SIEMENS

Gigaset C450 IP

Gigaset

Issued by
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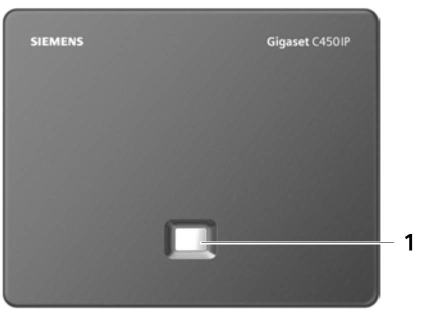
The handset at a glance

Base station at a glance



Handset keys

1 **Display** in idle status (example)
2 **Battery charge status**
(1/3 charged to fully charged)
flashes: battery nearly empty
flashes: battery charging
3 **Control key** (page 20)
4 **Display keys** (page 20)
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 **Message key**
Opens calls and message lists
Flashes: new message or new call
7 **Hash key**
Keypad lock on/off (press and hold, page 19)
Toggle between upper/lower case letters and digits for text entry (page 69)
8 **Alarm clock key** (page 39)
Activating/deactivating the alarm clock
9 **Microphone**
10 **Recall key** (not for VoIP connections)
Enter flash (press briefly)
Insert a pause (press and hold)
11 **Star key**
Ringtones on/off (press and hold in idle status)
12 **Connection socket for headset** (page 13)
13 **Handsfree key**
Switch between earpiece and handsfree mode
Lights up: handsfree talking activated
Flashes: incoming call
14 **Talk key**
Accept a 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)
15 **Signal strength**
(low to high)
flashes: no reception

Base station key

1 **Paging key**
Lights up: LAN connection active (phone is connected to router)
Flashes: Data transfer to LAN connection
Press **briefly**: start paging (page 35)
Press and **hold**: set base station to registration mode (page 34)

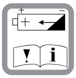
Contents

Safety precautions

Caution:
Read the safety precautions and the user guide before use.
Explain their contents to your children, and the potential hazards associated with using the telephone.

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

Fit only the **recommended rechargeable batteries (page 68)** of the same type! This means: do not use any other battery type or non-rechargeable batteries as this could result in significant health risks and personal injury.



Insert rechargeable batteries with the correct polarity, and use them according to this user guide (polarity symbols can be seen in the handset's battery compartment, page 6).

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 side of the handset to the ear when it rings or when you have activated on the handsfree function. 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 in bathrooms or shower rooms. The handset and base station are not splashproof (page 60).

Do not use your 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.

All electrical and electronic products should be disposed of separately from the municipal waste stream via designated collection facilities appointed by the government or the local authorities.
This crossed-out wheeled bin symbol on the product means the product is covered by the European Directive 2002/96/EC.
The correct disposal and separate collection of your old appliance will help prevent potential negative consequences for the environment and human health. It is a precondition for reuse and recycling of used electrical and electronic equipment.
For more detailed information about disposal of your old appliance, please contact your city office, waste disposal service or the shop where you purchased the product.

Note:
When the keypad lock is active you cannot even call emergency numbers!
Not all of the functions described in this user guide are available in all countries.

Gigaset C450 IP – more than just making calls

Your phone lets you make calls both via the fixed network and (cost effectively) via the Internet (VoIP) **without using a PC**.

And your phone can do much more besides:

- Press a button each time you make a call to indicate whether you want to make a call via the fixed network or via the Internet (page 17).
- Register up to **six** handsets on your base station. Your base station allows you to use one handset to make a call via the fixed network and another to make a call via the Internet at the same time.
- Configure the phone connection for VoIP without a PC. Your phone's connection wizard downloads general data about your VoIP provider from the Internet and guides you through entering your personal data (account). This makes it easy for you to start using VoIP (page 10).
- If necessary, establish any further VoIP settings on a PC. The phone has a Web interface (**Web configurator**) that can be accessed via your PC's Web browser (page 46).
- Assign your own password (system PIN) to protect your device and the Web configurator from unauthorised access (page 40).
- Send and receive SMS messages via the fixed network (page 28).
- Save 100 phone numbers on your handset (page 24).
- You can programme the keys of your phone with important phone numbers. The phone number is then dialled by simply pressing the respective key (page 25).
- Use the handsfree function to keep your hands free when making a call (page 18).
- Use your handset as an alarm clock (page 39).

Your Gigaset C450 IP has a protected operating system that offers increased security **against viruses** from the Internet.

Have fun using your new telephone!

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 with high voice quality with subscribers on the Internet, the fixed network or the mobile phone network.
- Your SIP provider will give you a personal number, with which you can be reached from the Internet, the fixed 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 or volume-based price
- Internet access, i.e. you need a router that will connect your phone to the Internet.
- Access to the services of a VoIP provider. Open an account with a VoIP provider.

First steps

Pack contents

The pack contains:
- one Gigaset C450 IP base station
- one Gigaset C45 handset
- one mains adapter for the base station
- one charging cradle incl. mains adapter
- one phone cord
- one Ethernet cable (LAN cable)
- two batteries
- one battery cover
- one belt clip
- one quick guide

Setting up the handset for use

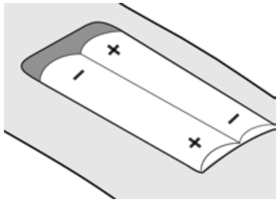
The display is protected with a plastic film. **Please remove the protective film!**

Inserting the batteries

Caution:
Use only the rechargeable batteries recommended by Siemens Home and Office Communication Devices GmbH & Co. KG on page 68! This means: on no account may conventional (non-rechargeable) batteries or other battery types be used, otherwise serious damage to health and property cannot be ruled out, e.g. the outer casing of the batteries could be destroyed or 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 (see figure).

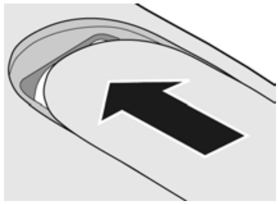
The polarity is indicated in the battery compartment.



The handset switches on automatically. You will hear a confirmation tone.

Closing the battery cover

▶ Place the cover on the battery compartment as shown in the diagram, then push it up until it clicks into position.



Opening the battery cover

▶ Press down on the battery cover below its upper end and slide the cover down.

Connecting the charging cradle

Connecting up the charging cradle and mounting it on the wall (if required) is described at the end of this user guide.

▶ To charge the batteries, leave the handset in the charging cradle.

Notes:
– Only place the handset in the charging cradle that is intended for it.
– If the handset has switched itself off because the batteries are flat and if it is then placed in the charging cradle, it will switch itself on automatically.

For questions and problems see page 61.

Initial charging and discharging of batteries

Battery charging is indicated in the top right of the display by a flashing battery icon , or . During handset operation, the battery icon indicates the charge status of the batteries (page 1).

The correct charge status can only be displayed when the batteries are first fully charged **and** discharged through use.

- To do this, leave the handset in the charging cradle without interruption until the battery icon stops flashing in the display (approx. 13 hours).
- Once the batteries are fully charged, remove the handset from the charging cradle and do not put it back again until the batteries are fully discharged.

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

Please note:

- Always repeat the charging and discharging procedure if you remove the batteries from the handset and insert them in again.
- The batteries may warm up as they are charging. This is not dangerous.
- After a while the charge capacity of the batteries will decrease for technical reasons.

Note:
You will find explanations for the symbols and typographical conventions used in this user guide in the appendix, page 60.

Setting the date and time

Menu → Settings → Date/Time

Enter the day, month and year with 6 digits and press OK. Use to move between the fields.

Enter hours and minutes with 4 digits (e.g. 0 7 1 5 for 07.15 hrs) and press OK. Use to move between the fields.

The date and time are shown in the handset's idle display page 1.

Registering the handset to the base station

Your handset is registered to the base station by default.

Instructions on how to register further handsets to the base station are given on page 34.

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.

- Place or hang the base station in a central position in your flat or house.

Please note:

- Never expose the telephone to heat sources, direct sunlight, 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 network and via VoIP, you must connect the base station to the fixed network and the Internet, see Figure 1.

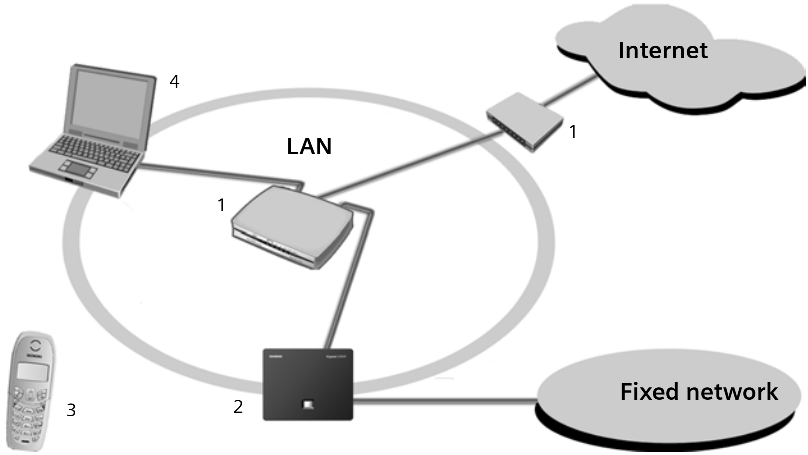


Figure 1 Connecting the phone to the fixed network and the Internet

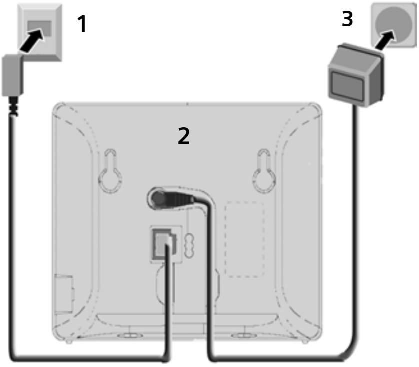
1 Internet connection:
Router with integrated modem or router and modem
2 Gigaset C450 IP base station
3 Gigaset C45 handset
4 PC in LAN

Follow the steps in the order given below:

1. Connect the base station with the phone connection
2. Connect the base station with the mains power supply
3. Connect the base station with the router

Connecting the base station with the fixed network and the mains power supply

▶ **Please first** connect the phone jack and **then** the mains adapter, as shown below.

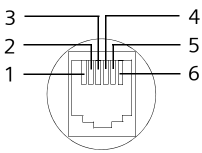


1 Phone jack with phone cord
2 Underside of the base station
3 Mains adapter 230 V

Please note:

- Keep the mains adapter **plugged in at all times** for operation, as the phone does not work without mains connection.
- If you buy a replacement phone cord from a retailer, ensure that the phone jack is connected correctly.

Correct phone jack assignment

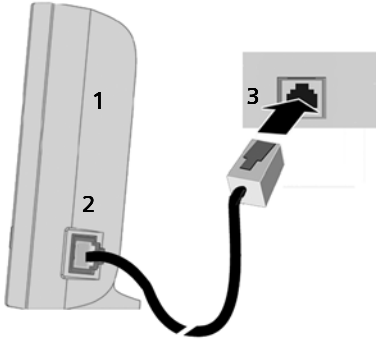


1 unused
2 unused
3 a
4 b
5 unused
6 unused

You can now use your phone to make calls via the fixed network and can be reached at your fixed network number!

Connecting the base station with the router

For Internet access you need a router, connected to the Internet via a modem (if necessary, this can be integrated in the router).



1 Side view of the base station
2 Network plug (LAN) with network cable
3 Router network plug

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

Making settings for VoIP telephony

Before you can use the Internet (VoIP) to phone any other users on the Internet, the fixed 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 a VoIP provider (e.g.via your PC) and set up an account. The provider must support the VoIP SIP standard.

The following phone settings are necessary in order for you to use VoIP. You will receive all information from your VoIP provider.

- Your user name with the VoIP provider, if this is required by the VoIP provider
- Your registration name
- Your password with the VoIP provider
- VoIP provider general settings

The connection wizard will help you with the settings.

Starting the connection wizard

Precondition: The base station is connected to the mains power supply and a router. Your router is connected to the Internet.

Tip: If VoIP (IP) is activated as the default connection for your phone (default setting see page 41), the phone will attempt to make a direct connection to the server belonging to your VoIP provider after the connection assistant is closed. If incorrect/ incomplete information means that the connection cannot be established, messages will be displayed (page 12).

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 is activated. Turn to page 43 to find out how to assign your phone a static IP address if necessary.

