

# Gigaset

## T640 PRO - T440 PRO

### Administration



# Contents

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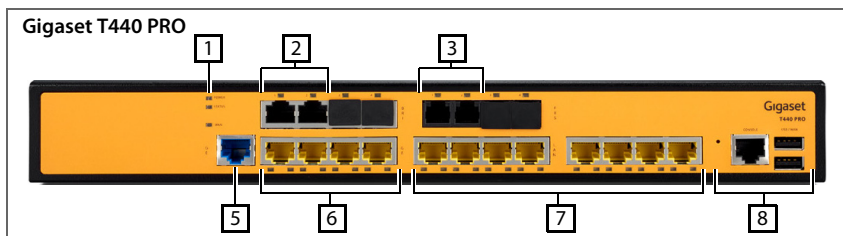
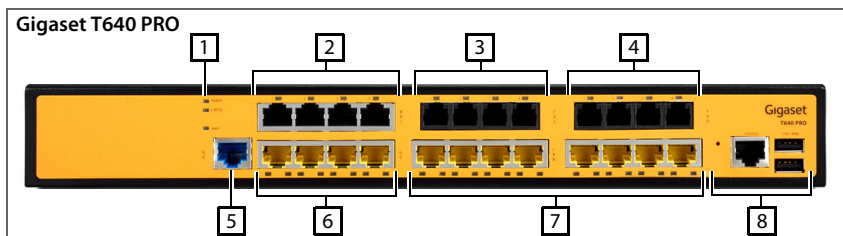
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# Overview

## Connections



1	POWER STATUS	LEDs for displaying the status
2	BRI	BRI ports for ISDN telephony (→ p. 8); Gigaset T640 PRO: 4 x; Gigaset T440 PRO: 2 x
3	FXS	Ports for connecting analogue devices (phone/fax) (→ p. 7); Gigaset T640 PRO: 4 x; Gigaset T440 PRO: 2 x
4	FXO	Connection to the analogue phone network (→ p. 7); Gigaset T640 PRO: 4 x
5	GE	Ethernet RJ-45 port (10/100/1000Base-T), currently not used
6	GE	4 x Gigabit Ethernet LAN ports (10/100/1000Base-T) for connecting IP phones, computers and IP switches (→ p. 7).
7	LAN	8 x Fast Ethernet LAN ports (10/100Base-TX) for connecting IP phones, computers and IP switches (→ p. 7).
8		Currently not used

Information on the current status of ports and interfaces → p. 74.



All LAN ports support PoE (Power over Ethernet) as per the IEEE 802.3af-2003 standard

## LEDs


LEDs on the front:

LED	Colour	State	Description
STATUS	Green	On	Device being used
		Flashing	Device being restarted
	Red	On	Fault on restart
POWER	Green	On	Power supply available
	–	Off	No power supply

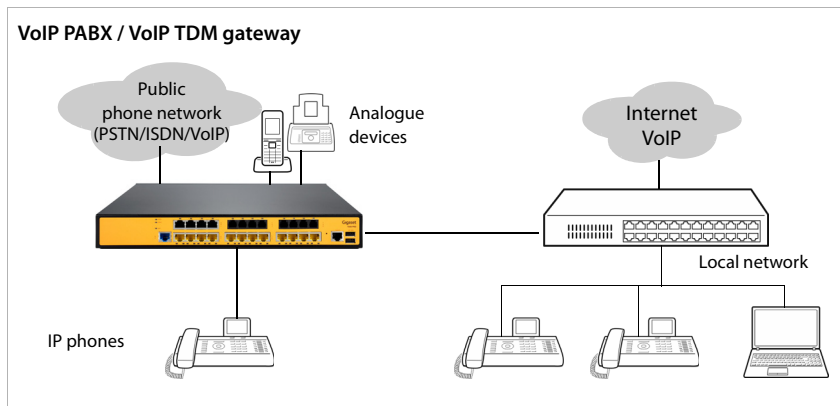
LED on port	Colour	State	Description
LAN on right (Ethernet)	Green	On	Ethernet connection established
		Flashing	Port sending data
	–	Off	No Ethernet connection
LAN on left (PoE)	Yellow	On	The LAN port is powering the device connected (e.g. an IP phone)
		Flashing Fast	Line overload or short-circuit detected – PoE not enabled
		Flashing Slow	PoE loading on port, but the device has insufficient current to provide the power required – PoE not enabled
	–	Off	No current on the output line – PoE not enabled
FXS	Green	On	The phone connected is busy
		Flashing	Extension being called
	Red	On	Fault – not operational due to connection fault or SPI fault (SPI=Serial Peripheral Interface)
	–	Off	Phone receiver down or device without power
FXO	Green	On	FXO line is busy to the PABX
		Flashing	PABX signalling
	Red	On	Fault – not operational due to connection fault or SPI fault
	–	Off	Phone receiver down or device without power
BRI	Green	On	Physical Layer (Layer 1) synchronised (normal mode)
	Red	On	Physical Layer (Layer 1) not synchronised
	–	Off	Trunk group not active

## Back



<b>1</b>	USB	USB ports for connecting peripherals for the server (e.g. mouse and keyboard)
<b>2</b>	VGA	VGA port for connecting a screen for the server
<b>3</b>	GE1/GE 2	Currently not used
<b>4</b>		Earthing screw ( → p. 8)
<b>5</b>	100-240V~1.5A 50-60Hz	Power connector ( → p. 8)
<b>6</b>		250V fuse

## Possible scenarios



## Setting up and connecting a device



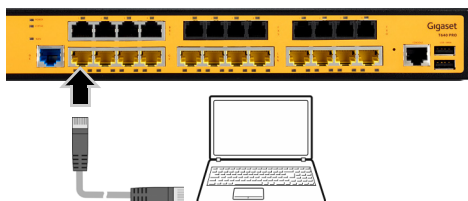
Observe the safety instructions and the information on the place of installation in the manufacturer documents provided.

The **installation instructions** supplied provide information on initial use of the PABX. Keep to the sequence described for initial use.

## Connecting

### Connect a computer

For the initial configuration, a computer needs to be connected directly to a LAN port on the device.



Straight-through Ethernet cable  
(not supplied)

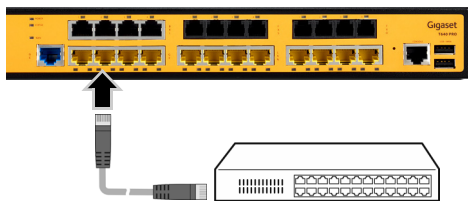
- ▶ Connect the network connector of the computer directly to the LAN port

### Establish a LAN connection

Use a LAN port to connect the PABX to the local network.



Remember that beforehand you must align the network configuration of the PABX to the settings of your network → p. 71.



CAT 5e or CAT 6 Ethernet cable  
(not supplied)

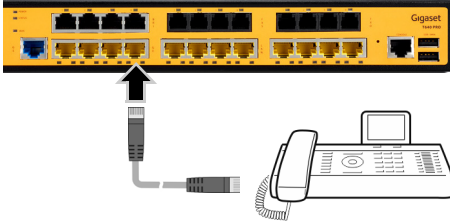
- ▶ Connect the LAN port (GE or FE) to a Gigabit Ethernet or Fast Ethernet network, e.g. using a port on a switch



## Connecting PoE clients to LAN ports

The device makes available 4 Gigabit Ethernet LAN ports (10/100/1000Base-T) and 8 Fast Ethernet LAN ports (10/100Base-TX) to connect Ethernet devices (such as IP phones).

The LAN ports support IEEE PoE standard 802.3af-2003. In addition to transmitting the other data, the connectors can power PoE-capable devices connected over the Ethernet cable. The LAN connectors automatically detect devices supporting the IEEE 802.3 standard, the device classification and the maximum power permitted.



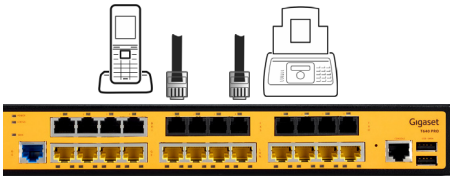
CAT 5e or CAT 6 Ethernet cable (not supplied)

- ▶ Connect the GE or FE LAN port to the network connection of an IP phone
- or
- ▶ Connect the IP phone directly to the Ethernet network

## Connecting analogue devices

Analogue phones, dial-up modems and fax devices can be connected to the FXS ports. These analogue devices can then be used for Internet telephony.

An FXS port supplies the line voltage and ringing current for the phones.



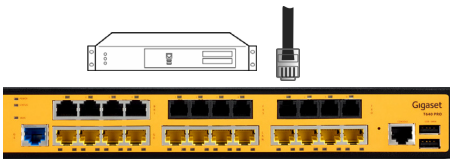
RJ11 phone cable (not supplied)

- ▶ Connect the FXS port to an analogue device (e.g. phone, dial-up modem or fax device)

## Connecting to the phone network or PABX

FXO ports are only available for the Gigaset T640 PRO.

An FXO port establishes the connection to the public phone network (PSTN) or an analogue PABX. An FXO port receives the line voltage and ringing current for the phones from the phone network or PABX (as with analogue phones). An FXO port is the interface between the analogue phone network or system and the Internet.

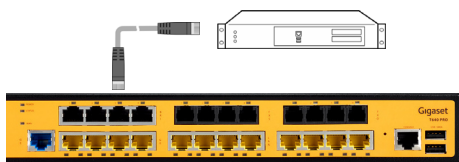


RJ11 phone cable (not supplied)

- ▶ Connect an FXO port to an analogue phone port (e.g. the phone network connector or PBX)

### Connecting ISDN lines

The BRI ports (Basic Rate Interface) are used to connect ISDN PABXs. Every BRI port can be configured as an end device connector (TE) (→ p. 47).



RJ11 phone cable (not supplied)

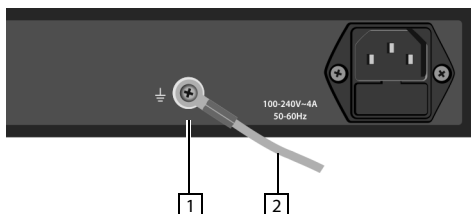
- ▶ Connect the BRI RJ-45 port to an ISDN device



A cable with a minimum rating of 0.14 A/mm is required for BRI port connections to the phone network to provide protection against electric shock and fire.

### Affixing the earthing cable

The device must be earthed using a standard earthing cable (1.5 A/mm as a minimum).



Earthing cable: 1.5 A/mm minimum (not supplied)

- ▶ Undo the earthing screw **1** on the rear of the housing
- ▶ Attach the earthing cable **2** to the earthing screw. Connect the other end of the earthing cable to a protective earth conductor.



The devices are classified as Class I EN60950 and UL60950 and must be permanently earthed.

### Connecting the power



**Only use the power cable supplied.**

- ▶ Connect the device to the mains power

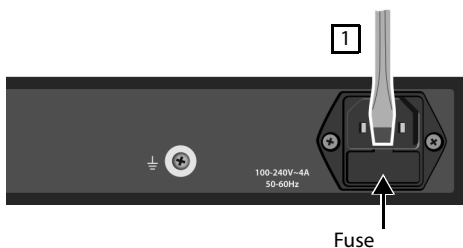
As soon as power is connected up, start the system. The **POWER LED** on the front lights up.

## Replacing the fuse

The device has a fuse for overvoltage protection. It is located below the power connector on the rear.



