to be reset to the default settings:

- Select the Delete button.

Fields without default settings are empty.

Please note

If you do not make any entries for a longer period, the connection to the Web configurator is automatically terminated. Unsaved entries are lost. If necessary, implement temporary security measures. You can subsequently continue the entry and make changes if necessary.

Activating a new connection

If you have configured a new VoIP connection, you must also activate it.

In the Connections list:

- Activate the relevant option in the Active column (☑ = activated).

Your phone will register itself with the VoIP provider using the relevant access data. Refresh the website (e.g. by pressing F5).

The Status Registered column will appear if registration was successful. You can now be reached on this VoIP phone number.

Please note

Once the new entry has been made, the VoIP phone number is assigned to each handset as a receive number. For how to adjust the assignment → Page 131.

Configuring the fixed line connection

You can assign a name to your fixed line connection. This name is used to display the connection on the handset and the Web configurator interface, e.g. during allocation of send and receive numbers (→ Page 131), for the call display (→ Page 44).

- Open the following Web page: Settings → Telephony → Connections.
- Select the Edit button in the Fixed Line Connection area.
- Enter your fixed line network number or the name of your choice (max. 16 characters) for your fixed line connection in the Connection Name or Number field. The default is "Fixed Line".

Configuring the Gigaset.net connection

Your phone is assigned a Gigaset.net phone number by default. As soon as you have connected your phone to the Internet, you can make calls using the

Gigaset.net and receive calls from other Gigaset.net subscribers, provided that your Gigaset.net connection has been activated. You can deactivate the Gigaset.net connection.

Activating/deactivating the Gigaset.net connection

- Open the following Web page: Settings → Telephony → Connections. The list of connections will be displayed (→ Bild 2 on Page 117).
- In the Gigaset.net area: use the option in the Active column to activate (☑) or deactivate (☐) the Gigaset.net connection.

Please note

If you do not use your Gigaset.net connection for six weeks, it is automatically deactivated. You cannot be reached for calls from Gigaset.net.

The connection is reactivated:

- as soon as you start a search in the Gigaset.net directory (→ Page 48) or
- make a call via Gigaset.net, i.e. dial a number ending in #9 (two attempts may be necessary) or
- activate the connection via the Web configurator as described above.

Activating/deactivating the STUN server of the Gigaset.net connection

The Gigaset.net connection is preconfigured in your phone. The Gigaset.net uses a STUN server as standard. In the sent data packets, Gigaset.net replaces the private IP address of your phone with its public IP address.

If you operate your phone behind a router with symmetrical NAT, STUN cannot be used. Otherwise, when making Gigaset.net calls you will not be able to hear the caller.

In this case, deactivate STUN for the Gigaset.net connection.

- Open the following Web page: Settings → Telephony → Connections.
- Select Edit in the Gigaset.net area.

STUN enabled

Select No to deactivate STUN.

Select Yes if you want your phone to use STUN.

- Select Set to save the changes.

Optimising voice quality for VoIP connections

You can make general and connection-specific settings to improve the voice quality for VoIP telephony.

- Open the following Web page: Settings → Telephony → Audio.

The voice quality for VoIP connections is mainly determined by the **voice codec** used for transferring the data and the available **bandwidth** of your DSL connection.

In the case of the voice codec, the voice data is digitalised (coded/decoded) and compressed. A "better" codec (better voice quality) means more data needs to be transferred, i.e. perfect voice data transfer requires a DSL connection with a larger bandwidth.

The following voice codecs are supported by your phone:

G.722
: Excellent voice quality. The **broadband** speech codec G.722 works at the same bit rate as G.711 (64 kbit/s per speech connection) but with a higher sampling rate. This allows higher frequencies to be played back. The speech tone is therefore clearer and better than for the other codecs (High Definition Sound Performance).

Gigaset S67H, S68H and SL37H handsets, for example, are HDSP-compatible.

G.711 a law / G.711 μ law

Excellent voice quality (comparable with ISDN). The necessary bandwidth is 64 kbit/s per voice connection.

G.726
: Good voice quality (inferior to that with G.711 but better than with G.729).

Your phone supports G726 with a transmission rate of 32 kbit/s per voice connection.

G.729
: Average voice quality. The necessary bandwidth is less than or equal to 8 kbit/s per voice connection.

Both parties involved in the telephone connection (caller/sender and receiver) must use the same voice codec. The voice codec is negotiated between the sender and the recipient when establishing a connection.

You can influence the voice quality by selecting (bearing in mind the bandwidth of your DSL connection) the voice codecs your phone is to use, and specifying the order in which the codecs are to be suggested when a VoIP connection is established.

Area: Settings for Bandwidth

The settings in this area influence all VoIP connections (VoIP phone numbers).

Allow 1 VoIP call only

You can usually make two VoIP calls at the same time on your phone. If, however, your DSL connection has a narrow bandwidth, there may be problems if two VoIP calls are made at the same time. The data is no longer transferred properly (long voice delay, data losses etc.).

- Select Yes next to Allow 1 VoIP call only to prevent any further parallel VoIP phone connections being established.

▶ If you wish to permit two VoIP connections, select No.

Please remember

If only one VoIP connection is permitted, the following VoIP network services will **no** longer be available:

- Call waiting
  Call waiting is not displayed during a call via VoIP
- External consultation call from a VoIP call
- Toggling and initiating a conference call via VoIP

Voice Quality

Default settings for the codecs used are stored in your phone: one setting optimised for narrow and one for wide bandwidths.

▶ Activate one of the options Optimized for low bandwidth / Optimized for high bandwidth if you wish to accept a default setting for all VoIP connections. The settings are shown in the Settings for Connections area and cannot be changed.

▶ Activate the Own Codec preference option if you wish to select and set connection-specific voice codecs yourself (see "Area: Settings for Connections").

Area: Settings for Connections

In this area you can make specific settings for each of your VoIP phone numbers.

You can make the following settings for each VoIP phone number configured on your phone:

Volume for VoIP Calls

Depending on the VoIP provider, it is possible that the received voice/earpiece volume is too low or too high, so that adjusting the volume via the handset is not adequate.

Specify whether the received volume range is too high or too low. The following options are available:

Low
: Voice/earpiece volume is too high. Activate this option to reduce the volume by 6 dB.

Normal
: The voice/earpiece volume does not need to be raised/lowered.

High
: Voice/earpiece volume is too low. Activate this option to increase the volume by 6 dB.

Voice codecs

Precondition: The Own Codec preference option is activated for the Voice Quality in the Settings for Bandwidth area.

Select the voice codecs your phone is to use, and specify the order in which the codecs are to be suggested when a VoIP connection is established via this VoIP phone number.

- Apply the voice codecs that your phone is to suggest for outgoing calls into the Selected codecs list.
  To do this, in the Available codecs list select the voice codec that you want to apply (you can mark several entries using the Shift key or the Ctrl key). Select <Add.
- Move the voice codecs that you do not want the phone to use into the Available codecs list.
  Select the voice codecs in the Selected codecs list (see above) and click on the Remove> button.
- Sort the voice codecs in the Selected codecs list into the order in which they should be suggested to the receiving device when a connection is established. To do this, use the Up and Down buttons.

When establishing a VoIP connection, the phone suggests the 1st voice codec in the Selected codecs list to the receiving device to begin with. If the receiving device does not accept this voice codec (e.g. because it is not supported), the 2nd voice codec on the list is suggested, and so on.

If the receiving device does not accept any of the voice codecs in the Selected codecs list, the connection is **not** established. An appropriate message will be displayed on the handset.

If the phone always starts by trying to establish a broadband connection, put the G.722 codec at the top of the Selected codecs list.

Please note

- You should only deactivate codecs (put them in the Available codecs list) if there is a particular reason. The more codecs that are deactivated, the greater the danger that calls will not be able to be established due to unsuccessful codec negotiations. In particular you can only establish broadband connections if you permit the G.722 codec.
- With incoming calls, all supported voice codecs are always permitted.

Area: Settings for Codecs

To save additional bandwidth and transmission capacity, on VoIP connections that use the G.729 codec you can suppress the transmission of voice packets in pauses ("Silence Suppression"). Then, instead of the background noises in your environment, your caller hears a synthetic noise generated in the receiver.

Please note: "Silence Suppression" can sometimes lead to deterioration in the voice quality.

- In the Enable Annex B for codec G.729 field, state whether the transmission of data packets during pauses should be suppressed when using the G.729 codec, (select Yes).

Saving settings on the phone

- Select Set to save the settings for the voice quality.

Please note

You should observe the following for good voice quality:

- When making calls using VoIP, avoid performing other Internet activities (e.g. surfing the Internet).
- Please note that voice delays can occur depending on the codec used and the network capacity utilisation.

Voice quality and infrastructure

With your Gigaset A580 IP, you have the opportunity to make calls with good voice quality via VoIP.

However, your phone's performance with VoIP – and therefore the voice quality – also depends on the properties of the entire infrastructure.

The following components from your VoIP provider may impact performance :

- Router
- DSLAM
- DSL transmission line and speed
- Connection paths over the Internet
- If applicable, other applications that also use the DSL connection

In VoIP networks, voice quality is affected by various things including the "quality of service" (QoS). If the entire infrastructure demonstrates QoS, voice quality is higher (fewer delays, less echoing, less crackling etc.).

If, for example, the router does not have QoS, then the voice quality is not as good. Please see the specialist documentation for further information.

Setting the telephone's default connection

The default telephone connection defines which line type (VoIP or fixed line network) will be used to dial numbers when you **briefly** press the talk key. The default connection is applied to all registered handsets.

- Open the following Web page: Settings → Telephony → Number Assignment.
- Enter the default connection in the Linetype for outgoing calls area. This can be done by selecting the VoIP or Fixed Line option.
- Now select Set to activate your settings.

Please note

- The default connection is only relevant when dialling numbers that are not subject to dialling plans and are entered without a line suffix.
- You can change the settings for the default connection via any registered handset (→ Page 99).

Assigning send and receive numbers to handsets

You can assign up to eight phone numbers on your phone: your fixed line network number, your Gigaset.net number and six VoIP numbers.

You can assign as many of these numbers as you like to each handset as receive numbers. Receive numbers determine which handset(s) will ring when a call is received.

You can assign one of your VoIP numbers to each handset as a (VoIP) send number. Send numbers define which VoIP account should be used in general to make and pay for outgoing VoIP calls. **Exceptions:**

- A phone number is dialled with a line suffix (→ Page 118) or
- A dialling plan has been defined for the phone number (→ Page 136).

The Gigaset.net number and fixed line network number are permanently assigned to each registered handset as send numbers.

Please note

A handset is assigned the following numbers after it is registered on the base station:

- Receive numbers: all phone numbers assigned to the phone (fixed line network, Gigaset.net and VoIP).
- Send numbers: the fixed line network number and the VoIP phone number that you entered at the start of the phone configuration.

- Open the following Web page: Settings → Telephony → Number Assignment.

The display shows all registered handsets. A list is displayed for each handset showing the phone numbers that are configured and activated for the phone. The connection names are shown in the Connections column. The fixed line network connection is always at the end of the list.

- Define a VoIP phone number as the send number for each handset. To do this, click the option following the phone number in the for outgoing calls column. The previous assignment will automatically be deactivated.

Please note

The fixed line network number is permanently assigned to each handset as a send number. This assignment cannot be deactivated. It ensures that emergency numbers can be called from every handset.

The Gigaset.net number is also permanently assigned to each handset.