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

▶ Press the message key ✉.

You will see the following display:

Yes	Press the display key to start the connection assistant.
	Enter the system PIN of the base station (the default is 0000) and press OK.

Note:
The connection wizard 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 wizard at any time via the menu (page 42).

Downloading VoIP provider data

The phone establishes a connection with the Siemens server on the Internet. Profiles with general access data for various VoIP providers can be downloaded here. The message **Select country** is displayed.

After changing the display:

Select country and press **OK**.

Select provider is displayed.

After changing the display:

Select VoIP provider and press **OK**.

The necessary data for your VoIP provider is downloaded and saved on the phone.

Note:
If the data for your VoIP provider is not offered for download, you must make the necessary settings with the Web configurator at a later stage (page 51)
Press the display key **Back**. You can then conduct the following steps with the connection assistant (see "Entering VoIP user data").

Entering VoIP user data

Depending on the VoIP provider, the following information will be requested in sequence:

Username:
If this is required by your provider, enter name and press **OK**.

Authentication Name:
Enter name and press **OK**.

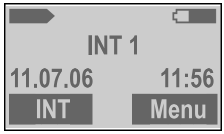
Authentication Password:
Enter password and press **OK**.

Note:
Please note when making these entries that the VoIP user data is case sensitive. If necessary, press and hold the # key to switch between upper and lower case and digits

Completing the VoIP settings

Once all the necessary entries have been made, the handset reverts to idle status.

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



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

Note:
To ensure that you can always be reached via the Internet, the router should be permanently connected to the Internet.

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, then either a fault has occurred or your information was incomplete:

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 modem/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.
  - Find the IP address using the handset menu: **Menu** → **Settings** → **Base** → **VoIP Configuration** → (enter system PIN) → **IP Configuration** → **IP Address**
  - Start the Web configurator with the IP address (page 46).
  - If no connection can be established: change the settings on the router (activate DHCP server) or the phone's (static) 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**. In particular, check your use of upper and lower case. To do this, open the following menu on your handset: **Menu** → **Settings** → **Base** → **VoIP Configuration**
- The server address for the VoIP server has not yet been entered, or has been entered incorrectly.
  - Start the Web configurator.
  - Open **Settings** → **Telephony** → **VoIP** Web page.
  - Edit the server address where necessary.

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), then it makes sense to switch off DHCP and assign the phone a static IP address (otherwise you may not be able to hear the other party during VoIP calls):
– Via the handset menu:
Menu → **Settings** → **Base** → **VoIP Configuration** → (enter system PIN) → **IP Configuration** → **IP Address**
Or
– Via the Web configurator:
- Open **Settings** → **IP configuration** Web page.
- 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 default gateway and DNS server via the Web configurator (page 50). The IP address for the router is generally entered here.

You will find other messages and possible measures in the Questions and answers section on page 61.

Belt clip and headset

By using a belt clip and headset (optional) you can easily make your handset a constant companion both inside the building and in its immediate vicinity.

Attaching the belt clip

There are notches for attaching the belt clip on the side of the handset at approximately the same height as the display.

- Press the belt clip onto the back of the handset so that the protrusions on the belt clip engage with the notches.

The tongue of the belt clip must face the battery compartment.

Connection socket for headset

You can use headsets with jack connectors. The following models have been tested and are therefore recommended: HAMA Plantronics M40, MX100 and MX150.

The transmission quality of other models cannot be guaranteed.

Menu trees

Phone menu

There are two ways to select a function:

Using number combinations ("shortcut")

- To open the main menu, press **Menu** with the handset in idle status.
- Enter the number combination that is in front of the function in the menu tree.
- **Example:** **Menu** 4 2 2 for "Set handset language".

Scrolling through the menus

- To open the main menu, press **Menu** with the handset in idle status.
- Scroll to the function with the control key and press **OK**.

1 SMS

1-1	Write Message					page 29
1-2	Incoming 00+00					
1-3	Outgoing					
1-6	Settings	1-6-1	Service Centres	1-6-1-1	Service Centre 1	page 31
				...	[to]	
				1-6-1-4	Service Centre 4	
		1-6-2	Status Report			
		1-6-3	Register to Service Centres			

2 Alarm Clock page 39

2-1	Activation	
2-2	Wake up time	

3 Audio Settings

3-1	Ringer Volume			page 38
3-2	Ringer Melody	3-2-1	External Calls	page 38
		3-2-2	Internal Calls	
		3-2-3	Alarm Clock	
3-3	Advisory Tones			page 39

3-4	Battery Low	3-4-1	Off			page 39
		3-4-2	On			
		3-4-3	During Call			

4 Settings

4-1	Date/Time					page 7
4-2	Handset	4-2-1	Display	4-2-1-1	Screensaver	page 37
				4-2-1-2	Colour Scheme	
				4-2-1-3	Contrast	
				4-2-1-4	Backlight	
		4-2-2	Language			page 37
		4-2-3	Auto Answer			page 37
		4-2-4	Register Handset			page 34
		4-2-5	Reset Handset			page 40
4-3	Base	4-3-1	Select Services Only displayed if **Default Line Type** = **fixed line** is set.	4-3-1-6	For All Calls	page 22
		4-3-2	System PIN			
		4-3-3	Base Reset			
		4-3-4	Additional Fea-tures	4-3-4-1	Dialling Mode	page 44
				4-3-4-2	Recall	page 44
				4-3-4-3	Repeater Mode	page 41
				4-3-4-5	Additional Emer-gency No.	page 41
		4-3-6	VoIP Configura-tion	4-3-6-1	Connection Assistant	page 42
				4-3-6-2	Select VoIP Pro-vider	
				4-3-6-3	Username	
				4-3-6-4	Authentication Name	
				4-3-6-5	Authentication Password	
				4-3-6-6	IP Configuration	

4-3-7	Default Line Type	4-3-7-1	IP	page 41
		4-3-7-2	fixed line	
4-3-8	Firmware Update			page 41

5	**Voice Mail**			page 33
5-1	Set Key 1	5-1-1	Network Mailbox	

Web configurator menu

Home			page 46
Settings			
	IP configuration		page 50
	Telephony		
		VoIP	page 51
		DTMF	page 56
		Dialing Plans	page 56
	Miscellaneous		page 57, page 58
Status			page 59

Making calls with VoIP and the fixed network

Making an external call

External calls are calls made via the public telephone network (fixed network) or via the Internet (VoIP). You can use the talk key when dialling to select what type of connection you want to use (fixed network or VoIP). One particular connection type is set as the default connection for your phone. The default is VoIP (for how to change the setting if required, see page 41).

- Enter the required number/IP address using the keypad.
- Press the talk key **briefly** if you want to make the call via the default connection.

Or:

- Press **and hold** the talk key if you want to make a phone call via the other connection type (the non-default connection).

Notes:
- If there are at least two handsets registered to your base station, you can use one to make a call via the fixed network and the other to make a call via the Internet (VoIP) at the same time.
- If you use a different GAP-compatible handset to the Gigaset C45, all calls will be made via the default 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.
- If you use VoIP to make a call to the fixed network, you may also have to dial the area code for local calls (depending on the VoIP provider). You can avoid this by entering the area code in the configuration of the base station (via the Web configurator, see page 56). It will then be inserted automatically for local calls.

Cancelling the dialling operation

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

Entering an IP address

If you are making a call via VoIP, you can enter an IP address instead of a phone number.

*	Press the star key to separate the parts of the IP address (e.g. 149*246*122*28).
#	If necessary, press the hash key to attach the SIP port number of the person you are calling (page 85) to the IP address (e.g. 149*246*122*28#5060).

Notes:
- Dialling with the directory (page 24) or last number redial list (page 26) saves repeated keying of phone numbers.
- You can assign a number from the directory to a key for speed dialling (page 25).
- You can edit or add to any phone number selected by means of quick dial or from the directory and use it for the current call.

Ending a call

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.

You can answer the call by:

- Pressing the talk key.
- Pressing the handsfree key.

If the handset is in the charging cradle and the **Auto Answer** function is activated (page 37), the handset will take a call automatically when you lift it out of the cradle.

If the ringtone is intrusive, press **Menu** **Silent**. You can accept the call so long as it is displayed on the screen.

Calling Line Identification

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

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

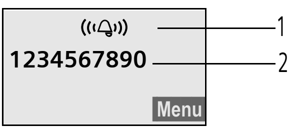
- Your fixed network provider supports CLIP, CLI:
  - CLI (Calling Line Identification): the caller's number is transmitted
  - CLIP (Calling Line Identification Presentation): the caller's number is displayed
- You have arranged CLIP with your network provider.
- The caller has arranged CLI with the network provider.

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

Call display

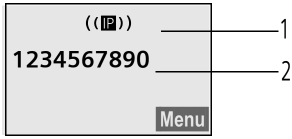
By means of the ringtone icon on the screen, you can decide whether the call is directed to your fixed network number or to your VoIP number.

Calls to your fixed network number



1 Ringtone icon
2 Number or name of caller

Calls to your VoIP number



1 Ringtone icon
2 Number or name of caller

Display when Calling Line Identification is withheld

For calls from the fixed network, the caller can withhold calling line identification or not request it. In this case the number is not displayed. The following is displayed in place of the number:

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

Handsfree talking

In handsfree mode, instead of holding the handset to your ear you can put it down, e.g. on the table in front of you, to allow others to participate in the call.

Activating/deactivating handsfree mode

Activating while dialling

Enter number and press briefly/press and hold the handsfree key to select the connection type (page 17).

▶ 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.

Switch handsfree on and off during a call.

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

- Press and hold the handsfree key while placing the handset in the charging cradle. If the handsfree key does not light up, press the key again.

For how to adjust the loudspeaker volume, see page 37.

Muting the handset

You can deactivate the microphone in your handset during an external call. The other party hears a wait melody.

Muting the handset

INT Press the display key.

Cancelling muting

Back Press the display key.

Dialling the emergency number

The default setting for your phone is that all numbers that are saved as emergency numbers are automatically dialled via the fixed network, irrespective of whether you press the talk key briefly or press and hold it.

You can deactivate this function via the Web configurator (**Dialing Plans**, page 57), e.g. if you use the phone without a fixed network. Ask beforehand, however, whether your VoIP provider supports emergency numbers.

Enter the emergency number and press the talk key.

Three emergency numbers have already been entered in your phone. You can set one more emergency number (page 41).

Notes:

- You can use the Web configurator to display which emergency numbers are saved on your phone (page 56).
- **Please note:** If you have used the Web configurator to deactivate the **Emergency calls always via fixed line** function and have entered an automatic area code for VoIP calls (page 56), the area code will also be prefixed to the emergency numbers when they are dialled via VoIP.

Operating the handset

Switching the handset on/off

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

You will hear the confirmation tone.

Activating/deactivating the keypad lock

\# Press and **hold** the hash key.

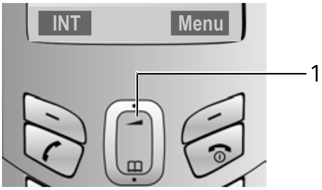
You will hear the confirmation tone. The icon appears in the display when the keypad lock is activated.

The keypad lock deactivates automatically when you receive a call and activates again after the call.

Note:

The handset displays an advisory message if you press a key by accident while the keypad lock is on. To deactivate the keypad lock, press and **hold** the hash key #.

Control key



1 Control key

In this user guide, the side of the control key that you must press in the given operating situation is shown in black (top, bottom). Example: for "press the top of the control key".

The control key has a number of different functions:

When the handset is in idle status

	Open the directory.
	Adjust the ringtone volume of the handset (page 38).

In lists and menus

/	Scroll up/down line by line.

In an input field

/	Move the cursor **left** or **right**.

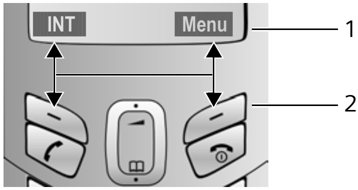
During an external call

	Open the directory.
	Adjust loudspeaker volume for earpiece and handsfree mode.

Display keys

The current display functions are shown in the bottom display line in reversed highlights. The function of the display keys changes depending on the particular operating situation.

Example:



1 Current display key functions
2 Display keys

The most important display symbols are:

Back	Go back one menu level or cancel the operation.
INT	Make an internal call (page 35).
Menu	Open the main menu or a context-dependent menu.
OK	Confirm highlighted selection.
◂C	Delete key: deletes one character at a time from right to left.