Only use replacement fuses of the same type and having the same rating.

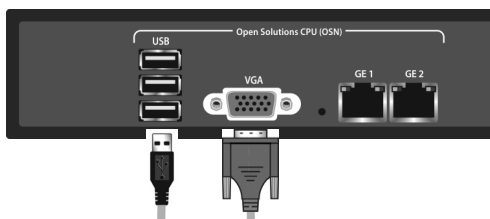


- ▶ Remove the mains connector from the device
- ▶ Use a small flat screwdriver to carefully open the fuse housing 1. ▶ Remove the fuse.
- ▶ Insert a new fuse into the housing and click it into place
- ▶ Connect to the power again and check the Power LED lights green

## Connecting peripherals to the server ports

(only for experts)

The Gigaset PABX software runs on a server integrated in the device. Normally no direct access to the server is necessary. All configuration and administration functions can be performed from the web interface. The connectors required are available on the back of the device should direct access be required (such as for a reinstallation or fault analysis). Computer peripherals (such as mouse, keyboard and monitor) can be connected here.



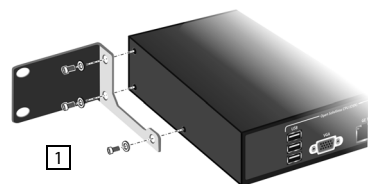
- ▶ **Connect a computer peripheral:** ▶ Plug a USB cable into one of the USB ports (type Standard-A) and connect it to the peripheral device.
- ▶ **Connect a monitor:** ▶ Use a VGA cable with a male connector (15-pin D-type) to connect the monitor to the VGA port.

## Installation

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### Installation in a server cabinet

The device is intended for installation in a server cabinet. Use the installation adapters supplied. Installation width: 19", height: 1 HE



- ▶ Attach the installation adapters to both sides of the housing [1]. Use the screws supplied.
- ▶ Position the device on a pre-installed shelf in the server cabinet



- ▶ Secure the ends of the installation adapters [2] to the vertical carriers of the server cabinet frame. Use the standard screws for server cabinets (not supplied).

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### Installation without server cabinet

If you are not installing the PABX in a server cabinet:

- ▶ To install the device so as to be non-slip, affix the rubber knobs to the underside of the housing

## Base configuration

\_product\_name PABXs feature comprehensive options for the use, administration and configuration of your phones.

This manual describes the **administration** for both devices.



The user instructions provide information on the phone functions and an introduction to the user interface.

For administration, the PABX features a web interface from which, using a browser on any computer, you can access your network.

**Prerequisite:** You have integrated the PABX into your local network as described in the installation instructions.



The PABX must be integrated into the local network infrastructure of the company. If, on delivery, the network configuration does not match your network environment, carry out a base configuration from a computer connected directly to the PABX (→ p. 6).

## Logging in

To log into the user interface, you need the IP address of the PABX and a user name with PIN.



IP address on delivery: 192.168.0.50  
Predefined user name for the Administrator:  
User name = **admin**, PIN = **0000**

- ▶ Open a standard browser on a computer
- ▶ In the address field, enter the IP address of the device . . . the login screen is displayed (**Login**)
- ▶ Enter the user name (**User name**) and associated PIN

### Installation Wizard

An Installation Wizard starts automatically on initial login to the user interface. This guides you step-by-step through important settings.

- ▶ Set language for Wizard ▶ **Next**
- ▶ A screen follows enabling you to load a backup  
On initial use: ▶ Skip screen with **Next**  
To restore the configuration from a backup file: ▶ Click **Yes** ▶ Select backup file ▶ **Next**



The configuration of the PABX must be stored in a backup file: ▶ **Administration** ▶ **System** ▶ **Backup** → p. 72

- ▶ Select country ▶ **Next** ... Country-specific settings are loaded, such as dial tone and ISDN & FXS/FXO parameters
- ▶ Change the PIN for Administrator ID **admin** ▶ **Next**

You can now exit the Wizard and configure the other settings later from the Administrator menu of the interface.



Running the Wizard through to the end is recommended however to connect the PABX to the local network and to get an exchange line to work successfully.  
Configure other settings with the Wizard: ▶ **Next**

Now end the Wizard: ▶ **Finish installation and reboot** ... The PABX is restarted. You can now log in with user name **admin** with the new PIN, and configure other settings.

### Other settings

#### Changing the network configuration

→ p. 68

- ▶ Enter the IP address of the PABX in the local network. The PABX requires a fixed IP address. This may have to be included in the configuration of the DHCP server in the network.
- ▶ Align to the network the settings for the subnetwork mask, the standard gateway (DHCP server) and the DNS servers

#### Disabling the DHCP server

→ p. 69

The PABX features an integrated DHCP server (this is enabled on delivery). If another DHCP server is active in the network, the PABX's DHCP server must be disabled.

- ▶ Disable the DHCP server using switch **Enable DHCP Server**

#### Configuring the email server

→ p. 70

An external email server must be set up to be able to send emails.

- ▶ Enable the sending of emails with switch **Email delivery** and enter the details for the SMTP server
- ▶ Or skip the step with **Next** to configure email later

### Configuring an outside line

→ p. 45

Configure an outside line. The PABX supports SIP, ISDN and analogue connections. Access details are available from your telephone provider.

- ▶ Select the type of outside line (SIP, ISDN or FXO) from the list next to **Outside line**, and enter the connection details

The connection configured is entered into the configuration as a SIP or TDM gateway. A gateway group for the connection is set up automatically. All incoming and outgoing connections are routed via this gateway group.

Later, you can set up other gateways and gateway groups from the Administrator menu, and align routing to the needs of the company.

### Entering users

→ p. 19

- ▶ Enter three users with **First name**, **Last name** and **Extension** for the users ... The users are created with user names **demo101**, **demo102**, **demo103** and PIN **0000**

### Ending the installation Wizard

You can re-check the settings on the last screen of the Wizard.

Change settings: ▶ Select tab ▶ Change settings as required ▶ Run Wizard again to end



Confirm settings: ▶ **Submit settings** ... The PABX is restarted



The network settings configured may mean you now have no access to the interface from the computer. ▶ Restore the connection with a new IP address as required

## Licensing

To use the PABX, the license made available on purchasing the system must be activated.

- ▶  **Administration** ▶ **System** ▶ **License**
- ▶ Click  and select the license file from your file system ▶ **Upload**

The system is now enabled.



Information on installing other licences is available from <http://wiki.gigasetpro.com> or contact your Gigaset partner.

## User interface

Once the basic configuration is complete and login is successful, the **Home** screen of the interface is opened.



Change PIN: → Personal profile, p. 15.

Detailed information on **Home** screen and **Menu**:

→ User instructions

### Logging out

Logging out of the user interface: ▶ Profile ▶ Logout

### Controls

Switch:  ON  OFF Enable/disable function

In lists: Edit entry Delete entry

Backup: Add and save entry

Help: Access online help

Request help information for this screen

### Screen navigation

Switch screen: ▶ Select required function from menu

Save changes: ▶ Click button **Save** (button is only active when changes need to be saved)

Exit screen without changes: ▶ Click button **Cancel**

### Filtering lists

A lot of information is shown in list form. You can filter lists to reduce the number of entries shown and to search for certain entries.

#### Alphabetic filter

A bar with the alphabet is shown above lists which can be filtered by alphabetic values.



▶ Click the letters in the ABC bar.

Only entries starting with the letter selected are displayed. Which field/fields is/are used for the filter is dependent on the list.


The user list for example is only filtered by name, and the contact list by name and first name. Clicking **A** in the user list shows the list of all users whose name starts with A. Clicking **A** in the contact list shows all contacts whose first or last name starts with A.





## Name/number filter

Different search fields are available depending on the list type to search for single or multiple entries, such as by **Name** or **Number** in a contact list, or by **MAC address** or **IP address** in the phone list.



- ▶ Enter one or more letters/digits in a search field ▶ Click  ... Only entries starting with what is entered in the search field are displayed

## Delete filter

- ▶ Click  ... The filter is deleted ▶ Click  ... The list is updated

## Personal profile

A personal profile containing the following information is set up for every user:

- **First name**, **Last name** and **E-mail address** as per user entry
- The **Extension** assigned to the user
- User interface language
- Any personal phone numbers

## Checking/adding personal details


- ▶  **Profile** ▶ **Personal data**

## Adding personal phone numbers

- ▶ Enter the phone numbers in fields **Mobile** and **Home** ▶ **Save** ... The numbers are transferred to your entry in the internal directory (→ p. 22)

## Add an image to be displayed as the caller image (CLIP image)

Formats: PNG, GIF, TIFF and JPG

- ▶ Click  ▶ Select the image from the file system on the computer or network ... The file name is entered in the text field ▶ **Save** ... The image is loaded and displayed

**Delete image:** ▶ Click  ▶ Confirm with **OK**

## Changing the PIN

- ▶  **Profile** ▶ **Change PIN** ▶ Enter current PIN ▶ Enter new PIN ▶ Repeat new PIN ▶ **Save**

## Changing the language


- ▶  **Profile** ▶ **Change language** ▶ Select the language required ▶ **Save**



The change of language also applies for the language settings on the phone.

## Pretending to be another user

Here you can change settings for users and check them when faults occur.

▶  Profile ▶ Impersonate ▶ Enter the user name you want to use as an alternative ▶ Save

## Administration menu overview

Users & extensions	Users & groups	Users	→ p. 19	
		Pickup groups	→ p. 22	
		Hunt groups	→ p. 23	
	Permissions	Permission groups	→ p. 25	
		GUI	→ p. 28	
	Global contacts	Contact list	→ p. 30	
		CSV import/export	→ p. 31	
	Queues		→ p. 33	
	IVR		→ p. 36	
	Audio files		→ p. 38	
	Hold music		→ p. 39	
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		Update		→ p. 65
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		Key profiles	→ p. 40	
		Provisioning parameters	→ p. 40	
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		TDM Gateways		
			FXS ports	→ p. 53
			FXO ports	→ p. 53
			BRI ports	→ p. 53
		Inbound routes	Inbound routes	→ p. 57
			Night answer service	→ p. 53
		Outbound routes		→ p. 59
		Call forward	Queues	→ p. 62
			Hunt groups	→ p. 63
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	Interfaces		→ p. 74	
	SIP-Status		→ p. 75	
		Diagnostics		
			System log	→ p. 75
			Telephony	→ p. 75
			Operating system	→ p. 75
		Intrusion detection	→ p. 75	
	Reboot & shutdown		→ p. 76	

## Users, groups and extensions

From the viewpoint of the PABX, a user name is assigned to an extension. A user is able to phone using the PABX when the user name is assigned an extension and the phone (with this extension) is registered with the PABX. If a user has multiple phones (e.g. a desktop phone and a DECT handset), an ID must be set up for every extension.

The PABX\_product\_name is delivered with a predefined user name for the Administrator. User name = **admin**, PIN = **0000**.

During set-up with the installation Wizard, three more user names (demo101 - demo103) are set up (→ p. 12) - which you can change or delete for internal purposes.