- Select the phone numbers for each handset (fixed line network, VoIP) that are to be assigned to the handset as receive numbers. To do this, click the option following the phone number in the for incoming calls column. Every handset can be assigned several phone numbers or no phone number (☑ = assigned).
- Now select Set to save your settings.

Please note

- If a VoIP phone number that has been assigned to a handset as a send number is deleted, the handset will automatically be assigned the first configured VoIP phone number.
- Calls made to a number that is not assigned to a handset as a receive number will not be signalled on any handset.
- If you have not assigned receive numbers to any of the handsets, calls to all connections will be signalled on all handsets.

Activating Call Forwarding for VoIP connections

You can forward calls to your VoIP numbers and to your Gigaset.net number.

You can forward calls to your VoIP numbers to any external number (VoIP, fixed line or mobile network number). The forwarding is done via a VoIP connection.

You can forward calls to your Gigaset.net number within the Gigaset.net, i.e. to another Gigaset.net number.

You can define if and when calls to your Gigaset.net number and some of your VoIP numbers (VoIP account) should be forwarded to this VoIP number.

You can also use the handset to define call forwarding and activate/deactivate it → Page 55.

▶ Open the following Web page: Settings → Telephony → Call Forwarding.

The display shows a list of all your configured VoIP connections and your Gigaset.net number.

Connections

Select the name you have assigned to the VoIP number, or select Gigaset.net.

When — Select when a call to this VoIP number should be forwarded: When busy / No reply / Always. Select Off to deactivate call forwarding.

Call number — Enter the phone number to which the calls should be forwarded. Please note that you may have to enter the area code when forwarding to a fixed line network number in the same area (depending on your VoIP provider and the setting for the automatic area code → Page 135).

The settings only affect the phone number selected in Connections.

Please note

For how to forward your fixed line network number → Page 54.

Setting DTMF signalling for VoIP

DTMF signalling, for example, is required to check and control some network mailboxes via digit codes.

To send DTMF signals via VoIP you must first define how key codes should be converted into and sent as DTMF signals: as audible information via the speech channel or as a "SIP Info" message.

Ask your VoIP provider which type of DTMF transmission it supports.

- Open the following Web page: Settings → Telephony → Advanced Settings.

Area: DTMF over VoIP connections

Make the required settings for sending DTMF signals.

- Activate Audio or RFC 2833, if DTMF signals are to be transmitted acoustically (in voice packets).
- Activate SIP Info if DTMF signals are to be transmitted as code.
- Now select Set to save your settings.

Please note

- The settings for DTMF signalling apply to all VoIP connections (VoIP accounts).
- DTMF signals cannot be transmitted in the audio path (Audio) on broadband connections (the G.722 codec is used).

Defining recall key functions for VoIP (hook flash)

Your VoIP provider may support special performance features. To make use of these features, your phone needs to send a specific signal (data packet) to the SIP server. You can assign this "signal" to your phone's recall key.

If you press the recall key during a VoIP call the signal will be sent to the server.

- Open the following Web page: Settings → Telephony → Advanced Settings.

Area: Hook Flash (R-key)

- In the Application Type fields, (maximum 31 characters) and Application Signal fields (maximum 15 characters) enter the data that you have received from your VoIP provider.
- Now select Set to save your settings.

The setting for the recall key applies to all registered handsets.

Defining local communication ports for VoIP

- Open the following Web page: Settings → Telephony → Advanced Settings.

Area: Listen ports for VoIP connections

Specify which local ports the telephone is to use for VoIP telephony. The ports must not be used by any other subscriber in the LAN.

SIP port

Specify the local communication port that the phone should use to send and receive signalling data. Specify a number between 1024 and 49152. The default port number for SIP signalling is 5060.

RTP port

Specify the local communication port that the phone should use to receive voice data. Enter an **even** number between 1024 and 49152. The port number must **not** be the same as the port number in the SIP port field. If you enter an odd number, the next lowest even number will be selected automatically (e.g. if you enter 5003, then 5002 is set automatically). The default port number for voice transmission is 5004.

Use random ports

Select Yes, if you do not want the phone to use fixed ports for SIP port and RTP port, but rather to use any free ports.

The use of random ports makes sense if you want several phones to be operated on the same router with NAT. The phones must then use different ports so that the router's NAT is only able to forward incoming calls and voice data to one (the intended) phone.

If you select No, the phone will use the ports specified in SIP port and RTP port.

- Now select Set to save your settings.

Configuring call forwarding via VoIP

If you are toggling calls via VoIP connections, you can connect the two external callers (provider-dependent). You can configure settings for this type of call forwarding.

- Open the following Web page: Settings → Telephony → Advanced Settings.

Area: Call Transfer

- Make your settings for call forwarding via VoIP in the following fields:

Transfer Call by On-Hook

If you select Yes, the external parties you are toggling between will be connected when you press the end call key ☎. Your connections with the callers will be terminated.

Preferred Refer To

Define the protocol (the contents of the "Refer To" information) that should be used with preference when forwarding calls:

Target’s contact info

This protocol is recommended for "closed" networks (internal company and business networks).

Original URL

This protocol is recommended when the base station is connected to the Internet via a router with NAT.

Automatic Refer To

If you select Yes, the base station will automatically attempt to determine the best protocol.

If the base station cannot determine the best protocol, it will use the protocol defined in Preferred Refer To.

- Now select Set to save your settings.

Setting area code predialling

On the base station, save the complete code (with international code) for the area in which you are using the phone.

For VoIP calls you must generally always dial the area code – even for local calls. You can save the need to dial the area code for local calls by setting your phone to prefix this code for all VoIP calls made in the same local area.

For calls made via VoIP, the area code entered is then prefixed to all numbers that do not start with 0 – even when dialling numbers from the directory and other lists.

Exception: Numbers for which you have defined dialling plans (→ Page 136).

- Open the following Web page: Settings → Telephony → Dialling Plans.

Area: Area Codes

Make the following settings here:

- From the Country list, select the country in which you are using your phone. This way the country code and the prefix of the area code are automatically set (in the International Prefix / Area Code and Local Prefix fields).
- In the Local Area Code field, enter the area code for your town without a prefix, e.g. 20 (for London).
- Select Yes next to Predial area code for local calls through VoIP to activate the function.

  Select No to deactivate the function. You will then need to enter the area code for local calls made via VoIP. Numbers in the directory must always contain the area code when dialling via VoIP.
- Select Set to save the settings.

Please remember

- The area code will also be prefixed to VoIP calls made to emergency numbers if there are **no** defined dialling plans for these numbers.
- The numbers of your network mailbox saved in the base station are **not** prefixed with an area code (→ Page 138).

Defining dialling plans – cost control

You can define dialling plans to reduce costs:

- You can define the connection (one of your VoIP accounts, the fixed line network) through which calls to specific numbers should be made and paid for.

  If you enter just a few digits (e.g. local area, national or mobile network code) any call to a number beginning with these digits will be made via the elected connection.
- You can block specific numbers, i.e. your phone will not establish a connection to these numbers (e.g. 0190 or 0900 numbers).

These dialling plans apply to all registered handsets. The settings for the default connection (briefly press/press and hold ☐) and the send numbers of handsets do not apply to numbers governed by a dialling plan.

Please note

You can override dialling plans, with the exception of blocks, as follows:

- Dial the number with a line suffix (e.g. 123456789#3, → Page 40).
- Before entering the number, define another connection type by pressing the Phone or IP display key (→ Page 92).
  If, for example, you press Phone, the number will be dialled via the fixed line network, even if the dialling plan states that a VoIP connection should be used.

Tips:

- Compare the rates for long-distance calls (especially for international calls) offered by your fixed line network and VoIP providers, and determine which connection should be used specifically for these countries/locations, e.g. a dialling plan for the Phone Number "0033" would apply to every call made to France.
- Use dialling plans to define that numbers starting with a call-by-call number are always made via your fixed line network connection. To do so, enter the call-by-call number in the Phone Number field.

Defining dialling plans

▶ Open the following Web page: Settings → Telephony → Dialling Plans.

Area: Dialling Plans

Specify dialling plans for your phone. Specify the following:

Phone Number

Enter the number or the first digits of the phone number (e.g. an area code) to which the dialling plan should apply (max. 15 digits).

Connection Type

The list shows all the VoIP connections that you have configured as well as your fixed line network connection. It also displays the name assigned to each connection.

- From the list, select the connection via which the number or numbers that start with the specified sequence of digits should be dialled.

Or:

- Select Block if the number or numbers that start with the sequence of digits should be blocked.

  The display will show Not possible! if an attempt is made to dial a blocked number.

Comment (optional)

You can enter a description of the dialling plan here (maximum of 20 characters).

- Select Add.

The dialling plan is activated immediately.

A new empty line for a new dialling plan will appear if your phone still has enough space to add further plans.

Please note

If dialling plans overlap, the one with the greatest concordance will apply.

Example:

There is a dialling plan for the number "02" and one for the number "023". If you dial "0231..." the second plan will apply; if you dial "0208..." the first plan will apply.

Examples

- You want to block your phone for all 09 numbers.

  Dialling plan:

  Phone Number = 0190 Connection Type = Block
- All calls to the mobile phone network should be made via your VoIP connection with provider B.

  Dialling plans:

  Phone Number = 017 Connection Type = IP3, provider B

  and the corresponding entries for "015" and "016".

Activating/deactivating dialling plans

- Select the option in the Active column to activate/deactivate the corresponding dialling plan (☑ = activated).

A deactivated dialling plan will not take effect until it is reactivated.

Deleting dialling plans

- Select Delete next to the dialling plan you wish to delete.

The dialling plan is deleted from the list immediately. The space in the list is released.

Please note

Predefined dialling plans set as defaults (for emergency numbers) **cannot** be deactivated and **cannot** be deleted.

Emergency numbers

Dialling plans for emergency numbers (e.g. the **local** emergency service number) are preset for certain countries. The fixed line network is determined as the Connection Type.

These dialling plans cannot be deleted, deactivated or blocked. However, you can change the Connection Type.

This should only be changed if the phone is not connected to the fixed line network. If you choose a VoIP connection, please make sure the VoIP provider supports calls to emergency numbers.

If no emergency numbers are set by default, you should define dialling plans for emergency numbers yourself and assign them to a connection of which you know that it supports emergency calls. Calls to emergency numbers are always supported by fixed line networks.

Warning

- Emergency numbers cannot be dialled if the keypad lock is activated. Before dialling, press **and hold** the hash key #, to release the keypad lock.
- If you have activated an automatic area code (→ Page 135) and if no dialling plan for emergency numbers is defined, the area code will also be prefixed to emergency calls made via VoIP.

Activating/deactivating network mailbox, entering numbers

Many fixed network providers and VoIP providers offer answer machines on the network – these are known as network mailboxes.

Each network mailbox accepts incoming calls made via the corresponding line (fixed line network or corresponding VoIP phone number).

You can enter the relevant network mailbox for each configured connection (VoIP, fixed line network) via the Web configurator. You can activate or deactivate the network mailbox for your VoIP connections.

- Open the following Web page: Settings → Telephony → Network Mailbox.

A list with all possible connections is displayed on the Web page. The names of the connections are displayed in the Connection column.

Entering numbers

- Enter the network mailbox number in the Call number column after the desired connection.

  With some VoIP providers your mailbox number will be downloaded together with the general VoIP provider data (→ Page 120), saved to your base station and displayed under Call number.
- Now select Set to save your settings.

Activating/deactivating the network mailbox

- You can activate (☑) and deactivate (☐) individual network mailboxes using the option in the Active column. Activating/deactivating is carried out by selecting the appropriate option. The change does not need to be saved.

