# Gigaset S675 IP, S685 IP, C475 IP:

# New and updated functions

This document is a supplement to the user guide for the following Gigaset VoIP phones:

Gigaset C475 IP, Gigaset S675 IP and Gigaset S685 IP

In the time since the user guides were completed, the functionality of these devices has been increased (firmware version 02.140 or higher). These changes are described in this document.

## Important information for the Gigaset S685 IP:

This document is a supplement to the user guide for Gigaset S685 IP devices manufactured before May 2009. The label in the handset's battery compartment will bear the inscription "Gigaset S68H" (no "S2").

The label is located on the bottom of the battery compartment.

The user guide can be found on the CD supplied with the device.

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# Overview of the new and updated functions

## New functions

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- You can display the text of e-mail messages on the handset.
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- You can deactivate your handset's microphone during an external call via the left display key. In this case, the other party cannot hear you, but you can still hear him/her. Network functions (e.g. external consultation calls, initiate/end conference call) that can no longer be accessed via the left display key due to this new function are now provided in the pop-up menu.
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- If you connect the telephone to a PABX, you can save the PABX access code on your phone.
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- You can now activate your fixed line connection as an alternative connection. If the telephone cannot establish a VoIP connection, it automatically attempts to establish a connection via the fixed line network.
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- As before, you can assign a VoIP provider feature to the R key. Alternatively, you can use the R key for transferring calls (call transfer via VoIP).
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- Depending on your router's NAT, you can activate/deactivate the STUN server for your Gigaset.net connection.
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## Updated/extended functions

- Changes to getting the phone started.
  For example, Gigaset.net assistant is only started when you first open the Gigaset.net directory.
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- Since several consecutive RTP ports are required for each VoIP connection, you can now specify a port number range for the RTP ports when configuring the telephone.
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- The signalling of calls made to a number that is not assigned to a handset as a receive number has changed.
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## New information on troubleshooting/problem analysis

- New functions (e.g. immediate download of a provider profile) have been added to the service information that you can use during an external call (e.g. with the Gigaset service).
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- If your phone is connected to a NAT router, the NAT can cause problems during VoIP telephony (especially if you connect multiple VoIP telephones to your router). Notes on resolving these problems can be found in these amendments.
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- The table of VoIP status codes that you can display on the screen has been extended. The extended table can be found in these amendments.
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## Function no longer in use

- When dialling, you can no longer select the line type by adding **#** or *****.
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# Description of new and updated functions

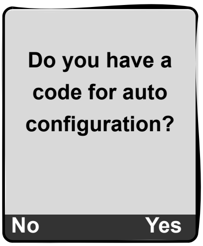
## Changes to phone setup process

This section amends the section "First steps – Making settings for VoIP telephony" in the user guide for your Gigaset VoIP phone.

The procedures for "Making settings for VoIP telephony" have changed as follows.

1. Auto-configuration:

   After you have started the installation assistant and entered the system PIN, the following is displayed:




If you have received an **auto-configuration code** (Activation Code) from your VoIP provider:

- Press the key below Yes on the display screen.

You are prompted to enter the code.

- Use the keypad to enter the auto-configuration code (max. 32 characters) and press OK.

All data necessary for VoIP telephony is loaded directly from the Internet to your phone. The handset returns to idle status. The configuration is complete.

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

- Press the key below No on the display screen.

The VoIP configuration is then performed as described in the user guide for your telephone.

2. Gigaset.net assistant:
   After you have completed the VoIP configuration, i.e. after entering your user data or the auto-configuration code for your VoIP account, the Gigaset.net assistant is **no longer** started (the step "Entering your name in the Gigaset.net directory" in the telephone user guide can be skipped). After you have entered your user data or the auto-configuration code, the handset reverts to idle status.

   The Gigaset.net assistant is started when you open the Gigaset.net directory for the first time (→ Page 7). You can then enter your name in the Gigaset.net directory.

3. If a firmware update for your telephone is available in the Internet:
   In this case, the message **New firmware available** is displayed if you press the flashing message key after starting your handset and connecting the base.

   Perform the firmware update (press the right display key **Yes**). Once the update has been completed (after approx. 3 minutes) the handset's idle display appears again and the message key ✉ flashes. If you press ✉, the following is displayed: **Start wizard for entry of VoIP connection data?**. You can then start the connection assistant as described in the user guide.

# Starting the Gigaset.net assistant when first opening the Gigaset.net-directory

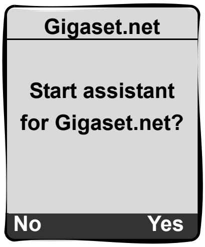
After setup, you can use the Gigaset.net assistant to enter your name in the Gigaset.net directory, i.e. create a Gigaset.net directory entry for your telephone. To do so, open the Gigaset.net directory.

**Precondition:** Your handset is in idle status.

Press and **hold**.

If necessary, select **Gigaset.net** from the list of available online directories and press **OK**.

The following appears in the handset's display:




- Press the display key **Yes** to start the assistant.

**Please note**

The Gigaset.net assistant is only opened the **first** time you open the Gigaset.net directory. If you press **No**, the assistant will be cancelled and not restarted. You can then enter your name using the Gigaset.net directory menu (**Options** → **Own details**).


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

If there is already an entry with this name, you are requested to enter a different name.

If an entry was created successfully in the Gigaset.net directory, the message "**Your user name has been added to Gigaset.net!**" is displayed.

The handset returns to idle status.

# Info Center – with the handset always online

You can use your Gigaset S67H, S68H or C47H handset to access online content on the Internet, i.e. request info services specific to your handset's display. The info services available are constantly updated. A preselection has already been made for your handset, but you can change these settings and add new services. Visit the Gigaset.net page www.gigaset.net on the Internet and compile your personal info services.

### Customising info services

- On the PC, open the page **Settings** → **Services** of the Web configurator for your handset.
- Click the link **gigaset.net/myaccount**.

Or:

- Enter the following address in the address field of your PC's web browser: www.gigaset.net
- Enter your Gigaset.net user ID and password on the Gigaset.net page. You will find your user ID and password on the Web configurator page "**Settings** → **Services**").

In both cases, a Web page is opened on which you can compile your info services.

## Starting Info Center, selecting info services

→ **Sel. Services** → **Info Center**

The menu of your Info Center, i.e. a list of user-specified info services (see above) is displayed. You can navigate between the info services.