Returning to idle status

You wish to return to idle status from anywhere in the menu:

- Press the end call key and hold.

Or:

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

Changes that you have not confirmed/ saved by pressing OK will be rejected.

For an example of the display in idle status, page 1.

Menu guidance

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

Main menu (first menu level)

▶ To open the main menu, press **Menu** with the handset in idle status.

Accessing a function

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

Or:

▶ Enter the number that is in front of the function in the menu tree (page 14).

The corresponding submenu (the next menu level) is opened.

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 number combination that is in front of the function in the menu tree (page 14).

A short press on the end call key returns you to the previous menu level / cancels the operation.

Correcting incorrect entries

- Navigate to the incorrect input with the control key if is displayed.
- Press **C** to delete the character to the left of the cursor.
- Insert new character to the left of the cursor.
- When entering the time and date etc., edit the flashing character.

You will find explanations for the symbols and typographical conventions used in this user guide in the appendix, page 60.

Network services

The following network services can currently only be used for making calls via the fixed network.

Note:
The Settings ➝ Base ➝ Select Services menu is only displayed if you have set the fixed network as your default connection (page 41).

Network services are functions that your network provider makes available to you. You have to request these services from your network provider.

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

Making 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.

Setting up call diversion

Menu ➝ Settings ➝ Base ➝ Select Services ➝ For All Calls ➝ Call Divert

Setting up call forwarding

All Calls / No Answer / When Busy
Select and press OK.

On — Select and press OK.

Enter number and press OK.

After confirmation from the fixed network:

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

Deactivating call forwarding

All Calls / No Answer / When Busy

Off — Select and press OK.

After confirmation from the fixed network:

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

Call forwarding is deactivated.

Activating/deactivating call waiting

When call waiting is activated, the caller will hear the ringing tone if you are already making a call. This call is announced acoustically and visually on your handset screen.

Accepting/rejecting call waiting, see page 23.

Menu ➝ Settings ➝ Base ➝ Select Services ➝ For All Calls ➝ Call Waiting

On / Off — Select and press OK.

After confirmation from the fixed network:

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

Functions during a call

Initiating ringback

You hear the busy tone.

Menu ➝ Ringback

Press the end call key.

Consultation

During a call:

Menu ➝ External Call

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

The number is dialled via the fixed network.

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

You have the following options:

- Toggling:
  - ▶ Use [key] to toggle between the participants.
  - – End call with active participant: **Menu** End Active Call.
- 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 [end call key].

Accepting a waiting call

Precondition: Call waiting is activated (page 22).

Menu → Accept Call Waiting

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

Note:
- Without CLIP a waiting call is only announced with a beep.
- If the first call was an internal call, the internal connection is ended.
- An internal call waiting is shown on the display. You can neither accept the internal call nor reject it.

Rejecting a waiting call

Menu → Reject Call Waiting

Select and press **OK**.

Favouring a waiting call

Menu → Favour Call Waiting

Select and press **OK**.

Using the directory and lists

The options are:

- Directory
- Last number redial list
- SMS list
- Calls list

You can save 100 entries in the directory.

You can create a personalised directory for your own individual handset. However, you can send the list or individual entries to other handsets (page 25).

Directory

In the **directory** you store numbers and matching names.

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

Length of an entry

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

Notes:
- You may also have to enter the phone number with the area code for local calls when making VoIP calls to the fixed network (depending on the VoIP provider). You should therefore always save phone numbers in the directory with the area code. Alternatively, you can also use the Web configurator to define an area code, which is automatically prefixed to all numbers that are dialled without an area code for calls via VoIP (see Defining dialling plans, page 56).
- You can assign a number from the directory to a key for quick dial (page 25).

Saving the first number in the directory

→ **Directory empty New Entry?**

Enter number and press **OK**.

Enter the name and press **OK**.

Notes:
- To find out how to enter IP addresses, turn to page 17.
- If you enter an asterisk (*) at the end of the number, the number is dialled via the non-default connection (page 41), even if you briefly press the talk key or store this number on a key for quick dial.

Saving a number in the directory

→ **Menu** → **New Entry**

Enter number and press **OK**.

Enter the name and press **OK**.

Selecting a directory entry

Open the directory.

You have the following options:

- Use to scroll to the entry until the required name is selected.
- Enter the first character of the name, or scroll to the entry with .

Dialling with the directory

→ (select entry; page 24)

Briefly press/press and hold the talk key. The number is dialled using the selected connection type (page 17).

Note:
You can only dial IP addresses via VoIP.

Managing directory entries

You have selected an entry (page 24).

Editing entries

Menu → Edit Entry

Edit the number if required, and press OK.

Edit the name if required, and press OK.

Assigning a key

You can assign keys **0** and **2** to **9** with a number. The number is then dialled by simply pressing a key.

Menu → Shortcut

Assign to the current entry for quick dial to a selected key.

Using other functions

→ (select entry; page 24) → Menu

The following functions can be selected with :

Use Number
Edit or add to a saved number. Then dial or use other functions with Menu.

Delete Entry
Delete selected entry.

Send Entry
Send a single entry to another handset (page 25).

Delete List
Delete **all** directory entries.

Send List
Send the complete list to another handset (page 25).

Using quick dial keys

▸ Press and **hold** the required quick dial key (page 25).

If an asterisk (*) is placed at the end of the relevant phone number, the number is dialled via the non-default connection; otherwise it is always dialled via the default connection set (page 41).

Sending the directory to another handset

Requirements:

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

→ (Select entry; page 24) → Menu
→ Send Entry / Send List

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

A successful transfer is confirmed by a message and confirmation tone on the receiving handset.

If you have sent a single entry, you can transfer another entry with OK.

Please note:

- Entries with identical numbers are not overwritten in the receiver 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 that are displayed in a list, e.g. the calls list or last number redial list, to the directory.

A number is displayed.

Menu → Copy to Directory

▸ Complete the entry (page 24).

Copying a number from the directory

You can open the directory in many operating situations e.g. to copy a number. Your handset need not be in idle status.

Open the directory.

Select an entry (page 24).

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, then the corresponding name will be displayed.

Dialling from the last number redial list

c	Press the key **briefly**.
s	Select an entry.
c	Briefly press/press and hold the talk key. The number is dialled using the selected connection type (page 17).

Managing entries in the last number redial list

c	Press the key **briefly**.
s	Select an entry.
Menu	Press the display key.

The following functions can be selected with s:

Use Number
(as in the directory, page 25)

Copy to Directory
An entry is transferred to the directory (page 25).

Delete Entry
(as in the directory, page 25)

Delete List
(as in the directory, page 25)

Opening lists with the message key

You can use the message key f to open the following lists:

- SMS list
- Network mailbox
  If your network provider supports this function and the network mailbox is configured for fast access (page 33).
- Calls list

An advisory tone sounds as soon as a **new message** arrives in a list. The message key f flashes. A message appears in the display.

When you press the flashing key f, you will see all the lists that contain new messages. If only one list contains new messages, this will be opened automatically.

Note:
If calls are saved in the network answering machine you will receive a message if the appropriate settings have been made (see the network mailbox instructions of your network provider).

Calls list

Precondition: CLIP (page 18)

The numbers of the last 30 outgoing calls are saved. Multiple calls from the same number are only saved once (the last call).

The calls list is displayed as follows:

Calls List: 01+02

Number of new entries + number of old, read entries

Opening the calls list

f → Calls List: 01+02

s Select entry.

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

List entry

Example of a list entry:

New Call	01/02
1234567890	
11.03.06	19:27
Back	Menu

- Status of entry

  In the calls list

  New Call: new missed call

  Old Call: entry already read
- Entry number

  01/02 means e.g.: first of a total of two entries.
- Number or name of caller

  You can add the number of the caller to the directory (page 25).
- Call date and time (if set, page 7).

Selecting from the calls list

→ Calls List: 01+02

Select entry.

Briefly press/press and hold the talk key. The number is dialled using the selected connection type (page 17).

Managing entries in the calls list

→ Calls List: 01+02

Select entry.

Menu Press the display key.

The following functions can be selected with :

Copy to Directory
Accept number in the directory (page 25).

Delete Entry (as in the directory, page 25)

Delete List (as in the directory, page 25)

Making cost-effective calls

Using the Internet (VoIP) is the preferred cost-effective way of making calls. If you make calls via the fixed network, select a network provider who offers very low call charges (call-by-call) or have the call duration displayed on your handset after the call.

Displaying the call duration

The duration of a call is displayed

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

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

SMS (text messages)

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

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

Your unit is supplied ready for you to send SMS messages immediately.

Requirements:

- Calling Line Identification (CLIP, page 18) is enabled for your phone connection.
- Your network provider supports SMS in the fixed network (information on this can be obtained from your network provider).
- You are registered with your service provider to send and receive SMS.

SMS messages are exchanged between SMS centres that are operated by service providers. You must enter the SMS centre in the phone through which you wish to send and receive. You can receive SMS from **every** SMS centre that is entered provided you have registered with your service provider. The registration wizard (page 28) will help you to register.

Your text message is sent through the SMS centre that is active. However, you can activate any other SMS centre to send a current message (page 31).

If no SMS centre has been entered, an error message will be displayed as soon as you try to send an SMS. Enter an SMS centre (page 31).

Please note:

- If your phone is connected to a PABX, please read page 31.
- You must be registered with your service provider to receive SMS messages.
- Each incoming SMS is signalled by a single ring (ringtone as for external calls). If you accept such an SMS call on the first ring, the SMS will be lost. To prevent this, suppress the first ringtone for all external calls (page 38).

Registering for SMS using the registration wizard

You can use the registration wizard to register with all service providers whose number you have entered to send and receive SMS messages.

Precondition:

- You must have saved a number for at least one SMS centre.

When you call up the SMS menu for the first time, the wizard automatically registers you with accessible SMS centres whose number you have entered. You can also use the registration wizard to register with SMS centres at a later time.

Menu → SMS (on 1st call)

Menu → SMS → Register to Service Centres (later)

Yes — Press the soft key to confirm the prompt.

You can now receive SMS messages from any of the SMS centres whose number you have entered (page 31).

Writing/sending an SMS

An SMS may contain up to 160 characters.

Writing/sending SMS

Menu → **SMS** → **Write Message**

Write an SMS. For how to enter the text, see page 69.

Menu **Send Text**

Select and press **OK**.

/ Enter the number with dialling code (including your local area code) from the directory or key it in manually, and press **OK**. For SMS to an SMS mailbox: put the mailbox ID at the **end** of the number.
The SMS is sent.

Note:
If you are interrupted by an external call while writing an SMS, the text is automatically saved in the draft message list.

SMS status report

Precondition: Your network provider supports this feature.

If you have activated the status report, you will receive an SMS with a confirmation message after sending.

Activating/deactivating a status report

Menu → **SMS** → **Settings**

Status Report

Select and press **OK** (✓ = on).

Reading/deleting a status report

- Open the incoming message list (page 30) and then:

Select SMS with the **State OK** or **State NOK** status.

Menu **Read SMS**

Select and press **OK** to read the status report. Scroll using .

Or:

Menu **Delete Entry**

Select and press **OK** to delete the status report.

Draft message list

In the draft message list, you can save, edit later and send an SMS.

Saving an SMS in the draft message list

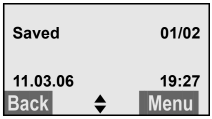
You write an SMS (page 29).

Menu → **Save Text**

Opening the draft message list

Menu → **SMS** → **Outgoing**

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



01/02: Current number/total number of SMS messages

Reading or deleting individual SMS messages

- Open the draft message list.

Select SMS.

Menu **Read SMS**

Select and press **OK** to read the message. Scroll in the SMS using .

Or:

Menu **Delete Entry**

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

Writing an SMS

You are reading an SMS in the draft message list.

Menu Press the soft key.

Write Message

Write and then send a new SMS (page 29) or save.

Deleting draft message list

- Open the draft message list.

Menu Delete List
Select and press **OK**.

OK Press the soft 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. Linked SMS messages are separated into individual SMS messages with max. 153 characters and saved as such 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**.

The display tells you if the SMS memory is full.

- Deleting SMS messages you no longer require (page 30).

Incoming message list

The incoming message list contains:

- All received SMS messages, starting with the latest.
- Messages that could not be sent on account of an error.