Please note

You need to have **requested** the network mailbox for your fixed line network connection from your fixed line network provider.

Messaging

Your Gigaset A580 IP base station has messaging functions that can be used via a messaging capable handset, e.g. a **Gigaset** S67H, S68H, SL37H or C47H handset, that you can register to your base station.

The following messaging functions are available:

- Messenger functions

  The messenger client in your base station enables **instant messaging** (immediate message transfer, chatting). The phone supports the XMPP messenger (Jabber).
- E-mail functions

  Your telephone checks at regular intervals whether new messages have arrived in your incoming e-mail server. New e-mail messages are displayed by means of e-mail notifications on a handset with messaging functionality (sender and subject of the e-mail).

Saving messenger access data

In order to be able to use your base station's messenger functions, you need to register a handset with messaging capability and save the access data for your messenger server in the phone.

Your base station is already registered with the Gigaset.net Jabber server. An account has been assigned to your phone. You can chat to other Gigaset.net subscribers via this account. To do this, you need to log on to the Gigaset.net messenger server via your PC's Web browser using only this account, and then create a buddy list (→ "Setting up Gigaset.net Jabber account", Page 141).

You can also register with another instant messaging provider that supports XMPP Messenger (Jabber).

In order for you to use your phone's messenger to "go online" and "chat" on the Internet, the access data of a messenger server must be saved on your phone.

You can define a Resource name and a Priority for your phone. Both are required if you are logged in (online) with the messenger server with several devices (phone, desktop PC and laptop) at the same time using the same Jabber ID.

The Resource name is used to distinguish between these devices. The phone cannot log in to the messenger server if it does not have a resource name.

You should assign a Priority as each message will only be sent to one device for each Jabber ID. The Priority determines which of the devices receives the message.

Example

You are online using one of your phone's handsets and your PC both at the same time. You have assigned your phone (Resource name "phone") Priority 5 and your PC (Resource name "PC") priority 10. In this case, any message addressed to your Jabber ID will be sent to your phone.

- Open the following Web page: Settings → Messaging → Messenger.
- In the Messenger Account field, select whether you wish to use the Gigaset.net Jabber server or another provider's messenger server (Other).

  The access data for Gigaset.net is already stored in the base station. It is displayed in Jabber ID, Authentication password and Jabber server. With this data you can also register with the Gigaset.net Jabber server through your PC.
- Enter the user ID (max. 50 characters) and password (max. 20 characters) that you used to register with the messenger server in the Jabber ID and Authentication password fields. If you have selected Messenger Account = Gigaset.net, the fields are preconfigured with your Gigaset.net account.
- In the Jabber server field, enter the IP address or the DNS name of the messenger server with which you are registered for instant messaging.

  Max. 74 alphanumeric characters.

  If you have selected Messenger Account = Gigaset.net, the field is preconfigured with the name of the Gigaset.net server.
- Enter the number of the communication port on the Jabber server in the Jabber server port field. The default port is 5222.

  If you have selected Messenger Account = Gigaset.net, the port number is preconfigured.
- Enter a resource name (max. 20 characters) in the Resource field.

  The default is: phone.
- Enter the priority for your phone in the Priority field. Select a number between -128 (highest priority) and 127 (lowest priority) for the priority.

  The default is: 5
- Select Set.

Setting up Gigaset.net Jabber account

Your phone is already registered with the Gigaset.net Jabber server. An account has been assigned to your phone.

In order to chat with other Gigaset.net subscribers via this account, you must transfer the required Gigaset.net subscribers to a contact list (buddy list) on your PC. You can use any conventional Jabber client for this (e.g. PSI, Miranda; see e.g. http://www.swissjabber.ch).

Carry out the following steps to use your Gigaset.net Jabber account:

- Start the Web configurator, open the Settings → Messaging → Messenger Web page and select the Messenger Account Gigaset.net field. Your account data is displayed in Jabber ID and Authentication password. You will need these to create a buddy list via the Jabber client on the PC.
- Start your Jabber client on the PC.
- Enter your Gigaset.net Jabber ID on the Jabber client as a new account. The Jabber ID consists of your Gigaset.net number and "@jabber.gigaset.net"

  Example: 12345678901#9@jabber.gigaset.net
- Then enter your Authentication password.

Please note

- Do **not** select the option "Create new account". Your Gigaset.net Jabber account has already been created in Gigaset.net.
- The option "SSL connection" must be **deactivated** in the Jabber client.

- Now you can enter Gigaset.net subscribers as contacts (buddies).

  For the Jabber ID of each subscriber, enter the subscriber's Gigaset.net number with "@jabber.gigaset.net" (example: 2141524901#9@jabber.gigaset.net).

A request to "Add to contact list" will be sent to the subscriber.

If the subscriber accepts this request, they will be added to your buddy list.

Please note

For information on how to go online with your messaging-capable handset and chat with or call buddies, please see the extensive user guides for the Gigaset S675 IP or C470 IP for example. These are available on the Internet at:

www.siemens.com/gigaset.

Making e-mail settings

You must store the address or DNS name of your incoming e-mail server and your personal access data in the phone and activate the e-mail check with the incoming e-mail server, so that the phone can establish a connection to the incoming e-mail server and connect to your mailbox.

- Open the following Web page: Settings → Messaging → E-Mail.
- Enter the user name (account name) agreed with the Internet provider (max. 50 characters) in the Authentication Name field.

- Enter the password agreed with the provider for accessing the incoming e-mail server (max. 50 characters; case sensitive) in the Authentication password field.
- Enter the name of the incoming e-mail server (POP3 server) (max. 74 characters) in the POP3 Server field. Example: pop.theserver.com.
- From the Check for new e-mail list select the time interval at which your phone should check if new messages have arrived in your incoming e-mail server. Select Never to deactivate the prompt. Select one of the other values to activate the prompt for new e-mail messages.
- Select the Set button to save the settings in your phone.

Please note

For information on how e-mail messages are displayed and opened on the messaging-capable handset, please see the extensive user guides for the Gigaset S675 IP or C470 IP for example. These are available on the Internet at:
www.siemens.com/gigaset.

Configuring info services/activating idle display

You can configure your registered handsets to display customised text information (e.g. weather reports, RSS feeds) on the idle display.

Customising info services

Please note

The weather report is preset.

- Open the following Web page: Settings → Services.

Area: Info Services Configuration

- Click on the www.gigaset.net/myaccount link
  or enter the URL into the address field of a different browser window.
  The Web page for Gigaset.net info services is opened.
- Enter the user ID and password that can be found in the Info Services Configuration area of the Web configurator Services page in the links on the Gigaset.net page.

This will open a website where you can compile your info service.

- Define which information should be sent regularly to your handset.

Activating display info services

You can display the selected text information on each Gigaset A58H, C38H, S67H, S68H, SL37H or C47H handset that is registered to your base station.

Using the Web configurator:

- Open the following Web page: Settings → Services.

Activate Info Services

- Select Yes to activate the display of text information.

Select No to deactivate display of text information for all registered handsets.

- Select the Set button to save the settings in your phone.

On the handset:

- Gigaset A58H and C38H:
  - Set the screensaver Info Services on the handset (→ Page 91).

  If up-to-date text information is available, these will be displayed as a scrolling message under the digital clock on the handset in idle display mode (→ Page 91).
- Gigaset S67H, S68H, SL37H and C47H
  - Set the screensaver Digital Clock on the handset.

  Text information is displayed as a full screen in idle display mode. It overwrites the digital clock.

Selecting and registering online directories for access

You can use online directories (net directory and classified directory) on your handset. You can use your telephone's Web configurator to define which online directory you wish to use.

You can also elect to display the name under which the caller making an incoming call is saved in the online or Gigaset.net directory (Display of caller's name) – in the call display on the handset and in the caller list.

Precondition: This function is supported by the provider of the online directory.

- Open the following Web page: Settings → Services.

  The settings are made in the Online Directory area.
- Select the provider whose online directory you wish to use from the Provider list. Select "---" if you do not wish to use an online directory.

The following fields are displayed depending on the Provider you select:

Display of caller's name

This is displayed if the provider supports transmission of the caller name from the online directory to the call display.

- Select On to update the display.

Authentication Name, Authentication password

These are displayed if you need to register with the provider to gain access to certain services:

- Some providers require you to register every time you want to access their online directory. They require registration with user name and password for access to the online directory. You will need to save this data to your base station.
- Other providers differentiate between standard and premium services. You can access standard services **without** entering user name and password. But you will have to register to use the premium services. You will need to save the access data in your base station to gain access to premium services.

- Enter the details you received from your provider in the Authentication Name and Authentication password fields.
- Select the Set button to save the settings in your phone.

Please note

- How to use online directories on handsets → Page 67.
- In the handset's list of online directories (press and **hold**), provider-specific names of the online directory are displayed. The standard names Online Directory and Yellow Pages are displayed in the local directory.
- If you select "– – –" from the Provider list, the entries for online and classified directories will not be displayed in the handset's list of online directories. The directory entries for the online directory and the classified directories are retained, but you cannot establish a connection with them.

Changing internal handset numbers and names

Each handset is **automatically** assigned an internal number (1 to 6) and an internal name ("INT 1", "INT 2" etc.) when it registers with the base station (→ Page 83).

The internal numbers and names of all registered handsets can be changed.

Please note

For information on how to change internal names and numbers, → Page 85.

- Open the following Web page: Settings → Handsets.

The names and internal numbers of all registered handsets are displayed in the Registered Handsets area.

- Select the handset whose number/name you want to change.
- **Changing numbers:** Select the internal number that you want to assign to the handset in the No. column of the handset. If a handset with this internal number already exists, you will also have to change the number allocation for this handset. The internal numbers 1-6 can only be assigned once each.
- **Changing names:** If necessary, change the name of the handset in the Name column. The name may contain up to 10 characters.
- If necessary, repeat the process for other handsets.
- Select Set to save the settings.

The changes are saved in the internal lists of all registered handsets. Handsets are sorted by their internal numbers in the internal list. The order of the handsets in the list can therefore be changed.

Please note

If an internal number has been entered twice, a message will appear. The internal numbers are not changed.

Loading and deleting handset directories to/from the PC.

The Web configurator has the following options for editing the directories of the registered handsets.

- ◆ Store the handset directories on a PC. Entries are stored in vCard format in a vcf file on the PC. You can edit these files with an ASCII editor (e.g. Notepad/ Editor in Windows Accessories) and load them onto any registered handset. You can also transfer directory entries to your PC address book (e.g. Outlook Express™ address book).
- ◆ Transfer contact details from your PC address book to handset directories. Export the contacts, e.g. with Outlook Express ™ to vcf files (vCards) and transfer them to handset directories using the Web configurator.
- ◆ Delete the directory on the handset. If you have edited the directory file (vcf file) on the PC and would like to use this modified directory on the handset, you can delete the current directory from the handset first.

  Tip: Back up the current directory on your PC before deleting it. You can then load it back onto the handset if the modified directory is affected by formatting errors and some, or all, of it cannot be loaded onto the handset.

Please note

- ◆ You can find information on the vCard format (vcf) on the Internet, e.g. under:
- ◆ www.en.wikipedia.org/wiki/VCard or www.de.wikipedia.org/wiki/VCard
- ◆ (You can set the display language at the bottom left-hand side in the navigation area of the Web page.)
- ◆ You can still load directories in tsv format from your PC to your handset.
- ◆ If you want to transfer a handset directory (vcf file) saved on the PC that contains numerous entries to a Microsoft Outlook™ address book, please note the following:
- ◆ Microsoft Outlook™ only ever transfers the first (directory) entry from the vcf file to its address book.