Example:


Select the info service and press **OK**.

To access certain info services (personalised services), it is necessary to register with a user name and password. In this case, enter your access data as described in the section "Registration for personalised info services" on Page 9.

### Messages when loading requested information

The information is loaded from the Internet. Wait a few seconds until the information is displayed. The display shows **Please wait**.

If the information for an info service cannot be displayed, one of the following messages appears:

**Requested page can't be reached.**

Possible causes of this are:

◆ Time limit exceeded (timeout) when loading the information, or
◆ the Internet server for the info service cannot be accessed.

▶ Check your Internet connection and repeat the request at a later time.

**Coding error on requested page**

The content of the requested info service is coded in a format which the handset cannot display.

**Can't display requested page**

General error when loading the info service.

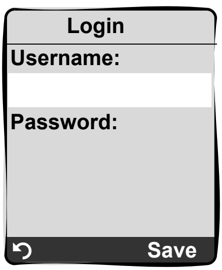
**Login failed.**

Registration has failed. Possible causes of this are:

◆ You have not entered your registration data correctly.
  ▶ Reselect the info service and repeat the registration process. Please remember that data is case sensitive.
◆ You are not entitled to access this info service.

## Registration for personalised info services

If a special registration with user name and password is required to access an info service, the following is displayed after the service has been called up (example):




| | |
|---|---|
| **Username** | Enter the user name that you have agreed with the info service provider. |
| **Password** | Enter the password associated with this user name. |
| **Save** | Press the display key to send the registration data. |

If registration was successful, the requested info service is displayed.

If registration failed, a message to this effect appears on the display → Messages when loading requested information, Page 9.

**Please note**

Please remember that registration data is case sensitive.

## Operating the Info Center

Depending on the type of info service requested, you can carry out the following actions:

### Scrolling within an info service

▶ You can use [down] to scroll downwards within an info service, and [up] to scroll up (back).

### Skipping back to the previous page

▶ Press the left display key [back].

### Skipping back to the Info Center menu

▶ **Briefly** press the end call key [end].

You want to go offline:

Press and **hold** the end call key [end], the handset returns to idle status.

### Selecting a hyperlink

◆ Hyperlink to further information:

If the page contains a hyperlink to further information, this is indicated by the ▶ icon.

If a page is opened using hyperlinks, the first hyperlink is highlighted.

▶ Using the control keys ([down] and/or [up]) as required, you can navigate to the hyperlink that you would like to select. The hyperlink is then highlighted by bars.

▶ Press the right display key **Link**, to open the relevant page.

◆ Hyperlink to a phone number:

If a hyperlink contains a phone number, you can copy the number to the local directory or call the number directly (Click-2-Call functionality).

▶ Select the hyperlink using the [down] and/or [up] keys, as required.

▶ You can identify a hyperlink of this type by the fact that **Call** is shown above the right display key.

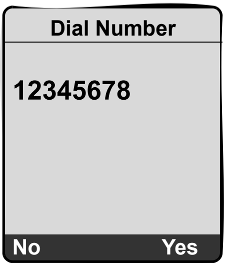
▶ Press **CopyToDir** if you want to copy the phone number to your handset's local directory.

Or:

▶ Press **Call** to call the stored number.

Depending on the provider, if you press **Call**,

– the number is either dialled directly or
– appears first in the display, in which case you must first confirm the number before it is dialled.
  Example:




▶ If you press **Yes**, the number is dialled.

Or:

▶ If you press **No**, the page with the hyperlink is displayed again. The number will **not** be dialled.

## Entering text

▶ If necessary, use to navigate to the line containing the field into which you want to enter text. The cursor flashes in the text field.
▶ Enter your text using the handset keys (for information on entering text see → the appendix to the user guide belonging to your phone).
▶ If necessary, navigate to other text fields to complete them or make a selection (see below).
▶ Press the right display key to complete the entry and send the data.

## Making selections

▶ If necessary, use to navigate to the line, in which you would like to make a selection.
▶ Press left or right on the control key several times to make the desired selection.
▶ Use to navigate to other selection fields and make your selection as described above.
▶ Press the left display key to complete the selection and send the data.

## Setting options

▶ Use to navigate to the line containing the option. The line is highlighted.
▶ Activate or deactivate the option via the control key (press right) or the right display key (e.g. **OK**).
▶ If necessary, navigate to other options or text fields to set or complete them.
▶ Press the left display key (e.g. **Send**) to complete the entry and send the data.

# Sending and receiving SMS (text messages) via VoIP

This section amends the chapter "SMS (text messages)" in the user guide for your Gigaset VoIP phone.

You can now use your telephone to send and receive SMS messages via the fixed line network and **VoIP**.

You can receive SMS messages (abbreviated: SMS) via all of your telephone's connections (with the exception of Gigaset.net). You must explicitly specify the (send) line via which the SMS messages are to be sent (fixed line network or one of your VoIP connections).

Precondition: Your fixed line network and VoIP providers support SMS functionality.

## Setting the send line

You define the send line when setting the SMS centres. For each individual SMS centre, you can specify which of your lines is to be used to send SMS messages when this SMS centre is activated as the send centre.

**Please note**

Before you specify one of your VoIP connections as the send line, check with your VoIP provider whether the SMS centre can be reached via the VoIP connection.
Not all VoIP providers support special phone numbers!

→ Messaging → SMS → Settings → Service Centres

Select SMS centre (e.g. Service Centr. 1) and press **OK**.

Scroll to the **Send via** line to select the send line to be used when sending SMS messages via this SMS centre. The fixed line network is preset by default.

**Edit** Press the display key.
A list of your phone's connections will be displayed. You can select from your fixed line network connection and all VoIP connections that you have configured. The standard names for the connections are displayed. IP1 to IP6, Fxd. ln.

**Fxd. ln. / IP1 / IP2 / ...**
Select the VoIP or fixed line connection and press **OK**.

**Save** Press the display key to save the changes.

**Please note**

- If you have selected a VoIP connection and the attempt to transmit the SMS messages fails, the SMS with error status is stored in the incoming message list. Even if you have activated your fixed line network connection as an alternative connection (→ Page 20), the telephone does not attempt to send SMS messages via the fixed line network.
- If you have selected a VoIP connection as a send line and this is deleted from the configuration, the first VoIP connection in the configuration will be used.