New SMS messages are signalled on all Gigaset C45 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):

SMS:
01+05

01+05: The number of new + the number of old, read messages

An entry in the list is displayed e.g. as follows:

New 01/02
1234567890
11.03.06 19:27
Back ⬍ Menu

01/02: Current number of the SMS currently in the display / total number of new SMS messages

Opening the incoming message list via the SMS menu

Menu → SMS → Incoming 01+05

Reading or deleting individual SMS messages

- Open the incoming message list.
- Continue as for reading/deleting individual SMS from the draft message list, page 29.

A new message which you have read acquires the status Old.

Deleting incoming message list

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

- Open the incoming message list.

Menu Press the soft key.

- Continue as described at "Deleting draft message list", page 30.

Replying to or forwarding SMS messages

You are reading an SMS (page 30).

Menu Press the soft key.

You have the following options:

Reply
Write and send a reply SMS directly (page 29).

Send Text
Forward the text of an SMS to another recipient (page 29).

Adding a number to the directory

Adding the sender's number

You are reading an SMS in the incoming message list.

Menu Press the soft key.

For further information, see page 25.

Note:
You can create a special directory for SMS messages within your main directory by putting a star (*) before the names.
An appended mailbox ID will be copied to the directory.

Setting SMS centre

You can set up a maximum of four SMS centres.

Entering/changing SMS centres

- 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 SMS centre (e.g. Service Centre 1) and press OK.

You have the following options:

Active Send Serv. Centre
If the SMS messages are to be sent through this SMS centre, press OK to activate the SMS centre (✓ = on). If a different SMS centre was active previously, then this will be deactivated. With SMS centres 2, 3 and 4, the setting only applies to the next SMS.

SMS
Enter the number of the SMS centre and press OK.

Sending an SMS through another SMS centre

- Activate the SMS centre (2 or 3) as the active send service centre (page 31).
- Send the SMS.

This setting only applies to the next SMS to be sent. After that, the setting returns to Service Centre 1.

SMS on a PABX

- You can only receive an SMS when the **Calling Line Identification** (page 18) **is forwarded** to the extension of the PABX **(CLIP)**. The CLIP of the phone number for the SMS centre is evaluated in your **Gigaset**.
- Depending on your PABX, you may have to add the access code (external line prefix) before the number of the SMS centre.

  If in doubt, test your PABX e.g. by sending an SMS to your own phone 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 **on ISDN PABXs** is only possible via the MSN number assigned to your base station.

Activating/deactivating the SMS function

When you switch off you cannot send or receive any SMS messages with your phone.

Settings which you have made for sending and receiving SMS messages (the numbers of SMS centres) and any entries in the incoming message and draft message lists will be retained even after the function is deactivated.

Menu 4 3 9 2 6

0 OK	Deactivate the SMS function.

Or:

1 OK	Activate the SMS function (default settings).

SMS troubleshooting

Error codes when sending

If an SMS cannot be sent for a longer period of time, it is moved to the incoming message list and given the status Error XX.

E0	Calling Line Identification permanently withheld (CLIR) or Calling Line Identification not activated.
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. You have not requested the CLIP service (Calling Line Identification Presentation, page 18). ▸ Ask your service provider to enable this service. 2. SMS transmission has been interrupted (e.g. by a call). ▸ Re-send the SMS. 3. Network provider does not support this feature. 4. No phone number or an invalid phone number is entered for the SMS centre activated as the active send service centre. ▸ Enter phone number (page 31).
You receive an incomplete SMS. 1. Your phone's memory is full. ▸ Delete old SMS messages (page 30). 2. The service provider has not yet sent the rest of the SMS.
The message is played back. 1. The "display call number" service is not activated. ▸ Ask your service provider to activate this function (chargeable). 2. Mobile phone operator and network SMS service provider have not agreed on a cooperation. ▸ Obtain information from your network SMS service provider. 3. Your terminal is recorded by your SMS provider as having no fixed network SMS functionality, i.e. you are no longer registered with the provider. ▸ Register your terminal (again) for SMS reception (page 28).
SMS messages are only received as voice messages 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. ▸ Register your terminal (again) for SMS reception (page 28).
You cannot access the SMS functions on your handset. Another handset that is registered at the same base station is currently using the SMS functions. ▸ Wait until there is no other handset using the SMS functions.

Using the network mailbox

Some fixed network providers and VoIP providers offer answering machines on the network – network mailboxes.

You can use the relevant network mailbox if you have **requested** it from your fixed network or VoIP provider.

The network mailbox only answers incoming calls made via the relevant line (fixed network or VoIP). To record all calls, you should therefore set up network mailboxes for both fixed network and VoIP.

> **Note:**
> You can only set up fast access to one of the network mailboxes.
> You can assign the number for the second network mailbox a quick dial digit in the directory (e.g. the **2** key) (page 25). The quick dial digit must be assigned for each handset.

Tip: A fixed network answer machine should always be controlled via the fixed network connection. If VoIP is set as the default connection on your phone, add an asterisk (*) to the end of the number of the network answer machine. The connection is then established via the fixed network.

Configuring the network mailbox for fast access

With fast access you can dial a network mailbox directly.

The network mailbox is preconfigured for fast access. You only need to enter the number of a network mailbox.

Configuring the network mailbox for fast access and entering the network mailbox number

Menu → Voice Mail → Set Key 1

Network Mailbox
: Select and press OK (✓ = fast access activated).

Enter the network mailbox number and press OK. The entry is saved.

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

Fast access is automatically activated.

To deactivate fast access, you must delete the number.

This setting is now valid for all registered handsets.

> **Note:**
> If no number has been entered: press and **hold** **1** to enter the number.

Calling the network mailbox

1 Press and **hold**. You are connected straight to the network mailbox.

If necessary, press the handsfree key. You hear the network mailbox announcement.

The number is dialled via the default connection.

> **Note:**
> If you have set an automatic area code (page 56), the area code is also prefixed to the number of the network answer machine if it does not start with 0 and is dialled via VoIP.

Viewing the network mailbox message

If a message arrives for you, you receive a call from the network mailbox. If you have requested Calling Line Identification, the display shows the network mailbox number. If you accept the call, the new messages are played back. If you do not accept the call, the network mailbox number will be saved in the missed calls list and the message key flashes (page 26).

Using several handsets

Registering handsets

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

Notes:
- If there are several handsets registered to your base station, you can use one to make a call via the fixed network and the other to make a call via the Internet at the same time.
- As a rule, all calls from a registered GAP handset are dialled via the connection type (fixed network or VoIP, see page 41) that has been set up as **Default Line Type**.
  If you want to establish a connection via the other connection type, enter a "*" (star) after the phone number. **Example:** 0498912345671234567*.

Registering another Gigaset C45 handset

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

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

If the handset has been registered successfully you will see the display key **INT** at the bottom left of the display. Otherwise repeat the procedure.

On the handset

Menu → **Settings** → **Handset** → **Register Handset**

Enter the system PIN of the base station (the default is 0000) and press **OK**. The display shows e.g. **Registering** and **Base** is flashing.

On the base station

Within 60 secs. press and **hold** the registration/paging key on the base station (page 1) (min. 1 sec.).

The handset is assigned the lowest unassigned internal number (1–6). If several handsets are registered to the base station, the internal number is shown in the display after registration, e.g. **INT 2**. This means that the handset has been assigned the internal number 2.

Notes:
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

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

On the handset

- Start to register the handset as described in its user guide.

On the base station

Press and hold the registration/paging key on the base station (page 1) (min. 1 sec.).

De-registering handsets

You can de-register any registered C45 handset from any registered handset.

INT Press the display key. All registered handsets are displayed.

Select the handset to be de-registered.

Menu Press the display key.

De-register Handset

	Select and press **OK**.
	Enter the base station system PIN (default setting: 0000).
OK	Press the display key to confirm the prompt.
	Press and **hold** (idle status).

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

Changing a handset's internal number

A handset is automatically assigned the lowest available number on registration. In the list of internal subscribers, the handset is sorted according to its internal number.

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

INT	Press the display key.
Menu	Press the display key.

Edit Handset Number

	Select and press **OK**.
	Select handset.
	Enter number (1–6).
OK	Press the display key to complete the operation.

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 at registration. You can change these names. The changed name is displayed in every handset's list.

INT	Press the display key.
	Select handset.
Menu	Press the display key.

Change Handset Name

	Select and press **OK**.
	Enter name (max. 10 characters) and press **OK**.

Locating a handset ("Paging")

You can locate your handset using the base station.

- Press the registration/paging key on the base station (page 1) **briefly**.
- All handsets will ring at the same time ("paging"), even if the ringtones are switched off.

Ending paging

/	**Briefly** press the registration/paging key on the base station (page 1) or press the talk key on the handset.

Making internal calls

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

Calling a specific handset

INT	Press the display key.
	Select handset and press the talk key.

Or:

	Enter the number of the handset.

Calling all handsets ("group call")

INT	Press the display key.
*	Press the star key.

Or:

Call All	Select and press the talk key.

Ending a call

	Press the end call key.

Note:
You can reject an internal call by pressing the end call key.

Transferring a call to another handset

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

INT Press the display key.
The external participant hears the hold music.

Select handset or Call All and press OK.

When an internal participant answers:

- If necessary announce the external call.

Press the end call key.

The call is transferred. If the internal participant does not answer, or their phone is in use, the call will automatically return to you.

Internal consultation calls

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

INT Press the display key.
The external participant hears the hold music.

Select handset or Call All and press OK.

When an internal participant answers you can speak to them.

Ending a consultation call

Menu Press the display key.

Back Select and press OK.

You are reconnected with the external participant.

Initiating a conference call

You are in an internal consultation call:

Menu Press the display key.

Conference Call
Select and press OK.

The internal subscriber called can end the conference call by pressing the end call key.

Accept call waiting during an internal call

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

Press the end call key to end the internal call.

Press the talk key to take the external call.

Handset settings

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

Change display language

You can view the display texts in different languages.

Menu → Settings → Handset → Language

The current language is indicated by ✓ .

Select a language and press OK.

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

If you accidentally choose a language you do not understand:

Menu 4 2 2

Press keys one after the other.

Select the correct language and press OK.

Setting the display

You have a choice of four colour schemes and several different contrasts. You can also set a screensaver and backlight.

Menu → Settings → Handset → Display

You have the following options:

Screensaver
There are four different screensavers and the settings No Screensaver or Digital Clock.

Colour Scheme
Four colour schemes. When the backlight is switched off, the display is shown in black and white regardless of the selected setting.

Contrast
You have a choice of several different contrasts.

Backlight
In Charger / Without Charger. Determines whether the backlight stays on permanently or is switched off after a certain time (✓ = permanently switched on).

Note:
If the backlight is switched on outside the charging cradle, the standby time for the handset is considerably reduced!

Activating/deactivating auto-answer

When this function is activated, when a call arrives 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).

Adjusting the loudspeaker volume

You can set the loudspeaker volume for handsfree talking to five different levels and the earpiece volume to three different levels. You can only adjust the loudspeaker volume during a call.

You are conducting an external call.

Press the control key.

Adjust the volume and press OK.

Note:
The handsfree volume can only be adjusted when this function is set.

If is assigned a different function e.g. toggling (page 22):

Menu Open menu.

Volume Select and press OK.

Make settings (see above).

Changing ringtones

- Volume:
  Five volume levels (1–5; e.g. Volume 2 =) and "crescendo" ring . With "crescendo" ring, the volume gets louder with every ring.
- Melody:
  List of pre-loaded ringtone melodies. The first three melodies are the "classical" ringtones.

You can also set different melodies for the following functions:

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

Setting the ringtone volume

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

Menu → Audio Settings → Ringer Volume

Or in idle status:

	Press **briefly**.

Then:

	Adjust the volume and press OK.
	Press and **hold** (idle status).

Setting ringtone melody

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

Menu → Audio Settings → Ringer Melody

External Calls / Internal Calls / Alarm Clock	Select and press OK.
	Select melody (✓ = on) and press OK.
	Press and **hold** (idle status).

Activating/deactivating the ringtone

You can turn off the ringtone on your handset before you take a call or while the phone is in idle status.You can take a call so long as it is displayed on the screen.

Deactivating the ringtone

*	Press the star key **and hold**, until the icon appears in the display.

Re-activating the ringtone

*	Press and **hold** the star key in idle status.

Activating/deactivating muting of the first ringtone

The phone identifies an incoming SMS from the first network signal.