Preconditions:

- ◆ The handset can send and receive directory entries.
- ◆ The handset is activated and is in idle status.

▶ Open the following Web page: Settings → Handsets.

The names of all registered handsets are displayed in the Directory area.

▶ Select the handset for which you want to save or edit the directory. To do this, click on the option before the handset.

Loading the directory file from the PC to the handset

- In the Transfer directory to handset area, specify the vcf file you want to load on to the handset (complete path name), or select Browse... and navigate your way to the file.
- Select Transfer button to start the transfer.

The display will show how many of the entries from the vcf file are being transferred to the directory.

Transfer rules

The directory entries from a vcf file that are loaded onto the handset will be added to the directory. If an entry already exists for a name, it will either be supplemented or a new entry for the name will be created. The process will not overwrite or delete any phone numbers.

Please note

Depending on your handset type, up to 3 entries with the same name will be created in the handset directory for each vCard – one entry per entered number.

Loading the directory from the handset to the PC

- Select Save in the Handset Directory area. A Windows dialogue box will be shown to save the file.
- Enter the directory on the PC (complete path name) in which the directory file is to be stored. Select Save or OK.

Deleting the directory

- Select Delete in the Handset Directory area.
- Confirm the security prompt Telephone directory of the selected handset will be deleted. Continue? with OK.

This deletes all the entries in the directory, including the entries for online directories.

Please note

For how to delete the directory on the handset → Page 65.

Directory file content (vcf file)

The following data (if available) is written into the vcf file for entry into the directory or transferred from a vcf file into the handset directory.

1 Last name
2 First name
3 Number
4 Number (office)
5 Number (mobile)
6 E-mail address
7 Date (YYYY-MM-DD) and time of the reminder call (HH:MM) separated by a

"T" (example: 2008-01-12T11:00).

8 Identification as VIP (X-SIEMENS-VIP:1)

Other information that a vCard may contain is not entered into the handset directory.

Example for an entry in vCard format:

BEGIN:VCARD
VERSION:2.1
N:Smith;Anna
TEL;HOME:1234567890
TEL;WORK:0299123456
TEL;CELL:0175987654321
EMAIL:anna@smith.com
BDAY:2008-01-12T11:00
X-SIEMENS-VIP:1
END:VCARD

Activating VoIP status message display

You can display VoIP status messages on your handset when there are VoIP connection problems. These messages provide you with information on the status of a connection and contain a provider-specific status code that helps the service team when they are analysing the problem.

- Open the following Web page: Settings → Handsets.
  The settings are made in the Miscellaneous area.
- Select Yes next to Show VoIP status on handset to activate the status message display.
  If you select No, no VoIP status messages are displayed.
- Select Set to save the changes.

Please note

A table with possible status codes and their meaning can be found in the appendix → Page 157.

Starting a firmware update

If necessary, you can load updates of the base station firmware onto your phone.

The server on which Siemens makes new firmware versions available for your base station is set by default. The URL of the Internet server is displayed in the Data server field.

You should only change this URL under exceptional circumstances (e.g. if requested to do so due to a malfunction). This address is also used to load provider information from the Internet. You should therefore make a note of the default URL before you overwrite it. Otherwise, you will only be able to reactivate the default URL by resetting the base station back to the default settings (→ Page 98).

Please note

- When updating from the Internet, checks are made to ensure that no **newer** version of the firmware exists. If this is not the case, the operation is terminated.
- The firmware is only loaded from the Internet if you have not entered a local file in the User defined firmware file field prior to the update.

Preconditions:

- No calls are being made via the fixed line network or VoIP.
- There is no internal connection between registered handsets or to GHC devices.
- The base station menu is not open in any of the handsets.

▶ Open the following Web page: Settings → Miscellaneous.

▶ Select the Update Firmware button.

The firmware is updated. This process can take up to 3 minutes.

Please note

You can also start the firmware update on the handset (→ Page 100).

Firmware update from local firmware file

In exceptional circumstances you may receive, for example, a firmware file from Service that you can upload from your PC to your telephone (e.g. because the firmware update via the Internet did not work).

Precondition: A Web server is running on the local PC (e.g. Apache).

▶ First load the firmware file onto your PC.

▶ In the User defined firmware file field enter the IP address of the PC in your local network and the complete path and name of the firmware file on the PC. Example: 192.168.2.105/A580IP/FW_file.bin.

▶ Select Set to save the changes.

▶ Select the Update Firmware button to start the update.

This setting is automatically used for **this particular** firmware update. The URL in the Data server field is saved and used again for subsequent firmware updates. You will have to re-enter the IP address and file name if you need to carry out another update with a firmware file on your local PC.

Please note

If an error arises during a firmware update from a local PC, the most recent version of the firmware is automatically downloaded from the Internet.

Activating/deactivating the automatic version check

When the version check is activated, the phone checks on a daily basis whether the Siemens configuration server has a new version of the phone firmware or of the file with the general settings for your VoIP provider.

If a new version is available, a notification is sent to the handset and the message key flashes. You can then carry out an automatic update of the firmware (→ Page 100) or of the VoIP provider settings (→ Page 102).

- Open the following Web page: Settings → Miscellaneous.
- Select Yes next to Automatic check for software/profile updates to activate the automatic version check.

  Select No if you do not want a version check to be carried out.
- Select Set to save the changes.

Copying the date/time from time server

The date and time are shown in the idle display of registered handsets. They are important, for example, for stating the correct time in the calls list and for the "alarm clock" function.

There are two methods for updating the time and date on your base station: manually with one of the registered handsets (→ Page 14) or automatically by synchronisation with a time server on the Internet.

Activate/deactivate synchronisation with a time server as follows:

- Open the following Web page: Settings → Miscellaneous.
- In the Automatic adjustment of System Time with Time Server field select Yes to activate synchronisation between base station and a time server. If you select No the base station will not adopt time settings from a time server. In this case you should set the time and date manually using a handset.
- The Last synchronisation with time server field shows the last time the base station compared the time and date settings with a time server.
- In the Time Server field, enter the Internet address or name of the time server from which the base station should adopt its time and date settings (maximum 74 characters). The time server "europe.pool.ntp.org" is set as default on the base station. You can overwrite the setting.
- From the Country list, select the country in which your base station is being operated.
- The Time Zone field shows the valid time zone for the Country. It shows the deviation between local time (not standard time) and Greenwich Mean Time (GMT).

  If a country is divided into various time zones, they will all appear in the list. Select the appropriate Time Zone for the base station from the list.

- The Automatically adjust clock to summer-time changes field will be displayed if your time zone differentiates between summer time and standard time.

  Select On if you want the time to change automatically to summer time or standard time when summer time begins and ends respectively.

  Select Off if you do not want to change to summer time.

  Please note: If the date and time are updated by a time server that automatically switches between summer time and standard time, you must always select Off here.
- Select the Set button to save the settings in your phone.

Once you have activated synchronisation, the time and date will be compared with a time server as soon as an Internet connection is established.

Synchronisation will usually occur once a day (at night) if synchronisation is activated. Any additional synchronisation will take place only after each new system start of the base station (e.g. after a firmware update or a power cut).

If you register a new handset on your base station it will assume the time and date of the base station without any additional synchronisation with the time server.

Date and time settings are transferred to every handset after synchronisation.

Please note

- The default time server "europe.pool.ntp.org" will remain stored in the base station even if you overwrite it. If you delete your time server from the Time Server field and synchronisation is still activated, the base station will continue to synchronise with the default time server. However, it will no longer appear in the Time Server field.
- If you have entered your own time server in the Time Server field and the base station is unable to synchronise for ten consecutive attempts, the base station will synchronise with the default time server.
- If you have deactivated synchronisation with a time server, and if the date and time are not set on any handset, then the base station will attempt to reference date and time settings from the CLIP information of an incoming call.

Querying the phone status

General information about your phone is displayed.

- In the menu list, select the Status tab.

The following information is displayed:

Area: IP Configuration

IP address
The phone's current IP address within the local network. For assigning the IP address → Page 115.

MAC address
The phone's device address.

Area: Software

Firmware version
Version of the firmware currently downloaded. You can download updates of the firmware on your phone (→ Page 100). Firmware updates are available on the Internet.

EEPROM version
Version of your phone's EEPROM storage chip (→ Page 182).

Service (Customer Care)

You can get assistance easily when you have technical questions or questions about how to use your device by using our online support service on the Internet at:

http://www.siemens.com/gigasetcustomercare

This site can be accessed at any time wherever you are. It will give you 24/7 support for all our products. It also a list of FAQs and answers plus user guides for you to download. You will also find frequently asked questions and answers in the **Questions and Answers** section of this user guide in the appendix.

If the device needs to be repaired, please contact one of our Customer Care Centers:

Abu Dhabi97 12 62 23 800
Argentina0800-888-9878
Australia1300 665 366
Austria05 17 07 50 04 (0,065 Euro/Min.)
Bahrain97 31 73 11 173
Belgium0 78 15 66 79
Bosnia Herzegovina033 276 649
Brazil
Grande Capitais e Regiões Metropolitanas:
4003 3020
R$ 0,14 (US$ 0,069)
Demais localidades: 0800 888 3020
R$ 0,20 (US$ 0,098)
Bulgaria02 873 94 88
Canada866 247 8758
China0 21 400 670 6007 (RMB 0.11)
Croatia016 10 53 81 (0,23 Kun)
Czech Republic233 032 727
Denmark35 25 86 00
Dubai97 14 39 69 944
Egypt202 7623441
Finland09 23 11 34 25
France01 56 38 42 00 (Appel national)
Germany01805 333 222
(0,14 Euro/Min. aus dem Festnetz der Deutschen Telekom.
Für Anrufe aus den Mobilfunknetzen können abweichende Preise gelten)
Greece801 1000 500 (0,026 Euro)
Hungary06 14 71 24 44 (27 Ft)
IndiaPlease refer to your local warranty card
Ireland18 50 77 72 77
Israel1 700 700 727
Italy199.15.11.15
Jordan00962 6 5625460/1/2
Kuwait00965-2464993
Latvia7 50 11 18
Lebanon00961-1236110
Luxembourg 40 66 61 56 40
Malaysia603 77124304
Malta+353 21 4940 632
Mexico01 80 07 11 00 03
Netherlands
0900-3333102 (0,25 Euro/min.)
New Zealand08 00 27 43 63
Norway22 70 84 00
Oman96 82 47 09 281
Poland0 801 140 160
Portugal(351) 808 781 223
Romania+4.021.529.7114.
Russia8 (495) 228 1312
Serbia0800 222 111
Singapore62 27 11 18
Slovak Republic
02 59 68 22 66 (4,428 sk)
Slovenija0 14 74 63 36
South Africa08 60 10 11 57
Spain902 103935
Sweden08 750 99 11
Switzerland
0848 212 000 (0,08 SFr./Min.)
Taiwan02 23 96 10 06
Thailand02 722 1118
Turkey0216 459 98 59
Ukraine+380-44-451-71-72
United Arab Emirates0 43 66 03 86
United Kingdom0 84 53 67 08 12
USA1-866 247-8758
(toll free)

Please have your record of purchase ready when calling.

Replacement or repair services are not offered in countries where our product is not sold by authorised dealers.

Questions and answers

If you have any questions about using your phone, visit us at www.siemens.com/gigasetcustomercare at any time. The table below contains a list of common problems and possible solutions.