## Note on writing, sending and receiving SMS messages etc.

Regardless of your send line settings (fixed line network or VoIP) you can write, send and receive SMS messages as well as request SMS notifications as described in the user guide for your phone ( → chapter "SMS (text messages)").

If your VoIP provider supports the relevant features, you can also use personal mailboxes.

**Please note**

Every SMS addressed to one of your numbers (VoIP or fixed line network) is displayed on all registered handsets with SMS functionality, even if the phone number addressed is not assigned to the handset as a receive number.

## Removed: SMS registration assistant

Specific registration with the SMS service centre is no longer necessary in the majority of cases. You are registered with an SMS centre as soon as you send an SMS via this SMS centre.

For this reason, the registration assistant is no longer available. The → **Messaging** → **SMS** → **Settings** → **Subscribe to SMS** menu options have been removed.

# Reading e-mail messages on the handset

This section amends the chapter "E-mail notifications" in the user guide for your Gigaset VoIP phone.

Your phone will notify you when new e-mail messages have been received on your incoming e-mail server. Using the handset, you can now display the sender, date/time of receipt, subject and message text for each e-mail in the inbox.

## Preconditions:

- You have set up an e-mail account with an ISP.
- The incoming e-mail server uses the POP3 protocol.
- You have saved the name of the incoming e-mail server and your personal e-mail access data (account name, password) in the phone ( → user guide for your phone, Web configurator page: **Settings** → **Messaging** → **E-Mail**).

## Opening the inbox

→ **Messaging** → **E-mail**

Or if new e-mail messages have been received (the message key flashes):

→ **E-mail:**

The telephone establishes a connection to the incoming e-mail server. A list (inbox) of e-mail messages that are stored there will be displayed.

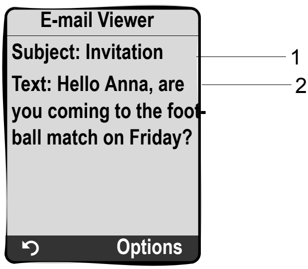
The sequence in which the e-mail messages are displayed is dependent on your POP3 server. Generally speaking, the new unread messages appear before old messages that have been read.

## Opening and reading e-mail messages

| | |
|---|---|
| | Select e-mail entry. |
| **View** | Press the display key. |

The subject (**Subject:**) and text (**Text:**) of the e-mail message are displayed. Any attachments to the e-mail are not displayed.

Example display:




1 **Subject** of the e-mail message. A maximum of 120 characters are displayed.
2 **Text** of the e-mail message (abbreviated if necessary).
A maximum of the first 640 characters of the subject and message text are displayed in total (Subject + Text + "Subject:" + "Text:" = 640 characters).

| | |
|---|---|
| | Press the display key to return to the inbox. |

**Please note**

- If the e-mail message contains more than just unstructured text, a brief message to this effect is displayed. The Subject of the message is then displayed.
- If the subject and/or message text are in HTML format, they may be displayed differently to how they appear on the PC e-mail client.

## Viewing e-mail sender's address

**Precondition:** You have opened the e-mail message to read (→ Page 14).

Options  Press the display key.

From  Select and press OK.

The sender's e-mail address is displayed in full (if necessary over several lines).

↺  Press the display key to return to the inbox.

Example:


## Deleting the e-mail message

You have opened the inbox:

⬍  Select e-mail entry.

Delete  Press the display key.

Or:

You have opened the e-mail message to read (→ Page 14) or request the e-mail sender's address to be displayed (→ Page 15):

Options  Press the display key.

Delete E-mail
Select and press OK.

The e-mail is deleted from the incoming e-mail server.

# Deactivating your handset's microphone

As well as muting the handset as described in the user guide (press left on the control key, the other party hears hold music), you can deactivate your handset's microphone during an external call. The other party cannot hear you, but you can still hear them. You can also deactivate the microphone during a conference call or when call swapping.

## Turning off the microphone

**Mute** Press the display key to deactivate the handset.
Your handset's microphone is deactivated. The display shows **Microphone is off**.

## Switching the microphone back on

**On** Press the display key to switch the microphone back on. The other party can hear you again.

**Please note**

The microphone is **automatically switched on** again in the following scenarios:

- If, during an external call (you have switched the microphone off), you establish a second connection, either by accepting a waiting call or by successfully connecting to an external/internal consultation call, the microphone is turned on. If you go back to the first party, the microphone remains **switched on**.
(If you reject a waiting call or are unable to connect to a consultation call, the microphone remains switched off.)
- If you deactivated the microphone while call swapping, it is reactivated for both connections, as soon as they are connected to the other caller.
- If you have deactivated the microphone during a conference call, the microphone is reactivated when you terminate the conference call by selecting
**Options** → **End Conference** (call swapping).

# Network services during an external call

This section amends the sections "Network services – Further network services in the fixed line network" and "Network services – Further network services for VoIP" in the user guide for your Gigaset VoIP telephone.

Some network services that were previously accessed via display keys are now provided via the context menu. To open the pop-up menu you must press the display key **Options**.

### Preconditions:

- Fixed line network: You have requested the following network services from your fixed line network provider.
- VoIP: Your phone permits two parallel VoIP connections.
  ( → user guide for your phone, Web configurator **Settings** → **Telephony** → **Audio**).

The following functions are affected

- **Consultation call**
  During an external call via VoIP or the fixed line network:
  - Press the display key **Options**.
  - Select **External Call** and press **OK**.
  - Enter a number or copy it from the directory and press **OK**.

  The first party is placed on hold and hears hold music.
- **Accepting a waiting call**
  **Precondition**: Call waiting is activated ( → user guide for your phone).
  You are conducting an external call via VoIP or the fixed line network. A second caller (waiting call) is signalled:
  - Press the display key **Options**.
  - Select **Accept waiting call** and press **OK**.

  The first party is placed on hold and hears hold music.
- **Initiating a conference call**
  You are call swapping and want to talk to both parties simultaneously:
  - Press the display key **Options**.
  - Select **Conference** and press **OK**.
- **Ending a conference call (call swapping)**
  - Press the display key **Options**.
  - Select **End Conference** and press **OK**.

## Rejecting a waiting call during a VoIP call

You can now reject a waiting call while conducting a conversation via VoIP.