You set up additional user names for PABX users:

- ◆ Gigaset T440 PRO: up to 40 users
- ◆ Gigaset T640 PRO: up to 80 users

Users can be assigned to different groups in line with their functions:

- ◆ Pickup groups: Members can accept the calls of other group members (→ p. 22)
- ◆ Hunt groups: All members are reachable on the same extension number (→ p. 23)
- ◆ Queue: Callers are kept in a queue and transferred to group members in line with definable rules (→ p. 33)
- ◆ Authorisation group: Members have access to a certain definable subset of user interface functions (→ p. 25)
- ◆ Provisioning group: The phones are assigned certain key profiles or special provisioning parameters (→ p. 44)



User names can only be changed or added once the license is activated successfully (→ p. 13)

## Managing users

- Administration ▶ Users & extensions ▶ Users & groups ▶ Users

Existing users are listed with their login name, correct name, extension and email address. Users registered with the PABX with their extension are denoted by a green dot ●.

Name 
Number

A B C D E F G H I J K L M N O P Q R S T U V W X Y Z

User <span style="font-size: 0.8em;">▼ ▲</span>	Name <span style="font-size: 0.8em;">▼ ▲</span>	Extension <span style="font-size: 0.8em;">▼ ▲</span>	E-mail <span style="font-size: 0.8em;">▼ ▲</span>		
● admin	Administrator	999999	admin@org.com	<input type="button" value="✎"/>	<input type="button" value="🗑"/>
● demo101	Martin Black	101	martin.black@org.com	<input type="button" value="✎"/>	<input type="button" value="🗑"/>
● demo102	Susan Brown	102	susan.brown@org.com	<input type="button" value="✎"/>	<input type="button" value="🗑"/>

### Filtering lists

- In fields **Name** and/or **Number**, enter a value ▶ Click  ... Only entries starting with what is entered are displayed

Value: one or more letters and/or digits

You can filter the list by **Name** and **Number** at the same time.

Example: **Name** = Ab, **Number** = 1; the filter returns all entries whose name starts with "Ab" and extension number with "1".



Parameters **First name** and **Last name** of the user entry are used for the filter name, i.e. filter "A" returns all entries whose first or last name starts with "A".

**Delete filter:** ▶ Click  in the field ... The value is cleared ▶ Click  ... The list is updated

### Sort list

- Click the arrows ▼ ▲ in a column heading ... The list is sorted in ascending or descending order



The **Name** column is sorted by last name.


### Delete list

- ▶ Click  ... The entry is deleted



The preconfigured **admin** user name cannot be deleted.

### Setting up new users / changing settings

- ▶ Set up new user: Click **Add user** (top right on screen)
- ▶ Change entry: Click  next to the entry

### Parameter

<b>User</b>	User name for logging into the user interface. It must be unique. Value: 2 - 50 alphanumeric chars. (lowercase letter and digits).
<b>Extension</b>	Extension number. Value: 2 - 10 digits An extension has a fixed assignment to a user. The value must therefore be unique. To use the extension on a phone, the user must enable the extension on this phone.
<b>First name / Last name</b>	First and last names of the user.
<b>PIN</b>	PIN for logging into the user interface. Value: 3 - 10 digits The user can change the PIN from the profile settings.
<b>SIP password</b>	Password for the VoIP account assigned to the user.
<b>Voicemail box</b>	Extension of the answering machine assigned to the user. The default value for this number is as displayed in field <b>Extension</b> . To activate the answering machine, the user must set up call divert to the answering machine. If the user wants to hear multiple answering machines: Enter the answering machines numbers separated by commas.
<b>E-mail</b>	Email address of user. The email address is used to provide notifications of arriving answering machine messages and for Fax2Mail. Fax2Mail is always enabled; notifications of answering machine messages can be set by the user ( <b>Menu ▶ Call Forwarding ▶ Call Forwarding ▶ Email notification on new voice messages</b> ).
<b>Language</b>	The setting determines the language for the phone, system announcements and the PABX user interface for the user. This setting can be changed by the user.

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**CLIP internal / CLIP external**

Call numbers of the user for the caller display (CLIP). The call numbers entered here are available to the user as a selection in menu **Service attributes**.

Any number of call numbers can be entered for internal and external. It must be possible to call the numbers entered.

**CLIP internal:**

Extension of the user or another extension to be displayed for internal calls.

**CLIP external:**

Numbers for outgoing external calls.



The numbers can only be entered on changing of the user settings, not for a new entry.

---

**Hide from phonebook**

Activated: The user is not entered into the internal directory of the PABX.

---

**Drop to operator**

Activated: Incoming calls for the user extension are diverted to a central number when not answered. ▶ Enter number in field **Operator extension**.

---

**Update Number (CLIP)**

Activated: The number is updated on a status change.

Example: The user has activated call divert. Caller A to the number of B is diverted to number C. The number of A is displayed at C.

---

**Reroute RTP stream**

Keep default (not activated) (→ p. 51).

---

**Provisioning group**

Assignment of the extension to a provisioning group (→ p. 44). All provisioning groups set up are made available.

▶ Select required group from the list

---

**Pickup groups**


Assignment of the extension to pickup groups (→ p. 22). All pickup groups set up are made available.


▶ Select the group(s) required

---

**User groups**

Assignment of the user to authorisation groups (→ p. 25). All authorisation groups of type **User** are made available.

Assign user: ▶ Click 


Remove user from group: ▶ Click 

Assignments to **All Users** and **All visible** users cannot be deleted.

---

**Image**

Image shown as a CLIP image.

▶ Click  ▶ Select the image from the file system on the computer or network ... The file name is entered in the text field ▶ **Save** ... The image is loaded and displayed

**Formats:** PNG, GIF, TIFF and JPG

---



Only **User**, **Extension** and **PIN** are mandatory parameters. All other settings are optional and can be added later.

The user is included in the internal contact list (**Menu ▶ Contacts ▶ Internal**) provided this is not explicitly inhibited with parameter **Hide from phonebook**.

Users must register their phones with the PABX (together with the extension assigned) as follows:

- ▶ On the phone, press buttons   <Extension> ▶ Press the hands-free key  ▶ Enter the PIN ▶ Confirm with

## Group management

Your Gigaset PABX features pickup groups and hunt groups to handle calls which can be accepted by more than one person.

### Pickup groups

Call pickup enables a user to accept a call for another subscriber. For this, the user presses a button on the phone which is specially reserved for this purpose and assigned the "Group pickup" function. Users belonging to the same call pickup group can accept calls for every group member.

- ▶ **Administration ▶ Users & extensions ▶ Users & groups ▶ Pickup groups**

Pickup groups already set up are listed with name and the number of its members.

Pickup Groups				
ID	Group	Members		
1	Accounting	1		
2	Sales	0		


Group

### Setting up a new pickup group

- ▶ Enter the name for the group in field **Group** ▶ Click ... The group is entered without members



## Adding/removing group members

▶ Click  next to the group entry ... The users assigned to the group are listed

**Adding a user:** ▶ Select an entry from the list **User** ▶ Click 

**Removing a user:** ▶ Click  next to the user entry



You can also assign a user to a call pickup group by editing the relevant user entry and enabling the checkbox next to the group in area **Pickup groups** (→ p. 19).

Users must belong to a user group with authorisation **Group pickup** (→ p. 25). Group **All Users** has this authorisation as standard.







For call pickup, a button profile must be used to assign a button on the phone with parameter **Group pickup** (→ p. 41). Enter the group ID of the pickup group in field **Number/Data**.


## Hunt groups

A hunt group bundles multiple numbers (extensions), which can then be called using one number. Incoming calls for the number of a hunt group are connected through directly to all the extensions.

▶  **Administration** ▶ **Users & extensions** ▶ **Users & groups** ▶ **Hunt groups**

Hunt groups already set up are displayed with group number (extension), name and other settings.


Edit Hunt Groups							
Extension	Title	Display prefix	Call scheme	Group busy	Members		
100	Hotline	14646	linear	no	2		
101	Reception	12345	parallel	yes	0		

Extension  Title  Add prefix  Call scheme  Group busy  


### Setting up a hunt group

#### Parameter:


<b>Extension</b>	Hunt group extension. Incoming calls to this extension are put through to all extensions of group members.
<b>Title</b>	Group name
<b>Display prefix</b>	Is displayed on the phone to indicate the call is arriving over the hunt group.
<b>Call scheme</b>	<b>linear</b> The extensions of group members ring in succession. Calls are forwarded to the first extension. If this line is busy or the phone is not picked up within a specified time, the other extensions are called in succession. <b>parallel</b> All extensions ring at the same time. As soon as one extension user picks up, the parallel ringing stops.
<b>Group busy</b>	<b>Activated:</b> If one extension in the group is busy, the call is not forwarded to the next. This makes sense for example when the hunt group comprises a phone and a mobile device used by the same user. If the user is talking on one of the phones, this call is not interrupted. <b>Not activated:</b> If one extension in the group is busy, the call is forwarded to the next free one.
<b>Members</b>	Number of group members

- ▶ Enter the parameters for the group ▶ Click  ... The group is set up

#### Adding/removing group members


- ▶ Click  next to the group entry ... The users assigned to the group are listed

#### Adding a user

- ▶ Select an entry from list **User** ▶ Enter the value for **Timeout** ▶ Click  ... The user is assigned to the group, the user extension belongs to the hunt group

**Timeout:** Time (in seconds) after which for call sequence **linear** the call is forwarded to the next extension (default = 5 seconds).

#### Removing a user

- ▶ Click  next to the user entry.



Screen **Administration** ▶ **Routes** ▶ **Call forward** ▶ **Hunt groups** enables you to define rules for diverting calls to a hunt group (→ p. 63).

## Authorisations

Menu **Permissions** enables two kinds of authorisations to be specified, and users and user groups to be assigned.

- ◆ **Permission groups** (→ p. 25) provides different ways to structure how the phone, the directories and the queues are used for different users. It is possible for example to set up authorisation groups for international calls, call forwarding and editing the company directory. Complex user structures such as the executive-secretary function, and different strategies for actioning queues, can also be implemented with authorisation groups.
- ◆ **GUI groups** (→ p. 28) provide the ability to differentiate access to different functions of the user interface for user groups. For example, a user without Administrator rights could be assigned the permission to manage users.

## Permission groups

An authorisation group comprises a defined number of authorisations. An authorisation group is assigned to user names (→ p. 21) or queues (→ p. 33).

Predefined authorisation groups:

<b>Admins</b>	Group for the Administrator. This group is assigned to user name <b>admin</b> . This assignment cannot be deleted.
<b>All invisible users</b>	This group contains all users not shown in the internal directory. This group is assigned to user name <b>admin</b> . This assignment cannot be deleted.
<b>All users</b>	Default group for users. This group is assigned to all users, including <b>admin</b> . This assignment cannot be deleted.
<b>All visible users</b>	This group contains all users shown in the internal directory. This group is assigned by default to all users, except <b>admin</b> . This assignment cannot be deleted.
<b>All queues</b>	Default group for queues

Authorisation groups can be changed and redefined.

- ▶ First define an authorisation model for your company.
  - Examples:
    - Dialling control for local, national and international calls:  
Set up authorisation groups Local, National and International, and select these groups accordingly for outgoing routing (→ p. 59)
    - Users who may use the "Intercom" feature (executive-secretary combination)
    - Users who may edit the global directory
- ▶ Assign the user more sub-authorisations depending on the authorisation model

### Setting up a new authorisation group

- ▶ Administration ▶ Users & extensions ▶ Permissions ▶ Permission groups

Existing groups are listed with **Name**, **Type** and **Members** (number).





Do not delete the predefined groups as you may prevent users from accessing the functions of the PABX.