Menu 4 3 9 1 9

1 OK	First ringtone muted.

Or:

0 OK	First ringtone audible.

Please note:

If first ringtone muting is deactivated, then every incoming SMS will be signalled by a ringtone. If you take this type of "call" at the first ringtone, you will lose the SMS.

Advisory tones

Your handset uses 'advisory tones' to tell you about different activities and statuses. You can activate or deactivate the following tones:

- **Advisory tones**:
  - **Key click**: every key press is confirmed.
  - **Confirmation tone** (rising tone sequence): at end of entry/setting, when replacing handset in the charging cradle and when an SMS is received or a new entry is made in the calls list.
  - **Error tone** (descending tone sequence): when you make an incorrect entry.
  - **Menu end tone**: when scrolling at the end of a menu.
- **Battery low beep**: the battery requires charging.

You cannot deactivate the confirmation tone for placing the handset in the charging cradle.

Activating/deactivating advisory tones

Menu → Audio Settings → Advisory Tones

Select and press OK (✓ = on).

All advisory tones are activated or deactivated.

Setting the battery low beep

Menu → Audio Settings → Battery Low

On / Off / During Call

Select and press OK (✓ = on). The battery low beep is activated or deactivated or sounds during a call.

Using the handset as an alarm clock

Activating/deactivating the alarm clock

Menu → Alarm Clock → Activation (✓ = on)

Or:

⏲ Press the alarm clock key.

After you activate the alarm clock, the menu for setting the wake up time opens automatically (page 39).

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

Setting the wake up time.

Menu → Alarm Clock → Wake up time

Enter the wake up time in hours and minutes, then press OK.

When the alarm clock rings...

Alarm repeat after 5 minutes

Snooze Press the display key or any key.

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

Switching off the alarm clock for 24 hours

Off Press the display key.

Restoring the handset default settings

You can reset any individual settings and changes that you have made. 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 to confirm.

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

Cancel the reset with .

Base station settings

The base station settings are carried out using a registered Gigaset C45 handset.

Changing the system PIN

You have to enter the system PIN when registering a handset to the base station.

You can change the base station's 4-digit default system PIN ("0000") to a 4-digit PIN known only to yourself.

Menu → Settings → Base → System PIN

Enter 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 numbers entered.

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

Restoring the base station to factory settings

Resetting the base station via the menu

The individual settings are reset. Only the date and time and the system PIN are retained. The handsets are still registered.

Menu → Settings → Base → Base Reset

OK Press the display key to confirm.

Resetting the base station using a key on the base station

All individual settings and the system PIN are reset. The system PIN is "0000" again. **All handsets registered above and beyond the delivery scope are deregistered.**

- Remove the cable connections from the base station to the router and fixed network.
- Remove the base station mains unit from the socket.
- Press and hold the registration/paging key (page 1).
- Plug the mains unit back into the power socket.
- Press and hold the registration/paging key (at least 2 sec.).
- Release the registration/paging key. The base station has now been reset.

Activating/deactivating repeater mode

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

Precondition: a repeater is registered.

Menu → Settings → Base → Additional Features → Repeater Mode

Select and press OK (✓ = on).

Setting up an emergency number

Emergency numbers have been preset in your phone. They cannot be amended. In addition, you can specify you own emergency number.

Note:
In default setting for your phone is that emergency numbers are automatically dialled via the fixed network. You can change this setting (page 56).

Menu → Settings → Base → Additional Features → Additional Emergency No.

Enter the system PIN and press OK.

If an additional emergency number has been saved, it is displayed.

Enter emergency number and press OK.

Set default connection

You can make settings according to whether you want to make calls via VoIP or fixed network by default.

Menu → Settings → Base → Default Line Type

IP / fixed line

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 a call via the other connection type.

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 website is preconfigured in your phone.

As an alternative to uploading the firmware via the Internet, it can also be loaded from a local PC. You can specify the PC via the Web configurator (page 57). This setting applies only to the next firmware update.

Precondition:

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

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

Starting firmware update

Menu → Settings → Base → Firmware Update

Enter base station system PIN (default setting: 0000).

The base station establishes a connection to the Internet or to the local PC.

Yes — Press display button to start the firmware update.

Notes:
- A firmware update can last up to 3 minutes. During the update, the handset loses the connection to the base station. When the update has been successfully completed, the handset re-establishes the connection to the base station.
- If the update is carried out from the Internet, a check is made to ensure that there is not a more recent version of the firmware available. If this is not the case, the operation is terminated and a message is issued to that effect.
- If an error occurs while firmware is being updated from a local PC, the most recent version of the firmware is automatically downloaded from the Internet.

Making VoIP settings

In order to be able to use VoIP, you must set a few parameters for your base station. You can set all parameters easily via a PC connected to your network (see page 46).

Using the connection wizard

The connection wizard starts automatically the first time your handset and base station are used. You can also start the connection wizard via the menu:

Menu → Settings → Base → VoIP Configuration (enter system PIN) → Connection Assistant

For how to enter VoIP settings using the connection wizard, see page 11.

Changing settings without the connection wizard

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

Downloading your VoIP provider's settings

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

Menu → Settings → Base → VoIP Configuration (enter system PIN) → Select VoIP Provider

The phone establishes a connection to the Internet.

▲▼	Select country and press OK.
▲▼	Select VoIP provider and press OK.

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

If errors occur during a download, see page 61.

Note:
You can make and adapt the general settings for your VoIP provider manually via your phone's Web configurator, see page 51.

Entering/changing VoIP user data

The VoIP settings must also be extended for your personal data. You will receive all necessary data from your VoIP provider.

Note:
To enter text see page 69.

Menu → Settings → Base → VoIP Configuration

⌨	Enter the system PIN and press OK.
Username / Authentication Name / Authentication Password	Select and press OK.
⌨	Enter/change user data and press OK.

Enter Caller ID for you VoIP provider account as the Username. The Username is mainly identical to your Internet phone number (the first part of your SIP address see page 52).

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.

Tip: 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 IP address can be assigned to the base station (by the router) automatically 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, you assign the base station a static IP address. This may be necessary regardless of your network configuration.

Note:
For how to make the local network settings on the Web configurator, turn to page 50.

Activating/deactivating dynamic assignment

Menu → **Settings** → **Base** → **VoIP Configuration** (enter system PIN) → **IP Configuration**

dynamic IP address (✓ = on)
Select and press **OK** to change the current settings.

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

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 (page 82) if you have deactivated dynamic assignment.

192.168.2.2 has been preset by default.

Menu → **Settings** → **Base** → **VoIP Configuration** (enter system PIN) → **IP Configuration** → **IP Address**

The current IP address is displayed.

If necessary, enter IP address and press **OK**.

Note:
For notes on the IP address, please see page 50 and the glossary on page 82.

Viewing/changing subnet mask

You can only change the subnet mask (page 86) if you have deactivated dynamic assignment.

255.255.255.0 has been preset by default.

Menu → **Settings** → **Base** → **VoIP Configuration** (enter system PIN) → **IP Configuration** → **Subnet Mask**

The current subnet mask is displayed.

If necessary, enter subnet mask and press **OK**.

Note:
For notes on the subnet mask, please see page 50 and the glossary on page 86.

Activating/deactivating display of VoIP status codes

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

Activate the function e.g. if you have problems with VoIP connections. You will receive a provider-specific status code, which supports the service when the problem is analysed.

Menu → Settings → Base → VoIP Configuration (enter system PIN) → IP Configuration

Status on HS (✓ = on)
Select and press OK.

Notes:

- For how to make the setting on the Web configurator, see page 59.
- A table with possible status codes and their meaning can be found in the Appendix on page 63.

Check the base station MAC address

Depending on your network configuration, it may be that you have to enter your base station MAC address e.g. into your router's access control list. You can check your base station MAC address:

Menu 4 3 9 2 0

The base station MAC address is displayed.

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

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 concern fixed 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 → Additional Features → Dialling Mode

Tone / Pulse
Select and press OK (✓ = on).

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

Setting the flash time

You can set the flashing time.

Menu → Settings → Base → Additional Features → Recall

Select flashing time and press OK.

The current setting is marked with ✓.

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 **4 3 9 1 6**

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

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

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 **4 3 9 1 1**

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

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

Switching temporarily to tone dialling (DTMF)

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

Precondition: You are currently conducting an external call via the fixed network or you have dialled an external fixed 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

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

Note:
Depending on your VoIP provider, it is possible that you will be unable to change individual settings in the Web configurator.

Configuring the phone via your PC

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 to each other via a router.

Notes:

- The phone is **not** blocked while you make 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.

With the Web configurator on your phone you have the following options:

- Configure your phone access to the local network (IP address, gateway to the Internet).
- Configure your phone for VoIP.
- Set the data server for firmware updates and load new firmware onto the phone if necessary.
- Obtain information about your phone's status (firmware version, MAC address etc.)

Connecting your PC to the Web configurator

- Launch the Web browser on your PC.
- Enter the phone's IP address in the address field of the Web browser, e.g. http://192.168.1.10.
- Press the return key.

A connection is established to the phone's Web configurator.

Note:
Your phone's IP address can change if you have activated dynamic IP address assignment (page 50).
You can check the phone's current IP address on the handset (page 43).

Registering, setting the Web configurator language

Once you have successfully established the connection, the Web page **Login** will be displayed in the Web browser.

You can select the language you want the menus and Web configurator dialogs to be displayed in. The language that is currently selected is displayed in the top field of the Web page.

- If necessary, click on ▾ to open the list of available languages.
- Select the language.
- In the bottom field of the Web page, enter your phone's system PIN (default setting: 0000) to access the Web configurator functions.
- Click on **OK**.

Once you have successfully registered, a **Home** opens with general information on the Web configurator.

Notes:

- If you have forgotten your system PIN, you must restore your device's factory settings. Ensure that all other settings are also restored (page 40).
- If you do not make any entries for a lengthy period (approx. 10 min.), you will be automatically de-registered. The next time you try to make an entry or open a Web page, the Web page **Login** will be displayed. Enter the system PIN again to re-register.
- Entries that had not yet been saved on the phone before automatic de-registration are lost.

De-registering

In the menu bar (page 48) at the top right of every Web page in the Web configurator, you will see the command **Log Off**. Click on **Log Off** to de-register from the Web configurator.

Caution:
Always use the command **Log Off** to end the connection to the Web configurator. If, for example, you close the Web browser without de-registering 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 the diagram below.

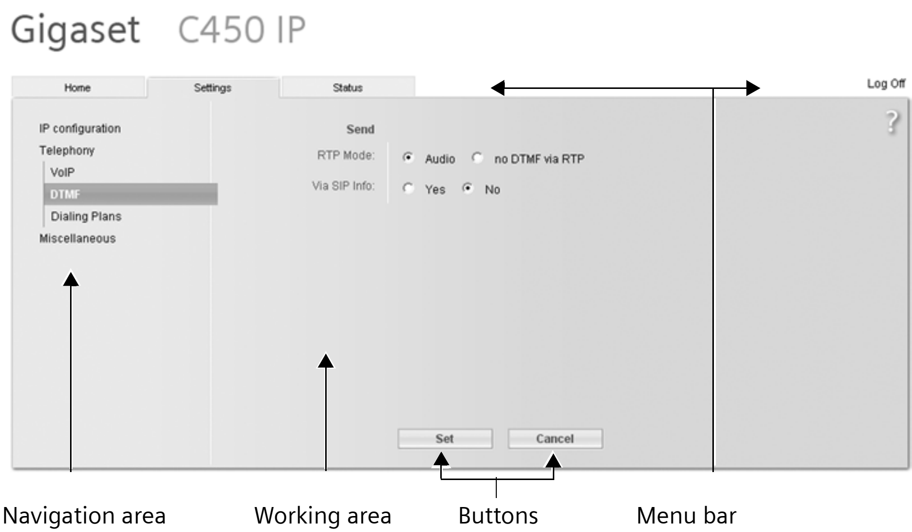


Figure 2 Example of the structure of a Web page

Menu bar

In the menu bar, the Web configurator menus are given 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 50)

  This menu allows you to make settings on your phone.

- **Status** (page 59)

  This menu gives you information about your phone.

If you click on the **Settings** menu, a list with this menu's functions is displayed in the navigation area (see below).

You will find the **Log Off** function to the right of the menu bar on every Web page (page 47).

Navigation area

In the navigation area, the functions of the menu selected in the menu bar (page 48) are listed.