You are conducting an external call via a VoIP connection. A second caller (waiting call) is signalled:

**Options** → **Reject waiting call**
Select the above and press **OK** to reject the waiting call.

# Operating the base on on the PABX – Setting access codes (external line prefixes)

This section amends the chapter "Operating the base on the PABX" in the user guide for your Gigaset VoIP phone.

Depending on your PABX, you must dial an access code before making external calls in order to obtain an external line. You can store this access code in your phone. For example, the access code is then automatically placed before the numbers selected from the calls list.

→ Settings → Base → Add. Features

**Access Code**

Select and press **OK**.

Enter or edit the access code (maximum three digits) and press **OK**.

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

If an access code is set, the following applies:

- The access code is added automatically when dialling from the calls list/answering machine list and when dialling emergency numbers and SMS centre numbers.
- When dialling numbers manually and dialling numbers from the directory you must add the access code yourself.

# Entering an HTTP proxy server (only when connected to an internal company network)

This section amends the chapter "Web configurator – IP configuration" in the user guide for your Gigaset VoIP phone.

Direct connections between network subscribers and the Internet are often not permitted within internal company or organisation networks (intranet). In such cases, all HTTP calls from the network are "transferred" by a proxy server. The proxy server is a computer or program within the network.

If your phone is connected to such a network, you must store the address of this HTTP proxy server on the phone and activate handling of HTTP calls via the HTTP proxy server.

Only then will you be able to access, for example, Gigaset.net directories, use the Info Center or obtain weather information etc. in idle display (information services).

- Open the following Web page: **Settings** → **IP Configuration**.

## Area: HTTP proxy

**Enable proxy**

- Click the **Yes** option if your phone is to handle HTTP calls via your network's HTTP proxy server.
- If you select **No**, the phone will attempt to access the Internet directly.

**Proxy server address**

- Enter the URL of the proxy server to which your phone is to send HTTP calls. The proxy server then creates the connection to the Internet.

**Proxy server port**

- Enter the communication port used on the HTTP proxy server (number between 0 and 55000). It is mainly port 80 that is used.

- Now select **Set** to save your settings.

## Activating/deactivating the STUN server of the Gigaset.net connection

This section amends the chapter "Web configurator – Configuring the Gigaset.net connection" in the user guide for your Gigaset VoIP phone.

The Gigaset.net connection is preconfigured in your phone. The Gigaset.net uses a STUN server as standard. In the sent data packets, Gigaset.net replaces the private IP address of your phone with its public IP address.

If you operate your phone with a symmetrical NAT router, STUN cannot be used. Otherwise, when making Gigaset.net calls you will not be able to hear the caller.

In this case, deactivate STUN for the Gigaset.net connection.

- Open the following Web page: **Settings** → **Telephony** → **Connections**.
- Select **Edit** in the **Gigaset.net** area.

**STUN enabled**

- Select **No** to deactivate STUN.
- Select **Yes** if you want your phone to use STUN.

- Select **Set** to save the changes.

# Activating the fixed line network connection as an alternative connection

You can activate the fixed line network connection on your phone as an alternative connection. If an attempt to establish a connection via VoIP then fails, an attempt is made automatically to establish the connection via the fixed line network.

An alternative connection would be used in the following cases:

- your VoIP connections are busy
- the SIP server for the VoIP connection cannot be accessed
- the dialled VoIP connection has not yet been configured or has not been configured correctly (e.g. incorrect password)
- the base does not have a connection to the Internet, e.g. because your router is deactivated or not connected to the Internet.

**Exceptions**

- SMS messages that are to be sent via a VoIP connection are **not** sent via the fixed line network connection as an alternative. The SMS message is stored in the incoming message list with an error status. Your handset's message key will flash.
- If you enter a VoIP line suffix (#1 to #6) or press the IP display key before dialling, the connection is **not** established over the fixed line network as an alternative.
- If you dial a URI or IP address instead of a phone number, the connection cannot be established via the fixed line network.

▶ Open the following Web page: **Settings** → **Telephony** → **Number Assignment**.

### Area Default Connection

▶ If you want to activate the fixed line network connection as an alternative connection, click the **Yes** option next to **Automatic Fallback to Fixed Line**. Select **No** to deactivate the function.

▶ Now select **Set** to activate your settings.

# R key function for VoIP – Hook flash/call diversion

This section replaces/amends the sections "Web configurator – Defining recall key functions for VoIP (hook flash)" in the user guide for your Gigaset VoIP phone.

Using your phone's Web configurator, you can assign a special feature of your VoIP provider to the R key. Alternatively, you can use the R key for call diversion (call transfer).

## Assigning the signal for a provider feature to the R key

To be able to use a special feature of your VoIP provider, your phone must send a specific signal (data packet) to the SIP server. You can assign this "signal" to your phone's R key.

If you press the R key during a VoIP call the signal will be sent to the server.

**Precondition:**

- DTMF reminders via SIP info messages is activated, i.e. the **SIP Info** option on this web page is activated.
- The R key is not used for call transfer, i.e. **Use the R key to initiate call transfer with the SIP Refer method.** = **No** is set for call transfer (→ Page 22).

If one of these preconditions is not fulfilled, the field in the **Hook Flash (R-key)** area is hidden.

▶ Open the following Web page: **Settings** → **Telephony** → **Advanced Settings**.

**Area Hook Flash (R-key)**

▶ In the **Application Type** (maximum 31 characters) and **Application Signal** fields (maximum 15 characters), enter the data that you have received from your VoIP provider.

▶ Now select **Set** to save your settings.

The setting for the R key applies to all registered handsets.

## Configuring the R key for call diversion (call transfer)

If you are transferring calls via VoIP connections, you can connect the two external callers (provider-dependent). You can configure settings for this type of call transfer.

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

### Area Call Transfer

**Use the R key to initiate call transfer with the SIP Refer method.**

- If you select **Yes**, the external parties you are toggling between will be connected when you press the R key R. Your connections with the parties will be terminated.

- Now select **Set** to save your settings.

**Please note**

- You can also activate the **Transfer Call by On-Hook** option. In this case, the two external parties are connected with one another when you press the end call key. To do so, you must use the Web configurator to define the preferred protocol to be used for call diversion ( → user guide for your phone).
- If you have deactivated both options, i.e. both **Use the R key to initiate call transfer with the SIP Refer method.** and **Transfer Call by On-Hook**, you can also divert a VoIP call using **Options** → **Call Transfer**.