- ▶ Enter a name for the group in field **Name** ▶ Select the group type from list **Type**:

**User:** For an authorisation group to be assigned to a user name

**Queue:** For an authorisation group to be assigned to a queue

- ▶ Click  ... The group is entered in the list

- ▶ Click  next to the group entry

Edit permission group: All users

ID	Name	Type	Members
16	<input type="text" value="All users"/>	Queue	0

Permission apply to group

Permission  apply to

Type Member

Member

### Adding authorisations

- ▶ Select the authorisation from list **Permission**

The list makes available these predefined authorisations:

- **Call waiting**
- **Call forwarding, Override call forward**
- **Busy Lamp Field**
- **Allow CLIP, Allow CLIR, Allow DND**
- **Direct Pickup**
- **Monitor own queues**
- **Fax server**
- **Show GUI** (assign authorisations to a GUI group → p. 28)
- **Edit global contacts**
- **Group pickup**

- Hot desking
  - Impersonate
  - Ringtone configuration
  - Member of internal phonebook
  - Set night answer service
  - Allow intercom
  - Allow reminder
  - Allow voicemail configuration
  - Queue logon/logoff
  - Call forward queues
  - Monitor queues
- ▶ In list **apply to**, select the group to which the authorisation is to apply ▶ Click **+** ... The authorisation is shown in the list



You need not assign authorisations to an authorisation group for outgoing routing (e.g. restriction to local calls, and time or number restrictions). You only create the group itself and assign the required members. This group is then assigned to a routing rule (→ p. 60).

## Adding members

- ▶ Enter the user name in the field ▶ Click **+**

### Example: Call pickup rules for executive's office and secretariat

The acceptance of calls can be regulated specifically using authorisations and pickup groups.

- ▶ Create authorisation group "Secretariat" with authorisations **Call forwarding** and **Intercom**, and apply them both to group "Secretariat"
- ▶ Create authorisation group "Executive's office" with authorisation **Intercom**, applied to group Executive's office and **Override call forward**, applied to group "Secretariat"
- ▶ Assign "Executive's office" and "Secretariat" to a shared pickup group (→ p. 22)
- ▶ Create key profiles for the two phone types (→ p. 41), with key assignments **Intercom** and **Group pickup**, and assign these key profiles to the provisioning groups for the executive's office and secretariat (→ p. 44)
- ▶ **Call waiting** must be enabled for "Secretariat" (from menu **User Settings**)

A caller dials the number of the queue (→ p. 33). If the executive's office phone is registered in the queue, the call is routed to this phone. If the executive's office phone is not registered in the queue, the call is routed to the secretariat's phone. If the secretariat's phone is busy, enabled call divert can be ignored from the executive's office and the call is taken on the secretariat's phone. The shared pickup group now means the call can be accepted by the executive's office.

### GUI groups


Which functions (modules) of the user interface are available to a user is specified by the affiliation to GUI groups.

There are two predefined GUI groups:

**New admin GUI** For the Administrator, assigned by default to authorisation group **Admins**

**New user GUI** For users without Administrator authorisation, assigned by default to authorisation group **All Users**

You can set up more GUI groups. They are made available for selection when authorisation groups are set up.

▶  Administration ▶ Users & extensions ▶ Permissions ▶ GUI

Existing GUI groups are displayed with the name and number of modules permitted for the group.

**New admin GUI 63/115** means for example, for authorisation group **New admin GUI**, 63 of 115 possible GUI modules are enabled.


GUI groups		
Title	Modules	
My GUI	9/98	<input type="text" value=""/> <input type="text" value=""/>
New admin GUI	64/98	<input type="text" value=""/> <input type="text" value=""/>
New user GUI	54/98	<input type="text" value=""/> <input type="text" value=""/>

Title


#### Setting up a new GUI group

▶ In field **Title**, enter the name for the group ▶ Click  ... The group is entered without authorisation

#### Deleting a GUI group

▶ Click  in the line for the group ... The group is deleted

## Adding/deleting modules

- ▶ Click  next to an entry ... All GUI modules are listed

The positioning of the modules corresponds to the layout of the user interface. Lower-level modules are denoted by the relevant number of dashes (-, --, ---), and indented.

To enable a module, the associated higher-level modules must also be enabled. When a higher-level module is disabled, all lower-level modules are also disabled regardless of their own setting.

- ▶ Enable/disable a module using the **ON/OFF** switch
- ▶ Click **Save** to save the settings

Example		
- User Settings		<input checked="" type="checkbox"/> ON
-- Service attributes	Service attributes	<input checked="" type="checkbox"/> ON
-- Key assignments	Key assignments	<input checked="" type="checkbox"/> ON
-- Ringtones	Ringtones	<input type="checkbox"/> OFF
-- CSV import/export		<input checked="" type="checkbox"/> ON
--- CSV import	CSV import	<input type="checkbox"/> OFF
--- CSV export	CSV export	<input checked="" type="checkbox"/> ON



The **Login** and **Logout** functions should be allowed for a GUI group as a minimum because otherwise no activities at all are then possible.

## Assigning a GUI group to users

- ▶ Create the authorisation group, applying authorisation **Show GUI** to the GUI group and adding it to the authorisation group
- ▶ Enable the authorisation group for the user(s) to be assigned this GUI authorisation (→ p. 21)

## Global contacts

In **Menu** → **Contacts**, the interface makes available three directories to the user: **Internal**, **Global** and **Private**. The entries for the **global** directory can be entered manually or imported from a CSV file.

### Manually creating a contact list





▶  **Administration** ▶ **Users & extensions** ▶ **Global contacts** ▶ **Contact list**

All existing directory entries are displayed.

Contacts

Name  Number

A B C D E F G H I J K L M N O P Q R S T U V W X Y Z

First name	Last name	Company	Work	Mobile	Home		
Susan	Black	Gigaset	+49 89 12341234				
James	Brown	Gigaset	01234243234	5353453	34534534		




Parameters **First name** and **Last name** are used for the name filter. The number filter uses parameters **Work**, **Mobile** and **Home**.

Filtering lists → p. 14.

### Creating a new contact

▶ Click **Add Contact** in the top right of the screen


#### Parameter:

<b>First name</b>	First name of contact
<b>Last name</b>	Last name of contact
<b>Company</b>	Company or organisation
<b>Office</b>	Work phone number
<b>Mobile</b>	Mobile number
<b>Home</b>	Private number
<b>Image</b>	Image shown when this contact rings. <ul style="list-style-type: none"> <li>▶ Click  ▶ Select the image from the file system on the computer or network ... The file name is entered in the text field ▶ <b>Save</b> ... The image is loaded and displayed</li> </ul>
	<b>Formats:</b> PNG, GIF, TIFF and JPG



## Exporting contacts

You can export contacts to the global directory and import them again as required, or use them on another system. CSV (Comma Separated Value) files are used for this.

- ▶  Administration ▶ Users & extensions ▶ Global contacts ▶ CSV import/export ▶ CSV export

Export contacts

Encoding	<input style="width: 100%;" type="text" value="UTF-8"/> ▼
Seperator	<input style="width: 100%;" type="text" value="Semicolon"/> ▼
Header	<input type="checkbox"/> OFF

- ▶ **Encoding** (UTF8 or ISO) and **Seperator** (comma or semicolon) must be selected for the export file
- ▶ **Header** must be enabled if the first line in the file is to have a header
  - Enabled: the first line in the export contains  
**First name,Last name,Company,Work,Mobile,Home**
  - Disabled: Only the contacts are exported
- ▶ Start export: ▶ Click **Download CSV** ▶ Select the destination for the file in the file system and enter a file name  
The default is prv\_pb\_<ID>.csv



The file name and the destination for the file can only be selected and entered when the settings in the browser permit the downloading of files.

### Importing contacts

You can import contacts saved with an export, or import into the global directory from other directories.

An import file with the contacts must be stored locally on your computer or in the network.

File format: First name,Last name,Company,Work,Mobile,Home

Separator: Comma, semicolon or tab

Example with semicolons and header:


First name;Last name;Company;Work;Mobile;Home


Peter;Brown;Company;123456789;01784567;083416786

Susan;Black;Org;987654321;015679787878

- ▶  Administration ▶ Users & extensions ▶ Global contacts ▶ CSV import/export ▶ CSV import

Import contacts

Import file	<input type="text" value="prv_pb_demo.csv"/> 
Encoding	<input type="text" value="UTF-8"/> ▼
Seperator	<input type="text" value="Semicolon"/> ▼
File includes header	<input type="checkbox"/> OFF

- ▶ Click  ▶ Select the file from the file system of the computer or network
- ▶ Select character encoding (UTF8 or ISO) ▶ Select separator used
- ▶ **File includes header**
  - Enabled: the first line in the file is not entered as a contact
  - Disabled: The first line is taken as a contact entry
- ▶ Click **Upload** ... The entries are displayed as a table for checking
- ▶ Click ... The entries are added to the personal directory. This also happens when a contact with identical details is already in the directory.







Even identical entries are added to the directory.

## Queues

The **Queues** module makes available an additional group type. It enables a user group to be assigned a queue function for incoming calls. Once a caller is in the queue, checks are performed at specified intervals whether and to which member of the queue group the call can be put through. Members of queue groups are called agents.

▶  **Administration** ▶ **Users & extensions** ▶ **Queues**

Queues

Queue	Title	Max. callers		
5000	Service	5		
5001	Hotline	10		

Title



Screen **Administration** ▶ **Routes** ▶ **Call forward** ▶ **Queues** enables you to specify rules for diverting calls to a queue (→ p. 62).


### Setting up a new queue

▶ In field **Title**, enter the name for the queue ▶ Click **+**

#### Parameter:

<b>Title</b>	Queue name
<b>Extension</b>	Queue extension
<b>Hold music</b>	<p>Callers hear whilst waiting the PABX's hold music or custom hold music.</p> <ul style="list-style-type: none"> <li>▶ Select the hold music class from the options list. Hold music class → p. 39</li> </ul> <p><b>Default:</b> A caller hears the default hold music.</p> <p><b>Ring instead of hold music:</b> The caller hears the dialling tone.</p>
<b>Greeting</b>	<p>Select an audio file for the introductory greeting - which is played once before the call is passed to the queue.</p> <ul style="list-style-type: none"> <li>▶ Choose an audio file from the options list. All available audio files are shown.</li> </ul>

---

<b>Announce hold time</b>	<p>Interim announcement interrupting the hold music and announcing the anticipated wait time . This announcement is internal to the system and cannot be changed.</p> <p><b>Yes:</b> The wait time is announced at regular intervals</p> <p><b>Once:</b> The wait time is announced only once</p> <p><b>No:</b> The wait time is not announced</p>
<b>High load announcement</b>	<p>Interim announcement interrupting the hold music and informing the caller of a high number of calls.</p> <ul style="list-style-type: none"><li>▶ Choose an audio file from the options list. All available voice files are shown.</li></ul> <p><b>None:</b> The caller is not informed</p> <p><b>if more than xxx callers waiting:</b> A caller is only notified when more callers than the number specified are waiting.</p>
<b>Wrap-up time</b>	<p>Pause (in seconds) for agents before the next call is put through. The <b>Wrap-up time</b> starts when call pickup is ended.</p>
<b>Pause agent on no answer</b>	<p>Specifies if, following a pause, another attempt is made to put through a call to an agent when none of the group members takes this call.</p>
<b>Pause agent on hangup</b>	<p>Specifies whether an agent has a pause after every call (ringing). The pause starts after the ringing, so after the <b>Ring time per agent</b>. The agent cannot be called during this break, regardless of the ringing strategy and when he/she took his/her last call. If an agent has just actioned a call (answered and ended it), and the remaining time for the ring break is longer than the <b>Wrap-up time</b>, the ring break comes into effect.</p>
<b>Weight</b>	<p>Queue priority. Value range: 0 – 255.</p> <p><b>Weight</b> specifies which queue is prioritised and when free agents are available. The higher the value, the higher the prioritisation over others.</p>
<b>Ring time per agent</b>	<p>Time after which a call attempt to an agent is cancelled. Entering 0 means the maximum duration of 3,600 seconds is used. For the <b>Least recent</b> call strategy, <b>Ring time per agent</b> determines when the next agent is called.</p>
<b>Max. callers</b>	<p>Maximum number of callers kept in a queue. Once the maximum number is reached, subsequent callers hear the engaged signal.</p> <p> For certain queue statuses (such as full and no agent answering), call divert to another number, an announcement or an answering machine can be set up (→ p. 62).</p>