If you click on 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 click on the function. The relevant page for the first subfunction is displayed in the working area

Working area

Depending on the function selected, information or dialog 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. entering special characters or certain value ranges.
- To open a list, click on . You can choose between default values.
- To activate options, click on . The previously activated option is deactivated. The active option is marked with .

Applying changes

As soon as you have made your change on a page, activate the new setting on the phone by clicking on **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.

Caution:
Changes that have not been saved on your phone are lost if you move to another Web page or if the Web configurator is terminated, e.g. due to the time limit (page 47).

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 on 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:

Entering your own emergency number

Settings → **Telephony** → **Emergency numbers**

To open this Web page, carry out the following steps after registration:

- Click on the **Settings** menu in the menu bar.
- Click on the **Telephony** function in the navigation area.

  The subfunctions of **Telephony** are displayed in the navigation tree.
- Click on the **Emergency numbers** subfunction.

Setting phone with Web configurator

You can make the following settings using the Web configurator:

- ◆ Connecting your phone to the local network (page 50)
- ◆ Configuration for VoIP telephony (page 51)
- ◆ User-specific dialling plans (page 56)
- ◆ Data server for firmware update downloads (page 57)
- ◆ Display of VoIP status codes on the handset (page 58)

IP configuration

Assign IP address

Make the necessary settings for operating your phone in your local network and to connect it to the Internet. For more detailed explanations on the individual components/terms, see the glossary (page 78).

▸ Open **Settings** → **IP configuration** Web page.

▸ 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 IP address for your phone. A static IP address is useful, for example, when 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 you 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 block 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. In addition, it must 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 in the address block 192.168.0.1 – 192.168.255.254.

The usual address for the subnet mask 255.255.255.0 is preconfigured in the default settings.

Default gateway

Enter the IP address for the standard gateway, by means of which the local network is connected with the Internet. This is generally the local (private) IP address for your router. 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 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.

Click on **Set** to save the changes.

Click on **Cancel** to reject the changes.

Allow access from other networks

The default setting for you phone is 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 to deactivate remote access again if you no longer need it.

- Open **Settings** → **IP configuration** Web page.
- In the **Remote Management** area, activate the option **Yes** to permit access from other networks.

  To deactivate remote access, click on 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 (standard 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.

VoIP telephony

Make the settings that your phone requires to access your provider's SIP server. For the majority of VoIP providers, you can make the most important setting on your handset (page 42). The Web configurator allows you to extend the possibility of these settings.

If your VoIP provider general settings are not available for download in the provider list on the Internet, you must make these settings using the Web configurator as follows.

- Open **Settings** → **Telephony** → **VoIP** Web page.
- In the working area, enter the configuration data as listed below into the areas **SIP**, **Listen ports**, **Network** and **Voice codecs**.

Area: SIP

Enter the configuration data that is necessary for accessing your VoIP provider's SIP service. You will receive this data from your VoIP provider.

Authentication Name
Specify the registration or authentication Id agreed with your VoIP provider. 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 / Confirm authentication password
Enter the password that you have agreed with your VoIP provider in the **Authentication password** field. The phone needs the password when registering with the SIP proxy/registrar server. The password is concealed when entered. Re-enter the password in the **Confirm authentication password** field.

Username

Enter the caller ID for your VoIP provider account. This ID is usually identical to the first part of your SIP address (URI, your Internet phone number).

Example: If your SIP address is "987654321@provider.com", enter "987654321" in **Username**.

Domain

Specify the last part of your SIP address (URI) here.

Example: For the SIP address "987654321@provider.com", enter "provider.com" in **Domain**.

Display name (optional)

Enter any name that should be shown in the other party's display when you call him via the Internet (example: Anna Sand). All characters in the UTF8 character set (Unicode) are permitted. This name must not exceed 32 characters

If you do not enter a name, **Username** is displayed.

Ask your VoIP provider if this feature is supported.

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.
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.

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.

Area: Listen ports

Specify the phone's local ports for VoIP telephony here. 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.

> **Note:**
> Ports 0 to 1023 should not be used, because these are often used by standard applications.

RTP port

Specify the local communication port that the phone should use to send and 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 even number just below it will be set (e.g. if you enter 5003, 5002 is set).
The default port number for voice transmission is 5004.

> **Note:**
> Ports 0 to 1023 should not be used, because these are often used by standard applications.

Use random ports

Click on the option **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 click on **No**, the phone will use the ports specified in **SIP port** and **RTP port**.

Area: Network

If your phone is connected to a router with NAT (Network Address Translation) and/or Firewall, you must make a few 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 53), 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

Click on **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.

If you selected the option **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.

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 smaller than the NAT session timeout.

As a rule you should not 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 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, just as with **Never**.

Outbound proxy

Enter the (fully qualified) DNS name or the IP address of your provider's outbound proxy.

> **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.

Area: Voice codecs

You can influence the quality of your VoIP connections via the parameters in the **Voice codecs** area. In addition to the language codecs used, you can activate/deactivate "Silence Suppression" and specify the language and earpiece volume (**VoIP Volume**).

Your phone supports various voice Codecs for digitalising (coding and decoding) voice data. The voice Codec used on a phone connection has a significant influence on the voice quality, e.g. through the time need to code/decode (voice delay). The choice of voice Codec is a compromise between voice quality and the necessary bandwidth.

Both sides of a phone connection (caller/sender side and receiver side) must be using the same voice Codec. The voice Codec is negotiated between the sender and the recipient when establishing a connection.

Set the voice Codec that your phone suggests when establishing a VoIP connection.

You can choose between the following voice Codecs supported by your phone:

G729

Average voice quality. The necessary bandwidth is less than 8 Kbit/s per voice connection.

To save additional bandwidth and transmission capacity, on VoIP connections that use Codec **G72** you can suppress the transmission of language packages in pauses ("Silence Suppression", **Enable Annex B for G729** option). Then, instead of the background noises in your environment, your caller hears a synthetic noise generated in the receiver.

Please note: "Silence Suppression" may mean a deterioration in the voice quality.

G711 a law/G711 μ law

Excellent voice quality (comparable with ISDN). The necessary bandwidth is 64 Kbit/s per voice connection.

G726

Good voice quality (inferior to that with G.711 but better than with G.729).

Your phone supports G.726 with a transmission rate of 32 Kbit/s per voice connection.

- In the **VoIP Volume** parameter, specify the amplification level of the voice and earpiece volume.

  With some VoIP providers it may be the case that the voice/earpiece volume is too low or too high. Volume regulation via the handset may then be insufficient.

  You can pre-adjust the volume via the **VoIP Volume** parameter. You specify whether the adjustable volume range on the handset should be raised or lowered. The following choices are available:

  Low

  Voice/earpiece volume is too low. If you activate this option the volume is raised by 6 dB.

  Normal

  The voice/earpiece volume does not need to be raised/lowered.

  High

  Voice/earpiece volume is too high. If you activate this option the volume is lowered by 6 dB.

- In the **Enable Annex B for G729** field, state whether, when using Codec **G729**, transmission of data packages for pauses is to be suppressed (**Yes**).

- Apply the voice Codecs that your phone suggests with outgoing calls into the **Selected codecs** list.

  Click in the **Available codecs** list on the voice Codec that you want to apply (you can mark several entries using the Shift key or the Ctrl key). Click on the **<Add** button.

- Move the voice Codecs that you do not want the phone to use into the **Available codecs** list.

  Also select the voice Codec in the **Available 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 receiver 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 receiver to begin with. If the receiver does not accept this voice Codec (e.g. because it does not support it), the 2nd voice Codec in the list is suggested etc.

If the receiver 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.

Notes:
- You should only deactivate Codecs (put them in the **Available codecs** list) if there is a particular reason. The more Codecs are deactivated, the greater the danger that calls will not be able to be established due to unsuccessful Codec negotiations.
- With incoming calls, all supported voice Codecs are always permitted.

Saving settings on phone

- Click on **Set** to save the changes.

If you want to reject the changes that have been made, click on **Cancel**. The Web page is re-loaded with the data saved on the phone.

Please note: If you do not make an entries for a lengthy 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.

Setting DTMF signalling

For example, DTMF signalling is required for playing and controlling some network answer machines via digit codes.

For VoIP specify how DTMF signals should be transmitted: as audible information in the voice 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** → **DTMF**.
- In the **RTP Mode** field, specify whether DTMF signals are to be transmitted acoustically (in voice packages). If so, activate **Audio**; otherwise, activate **no DTMF via RTP**.
- In the **Via SIP Info** field, specify whether DTMF signals are to be transmitted as code. Activate **Yes** or **No**.
- Now click on **Set** to save your settings.

Defining dialling plans

You can define user-specific dialling plans for your phone.

- Open the following Web page: **Settings** → **Telephony** → **Dialing Plans**.

Setting Area Code Predialling:

In VoIP calls you must generally always dial the area code – even for local calls.

You can save the annoying need to dial the area code for local calls by activating the **Area Code Predialling** function. In VoIP calls, 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.

- Enter your area code in the **Area Code** field, e.g. 089.
- Click on the **Yes** option next to **Predial area code for local calls through VoIP** to activate the function.

  If you click on **No** you must enter the area code even for local calls via VoIP. Numbers in the directory must always contain the area code for dialling via VoIP.
- Click on **Set** to save the settings.

Please note that if the option is activated, the area code is prefixed to all phone numbers that do not start with 0 and are dialled via VoIP. This is especially the case for numbers of the network answer machine (page 33) and, if the **Emergency calls always via fixed line** option is deactivated (see below), for emergency numbers.

Changing settings for dialling emergency numbers

The default setting for your phone is that emergency numbers are always dialled via the fixed network – irrespective of which connection type you select. The fixed network always supports emergency numbers (e.g. establishing connection to the **local** police emergency number).

These emergency numbers are already preconfigured in your phone's default settings. They are displayed on this website, but cannot be changed.

You can enter an additional emergency number.

You can deactivate the setting for emergency numbers to always be dialled via the fixed network.

> **Warning:**
> If you deactivate the **Emergency calls always via fixed line** option, make sure that your VoIP provider supports emergency numbers.

- Enter a phone number as an additional emergency number in the **User-editable number** field.
- If you click on **No** next to **Emergency calls always via fixed line**, the connection via the connection type that you indicate during dialling is established (e.g. by **pressing and holding** or **briefly pressing** the talk key).

  If you click on the **Yes** option, your phone always establishes the connection via the fixed network when dialling one of the emergency numbers (default setting).
- Click on **Set** to save the settings.

> **Note:**
> For how to change the emergency number, see page 41.

Specifying the server for firmware updates and starting the update

If necessary, you can load updates of the base station firmware onto your phone. You can either download the updates directly from the Internet or from a PC in your local network.

Using the Web configurator you can specify from where the firmware should be loaded.

- Open the following Web page: **Settings** ➔ **Miscellaneous**.

Download the firmware update directly from the Internet.

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.

The firmware is loaded from the Internet if you do not enter a local file in the **User defined firmware file** field before this update.

> **Notes:**
> – 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.
> – You should not change the URL for the Internet server because this address is also used to load provider information from the Internet. If you have entered another URL, you can re-activate the default URL by restoring the base station default settings (page 40).

Conducting the firmware update locally.

Precondition: A Web server runs on the local PC (e.g. Apache).

- First, load the desired version of the firmware from the Internet onto a local 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/C450IP/Firmware_Datei.bin.
- Click on **Set** to save the changes.

This setting is automatically used for the **subsequent** firmware update. The Internet server URL stays saved and is re-used for further firmware updates. If you want to use a local PC again for another update, then you have to re-enter the IP address and file name.

Notes:
- Updating via a PC in your LAN can make sense if you want to download the same version of the firmware again because of an error or if you want to first test the firmware for security reasons.
- 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.

Starting firmware update

Preconditions:
- No calls are being made via the fixed network or VoIP.
- There is no connection between registered handsets.
- The base station menu is not open in any of the handsets.

- Click on **Update Firmware**.

The firmware is updated. This process can take up to 3 minutes.

Notes:
You can also start the firmware update on the handset (page 41).

Activating display of VoIP status codes

Display VoIP status messages on your handset when there are VoIP connection problems. These messages give you information on the status of a connection and contain a provider-specific code that helps the service team when they are analysing the problem.

Note:
A table with possible status codes and their meaning can be found in the Appendix on page 63.

- Open the following Web page: **Settings** → **Miscellaneous**.
- Click on the **Yes** option after **Show VoIP status on handset** to activate status message display

  If you click on **No**, no VoIP status messages are displayed.
- Click on **Set** to save the changes.