# Defining local communication ports for VoIP

This section replaces the section "Web configurator – Defining local communication ports for VoIP" in the user guide for your Gigaset VoIP phone.

Specify which local communication ports (port numbers) the telephone is to use for VoIP telephony. The ports must not be used by any other subscriber in the LAN.

The following communication ports are used for VoIP telephony:

- **SIP port**
  Communication port via which the phone receives (SIP) signalling data
- **RTP port**
  RTP ports are used to receive voice and control data. Three consecutive, even port numbers are required for each VoIP connection.

You define a fixed number for the SIP port and a fixed number range for the RTP port or set your phone so that it uses free ports from a specified port number range ( → **Use random ports**).

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

### Area Listen ports for VoIP connections

**Use random ports**

- Click **No** if you want the phone to use the ports specified in the **SIP port** and **RTP port** fields.
- Select **Yes**, if you do not want the phone to use fixed ports for **SIP port** and **RTP port**, but rather to use any free ports from predefined ranges of port numbers.

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 divert incoming calls and voice data to one (the intended) phone.

### Use random ports = No

**SIP port**

- Specify the port number for the SIP port. Enter a number between 1024 and 49152 in the field.

The default port number for SIP signalling is 5060.

The port number specified must not be in the **RTP port** number range.

**RTP port**

- Specify a range of port numbers that are to be used as RTP ports. This range must be used in the LAN (router) for the phone.
- Enter the lowest port number in the left-hand field and the highest number in the right-hand field (numbers between 1024 and 55000).

**Size of the port number range:**

The difference between the port numbers must be at least **6** if you permit two simultaneous VoIP calls on your phone. It must be at least **4** if you only permit one VoIP call (→ user guide for your phone, Web configurator **Settings** → **Telephony** → **Audio**).

The lower of the port numbers in the range (in the left-hand field) must be an **even** number. If you enter an odd number, the next lowest even number will be selected automatically (e.g. if you enter 5003, then 5002 is set automatically).

The default port number for voice transmission is 5004.

### Use random ports = Yes

#### SIP port

- Enter the port number range from which the SIP port is to be dialled.
- Enter the lowest port number in the port number range in the left-hand field and the highest number in the right-hand field (numbers between 1024 and 49152).

This port number range must not overlap the range specified for **RTP port**.

The default range is 5060 to 5076.

#### RTP port

- Specify a range of port numbers from which the RTP ports are to be dialled.
- Enter the lowest port number in the port number range in the left-hand field and the highest number in the right-hand field.

The default range is 5004 to 5020.

- Now select **Set** to save your settings.

# Amendment to "Call signalling and number assignment"

The section amends the sections "Accepting calls", "Web configurator – Assigning send and receive numbers to handsets" and "Web configurator – Assigning receive numbers to the answering machine" in the user guide for your Gigaset VoIP phone.

## Signalling incoming calls

If you have **not** assigned any receive numbers, either to the answering machine or the registered handsets, calls to all connections will be signalled on all handsets.

If you **have** assigned receive numbers, your handset will only indicate calls to receiving numbers assigned to this handset. Please note the following cases:

- If the phone number is not assigned to a handset or an answering machine as a receive number, all calls to this number are signalled on all handsets.
- If the phone number is not assigned to a handset, but is assigned to the answering machine, the call is not signalled on any handset and is accepted by the answering machine.
- Calls to your phone's IP address are signalled on all handsets.

# Amendment to "Changing the display language"

The section amends the section "Handset settings – Changing the display language" in the user guide for your Gigaset VoIP phone.

## Parts of the menu are not displayed in the language selected...

...and three or more handsets are registered on your base. The language set on at least three handsets is not a standard language of the base. The standard base languages are: English, French, German, Italian, Spanish, Portuguese and Dutch.

## Cause:

On your base, display texts are only stored for the standard languages. In addition, these display texts can be stored in the base in two other languages or in another language for two different types of Gigaset handsets. When selecting the language on the handset, these texts are downloaded to the base from the Internet. If another non-standard language is set on a third handset, some display texts appear in one of the standard languages on this handset.

Both of the non-standard languages are saved on the base, which are set with the lower number of internal numbers.

If there is no further handset registered on the base whose type and language setting correspond to an additionally loaded language, then the memory is freed up. If necessary, the language set for another registered handset is loaded onto the base.

# Amendment to "Defining dialling plans"

This section amends the section "Web configurator – Defining dialling plans – Cost control" in the user guide for your Gigaset VoIP phone.

When defining dialling plans, the additional option **Use Area Codes** is now available ( → user guide for your phone, Web configurator page **Settings** → **Telephony** → **Dialling Plans**).

Activate the option **Use Area Codes** for all VoIP calls, if the automatic area code is to precede all phone number(s) for which the the dialling plan is defined.

You can define the automatic area code on the Web page **Settings** → **Telephony** → **Dialling Plans** under **Area Codes.**

**Please note**

In the case of dialling plans for emergency numbers, you should always deactivate the option **Use Area Codes**.

# Correction to "Checking the base MAC address"

This section replaces the section "Base settings – Automatic firmware update" in the user guide for your Gigaset VoIP phone.

To display the base MAC address, press the following keys in sequence while the handset is in idle status.

On the Gigaset S67H or S68H handset:

8 tuv 5 jkl 9 wxyz 7 pqrs 5 jkl

On the Gigaset C47H handset:

5 jkl 5 jkl 9 wxyz 7 pqrs 5 jkl

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

# Checking extended service information for the base

This section amends the section "Appendix – Checking service information" in the user guide for your Gigaset VoIP phone.

You may need the service information of your phone (base and handset) for Customer Services. The service information provided for the base has been extended.

## Base service information

**Precondition:** You are conducting an external call. The connection has been established for at least 8 seconds.

**Options** → **Service Info**

Confirm selection with **OK**.

In addition to the information specified in the user guide, the following information/functions are displayed. You can select them with [navigation key]:

**6:** Base device number. This contains additional information for the service employer.

**Unlock system**

Confirm selection with **OK**.
If necessary you can clear a provider-specific device lock with a corresponding code.

**Update profile**

Confirm selection with **OK**.
The current profiles of your VoIP providers (general provider data of all configured VoIP connections) are automatically loaded onto your phone. The general settings for all the VoIP connections are updated; profiles for these are available on the Gigaset server.

# Information on operating Gigaset VoIP telephones with Network Address Translation (NAT) routers

This section amends the section "Questions and answers" in the chapter "Customer care" in the user guide for your phone.