---

<b>Strategy</b>	<p>Specifies the method used to put incoming calls through to agent extensions:</p> <p><b>Round robin:</b> Every agent is assigned a time slot in which he/her is reachable. When the time is over, the agent is placed at the back of the agent list and next agent moves forward.</p> <p><b>Least recent:</b> The caller is put through to the agent who has been waiting the longest to take a call</p> <p><b>Random:</b> The agent is chosen randomly</p> <p><b>Fewest calls:</b> The agent with the lowest number of calls receives the call</p> <p><b>Ring all:</b> All free agents are called. The first one to pick up receives the call.</p>
<b>Enter</b>	<p>Specifies when a queue is enabled:</p> <ul style="list-style-type: none"> <li>- <b>Do not enter if no agent is logged on or no agent is available</b></li> <li>- <b>Do not enter if no agent is logged on</b></li> <li>- <b>Always</b></li> </ul>
<b>Leave</b>	<p>Specifies when a queue is disabled:</p> <ul style="list-style-type: none"> <li>- <b>When all agents log off</b></li> <li>- <b>When all agents log off or when no agent is available</b></li> <li>- <b>Never</b></li> </ul>

▶ Save the settings with **Save** ... The queue is entered in the list



You can record yourself, or upload, the audio files for the welcome message and announcements:



**Administration** ▶ **Users & extensions** ▶ **Audio files** (→ p. 36)

## Assigning users

Users register as agents for a queue with key sequence   <Queue extension> , and use    to remove themselves from.

Agents can also be assigned to a queue statically. Users are then added automatically and they cannot remove themselves.

Adding a user: ▶ Select the user from the list ▶ Click **+**

Removing a user: ▶ Click  next to the user entry



The function is only available when editing a queue, not when setting it up.

### IVR



An IVR makes it possible to navigate a caller through the phone system before the caller is connected to a particular person.

**Example:** A caller calls the service extension of your company - which is configured as an IVR. The caller hears an announcement prompting: "For question about your product, please press "1", for technical problems, please press "2", ... . One key press by the caller connects the caller to the correct person.

**Prerequisite:** You have at least one voice file you can use as an announcement. You must record this voice file, or load it onto the PABX, beforehand (→ p. 36).

---

### Setting up an IVR

- ▶  Administration ▶ Users & extensions ▶ IVR
- ▶ In field **Title**, enter the name for the IVR ▶ Click 

Parameters:

Parameter:

<b>Title</b>	Name of IVR
<b>Extension</b>	Extension of IVR. A call to this extension activates the IVR.
<b>Announcement file</b>	Audio file played as an announcement when the extension is called. This announcement includes for example instructions on how to use key codes for menu control. <ul style="list-style-type: none"><li>▶ Select the required audio file from the options menu</li></ul>
<b>Time to wait for input</b>	Time the system waits for a response from the caller (pressing of a button). The announcement is repeated if no response is received within this time. Specify the maximum number of repetitions with parameter <b>Repetitions</b> . What happens when afterwards there is still no response is defined in an interaction rule (→ p. 37).
<b>Repetitions</b>	Maximum number of times the announcement text is repeated.

---

## Key assignments

Specify the subsequent action when the caller presses 0 – 9, \* or #.

<b>None</b>	No action. The connection is not ended. The caller can press another button.
<b>Hang up</b>	The call is ended
<b>Go to extension</b>	The caller is diverted to another number or extension ▶ Enter the number/extension
<b>Voice message</b>	Another message is played and the call ended. ▶ Select the audio file from the options menu
<b>Repeat announcement</b>	The initial announcement is repeated once more

---

## Interaction rules

You specify with interaction rules the subsequent action when a caller does not respond at all or performs an invalid action.

**Parameter:**

<b>Hang up</b>	The call is ended
<b>Go to extension</b>	The caller is diverted to another number or extension ▶ Enter the number/extension
<b>Voice message</b>	Another message is played. ▶ Select the audio file from the options menu
<b>Repeat announcement</b>	The initial announcement is repeated once more

---

### Audio files



You require audio files for greeting text, wait time announcements and announcement text for IVRs. You can load audio files onto your PABX or record them yourself using a phone connected to the PABX.

Formats allowed: aif, aiff, wav, au, al, alaw, la, ul, ulaw, lu, gsm, cdr, mp3 and ogg

Maximum size: 20 MB

▶  **Administration** ▶ **Users & extensions** ▶ **Audio files**

Existing audio files are displayed with description and duration. The file name is used as the description. You can change the description at any time (this applies for files you record on the phone and audio files uploaded).

Play file: ▶ Click  ... The audio file is played over the computer speaker  
Repeat/pause playback: ▶ Click 

Delete file: ▶ Click 


---

### Recording an audio file

- ▶ In field **Extension**, enter the extension number of the phone for the recording ▶ Click **Record** ... The phone rings
- ▶ Pick up the receiver, or enable the hands-free function, and follow the instructions
- ▶ Refresh the display of screen **Audio files** (by clicking menu **Audio files** again for example) ... The new announcement is displayed
- ▶ In field **Description**, enter a name for the audio file

---

### Loading audio files

- ▶ Click  ▶ Select the audio file from the file system on the computer or in the network ... The file name is entered in the text field
- ▶ In field **Comment**, enter a description for the audio file ▶ Click **Upload** ... The file is loaded



## Hold music

Hold music is played for a caller in a queue waiting for a free agent (→ p. 33). Hold music is divided into certain classes. Every class can be assigned multiple audio files (played in sequence).

- ▶  Administration ▶ Users & extensions ▶ Hold music

Existing classes are displayed with name and number of files assigned.

- ▶ You can select hold music as the default music in the **default** column. This is used for a queue if **Default** is selected as the hold music.

Class **default** contains predefined music for the PABX. This music is used for queues provided no other music is available and selected. It cannot be changed or deleted.

### Defining new classes

- ▶ Enter the name for the hold music and click 


### Assigning/editing audio files

- ▶ Next to the entry for the class, click  ... Audio files already assigned are listed



Listening to an audio file:

- ▶ Click **Call** in column **Call assigned ext.**

or when the extension is not registered to the phone:

- ▶ Enter a phone extension in field **Call custom ext.** ▶ Click **Call** ... The phone with the extension specified rings ▶ Pick up the receiver or press the hands-free button 

Adding audio files:

- ▶ Click  ▶ Select the audio file from the file system on the computer or in the network ... The file name is entered in the text field
- ▶ Enter the description for the audio file ▶ Click  ... The file is loaded and entered in the list of audio files

# Provisioning

Provisioning means providing registered devices with configuration data. All devices are set up with Autoprovisioning when the PABX is started.

**Prerequisite:** You have assigned Option 114 with the IP address of the PABX for phones on the DHCP server (→ p. 69).

In Autoprovisioning, the parameters stored in the PABX are used as defaults for device configuration. The default settings cannot be changed.

You can overwrite one or more of these provisioning parameters, and the preconfigured key assignments for individual devices or device groups.

Changing default provisioning:

- ▶ Create a special key profile (→ p. 41) and/or provisioning profile (→ p. 43)
- ▶ Assign one or more profiles to a provisioning group (→ p. 44)
- ▶ Assign the provisioning group to users (→ p. 19)

On every restart of the phones registered for the users, the profile is sent and the relevant default values are overwritten.

---

## Phones

- ▶  Administration ▶ Provisioning ▶ Phones

This screen shows all phones known to the PABX. The PABX automatically detects all phones of the following device types when in the local network:

- Gigaset DE310 IP PRO
- Gigaset DE410 IP PRO
- Gigaset DE700 IP PRO
- Gigaset DE900 IP PRO
- Gigaset N510 IP PRO with a maximum of 6 handsets
- Gigaset N720 DM/IP PRO with a maximum of 100 handsets
- Maxwell 10

Because an analogue device can be connected to every FXS interface, these interfaces are also maintained and displayed as phones.

Phones					
MAC address		IP address		Phone type	
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	(all)	<input type="text"/>
MAC address	IP address	Phone type	User	Extension	Firmware version
7C:2F:80:1D:10:07-0:01	192.168.0.102	gigaset-de700	Administrator	950001	02.00.08
7C:2F:80:1D:10:07-0:02	192.168.0.100	gigaset-n510	Martin Black	950003	42.194
7C:2F:80:1D:10:07-0:03	192.168.0.101	gigaset-de900	Susan Brown	950004	02.00.08

The following information is displayed for every phone:

- ◆ MAC address and IP address
- ◆ Phone type
- ◆ User and Extension if the phone is registered. Phones not registered are given a default number on which they can be called.
- ◆ Current firmware version



The list also includes devices no longer in the network.

## Key profiles

A key profile can be defined for every device with function keys. If key assignments are already in the default settings for the PABX, they can be added to or overwritten.

Function keys are available on the following phones:



DE 410 IP PRO:	7 function keys, can be upgraded to 21 with expansion module
DE 700 IP PRO/ DE 900 IP PRO:	14 function keys, can be upgraded to 56 with a maximum of 3 expansion modules
Maxwell 10	100 programmable keys




When expansion modules are used, the PABX is not able to provide enough power to the phones over PoE. Use separate power adapters to power the phones.

- ▶ Administration ▶ Provisioning ▶ Key profiles

Key profiles

Profile	Phone type		
<input type="text"/>	Gigaset DE410 IP PRO		
Profile <input type="text"/>	<input type="button" value="Create new Profile"/>		

- ▶ Enter the name for a new profile in field **Profile** ▶ Click **Create new Profile** ... The profile is entered into the options lists (in alphabetical order)
- ▶ Select the required profile ▶ **Phone Type** ▶ Click 

You can allocate a key assignment for every possible function key (PK1 - PKn) on the phone type selected.



Users themselves are able to assign functions to the function keys of a phone (not the expansion modules). You can prevent this by locking the keys assigned a function.

- ▶ Enable function assignment for a key (PK1 - PKn) with the **ON/OFF** switch ▶ Select the function from the list:

- inherit -	The key keeps the default PABX assignment (if available)
<b>External destination</b>	Dial external number ▶ Enter the number in field <b>Number/Data</b>
<b>Extension</b>	Select the extension ▶ Enter the extension number in field <b>Number/Data</b>
<b>Group pickup</b>	Call pickup for those belonging to the same pickup group (→ p. 22) ▶ Enter the number in field <b>Number/Data</b>
<b>Intercom</b>	Dial a connection to an intercom extension ▶ Enter the extension number in field <b>Number/Data</b> <b>Prerequisite:</b> The extensions involved require authorisation <b>Allow intercom</b> (→ p. 25)
<b>Reminder</b>	The phone rings at the time specified. The key enables/disables the alarm. ▶ Enter the alarm in format <b>SS:MM</b> in field <b>Number/Data</b> This function is not supported on the Gigaset DE410 IP PRO.