Checking status information via your phone

General information about your phone is displayed.

▶ In the menu list, click on the **Status** register.

The following information is displayed:

IP configuration

IP address
: The phone's current IP address within the local network. For assigning the IP address, see page 50.

MAC address
: The phone's device address.

Software

Firmware version
: Version of the firmware currently downloaded. You can download updates of the firmware on your phone (page 41). Firmware updates are available on the Internet.

EEPROM version
: Version of your phone's EEPROM storage chip (page 80).

Appendix

Symbols and typographical conventions used

This section explains the meaning of certain symbols and typographical conventions that are used in this user guide.

	Enter digits or letters.
Menu	The display functions currently in the bottom display line are shown in reversed highlights. Press the relevant display key to launch the function.
	Press the control key up or down, e.g. when scrolling.

/ 0 / *△ etc.

Press the illustrated key on the handset.

External Calls / Internal Calls (example)

Select one of the menu functions (External Calls or Internal Calls) from the list and press OK.

Menu → Audio Settings → Ringer Melody (example)

Press Menu. Select Audio Settings with and press OK. Select Ringer Melody with and press OK.

Care

- Wipe the base station 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 the handset off and remove the batteries 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.

Questions and answers

If you have any questions about using your phone, visit us at any time at www.siemens.com/gigasetcustomercare. The table below contains a list of common problems and possible solutions.

Notes:
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 59). For how to check the MAC address displayed on your handset, turn to page 44.
- VoIP status code (page 63)
  For problems with VoIP connections, you should set VoIP status messages to be displayed on your handset. (page 42, page 58). 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 6).

The handset does not respond to a key press.

The keypad lock is activated.
▸ Press and **hold** the hash key # (page 19).

Base **flashes in the display.**

1. The handset is outside the range of the base station.
   ▸ Move the handset closer to the base station.
2. The base station is not switched on.
   ▸ Check the base station mains adapter (page 8).
3. An update of the base station firmware is currently being conducted (page 41/ page 57).
   ▸ Please wait until the update is complete.

Please Register **flashes in the display.**

The handset is not registered.
▸ Register the handset (page 34).

Handset does not ring.

The ringtone is switched off.
▸ Activate the ringtone (page 38).

You cannot hear a ring/dialling tone from the fixed network.

Base station's phone cord has been replaced.
▸ When purchasing a new cord, ensure that it has the correct pin connections (page 9).

When making calls from the fixed network, the caller's phone number is not displayed although CLIP (page 18) is set.

Phone number 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 43).

You have made a call via VoIP but cannot hear the other participant.

Your phone is connected to a router with NAT/ firewall.
▸ Your STUN server or outbound proxy settings are incomplete or incorrect. Check the settings (page 53, page 54).
▸ No outbound proxy is entered or the outbound proxy mode Never is activated (page 54) 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. The display shows **Server not accessible!**.

- First wait a few minutes. This is often a short-term event that corrects itself after a short time.

If the message is still 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. For example, set a ping command on 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. The display shows **SIP registration failed!**.

- 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. Your information for **Username**, **Authentication Name** and **Authentication Password** may be incomplete or incorrect.
   - Check your information. In particular, 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 establish a connection to the phone with your PC's Web browser.

- When establishing a connection, check the local phone IP address that has been entered. You can check the IP address on your handset (page 43).
- Check the LAN connections for the PC and phone.
- Check that your phone can be reached. Transmit a ping command to your phone, e.g. from your PC.
- 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 53).
- Your phone is not registered with the SIP service.
- You have entered the wrong user ID or an incorrect domain (page 52).

No firmware update or VoIP profile download is carried out.

1. If **Not possible! Try later!** is displayed, the VoIP line 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 use only firmware and downloads that are made available on the preconfigured Siemens server (page 57) 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 57). Correct the address. If necessary, reset the base station.
4. If **Transmission error XXX** is displayed, an error occurred in 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.

VoIP status codes

If you have problems with your VoIP connections, activate the **Status on HS** function (page 44, page 58). You will then receive a VoIP status code that will support you in problem analysis. Also enter the code during problem analysis by the Service department.

In the following tables you will find the meanings of the most important status codes and messages.

Status code	Meaning
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/she wants to make the connection.
0x301	Permanently redirected. The called party can no longer be reached under this number. The new number is transmitted 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 number dialled. The duration of redirecting is time-limited. The phone is also informed of the duration of redirecting.
0x305	The query is redirected to another "Proxy Server", e.g. to balance query loads. The phone will make the same query once again to another Proxy Server. This is not a redirection of the address per se.
0x380	Other service: The query or the call could not be made. But the phone is notified what other options there are to be able to connect the call.
0x400	Wrong call
0x401	Not authorised

Status code	Meaning
0x403	The requested service is not supported by the VoIP provider.
0x404	Wrong phone number. No subscriber to this number. Example: In 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	Calling partner cannot be reached (e.g. account cancelled).
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 "hops": The query was rejected because the service server (proxy) has decided that this query has already run through too many service servers. The maximum number was previously specified 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 can not be processed by the VoIP provider.

Status code	Meaning
0x486	The called party is busy.
0x487	General faults: The call was interrupted 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, which makes further execution of the query impossible. In this case, the caller or the phone displays the fault and repeats the query after a few seconds. 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, a 405 is sent 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.
0x503	The query cannot currently 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 of this.

Status code	Meaning
0x504	Time limit 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 the server or the SIP device uses that is 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 interrupted 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 subscribers.
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.

Voice quality and infrastructure

With your Gigaset C450 IP you have the possibility of making 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 VoIP provider components are just some of the things that can influence performance:

- Router
- DSLAM
- DSL transmission line and speed
- Connection paths over the Internet
- If necessary, other applications that also use the DSL connection

In VoIP networks, the voice quality, amongst other things, is influenced by the "quality of service" (QoS). If the entire infrastructure demonstrates QoS, voice quality is better (fewer delays, less echoing, less crackling etc.).

If, for example, the router does not have QoS, the voice quality is not as good. Please see the specialist documentation for further information.

Notes:
You should observe the following for good voice quality:
– When making calls using VoIP, avoid performing other Internet activities (e.g. surfing the net).
– Irrespective of the Codec used and the network capacity utilisation, note that voice delays can occur. Therefore, allow your VoIP calling partner to finish speaking. Do not interrupt him or her.

Searching for 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 sec.

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 you can call a service employee over the Internet without needing to be registered with a VoIP provider. This means that he/she can test online connections and VoIP telephony irrespective of the VoIP provider.

Service information of the handset

In the handset idle status:

- Press **Menu**.
- Enter * # 0 6 #.

The following information is displayed via the handset:

1: Serial number (IPUI)

2: Number of operating hours

3: Variant, version of handset software

Service (Customer Care)

We offer you support that is fast and tailored to your specific needs!

Our Online Support on the Internet:

www.siemens.com/gigasetcustomercare

This site can be accessed at any time wherever you are. It provides you with 24/7 support for all our products. It also provides interactive troubleshooting, 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 in the appendix to this user guide.

For fast and dependable assistance with any repairs or guarantee/warranty claims, contact our service centres.

Service centre:

801 11 11 11 6

Please have your proof of purchase ready when calling.

Replacement or repair services are not offered in countries where our product is not sold by authorised dealers.

Please address any questions about the DSL and cable connection to your Internet provider.

Authorisation

This device is designed for the analogic telephone connection in the greek telephone net.

Voice over IP telephony is possible with an additional modem via the LAN interface.

Siemens Home and Office Communication Devices GmbH & Co. KG hereby declares that the phone described in this user guide is in compliance with the essential requirements and other relevant provisions of European Directive 1999/5/EC (R&TTE).

If you require a copy of the original, visit the **website**: http://www.siemens.com/gigasetdocs.

CE 0682

Guarantee Certificate

Without prejudice to any claim the user (customer) may have in relation to the dealer, the customer shall be granted a manufacturer's Guarantee under the conditions set out below:

- In the case of new devices and their components exhibiting defects resulting from manufacturing and/or material faults within 24 months of purchase, Siemens shall, at its own option and free of charge, either replace the device with another device reflecting the current state of the art, or repair the said device. In respect of parts subject to wear and tear (e.g., batteries, keypads, casings), this warranty shall be valid for six months from the date of purchase.
- This Guarantee shall be invalid if the equipment defect is attributable to improper treatment and/or failure to comply with information contained in the user manuals.
- This Guarantee shall not extend to services performed by the authorised dealer or the customer themselves (e.g. installation, configuration, software downloads). User manuals and any software supplied on a separate data medium shall be excluded from the Guarantee.
- The purchase receipt, together with the date of purchase, shall be required as evidence for invoking the Guarantee. Claims under the Guarantee must be submitted within two months of the Guarantee default becoming evident.
- Ownership of devices or components replaced by and returned to Siemens shall vest in Siemens.
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- The duration of the Guarantee shall not be extended by services rendered under the terms of the Guarantee.
- Insofar as no Guarantee default exists, Siemens reserves the right to charge the customer for replacement or repair.
- The above provisions do not imply a change in the burden of proof to the detriment of the customer.

To invoke this Guarantee, please contact the Siemens telephone service. The relevant number is to be found in the accompanying user guide.

Specifications

Recommended rechargeable batteries

(Valid at the time of going to press)

Nickel-metal-hydride (NiMH):

- Sanyo Twicell 650
- Sanyo Twicell 700
- Sanyo NiMH 800
- Panasonic 700 mAh "for DECT"
- GP 550 mAh
- GP 700 mAh
- GP 850 mAh
- Yuasa Technology AAA Phone 600
- Yuasa Technology AAA Phone 700
- Yuasa Technology AAA 800
- VARTA Phone Power AAA 700 mAh

The handset is supplied with two recommended batteries.

Handset operating times/charging times

The following information relates to batteries with a capacity of 650 mAh.

Standby time	around 125 hours (5 days)
Talktime	around 13 hours
Charging time	around 7.5 hours

The operating and charging times apply only when using the recommended batteries.

Base station power consumption

Depending on current status, around 2.5 W.

General specifications

Interfaces	Fixed network, Ethernet
DECT standard	is supported
GAP standard	is supported
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 for operation	+5 °C to +45 °C; 20 % to 75 % relative humidity
Dialling mode	DTMF (touch tone dialling)/DP (dial pulsing)
Flashing time	250 ms
Codecs	G711, G726, G729AB with VAD/CNG
Quality of Service	TOS, DiffServ
Protocols	DECT, SIP, RTP, DHCP, NAT Traversal (STUN)
Base station dimensions	105 x 132 x 46 mm (L x W x D)
Dimensions, handset	141 x 53 x 31mm (L × W × H)
Base station weight	130 g
Weight of handset with battery	116 g

Writing and editing a text message

The following rules apply when writing a text message:

- The cursor is controlled with .
- Characters are added to the left of the cursor.
- Press the hash key # **briefly** to switch from "Abc" mode to "123", 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.
- Press the hash key # 3 times: shows the selection line of the characters assigned to the hash key.
- The first letter of the name of directory entries is automatically capitalised, followed by lower case letters.

Editing text

When you press a key and **hold** it, the characters of that key appear in the bottom display line and are highlighted one after the other. When you release the key the highlighted character is inserted into the input field. For how to enter special characters, see page 69.

The display briefly shows whether upper or lower case letters or digits are selected when you switch from one mode to the next: the bottom text line displays "abc -> Abc", "Abc -> 123" or "123 -> abc".

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 (shown here as ␣)
2. Digits (0–9)
3. Letters (alphabetical)
4. Other characters

To get 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 which you have preceded with an asterisk will move to the end of the directory.

Entering special characters

Standard characters

	1		0	*	#
	*)	**)			
1x	Space	Space	.	*	Abc--> 123
2x	1	↵	,	/	123 --> abc
3x	£	1	?	(	#
4x	$	€	!	)	@
5x	¥	£	0	<	\
6x	¤	$	+	=	&
7x		¥	-	>	§
8x		¤	:	%	
9x			¿		
10x			¡		
11x			"		
12x			'		
13x			;		
14x			_		

*) Directory and other lists
**) When writing an SMS

Greek

	1		0	*	#
	*)	**)			
1x	Space	Space	.	*	Abc--> 123
2x	1	↵	,	/	123 --> abc
3x	£	1	?	(	#
4x	$	€	!	)	@
5x		£	0	<	\
6x		$	+	=	&
7x			-	>	§
8x			:	%	
9x			"		
10x			'		
11x			;		
12x			_		

*) Directory and other lists
**) When writing an SMS

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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

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Small Parts Dispatch Com Bocholt

Email: kleinteileversand.com@siemens.com

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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.