Please note

To support the service team, it can be helpful if you have the following information to hand:

– Version of firmware, EEPROM and your phone's MAC address

- You can check this information with the Web configurator (→ Page 151). For how to display the MAC address on your handset → Page 105.
- VoIP status code (→ Page 157)
- For problems with VoIP connections, you should set VoIP status messages to be displayed on your handset (→ Page 102, Page 147). These messages contain a status code that helps when the problem is analysed.

The display is blank. 1. The handset is not switched on. ▸ Press **and hold** the end call key . 2. The battery is flat. ▸ Charge the battery or replace it (→ Page 12).
"Base" **flashes in the display.** 1. The handset is out of range of the base station or the base station's range has decreased because Eco mode is active. ▸ Move the handset closer to the base station. ▸ If necessary, deactivate Eco mode (→ Page 81). 2. The handset has been deregistered. ▸ Register the handset (→ Page 83). 3. The base station is not turned on. ▸ Check the base station's mains adapter (→ Page 18). 4. The base station firmware is currently being updated (automatically). ▸ Pease wait until the update is complete.
Handset does not ring. 1. The ringer is deactivated. ▸ Activate the ringer melody (→ Page 95). 2. Call forwarding set to All Calls. ▸ Deactivate call forwarding (fixed line network → Page 54; VoIP → Page 55/ Page 132).
You cannot hear a ringer melody/dialling tone from the fixed line network. The phone cord supplied has not been used or has been replaced by a new cord with the wrong pin connections. ▸ Please always use the phone cord supplied or ensure that the pin connections are correct when purchasing from a retailer (→ Page 165).

Error tone sounds after system PIN prompt

You have entered the wrong system PIN.

- Re-enter system PIN.

Have you forgotten the system PIN?

- Reset the base station to set the system PIN back to 0000 (→ Page 98).

The other party cannot hear you.

You have pressed the INT key or the INT display key. The handset is "muted".

- Press the display key to re-activate the microphone (→ Page 47).

When making calls from the fixed line network, the caller's phone number is not displayed although CLIP (→ ④④ is set.

Calling Line Identification is not enabled.

- The **caller** should ask his network provider to enable Calling Line Identification (CLI).

You hear an error tone when keying an input
(a descending tone sequence).

Action has failed/invalid input.

- Repeat the operation.
  Watch the display and refer to the user guide if necessary.

You cannot connect to the router and the phone is assigned a static IP address.

- Check on the router whether the IP address is already being used by another device in the LAN or belongs to the block of IP addresses that is reserved on the router for dynamic address assignment.
- If necessary, change the phone's IP address (→ Page 103).

You have made a call via VoIP but cannot hear the other caller.

Your phone is connected to a router with NAT/firewall.

- Your STUN server (→ Page 123) or outbound proxy (→ Page 124) settings are incomplete or incorrect. Check the settings.
- No outbound proxy is entered or the outbound proxy mode Never is activated (→ Page 124) and your phone is connected to a router with symmetric NAT or a blocking firewall.
- Port forwarding is activated on your router, but no permanent IP address has been assigned to your phone.

You cannot make calls via VoIP. Server not accessible is displayed.

- First wait a few minutes. This is often a short-term event that corrects itself after a short time.

If the message continues to be displayed, proceed as follows:

- Check whether your phone's Ethernet cable is correctly connected to the router.
- Check your router's cable connection to the Internet.
- Check whether the phone is connected to the LAN. Send a ping command, e.g. from your PC, to the phone (ping ␣ <local IP address of the phone>). It may be that no IP address could be assigned to the phone or a permanently set IP address is already assigned to another LAN subscriber. Check the settings on the router, you may have to activate the DHCP server.

You cannot make calls via VoIP. SIP registration failed is displayed.

- First wait a few minutes. This is often a short-term event that corrects itself after a short time.

The message may still be displayed for the following reasons:

1. The personal VoIP access data (Username, Authentication Name and Authentication Password) that you have entered is incomplete or incorrect.
   - Check your information. Particularly check your use of upper and lower case.
2. The general settings for your VoIP provider are incomplete or incorrect (incorrect server address).
   - Start the Web configurator and check the settings.

You cannot make calls via VoIP. VoIP config. error: xxx appears in the display (xxx = VoIP status code).

You are trying to make a call via a VoIP connection that is not properly configured.

- Start the Web configurator and check the settings. Possible status codes and their meanings are listed on Page 157.

The phone does not dial an entered number. The display shows Not possible!.

The number may be blocked (dialling plan).

- Open the Dialling Plans Web page of the Web configurator and delete or deactivate the block if necessary.

You cannot establish a connection to the phone with your PC's Web browser.

- When establishing a connection, check the phone's local IP address that has been entered. You can check the IP address on your handset.
- Check the LAN connections for the PC and phone.
- Check that your phone can be reached. Send a ping command, e.g. from your PC, to the phone (ping ␣ <local IP address of the phone>).
- You have tried to reach the phone via a secure http (https://...). Try again with http://....

You cannot be reached for calls from the Internet.

- There is no entry for your phone in your router's routing table. Check the settings for the NAT refresh time (→ Page 124).
- Your phone is not registered with the VoIP provider.
- You have entered the wrong user ID or an incorrect domain (→ Page 121).

No firmware update or VoIP profile download is carried out.

1. If Not possible, try later. is displayed, the VoIP connections may be busy or a download/update is already being carried out.
   - Repeat the process at a later time.
2. If File corrupt is displayed, the firmware or profile file may be invalid.
   - Please only use firmware and downloads that are made available on the preconfigured Siemens server (→ Page 147) or at www.siemens.com/gigasetcustomercare.
3. If Server not available is displayed, the download server may not be accessible.
   - The server is currently not accessible. Repeat the process at a later time.
   - You have changed the preconfigured server address (→ Page 147). Correct the address. If necessary, reset the base station.
4. If Transmission error XXX is displayed, an error has occurred during the transmission of the file. An HTTP error code is displayed for XXX.
   - Repeat the process. If the error occurs again, consult the Service department.
5. If Please check IP settings is displayed, your phone may not be connected to the Internet.
   - Check the cable connections between the phone and router and between the router and the Internet.
   - Check whether the phone is connected to the LAN, i.e. it can be reached at its IP address.

You cannot listen to or control a network mailbox.

VoIP:
Your VoIP provider does not support the type of DTMF signalling set up on your phone.
- Ask your VoIP provider which signalling it supports and change the settings on your phone (→ Page 132) if necessary.

When operating the base station within a PABX:
Your PABX is set for dial pulsing.
- Set your PABX to touch tone dialling.

No time is specified for a message in the calls list.

Date and time have not been set.
- Set date/time (→ Page 14) or
- Activate base station synchronisation with a time server on the Internet (→ Page 149).

VoIP status codes

If you have problems with your VoIP connections, activate the Status on HS function (→ Page 105, Page 147). You will then receive a VoIP status code that will support you during the problem analysis. Provide the code to the Service department during the problem analysis.

In the following tables you will find the meaning of the most important status codes and messages.

Status code	Meaning
0x31	VoIP config. error: IP domain not entered.
0x33	VoIP config. error: SIP user name (Authentication Name) not entered. This is shown, for example, when dialling with a line suffix, if no connection is configured for the suffix on the base station.
0x34	VoIP config. error: SIP password (Authentication password) not entered.
0x300	The called party can be reached under several phone numbers. If the VoIP provider supports this, a list of the phone numbers is transmitted as well as the status code. The caller can select to which number he wants to make the connection.
0x301	Permanently redirected. The called party can no longer be reached under this number. The new number is transferred to the phone together with the status code, and the phone then no longer accesses the old number but dials the new address immediately.
0x302	Temporarily redirected. The phone is informed that the called party cannot be reached under the dialled number. The call is redirected for a limited period. The phone is also notified of the length of the redirection.
0x305	The query is sent to a different "proxy server", e.g. to balance incoming queries. The phone will make the same query to another proxy server. This is not a redirection of the address per se.
0x380	Other service: The query or call could not be transferred. However, the phone is informed of the other options available to connect the call.
0x400	Wrong call
0x401	Not authorised
0x403	The requested service is not supported by the VoIP provider.
0x404	Wrong phone number. No connection on this number. Example: While making a local call you have not dialled the area code although your VoIP provider does not support local calls.
0x405	Method not permitted.
0x406	Not acceptable. The requested service cannot be provided.
0x407	Proxy authentication required.
0x408	The party cannot be reached (e.g. account has been deleted).

Status code	Meaning
0x410	The requested service is not available from the VoIP provider.
0x413	Message is too long.
0x414	URI is too long.
0x415	Query format is not supported.
0x416	URI is faulty.
0x420	Incorrect ending
0x421	Incorrect ending
0x423	The requested service is not supported by the VoIP provider.
0x480	The dialled number is temporarily unavailable.
0x481	The recipient is not available.
0x482	Double service query
0x483	Too many "jumps": The query was rejected because the service server (proxy) has decided that this query has already passed through too many service servers. The maximum number is defined beforehand by the original sender of the query.
0x484	Wrong number: In most cases this response means that you have simply omitted one or more digits in the phone number.
0x485	The URI dialled is not unique and cannot be processed by the VoIP provider.
0x486	The called party is busy.
0x487	General faults: The call was cancelled before a call was established. The status code confirms receipt of the interruption signal.
0x488	The server cannot process the query because the data entered in the media description is not compatible.
0x491	The server notifies that the query will be processed as soon as a previous query has been completed.
0x493	The server rejects the query because the phone cannot decrypt the message. The sender has used an encryption method that neither the server nor the receiver phone can decrypt.
0x500	The proxy or the receiving device has discovered a fault while executing the query. It is therefore impossible to execute the query. If this occurs, the caller or the phone displays the fault and repeats the query after a few seconds. If applicable, the number of seconds after which the query can be repeated may be transmitted to the caller or phone by the receiving device.
0x501	The query cannot be processed by the recipient because the recipient does not have the functionality that the caller requires. If the recipient understands the query but does not process it because the sender does not have the necessary rights or the query is not permitted in the current context, status code 405 is transmitted instead of 501.
0x502	In this case, the receiving device that transmits this error code is a proxy or a gateway and has received an invalid response from its gateway via which this query is to be processed.

Status code	Meaning
0x503	The query can currently not be processed by the receiving device or the proxy because the server is either overloaded or is being serviced. If it is possible for the query to be repeated in the foreseeable future, the server informs the caller or the phone.
0x504	Time limit exceeded at the gateway.
0x505	The server rejects the query because the indicated version number of the SIP protocol does not at least concur with the version that is used by server or the SIP device involved in this query.
0x515	The server rejects the query because the message exceeds the maximum permitted size.
0x600	The called party is busy.
0x603	The called party has rejected the call.
0x604	The called URI does not exist.
0x606	The communication settings are not acceptable.
0x701	The called party has hung up.
0x703	Connection cancelled because of time-out.
0x704	Connection interrupted because of a SIP error.
0x705	Wrong dialling tone
0x706	No connection established
0x751	Busy tone: No codec match between the calling and called party.
0x810	General socket layer error: User is not authorised.
0x811	General socket layer error: Wrong socket number
0x812	General socket layer error: Socket is not connected.
0x813	General socket layer error: Memory error.
0x814	General socket layer error: Socket not available - check IP settings/connection problem/VoIP setting incorrect.
0x815	General socket layer error: Illegal application on the socket interface.

Checking service information

You may need the service information of your phone (base station and handset) for Customer Services.

Base station service information

Precondition: You are conducting an external call. The connection has been established for at least 8 seconds.