In general no special telephone or router configuration is required when operating a Gigaset VoIP phone with a NAT router. The configuration settings described in this section are only necessary if you encounter one of the following problems.

### Typical problems caused by NAT

- No incoming calls are possible via VoIP. Calls to your VoIP phone number are not put through.
- Outgoing calls via VoIP are not connected.
- A connection is established with the other party, but you cannot hear them and/or they cannot hear you.

### Possible solution

1. Change the port numbers of the communication ports (SIP and RTP ports) on your telephone ( → "**1. Changing the port numbers for SIP and RTP on your VoIP phone**").
2. In some cases, you must also define port forwarding for the telephone's communication ports on the router ( → "**2. Setting port forwarding on the router**").

## 1. Changing the port numbers for SIP and RTP on your VoIP phone

**On your VoIP telephone**, define different (local) port numbers for the SIP and RTP ports (between 1024 and 49152).

- These numbers must not be used by any other application or host in the LAN and
- be considerably higher or lower than the SIP and RTP port numbers that you usually use (and are preset on the phone).

This procedure is particularly useful if additional VoIP phones are connected to the router.

### To change the SIP and RTP port numbers on your VoIP phone, proceed as follows:

- Connect your PC's browser to the Web configurator of the telephone and log in ( → user guide for your phone).
- Open the Web page **Settings** → **Telephony** → **Advanced Settings** and change the settings for the SIP and RTP ports (→ **Page 23**).

  To help you remember the new port numbers (e.g. for router configuration), you can choose numbers that are very similar to the standard settings, e.g.

| | | | |
|---|---|---|---|
| **SIP port** | 49060 | instead of | 5060 |
| **RTP port** | 49004 to 49010 | instead of | 5004 to 5010 |

- Save the changes on your telephone.
- Wait for the active VoIP connections to be re-registered. To do so, switch to the Web page **Settings** → **Telephony** → **Connections** to see the **Status** of your VoIP connections.
- Check to see whether the problem persists. If it does, perform step 2.

## 2. Setting port forwarding on the router

To ensure that your specified SIP and RTP port numbers are used on the WAN interface with the public IP address, you must define port forwarding rules for the SIP and RTP ports on the router.

### To define port forwarding on the router, proceed as follows:

The terms used in the following can vary from router to router.

To forward a port, you must make the following specifications (example):

| Protocol | Public port | Local port | Local host (IP) | |
|---|---|---|---|---|
| UDP | 49060 | 49060 | 192.168.2.10 | **for SIP** |
| UDP | 49004–49010 | 49004–49010 | 192.168.2.10 | **for RTP** |

**Protocol**
Enter **UPD** as the protocol to be used.

**Public port**
Port number/port number range on the WAN interface

**Local port**
The SIP and RTP port numbers set on the telephone.
In the new firmware version for Gigaset VoIP telephones, you can set a RTP port range. In this case, you must also define corresponding port forwarding for this range.

**Local host (IP)**
Local IP address of your phone in the LAN. You can see the phone's current IP address in the handset display by pressing the paging key on the base.

To enable the router to perform this port forwarding, the DHCP settings of the router must ensure that the telephone is always assigned the same local IP address – i.e. the DHCP does not change the IP address assigned to the telephone during operation. Alternatively, you can assign a fixed (static) IP address to the telephone ( → user guide for your phone). However, you must ensure that this IP address is not within the address range reserved for DHCP and is not assigned to any other LAN subscriber.

# Edited/extended table of VoIP status codes

This table replaces the table of VoIP status codes provided in the appendix of the user guide for your telephone.

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

| Status code | Meaning |
|---|---|
| 31 | IP configuration error:<br>IP domain not entered. |
| 33 | IP configuration error:<br>SIP user name (**Authentication Name**) not entered.<br>This is shown, for example, when dialling with a line suffix, if no connection is configured for the suffix on the base. |
| 34 | IP configuration error:<br>SIP password (**Authentication password**) not entered. |
| 300 | The called party can be reached under several phone numbers. If the VoIP provider supports this, a list of the phone numbers is transmitted as well as the status code. The caller can select to which number he wants to make the connection. |
| 301 | Permanently redirected.<br>The called party can no longer be reached under this number. The new number is transferred to the phone together with the status code, and the phone then no longer accesses the old number but dials the new address immediately. |
| 302 | Temporarily redirected.<br>The phone is informed that the called party cannot be reached under the dialled number. The call is redirected for a limited period. The phone is also notified of the length of the redirection. |
| 305 | The query is sent to a different "proxy server", e.g. to balance incoming queries. The phone will once again make the same query to another proxy server. This is not a redirection of the address per se. |
| 380 | Other service:<br>The query or call could not be transferred. But the phone is notified what other options there are to be able to connect the call. |
| 400 | Wrong call |
| 401 | Not authorised |
| 403 | The requested service is not supported by the VoIP provider. |
| 404 | Wrong phone number.<br>No connection on this number.<br>Example: In a local call you have not dialled the area code although your VoIP provider does not support local calls. |
| 405 | Method not permitted. |
| 406 | Not acceptable.<br>The requested service cannot be provided. |
| 407 | Proxy authentication required. |
| 408 | The party cannot be reached (e.g. account has been deleted). |

| Status code | Meaning |
|---|---|
| 410 | The requested service is not available from the VoIP provider. |
| 413 | Message is too long. |
| 414 | URI is too long. |
| 415 | Query format is not supported. |
| 416 | URI is faulty. |
| 420 | Incorrect ending |
| 421 | Incorrect ending |
| 423 | The requested service is not supported by the VoIP provider. |
| 480 | The dialled number is temporarily unavailable. |
| 481 | The recipient is not available. |
| 482 | Double service query |
| 483 | Too many "jumps":<br>The query was rejected because the service server (proxy) has decided that this query has already passed through too many service servers. The maximum number is defined beforehand by the original sender of the query. |
| 484 | Wrong number:<br>In most cases this response means that you have simply omitted one or more digits in the phone number. |
| 485 | The URI dialled is not unique and cannot be processed by the VoIP provider. |
| 486 | The called party is busy. |
| 487 | General faults:<br>The call was cancelled before a call was established. The status code confirms receipt of the interruption signal. |
| 488 | The server cannot process the query because the data entered in the media description is not compatible. |
| 491 | The server notifies that the query will be processed as soon as a previous query has been completed. |
| 493 | 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. |
| 500 | The proxy or the receiving device has discovered a fault while executing the query. It is therefore impossible to execute the query. If this occurs, the caller or the phone displays the fault and repeats the query after a few seconds. The number of seconds after which the query can be repeated may be transmitted to the caller or phone by the receiving device. |
| 501 | The query cannot be processed by the recipient because the recipient does not have the functionality that the caller requires. If the recipient understands the query but does not process it because the sender does not have the necessary rights or the query is not permitted in the current context, status code 405 is transmitted instead of 501. |
| 502 | In this case, the receiving device that transmits this error code is a proxy or a gateway and has received an invalid response from its gateway via which this query is to be processed. |