- ▶ In field **Label**, enter a description for the key assignment. It is used to create the keypad inserts.
- ▶ Lock the key to prevent changes by other users: ▶ Enable/disable the lock with the **ON/OFF** button in column **Locked?**
- ▶ Save the settings with **Save**



#### Assigning a key profile:

- ▶ Create a provisioning group containing this key profile → p. 44
- ▶ Assign the provisioning group to those users receiving this key assignment → p. 21

## Provisioning parameters

Provisioning profiles are only required in exceptional circumstances. They are used to store functions differing from the standard on certain devices.

An Excel file containing all possible parameters is on the Gigaset portal:

→ <https://teamwork.gigaset.com/gigawiki/display/GPPPO/Provisioning+step+by+step>

- ▶ Use the search function to search for **parameter-list** ▶ Select the most recent file from the files found; the date is part of the file name ▶ Download the file
- ▶ Open the Excel sheet for the phone whose settings you wish to change ▶ Select from the parameter list the parameters you wish to change

### Creating a provisioning profile

- ▶ **Administration** ▶ **Provisioning** ▶ **Provisioning parameters**

Provisioning parameters

Profile	Phone type		
<input style="width: 90%;" type="text"/>	<input style="width: 90%;" type="text" value="Gigaset DE410 IP PRO"/>	<input style="width: 20px; height: 20px;" type="button" value="✎"/>	<input style="width: 20px; height: 20px;" type="button" value="🗑️"/>
Profile <input style="width: 100%;" type="text"/>	<input type="button" value="Create new Profile"/>		

- ▶ Enter the name for a new profile in field **Profile** ▶ Click **Create new Profile** ... The profile is entered into the options lists (in alphabetical order)
- ▶ Select the required profile ▶ Select **Phone Type** ▶ Click

Edit provisioning parameters


Profile	Phone Type					
<input type="text" value="Test"/>	Gigaset N510 IP PRO					
Setting	Index	Value				
Setting	<input type="text"/>	Index	<input type="text"/>	Value	<input type="text"/>	<input type="button" value="+"/>

- ▶ Enter a parameter in field **Setting** ▶ Enter the index as required (the index is at the end of the parameter name in brackets) ▶ Enter the required value in field **Value** ▶ Click **+** ... The parameter is entered in the list
- ▶ Once you have entered all the parameters you wish to change: ▶ **Save**

## Provisioning groups

You can now create provisioning groups with the key and (as required) provisioning profiles you have created. Groups can have a hierarchical structure (i.e. a group can contain sub-groups). This makes it possible to differentiate further the allocation of key assignments within a group of users.

Example: You create key profile T1 - which is assigned to provisioning group P1 and only assigns the first 4 keys. Key profile T2 assigns keys 5 and 6 and is created as sub-group P2 under P1. You can now assign to users provisioning group P1 with key assignment T1, and other users P2 - comprising key assignments T1 and T2.

- ▶  **Administration** ▶ **Provisioning** ▶ **Provisioning groups**
- ▶ In field **Group**, enter a name for the group ▶ In field **Title**, specify an optional name for the group. This is then shown instead of the group name in the group configuration.
- ▶ From list **child of**, select a group under which the new group is a sub-group. All groups already created are made available. Groups on the topmost level are assigned to the **Root node**.
- ▶ **Key profile** and/or **Provisioning profile** must be selected from the lists
- ▶ Where applicable, select the number of available expansion modules for the phone type
- ▶ Click **+** ... The group is entered in the list and is now available for user configuration (→ p. 19)

## Routing

Your PABX features different ways to connect to a public phone network - over the Internet (SIP), using an analogue exchange line (FXO) and digital (ISDN). These connections must be configured in line with information provided by the phone provider responsible.

You have already set up a phone connection during the start-up phase (→ p. 11). The connection configured is entered into the configuration as a SIP or TDM gateway (FXO or ISDN). A gateway group for the connection is set up automatically. All incoming and outgoing connections are first routed via this gateway group.

You can set up other gateways and gateway groups from the Administrator menu, and align routing to the needs of the company.

### Configuration process

You require provider access details and any other information about the connection provided.



Help on setting up SIP accounts / trunks is available from:

<https://teamwork.gigaset.com/gigawiki/display/GPPPO/ITSP+SIP+Trunking>

- SIP:
- ▶ Set up at least one gateway group
  - ▶ Set up a SIP gateway for every provider account, and assign it to one gateway group

Analogue or ISDN:

- ▶ Configure a gateway for every available trunk line (FXO, ISDN)
  - All calls are routed internally over SIP. This is why SIP gateways are created automatically for configured FXO and ISDN gateways.
- ▶ Set up at least one gateway group, assign the SIP gateways
- ▶ Specify rules for incoming and outgoing calls



Different rules can be defined for incoming and outgoing calls, and for how call numbers are handled. For this, call number groups must be specified in the form of patterns, or regular expressions. An introduction to working with regular expressions is in the appendix (→ p. 77).

## TDM gateways

TDM (Time Division Multiplex) means digital phone technology for analogue and ISDN connections. You can deploy your PABX as a gateway between SIP (Internet telephony) and TDM, i.e. analogue or ISDN outside lines. If you want to use a TDM connection or connect up analogue devices, you must configure the connections.

An SIP gateway with internal registrar is automatically entered for every TDM gateway configured (→ p. 49).



Connections for analogue trunk lines (FXO) are only available on the Gigaset T640 PRO.

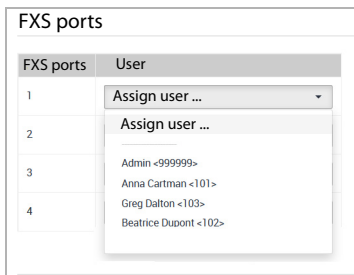
## FXS ports – analogue devices

On a Gigaset T640 PRO, you can only connect up to four (on a Gigaset T440 PRO up to two) analogue devices to the FXS ports (→ p. 7). You can assign devices to users on this screen.

▶ Administration ▶ Routes ▶ TDM Gateways ▶ FXS ports

▶ Select one user from the list for every FXS port to which a device is connected ▶ Save

The analogue devices automatically receive the extension of the respective users (→ p. 20). If a user has already registered a phone having this extension with the PABX, it is deregistered.



Information on the status of FXS ports is available on screen  
Administration ▶ System status ▶ Interfaces (→ p. 75)

## FXO ports – analogue trunk lines



FXO ports are only available on the Gigaset T640 PRO.

An FXO port establishes the connection to the public phone network (PSTN) or an analogue PABX (→ p. 7). You can combine FXO ports into one or more trunks, or connect a single line (outside line) to every port. Ports must be assigned continuously starting at 1.

▶ Administration ▶ Routes ▶ TDM Gateways ▶ FXO ports



### Setting up an FXO trunk

- ▶ Enter the trunk name ▶ +

Edit FXO Ports

Name	Prefix	FXO Port
<input style="width: 90%;" type="text"/>	<input type="checkbox"/> OFF	3 ▾

- ▶ Use the switch to specify whether a prefix is added at the front
- ▶ Select the number of ports to form the trunk. Only the number of ports still available is shown.
- ▶ **Save** ... The trunk is displayed with **ID**, the associated ports and the Name; a SIP gateway for the trunk is created

### Assigning call numbers

- ▶ Enter the associated call number for every FXO port assigned ▶ ✓



Information on the status of FXO ports is available on screen

**Administration** ▶ **System status** ▶ **Interfaces** (→ p. 75)

### BRI Ports

ISDN ports (BRI) are used to connect ISDN devices (→ p. 8). Every ISDN port can be configured as a Point-to-Multipoint or Point-to-Point connection

- ▶ **Administration** ▶ **Routes** ▶ **TDM Gateways** ▶ **BRI ports**
- ▶ Enter the trunk name ▶ +

#### Parameter:

<b>Country Code</b>	Country code, e.g. 49 for Germany
<b>Area code</b>	Local area code, e.g. 30 for Berlin
<b>PTP Pilot number</b>	Main number of a Point-to-Point connection How connection number and extension numbers are handled for incoming calls is configured in the gateway group with parameter <b>Inbound DIDs</b> (→ p. 55).
<b>National prefix</b>	Digit which must be prefixed for national calls, e.g. 0 (depends on country)
<b>International prefix</b>	Digits which must be prefixed for international calls, e.g. 00 (depends on country)

<b>Number substitution</b>	<p>Enabled: Suitable replacement rules are generated based upon the values in fields <b>Country Code</b>, <b>Area code</b>, <b>PTP Pilot number</b>, <b>National prefix</b> and <b>International prefix</b> to guarantee correct signalling of the call number type in the ISDN protocol (type National, International, etc.). This guarantees that numbers are displayed correctly at the called end.</p> <p>This is important most of all for connections with the <b>CLIP no screening</b> feature - because invalid numbers are quickly shown at the called end if signalling is incorrect.</p> <p>If the option is enabled, the details entered in the different call number fields may no longer be entered in the fields for the associated gateway group (above all Outbound caller ID), because this information is appended in addition to that from the TDM gateway. Once all fields are entered, only the extension may be signalled by the gateway group. If only the country code is specified, a combination of local area code, trunk number and extension must be signalled.</p>
<b>Layer 2 mode</b>	<p><b>PTP Pilot number:</b> For a Point-to-Point connection</p> <p><b>Point-to-Multipoint:</b> For a Point-to-Multipoint connection</p>
<b>Layer 3 mode</b>	<p>Protocol used for ISDN communication (country-specific)</p>
<b>Port</b>	<p>Number of ports belonging to this trunk.</p> <p>Ports must be assigned continuously starting at 1.</p>



- ◆ Not all number fields need to be completed. If a local area code is entered, a country code and the right national/international prefixes must be specified. In conjunction with the settings in **Routes ▶ Gateway groups**, they have a bearing on the flexibility of numbers which can be signalled.
- ◆ Diverted calls are signalled correctly when (inter)national prefixes and country code are entered correctly. The other fields have no bearing on diverted calls.
- ◆ Information on the status of ISDN ports is available on screen **Administration ▶ System status ▶ Interfaces** (→ p. 74)

## SIP gateways

At least one SIP gateway must be set up. If the Installation Wizard is used for start-up, a SIP gateway is already available for the trunk line configured.

If you use an Internet phone connection, you require the access details from your provider (ITSP).



Help on setting up SIP accounts/trunks is available from:

<https://teamwork.gigaset.com/gigawiki/display/GPPPO/ITSP+SIP+Trunking>

If you use an analogue or ISDN trunk line, configure the TDM gateways (→ p. 46). For every configured TDM gateway (FXO or ISDN trunk), a SIP gateway is automatically set up with the default rules, and displayed on screen SIP gateways. It is normally not necessary to make further changes to the configuration. It may be necessary to change the **Dial comand** parameter.

### ▶ Administration ▶ Routes ▶ SIP gateways

The SIP gateways already set up are displayed.