Menu → Service Info

Confirm selection with OK.

The following information is displayed:

1: Serial number of the base station (RFPI)

2: Serial number of your handset (IPUI)

3: Informs the service employees of the base station settings (in hex diagram), e.g. the number of registered handsets, repeater mode. The last 4 digits indicate the number of operating hours (hexadecimal).

4: Variant, version of the firmware (digits 3 to 5).

5: Gigaset.net number of your phone. With this number a service employee can call you over the Internet without you needing to be registered with a VoIP provider. This means that the employee can test online connections and VoIP telephony irrespective of the VoIP provider.

Handset service information

When the handset is in idle status:

Press the display key Menu.

▶ Press the following keys one after the other: * # 0 6 #

The information displayed on the handset includes:

1: Serial number (IPUI)

2: Number of operating hours

3: Variant, version of handset software

Authorisation

This device is intended for analogue phone lines in your network.

Voice over IP telephony is possible with an additional modem via the LAN interface.

Country-specific requirements have been taken into consideration.

We, Siemens Home and Office Communication Devices GmbH & Co. KG, declare that this device meets the essential requirements and other relevant regulations laid down in Directive 1999/5/EC.

A copy of the 1999/5/EC Declaration of Conformity is available at this Internet address:
http://www.siemens.com/gigasetdocs.

CE 0682

Environment

Our environmental mission statement

We at Siemens Home and Office Communication Devices GmbH & Co. KG carry social responsibility and are actively engaged in the interests of a better world. Our ideas, technologies and actions serve people, society and the environment. The aim of our global activity is to secure sustainable life resources for humanity. We are committed to a responsibility for our products that comprises their entire life cycle. The environmental impact of products, including their manufacture, procurement, distribution, use, service and disposal, are already evaluated during product and process design.

Further information on environmentally friendly products and processes is available on the Internet under gigaset.siemens.com.

Environmental management system

Siemens Home and Office Communication Devices is certified pursuant to the international standards EN 14001 and ISO 9001.

ISO 14001 (Environment): certified since September 2007 by TüV SÜD Management Service GmbH.

ISO 9001 (Quality): certified since 17/02/1994 by TüV Süd Management Service GmbH.

Ecological energy consumption

The use of ECO DECT (→ Page 81) saves energy and is an active contribution towards protecting the environment.

Disposal

Batteries should not be disposed of in general household waste. Observe the local waste disposal regulations, details of which can be obtained from your local authority or the dealer you purchased the product from.

All electrical and electronic equipment must be disposed of separately from general household waste using the sites designated by local authorities.

If a product displays this symbol of a crossed-out rubbish bin, the product is subject to European Directive 2002/96/EC.

The appropriate disposal and separate collection of used equipment serve to prevent potential harm to the environment and to health. They are a precondition for the re-use and recycling of used electrical and electronic equipment.

For further information on disposing of your used equipment, please contact your local authority, your refuse collection service or the dealer you purchased the product from.

Appendix

Care

Wipe the base station, charging cradle and handset with a **damp** cloth (do not use solvent) or an antistatic cloth.

Never use a dry cloth. This can cause static.

Contact with liquid ⚠

If the handset has come into contact with liquid:

- **Switch off the handset and remove the battery pack immediately**.
- Allow the liquid to drain from the handset.
- Pat all parts dry, then place the handset with the battery compartment open and the keypad facing down in a dry, warm place **for at least 72 hours** (**not** in a microwave, oven etc.).
- **Do not switch on the handset again until it is completely dry.**

When it has fully dried out, you will normally be able to use it again.

Specifications

Recommended batteries

Technology:
Nickel-metal-hydride (NiMH)

Size: AAA (Micro, HR03)

Voltage: 1.2 V

Capacity: 550–1000 mAh

We recommend the following battery types, because these are the only ones that guarantee the specified operating times, full functionality and long service life:

- GP 700 mAh
- Yuasa Phone 700 mAh
- Yuasa Phone 800 mAh
- Yuasa AAA 800
- Peacebay 600 mAh

The device is supplied with two recommended batteries.

Handset operating times/charging times

The operating time of your Gigaset depends on the capacity and age of the batteries and the way they are used. (All times are maximum possible times).

	Capacity (mAh) approx.			
	550	650	800	1000
Standby time (hours)	180	210	265	330
Talktime (hours)	23	25	33	41
Operating time for 1.5 hrs of calls per day (hours)	80	95	115	145
Charging time, base station (hours)	8	10	12	15
Charging time, charging cradle (hours)	6	7	9	11

At the time of going to print, batteries up to 800 mAh were available and had been tested in the system. Due to the constant progression in battery development, the list of recommended batteries in the FAQ section of the Gigaset Customer Care pages is regularly updated:

www.siemens.com/gigasetcustomercare

Base station power consumption

The power consumption for the base station is approx. 1.3 watt.

Power consumption of the charging cradle

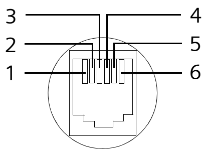
Charging (max. charge current):	approx. 1.4 watt
Sustained charge:	approx. 0.6 watt
Not charging (only the network adaptor):	approx. 0.3 watt

General specifications

Interfaces	Fixed line network, Ethernet
No. of channels	60 duplex channels
Radio frequency range	1880-1900 MHz
Duplex method	Time multiplex, 10 ms frame length
Channel grid	1728 kHz
Bit rate	1152 kbit/s
Modulation	GFSK
Language code	32 kbit/s
Transmission power	10 mW, average power per channel
Range	up to 300 m outdoors, up to 50 m indoors
Base station power supply	230 V ~/50 Hz
Environmental conditions in operation	+5°C to +45°C; 20% to 75% relative humidity
Codecs	G.711, G.726, G.729AB with VAD/CNG, G.722
Quality of Service	TOS, DiffServ
Protocols	DECT, GAP, SIP, RTP, DHCP, NAT Traversal (STUN), HTTP
Dialling mode	DTMF (touch tone dialling)/DP (dial pulsing)

Pin connections on the telephone jack

◆ If you buy a replacement phone cord from a retailer, make sure that the phone jack has the correct pin assignment.



1 unused
2 unused
3 a
4 b
5 unused
6 unused

Writing and editing text

The following rules apply when writing text:

◆ Use to move the cursor to the right or left.

◆ Characters are inserted to the left of the cursor.

◆ The first letter of the name of directory entries is automatically capitalised, followed by lower case letters.

Writing text/names

Press the relevant key several times to enter the corresponding letters/characters.

Standard characters

	1x	2x	3x	4x	5x	6x	7x	8x	9x	10x
1	␣ [1)]	↵ [2)]	1							
2 abc	a	b	c	2	ä	á	à	â	ã	ç
3 def	d	e	f	3	ë	é	è	ê		
4 ghi	g	h	i	4	ï	í	ì	î		
5 jkl	j	k	l	5						
6 mno	m	n	o	6	ö	ñ	ó	ò	ô	õ
7 pqrs	p	q	r	s	7	ß				
8 tuv	t	u	v	8	ü	ú	ù	û		
9 wxyz	w	x	y	z	9	ÿ	ý	æ	ø	å
0 +	.	,	?	!	0					

1)Space
2)Line break

When you press and **hold** a key, the characters of that key appear in the display and are highlighted one after the other. When you release the key, the highlighted character is inserted into the input field.

Setting upper/lower case or digits

Press the hash key # **briefly** to switch from "Abc" mode to "123" and from "123" to "abc" and from "abc" to "Abc" (upper case: 1st letter upper case, all others lower case). Press the hash key # **before** entering the letter.

You can see briefly in the display whether upper case, lower case or digits is selected.

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  http://www.siemens.com/gigaset/developer

For more information and Internet links to the source text of the free software, see the Online Support pages on the Internet at:

www.siemens.com/gigasetcustomercare

If it is not already supplied with the product, you can request the source text, including copyright notices, from Siemens. There is a charge to cover the cost of copying and postage. Please submit this request by e-mail or fax to the following address or fax number within 3 years of purchasing this product. Please state the exact device type plus the version number of the installed device software.

Small Parts Dispatch Com Bocholt

E-mail: kleinteileversand.com@siemens.com

Fax: 0049 2871 / 91 30 29

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[This is the first released version of the Lesser GPL. It also counts as the successor of the GNU Library Public License, version 2, hence the version number 2.1.]

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When a program is linked with a library, whether statically or using a shared library, the combination of the two is legally speaking a combined work, a derivative of the original library. The ordinary General Public License therefore permits such linking only if the entire combination fits its criteria of freedom. The Lesser General Public License permits more lax criteria for linking other code with the library.

We call this license the "Lesser" General Public License because it does less to protect the user's freedom than the ordinary General Public License. It also provides other free software developers less of an advantage over competing non-free programs. These disadvantages are the reason we use the ordinary General Public License for many libraries. However, the Lesser license provides advantages in certain special circumstances.

For example, on rare occasions, there may be a special need to encourage the widest possible use of a certain library, so that it becomes a de-facto standard. To achieve this, non-free programs must be allowed to use the library. A more frequent case is that a free library does the same job as widely used non-free libraries. In this case, there is little to gain by limiting the free library to free software only, so we use the Lesser General Public License.

In other cases, permission to use a particular library in non-free programs enables a greater number of people to use a large body of free software. For example, permission to use the GNU C Library in non-free programs enables many more people to use the whole GNU operating system, as well as its variant, the GNU/Linux operating system.

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Accessories

Gigaset handsets

Upgrade your Gigaset to a cordless PABX:

Gigaset A58H handset

- Illuminated graphic display
- Illuminated keypad
- Handsfree talking
- Polyphonic ringer melodies
- Directory for around 150 entries
- SMS (precondition: CLIP must be enabled)

www.siemens.com/gigasetA58H

Gigaset C38H handset

- Illuminated graphical colour display (65k colours)
- Illuminated keypad
- Handsfree talking
- Polyphonic ringer melodies
- Directory for around 150 entries
- SMS (precondition: CLIP must be enabled)
- Headset socket

www.siemens.com/gigasetC38H

Gigaset C47H handset

- Illuminated graphical colour display (65k colours)
- Illuminated keypad
- Handsfree talking
- Polyphonic ringer melodies
- Directory for around 150 entries
- SMS (precondition: CLIP must be enabled)
- Headset socket
- Room monitor

www.siemens.com/gigasetC47H

Gigaset S67H or S68H handset

- HDSP ready
- Illuminated graphical colour display (65k colours)
- Illuminated keypad
- Handsfree talking
- Polyphonic ringer melodies
- Directory for around 250 entries
- Picture CLIP
- SMS (precondition: CLIP must be enabled)
- Headset socket
- Bluetooth (Gigaset S68H only)
- Room monitor

www.siemens.com/gigasetS67H

www.siemens.com/gigasetS68H

Gigaset SL37H handset

- HDSP ready
- Illuminated graphical colour display (65k colours)
- Illuminated keypad
- Handsfree talking
- Polyphonic ringer melodies
- Directory for around 250 entries
- Picture CLIP
- SMS (precondition: CLIP must be enabled)
- PC interface e.g. for managing directory entries, ringer melodies and screensavers
- Headset socket
- Bluetooth
- Room monitor
- Walky-talky function

www.siemens.com/gigasetSL37H

Gigaset S45 handset

- Illuminated colour display (4096 colours)
- Illuminated keypad
- Handsfree talking
- Polyphonic ringer melodies
- Directory for around 150 entries
- SMS (precondition: CLIP must be enabled)
- Headset socket
- Room monitor

www.siemens.com/gigaset

Gigaset repeater

The Gigaset repeater can be used to increase the reception range of your Gigaset handset to the base station.

www.siemens.com/gigasetrepeater

Gigaset HC450 – door intercom for cordless phones

- Use the intercom from your cordless phone - no need for a fixed home phone
- Intuitive user functions using the display keys (open door, switch on entry light)
- Simple to configure using the handset menu
- Forward to outside phone numbers (intercom feature)
- Simple to install and register with the Gigaset system
- Replaces existing call button – no further cable is needed
- Supports the existing doorbell and standard door opener
- Configuration options for the second ringer key (separate intercom call, activating entrance lighting, or function such as first ringer key)

www.siemens.com/GigasetHC450

All accessories are available from your phone retailer.