| Status code | Meaning |
|---|---|
| 503 | The query cannot be processed by the receiving device or the proxy at present 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. |
| 504 | Time limit exceeded at the gateway. |
| 505 | The server rejects the query because the indicated version number of the SIP protocol does not concur with at least the version that is used by the server or SIP device involved in this query. |
| 515 | The server rejects the query because the message exceeds the maximum permitted size. |
| 600 | The called party is busy. |
| 603 | The called party has rejected the call. |
| 604 | The called URI does not exist. |
| 606 | The communication settings are not acceptable. |
| 701 | The called party has hung up. |
| 702 | VoIP socket error |
| 703 | Connection cancelled because of timeout. |
| 704 | Connection interrupted because of a SIP error. |
| 705 | SIP memory error. |
| 706 | SIP transaction memory error. |
| 751 | Busy tone:<br>No codec match between the calling and called party. |
| 810 | General socket layer error. |
| 811 | General socket layer error: Wrong socket number |
| 812 | General socket layer error: Socket is not connected. |
| 813 | General socket layer error: Memory error |
| 814 | General socket layer error: Socket not available – check IP settings/connection problem/VoIP setting incorrect. |
| 815 | General socket layer error:<br>Illegal application on the socket interface. |
| 922 | No DNS server known. |
| 923 | DNS name resolution failed. |
| 924 | Insufficient resources for DNS name resolution. |
| 925 | URL error. |

# Deleted function: "Send line selection for outgoing calls with */#"

This section relates to the selection of default or non-default connections by adding # or * to the dialled number.

If your purchased telephone came with firmware version 02.140 or later already installed (Manufactured after May 2009), this function is not available. With these devices, it is no longer possible to select the non-default connection by adding an asterisk (*) to the dialled number or to select the default connection by adding a hash symbol (#).

However, you can still use the line suffix to select the send line when dialling. If you add #0 to the number, it is dialled via the fixed line network. If you add #1, #2, ..., #6, the number is dialled via the corresponding VoIP connection. Further information about this can be found in the operating instructions for your telephone.

## Dialling with the quick dial keys

If you have assigned a phone number to a number key on the handset as a quick dial number, it is dialled via the default connection if no line suffix is specified. Exception:
A dialling plan has been defined for the number

# Handset menu overviews

## Gigaset S67H to Gigaset S675 IP, Gigaset S68H to Gigaset S685 IP

New and updated menus and submenus are marked in **orange**.

Please note that a few digit combinations (shortcuts) for quick entry to the submenus have also changed. They are also marked in **orange**.

**1 Messaging**

| | | | | | |
|---|---|---|---|---|---|
| 1-1 | SMS | **An SMS mailbox (general or private) activated without a PIN** | | | |
| | | 1-1-1 | New SMS | | |
| | | 1-1-2 | Incoming (0) | | |
| | | 1-1-3 | Draft (0) | | |
| | | **An SMS mailbox activated with a PIN or 2-3 mailboxes** | | | |
| | | 1-1-1 | Mailbox | 1-1-1-1 | New SMS |
| | | | | 1-1-1-2 | Incoming (0) |
| | | | | 1-1-1-3 | Draft (0) |
| | | 1-1-2<br>to<br>1-1-4 | Mailbox 1<br>Mailbox 2<br>Mailbox 3 | 1-1-2-1<br>to<br>1-1-4-1 | New SMS |
| | | | | 1-1-2-2<br>to<br>1-1-4-2 | Incoming (0) |
| | | | | 1-1-2-3<br>to<br>1-1-4-3 | Draft (0) |
| | | 1-1-6 | Settings | 1-1-6-1 | Service Centres |
| | | | | 1-1-6-2 | SMS Mailboxes |
| | | | | 1-1-6-3 | Notify Number |
| | | | | 1-1-6-4 | Notify Type |
| 1-2 | E-mail | | | | |

| | | | | | | |
|---|---|---|---|---|---|---|
| 1-3 | Messenger | 1-3-1 | Buddies | | | |
| | | 1-3-2 | User Status | 1-3-2-1 | Change Status | |
| | | | | 1-3-2-2 | Info | |
| | | 1-3-3 | Messages | | | |

**2 Sel. Services**

| | | | | |
|---|---|---|---|---|
| 2-1 | Info Center | → Page 8 | | |
| 2-2 | VoIP | 2-2-6 | Call Diversion | |
| | | 2-2-7 | Call Waiting | |
| 2-3 | Fixed Line | 2-3-6 | Call Diversion | *) Menu item **Withhold No.** is no longer available. It is replaced by 2-6 **Next Call**. |
| | | 2-3-7 | Call Waiting | |
| 2-4 | Ringback Off | | | |
| 2-5 | Always anon. | | | |
| 2-6 | Next Call | | | |

**3 Calls List**

**4 Add. Features**

| | | | | |
|---|---|---|---|---|
| 4-3 | Room Monitor | | | |
| 4-4 | Data Transfer | 4-4-2 | Bluetooth | Only for Gigaset S68H |
| | | 4-4-3 | Directory | |
| 4-6 | Missed Appts. | | | |

**5 Alarm Clock**

**6 Calendar**

**7 Resource Dir.**

| | | |
|---|---|---|
| 7-1 | Screensavers | |
| 7-2 | Caller Pictures | |
| 7-3 | Sounds | |
| 7-4 | Capacity | Only for Gigaset S68H |