▶ Enter the name for a new gateway ▶ **Create new gateway**

#### Parameter:

<b>Registrar</b>	Registration server of the provider; <b>internal</b> is entered automatically for TDM gateways
<b>Proxy</b>	Proxy server (if used)
<b>User</b>	Use name as per provider specifications, e.g. call number
<b>Password</b>	Password as per provider specifications
<b>Allow outbound calls</b>	Enabled: calls may also be made over the gateway
<b>Register</b>	Enabled: the PABX makes contact with the provider to register the SIP account (mandatory for ITSP)
<b>Language</b>	Announcement language
<b>Dial comand</b>	Format in which numbers are sent SIP/{prefix}{number:1}@{gateway}
	{prefix} is replaced by parameter <b>Add prefix</b> if defined (→ Outbound routes, p. 59)
	{number:1} <b>number</b> is replaced by the number dialled :1 removes the first digit of the number selected, e.g. when 0 must be dialled first to get an outside line
	{gateway} is replaced by the name of the SIP gateway

Examples:

SIP/{number}@{gateway} = dial without dialling for an outside line 0  
Extension dials 05251 123456;

"Calling SIP/05251123456@gw\_1\_siptrunk (SIP Trunk)" is sent

SIP/{number:1}@{gateway} = dial with dialling for an outside line 0  
Extension dials 0-05251 123456;

"Calling SIP/05251123456@gw\_1\_siptrunk (SIP Trunk)" is sent

SIP/{number}@{gateway} = dial in local network without local area code  
05251 and dialling for an outside line 0

Extension dials 0-123456

"Calling SIP/123456@gw\_1\_siptrunk (SIP Trunk)" = invalid number is sent

SIP/{prefix}{number}@{gateway} = dial in local network without local area  
code 05251 and dialling for an outside line 0

Prefix 05251 must be entered in outgoing routing.

Extension dials 0-123456

"Calling SIP/05251123456@gw\_1\_siptrunk (SIP Trunk)" is sent

---

### Source of destination number

#### INVITE request line

The destination number is taken from the **Invite** request made to the SIP server by the connection request.

#### To: header

The destination number is taken from field **To:** in the SIP header.

---

### Group

Gateway group to which the SIP gateway is assigned (→ p. 53).

A gateway can only be used once it belongs to a gateway group.

---

### Port

Port number for SIP communication;

Default 5060 (default SIP port)

---

### NAT

Default setting: **Yes**

Connection problems may arise if you connect the PABX to a router with NAT firewall.

Try the following settings:

#### Force rport

Causes the SIP server to return a response to the source (IP address/port) of the connection request.

#### comedia only

Symmetric NAT traversal. Enables the PABX to determine address information (IP address/port) from received data packets of the destination subscriber.

If the settings do not result in improvement, you may need to change the NAT settings of the router.

---

<b>Reroute RTP stream</b>	<p>The PABX normally tries to take the direct path from Subscriber A to Subscriber B for the data flow (RTP). If the system needs to respond to input during a call (such as control from the key codes) or the devices are in a network behind a NAT firewall, the server must act as a proxy. This parameter is used to change the RTP data flow.</p> <p>Not enabled (default):  Only divert the RTP media flow when the subscribers are not behind a NAT and this can be detected by the server. This means the PABX always acts as a proxy. The setting should not be changed.</p> <p>Enabled:  The server tries to establish direct RTP data flow between the two partners.</p>
<b>Check availability</b>	<p>Enabled: The PABX checks whether a call is possible with SIP, i.e. whether there is Internet access and the SIP server is available. If yes, the connection is established over SIP, if not, the PABX tries to establish the connection via another gateway in the same gateway group (such as an ISDN connection).</p>
<b>Simultaneous calls</b>	<p>Number of calls which can be made over the gateway at the same time. Default: 0 = unlimited.</p> <p>Normally limited in the provider contract.</p>
<b>DTFM mode</b>	<p>DTMF signalling (Dual Tone Multi Frequency) is required for example to poll and control some network mailboxes and to control automatic information systems using digit codes. To send DTMF signals over VoIP, you must define how the key codes are converted to signals and sent.</p> <p>Ask your provider which type of transmission it supports.</p>
<b>From user</b>	<p>User name assigned by the provider. Often the same as the call number (parameter <b>User</b>), but can be different.</p>
<b>From Domain</b>	<p>Domain name of the provider, almost always identical to <b>Registrar</b>. This information is available from your provider.</p>
<b>T38 support</b>	<p>T.38 is a protocol for sending faxes over data networks. Enable the function when faxes are to be sent over the gateway, and the provider supports this protocol.</p>
<b>Insecure</b>	<p>(provider-dependent)</p> <p>Authentication of the partner for a connection request (<b>INVITE</b>).</p> <p><b>no - Deactivated (default)</b></p> <p><b>port - Ignore port number</b>  The user context on the server is used to check the IP address (not the port number)</p> <p><b>invite - no authentication on incoming invites</b>  The user context on the server is used to check the IP address and port number (no authentication with password)</p> <p><b>port,invite - Options port and invite</b>  No authentication</p>

---

### Update remote party ID (CLIP) (provider-dependent)

**no - Deactivated (default)**

When the provider expects the remote party ID in the header:

**Use Remote-Party-ID header or**

**Use P-Asserted-Identity header**

The PABX adds an RPI header to the connection request.

---

### Trust remote party ID (provider-dependent)

**no - Deactivated (default)**

Do not trust the remote party ID for incoming calls

**Trust Remote-Party-ID**

Trust the remote party ID for incoming calls

The PABX copies the number from the **RPI** header (instead of from the **From** header).

---

### Codecs

(provider-dependent) The voice quality of VoIP connections is dependent on the voice codec used for data transmission, and so the bandwidth of your DSL connection (the better the codec the more data needs to be sent).

► Enable the voice codecs to be used by the gateway

Observe the specifications from your provider.

Priority when using codecs: from left to right, and top to bottom.

---

### Allowed IP subnet

Specify the subnets to which calls are allowed.

Default: 0.0.0.0/0 all subnets are allowed

Format: IP address / subnet mask

The subnet mask determines how many bits of the IP address specified are included:

32 All bits are included

Example: 192.168.1.1/32, only IP address 192.168.1.1 may be called

24 The first 24 bits are included

Example: 192.0.2.0/24, all IP addresses in network 192.0.2.\* may be called

16 The first 16 bits are included

Example: 192.168.0.0/16, all IP addresses in network 192.168.\*.\* may be called

8 The first 8 bits are included

Example: 192.0.0.0/8, all IP addresses in network 192.\*.\*.\* may be called

---



Information on the status of Ethernet ports is available on screen

**Administration** ► **System status** ► **Interfaces** (→ p. 75)





## Gateway groups

In a gateway group, you combine multiple gateways and define common rules for incoming and outgoing calls. At least one gateway group must be set up so the system can be used for making calls. A gateway (SIP, FXO or ISDN) must be assigned to a gateway group.

If the Installation Wizard is used for start-up, a gateway group is already set up with the default settings for the trunk line configured.

- ▶  Administration ▶ Routes ▶ Gateway groups

Gateway group

Gateway group	Gateways	Caller IDs		
comp_sip1 (comp-sip1)	2	0		
comp_sip2 (comp-sip2)	1	1		

Gateway group

- ▶ Enter the name for the group ▶ Create new group

## Routing

For the transfer of numbers to work correctly in the incoming and outgoing directions, the search/replace patterns for numbers must be changed in line with your location or the call number block provided.

The Installation Wizard defines default rules (depending on the specifications on setting up the trunk line). These can be changed as required.

Check the automatically created entries for the gateway group.

Edit gateway group	
Title	<input type="text" value="comp_trunk1"/>
Permit inbound calls	<input type="checkbox"/> OFF
Outbound caller ID	Search/replace pattern for outbound caller ID (1) s/ <input type="text" value="^(*)"/> / <input type="text" value="\$1"/> /
Asserted Identity	Search/replace pattern for asserted identity (1) s/ <input type="text"/> / <input type="text"/> /
Inbound DIDs	Search/replace pattern to cut prefixes (2) s/ <input type="text"/> / <input type="text"/> /
Inbound caller ID	Search/replace pattern for inbound caller ID (3) s/ <input type="text"/> / <input type="text"/> /
Gateways	-



Help on testing the interoperability of your SIP trunk is available from:

<https://teamwork.gigaset.com/gigawiki/display/GPPPO/ITSP+SIP+Trunking>

### Parameter:

Permit inbound calls	Enabled: calls may also be accepted over the gateway group
Outbound caller ID	Pattern for handling numbers for outgoing calls. The number replaced is sent to the called party. The caller can be called back on this number.
Search/replace pattern for asserted identity (1)	With some providers, the SIP header contains the caller's extension for identification (Asserted Identity). You use this parameter to specify what to do with this information for outgoing calls.



<b>Inbound DIDs</b>	Pattern for handling the prefix for incoming calls. Determines how an extension is forwarded internally, i.e. how the correct extension is reached.
<b>Inbound caller ID</b>	Pattern for handling the number for incoming calls. Determines the number on which the called party can call back the caller.
<b>Gateways</b>	Gateways belonging to the group. The gateways are only displayed here. They are assigned on screen <b>Administration ▶ Routes ▶ SIP gateways (→ p. 49)</b>

### Search/replace pattern

Numbers matching the pattern entered in the top field (s/) are replaced by numbers defined in the lower field. Regular expressions are used for the definition.

#### Examples of outgoing numbers:

The international format must be used to send all calls from all extensions.

^(.*)	All numbers
00498912345678	are replaced by number 00498912345678
	0049 or +49 can be used to specify the international format.

The company number is sent with local area code and extension

^\\d\\d\\d\\d(\\d)	Calls in format 12345, where 5 is used as variable \$1
052512088\$1	are replaced by number 052512088\$1, \$1 is replaced by the bracketed value of the search result

#### Examples of outgoing connection numbers (extension):

The connection number should not be sent.

^(.*)	All connection numbers
	are replaced by nothing.

Connection number 10 (switchboard) should always be sent

^(.*)	All connection numbers
10	are replaced by 10.

The number of the connection is sent.

^(.*)	All connection numbers
\$1	are sent

## Routing

### Example of prefix substitution for incoming numbers:

The last digit of the internal 3-digit number is replaced by a 1-digit extension.

498912345678(\d)      Numbers in format 4989123456781, where 1 stands for a 1-digit extension and is used as variable \$1

10\$1                      are replaced by 498912345678101.

### Examples of incoming numbers:

0 for dialling an outside line should be placed in front of incoming calls.