SIEMENS Original Accessories

Only use original accessories. This will avoid possible health risks and damage to property, and also ensure that all the relevant regulations are complied with.

Glossary

A

ADSL
Asymmetric Digital Subscriber Line
Special form of DSL.

ALG
Application Layer Gateway
NAT control mechanism of a router.

Many routers with integrated NAT use ALG. ALG lets the data packets in a VoIP connection pass and adds the public IP address of the secure private network.

The router's ALG should be deactivated if the VoIP provider offers a STUN server or an outbound proxy.

→ Firewall, NAT, Outbound proxy, STUN

Authentication
Restriction of access to a network/service by use of an ID and password to log in.

Automatic ringback
→ Ringback when the number is busy.

B

Block dialling
Enter the complete phone number, and correct it if necessary. Then pick up the handset or press the handsfree key to dial the phone number.

Broadband Internet access
→ DSL.

Buddy
Subscriber with whom you exchange brief messages on the Internet in real time (chatting).

→ Instant messaging.

C

Call Diversion
→ Call forwarding.

Call forwarding
CF
Automatic forwarding (CF) of a call to a different telephone number. There are three kinds of call forwarding:

- CFU, Call Forwarding Unconditional
- CFB, Call Forwarding Busy
- CFNR, Call Forwarding No Reply

Call waiting = CW.

VoIP provider feature. A beep during a call indicates that another caller is waiting. You can accept or reject the second call. You can activate/deactivate the feature.

CF Call forwarding

→ Call forwarding.

Chatting Form of communication on the Internet. During a chat, brief messages are exchanged between the communicating parties in real time. Chatting in this sense is understood to be a written form of communication.

Client Application that requests a service from a server.

Codec Coder/decoder

Codec is a procedure that digitises and compresses analogue voice before it is sent via the Internet, and decodes – i.e. translates into analogue voice – digital data when voice packets are received. There are different codecs, with differing degrees of compression, for instance.

Both parties involved in the telephone connection (caller/sender and recipient) must use the same codec. This is negotiated between the sender and the recipient when establishing a connection.

The choice of codec is a compromise between voice quality, transmission speed and the necessary bandwidth. A high level of compression, for example, means that the bandwidth required for each voice connection is low. However, it also means that the time needed to compress/decompress the data is greater, which increases execution time for data in the network and thus impairs voice quality. The time required increases the delay between the sender speaking and the recipient hearing what has been said.

COLP / COLR

Connected Line Identification Presentation/Restriction

Service characteristic of a VoIP connection for outgoing calls.

COLP displays the phone number accepting the call on the calling party's display unit.

The number of the party accepting the call is different to the dialled number, e.g. if the call is forwarded or transferred.

The called party can use COLR (Connected Line Identification Restriction) to prevent the number from appearing on the calling party's display.

Connection assistant

The connection assistant starts automatically when you set the handset and base station up for the first time, or when you try to connect to the Internet before making the necessary settings.

Connection wizard

→ Connection assistant.

Consultation call

You are on a call. With a consultation call, you interrupt the conversation briefly to establish a second connection to another participant. If you terminate the connection to this participant immediately, then this was a consultation call. If you switch between the first and second participant, it is called **Toggling**.

CW

Call waiting

→ **Call waiting**.

D

DHCP

Dynamic Host Configuration Protocol

Internet protocol that handles the automatic assignment of **IP addresses** to **Network subscriber**. The protocol is made available in the network by a server. A DHCP server can, for example, be a router.

The phone contains a DHCP client. A router that contains a DHCP server can assign the IP addresses for the phone automatically from a defined address block. The dynamic assignment means that several **Network subscriber** can share one IP address, although they can only use it alternately and not simultaneously.

With some routers you can specify that the IP address for the phone is never changed.

Displayed name

VoIP provider feature. You can specify any name that is to be shown to the other party during a call instead of your phone number.

DMZ (Demilitarised Zone)

DMZ describes a part of a network that is outside the firewall.

A DMZ is set up, as it were, between a network you want to protect (e.g. a LAN) and a non-secure network (e.g. the Internet). A DMZ permits unrestricted access from the Internet to only one or a few network components, while the other network components remain secure behind the firewall.

DNS

Domain Name System

Hierarchical system that permits the assignment of **IP addresses** to **Domain names** that are easier to note. This assignment has to be managed by a local DNS server in each (W)LAN. The local DNS server determines the IP address, if necessary by enquiring about superordinate DNS servers and other local DNS servers on the Internet.

You can specify the IP address of the primary/secondary DNS server.

→ **DynDNS**.

Domain name

Name of one (of several) Web server(s) on the Internet (e.g. Siemens Home). The domain name is assigned to the relevant IP address by DNS.

DSCP

Differentiated Service Code Point

→ **Quality of Service (QoS)**.

DSL
Digital Subscriber Line

Data transfer technology that allows Internet access with, for e.g. **1.5 Mbps** over a conventional telephone line. Preconditions: DSL modem and the appropriate service offered by the Internet provider.

DSLAM
Digital Subscriber Line Access Multiplexer

The DSLAM is a switch cabinet in an exchange at which all subscriber connectors converge.

DTMF
Dual Tone Multi-Frequency

Another description for dual tone multi-frequency dialling (DTMF).

Dynamic IP address

A dynamic IP address is assigned to a network component automatically via **DHCP**. The dynamic IP address for a network component can change every time it registers or at certain time intervals.

→ **Static IP address**

DynDNS
Dynamic DNS

Domain names and IP addresses are assigned via **DNS**. For **Dynamic IP addresses** this service is enhanced with "Dynamic DNS". This permits the use of a network component with a dynamic IP address as a **Server** on the **Internet**. DynDNS ensures that a service can always be addressed on the Internet under the same **Domain name** irrespective of the current IP address.

E

ECT
Explicit Call Transfer

Participant A calls participant B. The participant puts the connection on hold and calls participant C. Rather than connect everyone in a three-party conference, A now transfers participant B to C and hangs up.

EEPROM
Electrically Eraseable Programmable Read Only Memory

Memory building block in your phone with fixed data (e.g. default and customised settings) and data saved automatically (e.g. entries to the list of callers).

Ethernet network

Wired **LAN**.

F

Firewall

You can use a firewall to protect your network against unauthorised external access. This involves combining various measures and technologies (hard and/ or software) to control the flow of data between a private network you wish to protect and an unprotected network (e.g. the Internet).

→ **NAT**.

Firmware

Device software in which basic information is saved for the functioning of a device. A new version of the firmware can be loaded into the device's memory (firmware update) to correct errors or update the device software.

Flat rate
Billing system for an **Internet** connection. The Internet provider charges a set monthly fee. There are no additional charges for the duration of the connection or number of connections.

Fragmentation
Data packets that are too big are split into smaller packets (fragments) before they are transferred. They are put together again when they reach the recipient (defragmented).

Full duplex
Data transmission is a mode in which data can be sent and received at the same time.

G

G.711 a law, G.711 μ law
Standard for a **Codec**.

G.711 delivers a very good voice quality that corresponds to that in the ISDN fixed line. As there is little compression, the necessary bandwidth is around 64 kbit/s per voice connection, but the delay caused by coding/decoding is only 0.125 ms.

"a law" describes the European standard and "μ law" describes the North American/Japanese equivalent.

G.722
Standard for a **Codec**.

G.722 is a **broadband** language codec with a bandwidth of 50 Hz to 7 kHz, a net transmission rate of 64 kbit/s per language connection and integrated speech pause recognition and comfort noise generation (silence suppression).

G.722 delivers very good voice quality. A higher sampling rate provides clearer and better voice quality than other codecs and enables a speech tone in High Definition Sound Performance (HDSP).

G.726
Standard for a **Codec**.

G.726 delivers a good voice quality. It is inferior to the quality with codec **G.711** but better than with **G.729**.

G.729A/B
Standard for a **Codec**.

The voice quality is more likely to be lower with G.729A/B. As a result of the high level of compression, the necessary bandwidth is only around 8 kbit/s per voice connection, but the delay is around 15 ms.

Gateway
Connects two different **Networks**, e.g. a router as an Internet gateway.

For phone calls from **VoIP** to the telephone network, a gateway has to be connected to the IP network and the telephone network (gateway/VoIP provider). It forwards calls from VoIP to the telephone network as required.

Gateway provider
→ **SIP provider**.

Global IP address
→ **IP address**.

GSM
Global System for Mobile Communication

Originally, European standard for mobile networks. GSM can now be described as a worldwide standard. In the USA and Japan national standards are now more frequently supported than in the past.

H

Headset
Combination of microphone and headphone. A headset makes handsfree talking more convenient. There are headsets available that can be connected to the handset by a cable.

HTTP proxy

Server via which the **Network subscriber** can process their Internet traffic.

Hub
Uses one **Infrastructure network** to connect several **Network subscriber**. All data sent to the hub by one network subscriber is forwarded to all network subscribers.

→ **Gateway**, **Router**.

I

IEEE
Institute of Electrical and Electronics Engineers

International body that defines standards in electronics and electro-technology, concerned in particular with the standardisation of LAN technology, transmission protocols, data transfer rate and wiring.

Infrastructure network

Network with central structure: all **Network subscriber** communicate via a central **Router**.

Instant messaging

Service that uses a client program to allow chatting in real time, i.e. to send brief messages to other subscribers on the Internet.

Internet
Global **WAN**. A series of protocols have been defined for exchanging data, known by the name TCP/IP.

All **Network subscriber** are identifiable via their **IP address**. **DNS** assigns a **Domain name** to the **IP address**.

Important services on the Internet include the World Wide Web (WWW), e-mail, file transfer and discussion forums.

Internet Service Provider

Enables access to the Internet for a fee.

IP (Internet Protocol)

TCP/IP protocol on the **Internet**. IP is responsible for addressing subscribers in a **Network** using **IP addresses** and transfers data from the sender to the recipient. IP determines the paths (routing) along which the data packets travel.

IP address

A unique address for a network component within a network based on the TCP/IP protocols (e.g. LAN, Internet). On the **Internet**, domain names are usually assigned instead of IP addresses. **DNS** assigns the corresponding IP address to the domain name.

The IP address has four parts (decimal numbers between 0 and 255) separated by full stops (e.g. 230.94.233.2).

The IP address is made up of the network number and the number of the **Network subscriber** (e.g. phone). Depending on the **Subnet mask**, the front one, two or three parts make up the network number and the rest of the IP address addresses the network component. The network number of all the components in any one network must be identical.

IP addresses can be assigned automatically with DHCP (dynamic IP addresses) or manually (static IP addresses).

→ **DHCP**.

IP pool range

Range of IP addresses that the DHCP server can use to assign dynamic IP addresses.

L

LAN

Local Area Network

Network with a restricted physical range. A LAN can be wireless (WLAN) and/or wired.