## 8 Settings

| | | | | | |
|---|---|---|---|---|---|
| 8-1 | Date/Time | | | | |
| 8-2 | Audio Settings | 8-2-1 | Handset Volume | | |
| | | 8-2-2 | Ringer Settings | 8-2-2-1 | Ext. Calls |
| | | | | 8-2-2-2 | Internal Calls |
| | | | | 8-2-2-3 | Appointments |
| | | | | 8-2-2-4 | All |
| | | 8-2-3 | Advisory Tones | | |
| 8-3 | Display | 8-3-1 | Screen Saver | | |
| | | 8-3-2 | Colour Scheme | | |
| | | 8-3-3 | Contrast | | |
| | | 8-3-4 | Backlight | | |
| 8-4 | Handset | 8-4-1 | Language | | |
| | | 8-4-2 | Auto Answer | | |
| | | 8-4-3 | Register H/Set | | |
| | | 8-4-4 | Select Base | | |
| | | 8-4-5 | Area Codes | | |
| | | 8-4-6 | Reset Handset | | |
| 8-5 | Base | 8-5-1 | Calls List Type | 8-5-1-1 | Missed Calls |
| | | | | 8-5-1-2 | All Calls |
| | | 8-5-2 | Music on hold | | |
| | | 8-5-3 | System PIN | | |
| | | 8-5-4 | Base Reset | | |
| | | 8-5-5 | Add. Features | 8-5-5-1 | Repeater Mode |
| | | | | 8-5-5-2 | Access Code → Page 18 |
| | | | | 8-5-5-3 | Eco Mode |
| | | 8-5-6 | Local Network | | |
| | | 8-5-8 | Software Update | | |

| | | | | | |
|---|---|---|---|---|---|
| 8-6 | Voice Mail | 8-6-1 | Local AM | 8-6-1-1 | Ans Machine |
| | | | | 8-6-1-2 | Call Screening |
| | | | | 8-6-1-3 | Announcements |
| | | | | 8-6-1-4 | Message Length |
| | | | | 8-6-1-5 | Recording Quality |
| | | | | 8-6-1-6 | Ring Delay |
| | | 8-6-2 | Network AM(s) | 8-6-2-1 | Net AM Fxd. ln. |
| | | | | 8-6-2-2 | Net AM IP1 |
| | | | | : (dependent on the number of configured VoIP phone numbers and receive numbers on the handset) | |
| | | | | 8-6-2-7 | Net AM IP6 |
| | | 8-6-3 | Set Key 1 | Local AM | |
| | | | | Net AM Fxd. ln. | |
| | | | | Net AM IP1 | |
| | | | | : (dependent on the number of configured VoIP phone numbers and receive numbers on the handset) | |
| | | | | Net AM IP6 | |
| 8-7 | Telephony | 8-7-1 | Default Line | 8-7-1-1 | VoIP |
| | | | | 8-7-1-2 | Fixed Line |
| | | 8-7-2 | Connection Assist. | | |
| | | 8-7-6 | Fixed Line | 8-7-6-1 | Dialling Mode |
| | | | | 8-7-6-2 | Recall |
| | | 8-7-7 | VoIP | Enter system PIN | Show Stat. on HS |
| | | | | | Select Provider |
| | | | | | Provider Registr. |

# Gigaset C47H to Gigaset C475 IP

New and updated menus and submenus are marked in **orange**.

Please note that a few digit combinations (shortcuts) for quick entry to the submenus have also changed. They are also marked in **orange**.

1 ✉ **Messaging**

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 1-1 | SMS | **An SMS mailbox (general or private) activated without a PIN** | | | | | |
| | | 1-1-1 | New SMS | | | | |
| | | 1-1-2 | Incoming (0) | | | | |
| | | 1-1-3 | Draft (0) | | | | |
| | | **An SMS mailbox activated with a PIN or 2-3 mailboxes** | | | | | |
| | | 1-1-1 | Mailbox | 1-1-1-1 | New SMS | | |
| | | | | 1-1-1-2 | Incoming (0) | | |
| | | | | 1-1-1-3 | Draft (0) | | |
| | | 1-1-2 to 1-1-4 | Mailbox 1 Mailbox 2 Mailbox 3 | 1-1-2-1 to 1-1-4-1 | New SMS | | |
| | | | | 1-1-2-2 to 1-1-4-2 | Incoming (0) | | |
| | | | | 1-1-2-3 to 1-1-4-3 | Draft (0) | | |
| | | 1-1-6 | Settings | 1-1-6-1 | Service Centres | | |
| | | | | 1-1-6-2 | SMS Mailboxes | | |
| | | | | 1-1-6-3 | Notify Number | | |
| | | | | 1-1-6-4 | Notify Type | | |

| | | | | | |
|---|---|---|---|---|---|
| 1-2 | E-mail | | | | |
| 1-3 | Messenger | 1-3-1 | Buddies | | |
| | | 1-3-2 | User Status | 1-3-2-1 | Change Status |
| | | | | 1-3-2-2 | Info |
| | | 1-3-3 | Messages | | |

**2 Sel. Services**

| | | | |
|---|---|---|---|
| 2-1 | Info Center | | |
| 2-2 | VoIP | 2-2-6 | Call Diversion |
| | | 2-2-7 | Call Waiting |
| 2-3 | Fixed Line | 2-3-6 | Call Diversion |
| | | 2-3-7 | Call Waiting |
| 2-4 | Ringback Off | | |
| 2-5 | Always anon. | | |
| 2-6 | Next Call | | |

→ Page 8

*) Menu item **Withhold No.** is no longer available. It is replaced by 2-6 **Next Call**.

**3 Alarm Clock**

**4 Add. Features**

| | |
|---|---|
| 4-3 | Room Monitor |

## 5 Settings

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 5-1 | Date/Time | | | | | | |
| 5-2 | Audio Settings | 5-2-1 | Handset Volume | | | | |
| | | 5-2-2 | Ringer Settings | 5-2-2-1 | Ext. Calls | | |
| | | | | 5-2-2-2 | Internal Calls | | |
| | | | | 5-2-2-3 | All | | |
| | | 5-2-3 | Advisory Tones | | | | |
| 5-3 | Display | 5-3-1 | Screen Picture | | | | |
| | | 5-3-2 | Colour Scheme | | | | |
| | | 5-3-3 | Contrast | | | | |
| | | 5-3-4 | Backlight | | | | |
| 5-4 | Handset | 5-4-1 | Language | | | | |
| | | 5-4-2 | Auto Answer | | | | |
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| | | | | | |
|---|---|---|---|---|---|
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