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This option is useful when you wish to copy part of the code of the Library into a program that is not a library.

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However, linking a "work that uses the Library" with the Library creates an executable that is a derivative of the Library (because it contains portions of the Library), rather than a "work that uses the library". The executable is therefore covered by this License.

Section 6 states terms for distribution of such executables.

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Whether this is true is especially significant if the work can be linked without the Library, or if the work is itself a library. The threshold for this to be true is not precisely defined by law.

If such an object file uses only numerical parameters, data structure layouts and accessors, and small macros and small inline functions (ten lines or less in length), then the use of the object file is unrestricted, regardless of whether it is legally a derivative work. (Executables containing this object code plus portions of the Library will still fall under Section 6.)

Otherwise, if the work is a derivative of the Library, you may distribute the object code for the work under the terms of Section 6.

Any executables containing that work also fall under Section 6, whether or not they are linked directly with the Library itself.

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a) Accompany the work with the complete corresponding machine-readable source code for the Library including whatever changes were used in the work (which must be distributed under Sections 1 and 2 above); and, if the work is an executable linked with the Library, with the complete machine-readable "work that uses the Library", as object code and/or source code, so that the user can modify the Library and then relink to produce a modified executable containing the modified Library. (It is understood that the user who changes the contents of definitions files in the Library will not necessarily be able to recompile the application to use the modified definitions.)

b) Use a suitable shared library mechanism for linking with the Library. A suitable mechanism is one that (1) uses at run time a copy of the library already present on the user's computer system, rather than copying library functions into the executable, and (2) will operate properly with a modified version of the library, if the user installs one, as long as the modified version is interface-compatible with the version that the work was made with.

c) Accompany the work with a written offer, valid for at least three years, to give the same user the materials specified in Subsection 6a, above, for a charge no more than the cost of performing this distribution.

d) If distribution of the work is made by offering access to copy from a designated place, offer equivalent access to copy the above specified materials from the same place.

e) Verify that the user has already received a copy of these materials or that you have already sent this user a copy.

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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.

See also: **Firewall**, **NAT**, **Outbound proxy**, **STUN**.

Authentication
Restriction of access to a network/service by use of a password to log in.

Automatic ringback
See **Ringback when the number is busy**.

B

Block dialling
Enter the complete phone number, and correct it if necessary. Then pick up the receiver or press the handsfree key to dial the phone number.

Broadband Internet access
See **DSL**.

C

Call forwarding
CF

Automatic forwarding 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
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

See **Call forwarding**.

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 that vary, for instance, according to the level of compression.

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.

Consultation call

You are making a call. With a consultation call, you interrupt the conversation briefly to establish a connection to another participant. If you terminate the connection to this participant immediately, then this was an enquiry call. If you switch to and fro between the first and second participants, it is called **Toggling**.

CW

See **Call waiting**.

D

DHCP

Dynamic Host Configuration Protocol

Internet protocol that regulates the automatic assignment of **IP addresses** to **Network subscribers**. The protocol is made available in the network by a server. A DHCP server can e.g. 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. This dynamic assignment means that several **Network subscribers** can share one IP address, although they 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 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 of superordinate DNS servers and other local DNS servers in the Internet.

You can specify the IP address of the primary/secondary DNS server.

See also: **DynDNS**.

Domain name

Name of one (or 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

See **Quality of Service (QoS)**.

DSL

Digital Subscriber Line

Data transfer technology which allows Internet access at e.g. **1.5 Mbps** over conventional phone lines. Requirements: 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

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 in certain time intervals.

See also: **Static IP address**

DynDNS
Dynamic DNS

DNS is used to assign domain names and IP addresses. For **Dynamic IP Addresses** this service is now enhanced with "Dynamic DNS". This permits the use of a PC with a changing IP address as a **Server** on the **Internet**. DynDNS ensures that a service in the Internet can always be addressed under the same **Domain name** irrespective of the current IP address.

E

ECT
Explicit Call Transfer

Participant A calls Participant B. He 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

Your phone's storage chip with fixed data (e.g. user-specific device settings made at the factory) and automatically saved data (e.g. caller list entries).

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).

See also: **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
System of billing 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 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 network. As there is little compression, the necessary bandwidth is approx. 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.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 rather less with G.729A/B. As a result of the high level of compression, the necessary bandwidth is only approx. 8 Kbit/s per voice connection, but the delay is approx. 15 ms.

Gateway

Connects two different **Networks** with one another, e.g. router as 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

See **SIP Provider**.

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 comfortable. There are headsets available which are connected to the handset by a cable.

Hold music

Music on hold

Music is played while you are making a **Consultation call** or **Toggling**. The waiting participant hears music while on hold.

HTTP proxy

Sever via which the **Network subscribers** can process their Internet traffic.

Hub

Connects several **Network subscribers** in one **Infrastructure network**. All data sent to the hub by one network subscriber is forwarded to all network subscribers.

See also: **Gateway**, **Router**.

I

IEEE

Institute of Electrical and Electronics Engineers

International body that defines standards in electronics and electrotechnology, concerned in particular with the standardisation of LAN technology, transmission protocols, data transfer rate and wiring.

Infrastructure network

Network with central structure: all **Network subscribers** communicate via a central **Router**.

Internet

Global **WAN**. A series of protocols have been defined for exchanging data, known by the name TCP/IP.

Every **Network subscribers** is identifiable via its **IP address**. **DNS** assigns a **Domain name** to the **IP address**.

Important services on the Internet include the World Wide Web (WWW), Email, 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 the addressing of subscribers in a **Network** using **IP addresses**, and routes 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 on the basis of 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 consists of four parts (decimal numbers between 0 and 255) separated by points (e.g. 230.94.233.2).

The IP address is made up of the network number and the number of the **Network subscribers** (e.g. phone). Depending on the **Subnet mask**, the front one, two or three parts make up of the network number and the rest of the IP address addresses the network components. 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).

See also: **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.

See also: **IP address**.

Local SIP Port

See **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 is composed of six 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.

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 subscribers

Devices and computers 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.

See also: **STUN** and **NAT**.

P

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** across 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 subscribers**. 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 identifies the recipient or sender of a data packet within a network.

Pre-dialling

See **Block dialling**.

Private IP Address

See **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 the **IP address**/**Domain name** 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 or router) have a public and a local IP address.

See also: **IP address**, **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, bandwidth reservation and 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. Such items as melodies and screen pictures are saved in the RAM after being loaded onto the phone via the Web configurator.

Registrar

The registrar manages the **Network subscribers** 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 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 his receiver the connection is made automatically.

ROM

Read Only Memory

A type of memory that can only be read, as opposed to RAM which can be both read and written.

Router

Routes data packets within a network and between different networks via the quickest route. Can connect **Ethernet network** and WLAN. Can be the **gateway** to the Internet.

Routing

Routing is the transmission of data packets to another subscriber in your network. On its way to the recipient, the data packet is sent from one router to the next until it reaches its destination.

If data packets were not forwarded in this way, a network like the Internet would not be possible. Routing connects the individual network to this global system.

A router is a part of this system; it transmits data packets both within a network and from one network to the next. Transmission 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** by means of which voice data packets are sent and received for VoIP.

S

Server

Makes a service available to other **Network subscribers** (**Clients**). The term can indicate a computer/PC or an application. A server is addressed via the **IP address**/**Domain name** 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

See **URI**.

SIP port/Local SIP port

(Local) **Port** by means of which SIP signalling data is sent and received for VoIP.

SIP Provider

See **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 never changes.

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 by symmetric NATs.

See also: **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 subscribers**. The proportion of the IP address made up of the network number is determined in the subnet mask. For 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 in.

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 communication partners (applications).

See also: **UDP**, **TCP**, **TLS**.

U

UDP
User Datagram Protocol

Transport Protocol. Unlike **TCP**, **UDP** is a non session-based protocol. It does not establish a fixed connection. The data packets (datagrams) are sent as Broadcast. The recipient is solely responsible for making sure the data is received. The sender is not notified about whether it is received.

URI
Uniform Resource Identifier

Character string used to identify resources (e.g. Email recipient, http://siemens.com, files).

On the **Internet** URIs are used as a unique identification for resources. URIs are also described as an SIP address.

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

See **User recognition**.

User recognition

Name/number combination for access e.g. to your VoIP account.

V

Voice Codec

See **Codec**.

VoIP

Voice over Internet Protocol

Calls are no longer established and transmitted via the telephone network, but via 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 radio) 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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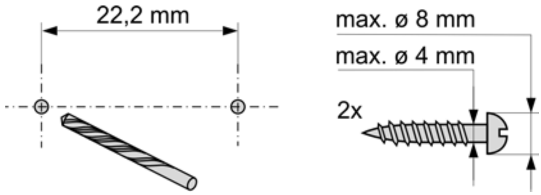
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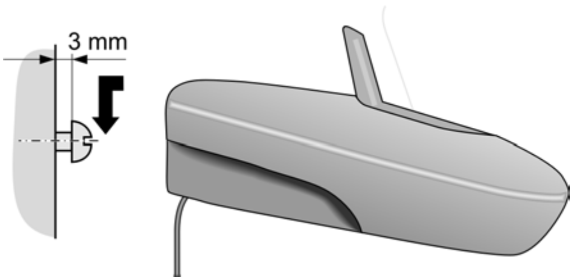
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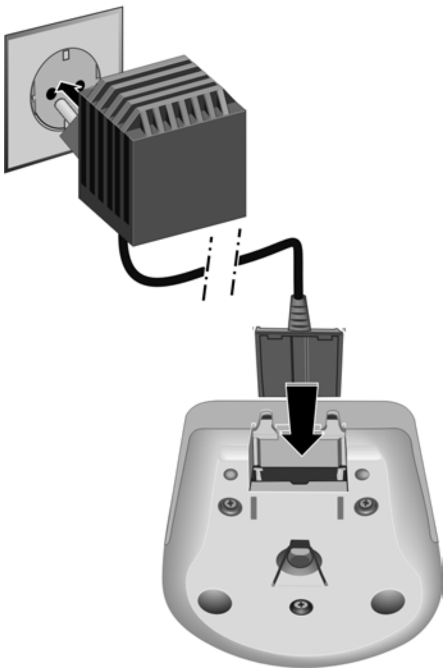

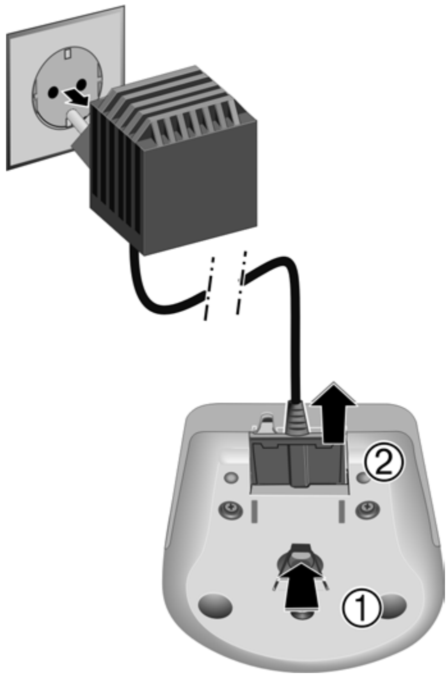

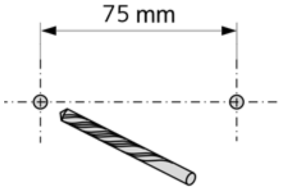
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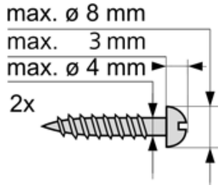
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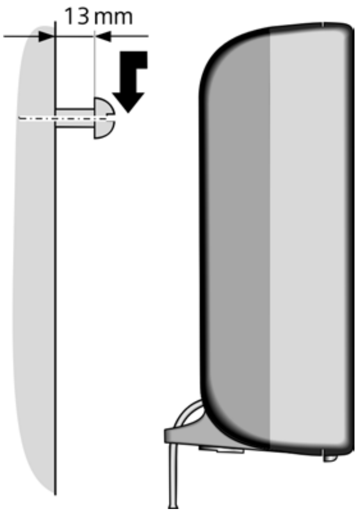
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