Local IP address

The local or private IP address is the address for a network component in the local network (LAN). The network operator can assign any address he or she wants. Devices that act as a link from a local network to the Internet (gateway or router) have a public and a private IP address.

→ **IP address**.

Local SIP Port

→ **SIP port/Local SIP port**.

M

MAC address

Media Access Control Address

Hardware address by means of which each network device (e.g. network card, switch, phone) can be uniquely identified worldwide. It consists of 6 parts (hexadecimal numbers) separated by a "-" (e.g. 00-90-65-44-00-3A).

The MAC address is assigned by the manufacturer and cannot be changed.

Mbps

Million bits per second

Unit of the transmission speed in a network.

MRU Maximum Receive Unit

Defines the maximum user data volume within a data packet.

MTU Maximum Transmission Unit

Defines the maximum length of a data packet that can be carried over the network at a time.

Music on hold

Music that is played while you are on a **Consultation call** or **Toggling**. The waiting participant hears music while on hold.

N

NAT Network Address Translation

Method for converting (private) **IP addresses** to one or more (public) IP addresses. NAT enables the IP addresses of **Network subscribers** (e.g. VoIP telephones) in a **LAN** to be concealed behind a shared IP address for the **Router** on the **Internet**.

VoIP telephones behind a NAT router cannot be reached by VoIP servers (on account of the private IP address). In order to "get around" NAT, it is possible to use (alternatively) **ALG** in the router, **STUN** in the VoIP telephone, or for the VoIP provider to use an **Outbound proxy**.

If an outbound proxy is made available you must allow for this in the VoIP settings for your phone.

Network Group of devices. Devices can be connected in either wired or wireless mode.

Networks can also differ in range and structure:

- Range: local networks (**LAN**) or wide-area networks (**WAN**)
- Structure: **Infrastructure network** or ad-hoc network

Network subscriber

Devices and PCs that are connected to each other in a network, e.g. servers, PCs and phones.

O

Outbound proxy

Alternative NAT control mechanism to STUN and ALG.

Outbound proxies are implemented by the VoIP provider in firewall/NAT environments as an alternative to **SIP proxy server**. They control data traffic through the firewall.

Outbound proxy and STUN servers should not be used simultaneously.

→ **STUN** and **NAT**.

P

Paging (handset search)

A base station function to locate registered handsets. The base station establishes a connection to every registered handset. The handsets start to ring. Paging is activated by briefly pressing the button on the base station and is deactivated by briefly pressing the same button again.

PIN
: Personal Identification Number

Protects against unauthorised use. When the PIN is activated, a number combination has to be entered in order to access a protected area.

You can protect your base station configuration data with a system PIN (4-digit number combination).

Port
: Data is exchanged between two applications in a **Network** via a port.

Port forwarding
: The Internet gateway (e.g. your router) forwards data packets from the **Internet** that are directed to a certain **Port** to the port concerned. This allows servers in the **LAN** to offer services on the Internet without you needing a public IP address.

Port number
: Indicates a specific application of a **Network subscriber**. Depending on the setting in the **LAN**, the port number is permanently assigned or else it is newly assigned with each access.

The combination of **IP address**/**Port** number uniquely identifies the recipient or sender of a data packet within a network.

Pre-dialling
: → **Block dialling**.

Private IP address
: → **Public IP address**.

Protocol
: Describes the agreements for communicating within a **Network**. It contains rules for opening, administering and closing a connection, about data formats, time frames and possible error handling.

Proxy/Proxy server
: Computer program that controls the exchange of data between **Client** and **Server** in computer networks. If the phone sends a query to the VoIP server, the proxy acts as a server towards the phone and as a client towards the server. A proxy is addressed via **Domain name**/**IP address** and **Port**.

Public IP address

The public IP address is the address for a network component on the Internet. It is assigned by the Internet Service Provider. Devices that act as a link from a local network to the Internet (gateway, router) have a public and a local IP address.

→ **IP address** and **NAT**

Q

Quality of Service (QoS)

Describes the Quality of Service in communication networks. Differentiations are made between various Quality of Service classes.

QoS influences the flow of data packets on the Internet, e.g. by prioritising data packets, reserving bandwidth and data packet optimisation.

In VoIP networks, QoS influences the voice quality. If the whole infrastructure (router, network server etc.) has QoS, the voice quality is better, i.e. fewer delays, less echoing, less crackling.

R

RAM

Random Access Memory

Memory in which you have reading and storage rights. Items such as melodies and screen pictures are saved in the RAM after you have loaded them onto the phone via the Web configurator.

Registrar

The registrar manages the **Network subscriber's** current IP addresses. When you register with your VoIP provider, your current IP address is saved on the registrar. This means you can also be reached when on the move.

Ringback when the call is not answered

= CCNR (Completion of Calls on No Reply). If a participant does not respond when called, a caller can arrange an automatic ringback. As soon as the destination phone has completed a call and is free again, the caller is rung back. This feature must be supported by the exchange. The ringback request is automatically cancelled after about 2 hours (depending on the VoIP provider).

Ringback when the number is busy

= CCBS (Completion of Calls to Busy Subscriber). If a caller hears the busy tone, he or she can activate the ringback function. As soon as the connection is free the caller is rung back. As soon as the caller lifts the receiver the connection is made automatically.

ROM

Read Only Memory

A type of memory that can only be read.

Router

Routes data packets within a network and between different networks via the quickest route. Can connect **Ethernet networks** and WLAN. Can be a **Gateway** to the Internet.

Routing
Routing is the transfer of data packets to another subscriber in your network. On their way to the recipient, the data packets are sent from one router to the next until they reach their destination.

If data packets were not forwarded in this way, a network like the Internet would not be possible. Routing connects the individual networks to this global system.

A router is a part of this system; it transfers data packets both within a local network and from one network to the next. Transfer of data from one network to another is performed on the basis of a common protocol.

RTP
Realtime Transport Protocol

Global standard for transferring audio and video data. Often used in conjunction with UDP. In this case, RTP packets are embedded in UDP packets.

RTP port
(Local) **Port** that is used to send and receive voice data packets for VoIP.

S

Server
Provides a service to other **Network subscriber** (**Client**s). The term can indicate a computer/PC or an application. A server is addressed via **IP address** / **Domain name**s and **Port**.

SIP (Session Initiation Protocol)
Signalling protocol independent of voice communication. Used for establishing and ending a call. It is also possible to define parameters for voice transmission.

SIP address
→ **URI**.

SIP port/Local SIP port
(Local) **Port** that is used to send and receive SIP signalling data for VoIP.

SIP provider
→ **VoIP provider**.

SIP proxy server
IP address of your VoIP provider's gateway server.

Static IP address
A static IP address is assigned to a network component manually during network configuration. Unlike a **Dynamic IP address**, a static IP address does not change.

STUN
Simple Transversal of UDP over NAT

NAT control mechanism.

STUN is a data protocol for VoIP telephones. STUN replaces the private IP address in the data packets of the VoIP telephone with the public address of the secure private network. To control data transfer, a STUN server is also required on the Internet. STUN cannot be implemented with symmetric NATs.

→ **ALG**, **Firewall**, **NAT**, **Outbound proxy**.

Subnet
Segment of a **Network**.

Subnet mask

IP addresses consist of a fixed network number and a variable subscriber number. The network number is identical for all **Network subscriber**. The size of the network number part is determined in the subnet mask. In the subnet mask 255.255.255.0, for example, the first three parts of the IP address are the network number and the last part the subscriber number.

Symmetric NAT

A symmetric NAT assigns different external IP addresses and port numbers to the same internal IP addresses and port numbers – depending on the external target address.

T

TCP

Transmission Control Protocol

Transport protocol. Session-based transmission protocol: it sets up, monitors and terminates a connection between sender and recipient for transporting data.

TLS

Transport Layer Security

Protocol for encrypting data transmissions on the Internet. TLS is a superordinated **Transport protocol**.

Toggling

Toggling allows you to switch between two callers or between a conference call and an individual caller without allowing the waiting caller to listen to the call.

Transmission rate

Speed at which data is transmitted in the **WAN** or **LAN**. The transmission rate is measured in data units per unit of time (Mbit/s).

Transport protocol

Controls data transport between two communication partners (applications).

→ **UDP**, **TCP**, **TLS**.

U

UDP

User Datagram Protocol

Transport protocol. Unlike **TCP**, **UDP** is a non session-based protocol. UDP does not establish a fixed connection. The data packets ("datagrams") are sent as a broadcast. The recipient is solely responsible for making sure the data is received. The sender is not notified about whether it is received or not.

URI

Uniform Resource Identifier

Character sequence for identifying resources (e.g. e-mail recipient, http://siemens.com, files).

On the **Internet**, URIs are used as a uniform identification for resources. URIs are also described as SIP addresses.

URIs can be entered in the phone as a number. By dialling a URI you can call an Internet subscriber with VoIP equipment.

URL
Universal Resource Locator

Globally unique address of a domain on the **Internet**.

A URL is a subtype of **URI**. URLs identify a resource by its location on the **Internet**. For historical reasons the term is often used as a synonym for URI.

User ID
→ **User identification**.

User identification
Name/number combination for access, e.g. to your VoIP account.

V

Voice codec
→ **Codec**.

VoIP
Voice over Internet Protocol

Telephone calls are no longer placed and transmitted over the telephone network but over the **Internet** (or other IP networks).

VoIP provider
A VoIP, SIP or **Gateway provider** is an Internet service provider that provides a **Gateway** for Internet telephony. As the phone works with the SIP standard, your provider must support the SIP standard.

The provider routes calls from VoIP to the telephone network (analogue, ISDN and mobile) and vice versa.

W

WAN
Wide Area Network

Wide-area network that is unrestricted in terms of area (e.g. **Internet**).

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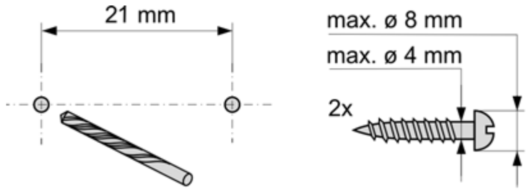
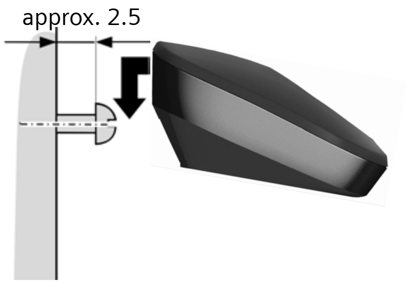
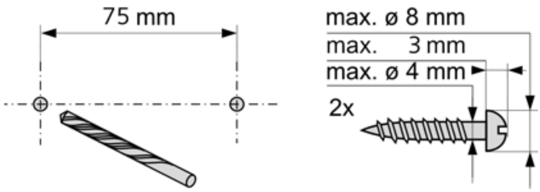
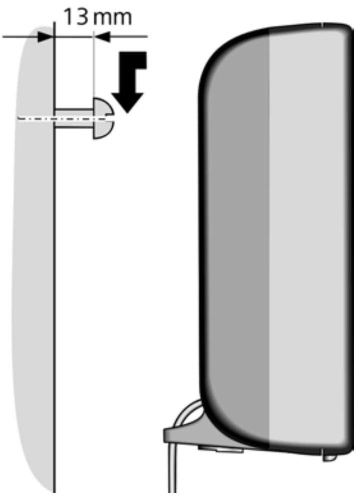
Mounting the charging cradle to the wall





Mounting the base station to the wall





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