^\(d\*)                      All incoming numbers, with the whole number used as variable \$1,

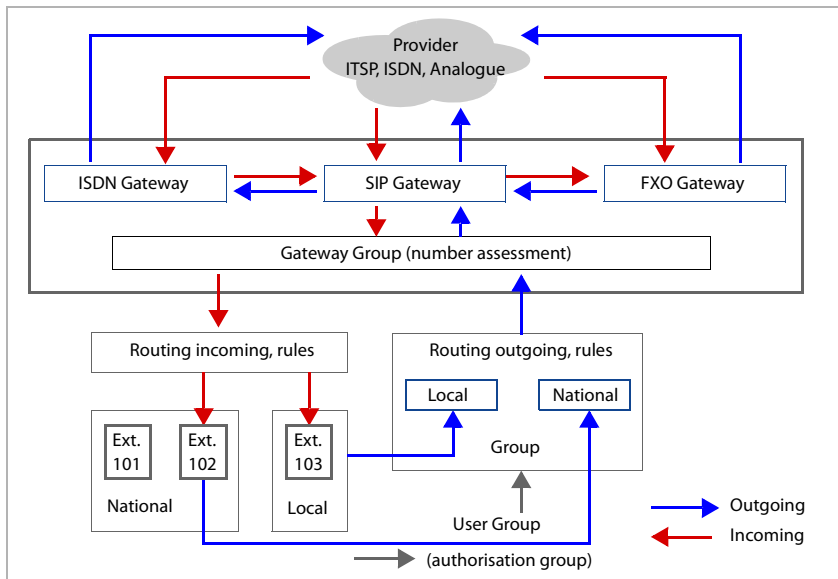
0\$1                              are replaced by 0 + the number



An introduction to working with regular expressions is in the appendix (→ p. 77).

## Routing

Routing specifies whether and how incoming and outgoing calls are forwarded to receiving parties through the system.



The routing for incoming and outgoing calls is already preconfigured by the Installation Wizard. For incoming calls, all recognised extension numbers from the gateway group are forwarded 1:1 by default as destination numbers.

Outgoing calls are passed 1:1 to the gateway group, and to the SIP gateway as per the gateway group configuration. For a TDM gateway with "dial for outside line" set for example, the SIP gateway sends the number to the exchange or TDM gateway in line with the variables in parameter **Dial comand**. The leading "0" is removed there because it is no longer required (→ p. 49).

Using the routing settings, you define rules according to which the forwarding of incoming and outgoing calls is to take place for certain times or people.

## Inbound routes

For incoming routing, you specify for a gateway group what the process is for incoming calls under different conditions, such as calls having certain numbers and calls during normal business hours, on public holidays, at night, etc.

You can create up to 9 different profiles for every gateway group configured.

▶  Administration ▶ Routes ▶ Inbound routes

### Diverting certain calls

The simplest form for incoming routing is diverting certain calls to a defined extension, e.g. an answering machine or secretariat.

▶ **Advanced options** = OFF

Inbound routes


---

Gateway group

sip trunk1 (sip\_trunk1) ▼

Advanced options

OFF

Rule	Number	Target	Profile	
<input type="text" value="Hotline"/>	<input type="text" value="062345678"/>	<input type="text" value="101"/>	Profile 1 ▼	
<input type="text" value="Service"/>	<input type="text" value="062345679"/>	<input type="text" value="102"/>	- ▼	

- ▶ Select required gateway group from the list
- ▶ Enter the name for the profile in field **Rule** ▶ Select the profile number (1-9)
- ▶ Enter the numbers of incoming calls to be actioned with this profile
- ▶ Enter as the **Target** the extension to which calls to this number are diverted
- ▶ **Save**

### Time-controlled diverting of calls

With this you are able to specify different routes by business hours or public holidays.

- ▶ Select the required gateway group from the list
- ▶ **Advanced options** must be selected with the switch

Inbound routes

Gateway group

Advanced options  ON

Rule	Active	Date	Weekdays	Time	Profile	Pattern	Target																				
<input style="width: 100%;" type="text"/>	<input checked="" type="checkbox"/> ON	<input style="width: 50px;" type="text"/> bis <input style="width: 50px;" type="text"/>	<table style="font-size: 0.8em; border-collapse: collapse;"> <tr> <td>M</td><td>T</td><td>W</td><td>T</td><td>F</td> </tr> <tr> <td><input checked="" type="checkbox"/></td><td><input checked="" type="checkbox"/></td><td><input checked="" type="checkbox"/></td><td><input checked="" type="checkbox"/></td><td><input checked="" type="checkbox"/></td> </tr> <tr> <td>S</td><td>S</td><td></td><td></td><td></td> </tr> <tr> <td><input checked="" type="checkbox"/></td><td><input checked="" type="checkbox"/></td><td></td><td></td><td></td> </tr> </table>	M	T	W	T	F	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	S	S				<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>				<input style="width: 50px;" type="text"/> <input style="width: 50px;" type="text"/>	Profile 1 ▾	<input style="width: 100%;" type="text"/>	<input style="width: 50px;" type="text"/>
M	T	W	T	F																							
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>																							
S	S																										
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>																										

- ▶ In field **Rule**, enter a name for the profile ▶ Select the profile number (1-9)

- ▶ **Date**, **Weekdays**, and **Time** must be specified for when the rule is to apply

Example: Profile for weekday nights

**Date** Leave fields free: the setting applies the whole year

**Weekdays** Select all fields apart from **S** and **S**: the setting applies from Monday to Friday

**Time** Enter 20:00 to 06:00

- ▶ **Pattern** and **Target**

Incoming numbers can be routed to different destinations for local, national and international numbers depending on the search/replace pattern in the gateway group.

Enter here the required pattern for analysing incoming numbers and forward it as the selected extension number to the destination.

The transfer of the number to **Pattern** is dependent on the analysis of parameter **Inbound DIDs** in the gateway group (→ p. 55). A fixed internal number (extension) can be assigned as a destination.

Example: Number 004989123456702 is selected. 00498912345670 is the number of the company, 2 the extension

Configuration for **Inbound DIDs** in gateway group:

Search pattern `s/00498912345670(\d)` `(\d) = $1`

Replace pattern `/10$1/` `10$1 = 102`


Incoming routing:

Profile	Pattern	Desti- nation	Result
Day activation	^(.*)	\$1	The call is put through to extension 102
Night activation	^(.*)	101	The call is put through to extension 101

- ▶ Enable/disable the rule with the **Active** switch. Enabled: The rule can be enabled in the **Night answer service** tab for users.
- ▶ **Save**

## Night answer service

Enable/disable on this screen the profiles for incoming routing you have defined.

- ▶  **Administration** ▶ **Routes** ▶ **Inbound routes** ▶ **Night answer service**

Enable a profile for all users:

- ▶ Select the profile from list **Active Profile** ▶ **Save** ... The rules of the profile are applied for all users

Users can choose another profile if this is allowed:

- ▶ Enable profiles 1 – 9 using the buttons

Enabled profiles are made available for selection by users.

- ◆ From the interface

On the **Home** screen in the **Night answer service** area

- ◆ From the phone

Activate:      ...  (for profiles 1-9)

Deactivate:



Users wanting to use it from the phone must belong to a user group having authorisation "Enable/disable routing profile".

## Outbound routes

For external routing, you specify which users (or user groups) may dial which external numbers, and how numbers are put through for outgoing calls.

After start-up with the Installation Wizard, outgoing routing is set such that every user can make unrestricted calls to anyone 24 hours a day every workday, regardless of their user / user group.

Additional rules must be defined if you want to limit this. You may want to specify for example that certain parties may only make calls, or call at certain times, in their own local area network, nationally or internationally. You can also ensure for example that emergency calls can be made from all phones, regardless of whether the parties are registered or not.

- ▶  **Administration** ▶ **Routes** ▶ **Outbound routes**

## Routing

### Simple outgoing routing to certain number

The simplest form of outgoing routing is diverting certain outgoing calls over another gateway.

- ▶ **Advanced options** = OFF

Outbound Routes

Advanced options  OFF

Rule	Number	Gateway group	
<input type="text" value="SIP1"/>	<input type="text" value="00498912345678"/>	<input type="text" value="SIP1"/>	<input type="button" value="🗑️"/>
<input type="text" value="SIP1"/>	<input type="text" value="*"/>	<input type="text" value="SIP1"/>	

- ▶ In field **Rule**, enter a name for the rule
- ▶ Enter the number
- ▶ Select the gateway from the list under **Gateway group** to be used for routing these calls. All SIP gateways configured are shown.
- ▶ **Save**

### User group-dependent settings for outgoing calls

Outgoing calls can, depending on the authorisation group to which a user belongs, be restricted to certain times and numbers.

- ▶ **Advanced options** must be selected with the switch

Outbound Routes

Advanced options  ON

Rule	Active	Weekdays	Time	Pattern	Group	Gateway	Add prefix
<input type="text"/>	<input checked="" type="checkbox"/>	M T W T F ☑ ☑ ☑ ☑ ☑ S S ☑ ☑	<input type="text"/> <input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

- ▶ Enter the rule name in field **Rule**
- ▶ Specify **Weekdays** and **Time** for when the rule is to apply  
Example: Profile for weekday nights  
**Weekdays**      Select all fields apart from **S** and **S**: the setting applies from Monday to Friday  
**Time**            Enter 20:00 to 06:00

- ▶ In the **Pattern** field, specify the number which can be dialled  
Examples:  
 $\wedge[1-9]$  = only numbers within the same local area network, no area code  
 $\wedge 0[1-9]$  = only national numbers
- ▶ Select the authorisation group for which this rule is to apply. Users must be assigned this authorisation group → p. 26).
- ▶ Select the gateway(s) from the lists under **Gateway group** to be used for routing these calls. All SIP gateways configured are shown.
- ▶ The number specified in field **Add prefix** is placed before the number dialled if the dial command defined for the SIP gateway contains variable {prefix} (→ p. 49).
- ▶ Enable/disable the rule with the **Active** switch.
- ▶ **Save**

### Example:

User **Greg Dalton** may make calls to his own local area network (089 for Munich in the example), but not national or international calls.



The following settings are necessary:

- ◆ **Greg Dalton** is assigned to user group **Local** (→ p. 21)
- ◆ User group **Local** is identical to authorisation group **Local** (→ p. 26)
- ◆ On screen **Outbound Routes**, set up two rules with name **Local**  
 Rule 1: the user dials with area code  
**Pattern**  $\wedge 089$  = numbers to the local area network with code 089 are allowed  
 Rule 2: the user dials without area code  
**Pattern**  $\wedge[1-9]$  = Numbers with no dialling code are allowed  
 All other numbers dialled are rejected (engaged tone).

For both rules, assign group **Local** to the SIP gateway over which calls are to be routed. The associated gateway group determines the format of the number send.



The rules defined are evaluated from top to bottom.

Changing the order: ▶ Use the   buttons on the right of a rule to move it up or down. The buttons are displayed as soon as more than one rule is defined.

## Call diverts

Calls to extensions from a queue or hunt group are forwarded by default according to the rules specified for the queue or hunt groups.


This menu allows call diverts to be set up for certain cases or times.

### Call divert for queues

**Prerequisite:** Queues must be set up (→ p. 33).

▶ Administration ▶ Routes ▶ Call forward ▶ Queues

The screen shows all queues set up with extensions and names.

▶ Click  next to the queue for which you want to set up a divert

#### Destination numbers for call forwards

▶ Specify the destination numbers for the call divert. These numbers can then be used in the rule.

##### Default number

Number to be used mainly for call divert, e.g. always for external calls

##### Temporary number

Number to be used in special cases, e.g. for internal calls when nobody is in the queue

##### VM number (internal user)

User extension with activated answering machine. An answering machine is activated when the user has set up call divert to an answering machine (→ p. 20).

### Specify rules for call divert

Forward ...	always	full	timeout	empty
internal	-	Temporary	Announcement 3-	Temporary
external	-	Announcement 2	AM with announc-	Default
		after (s)	15	

▶ Setting up rules separately for internal and external calls:

When is call divert to apply?: ▶ Configure the settings in the required column. You can define call diverts in one, multiple or all columns.

##### always

All calls are diverted

##### full

Call divert when all extensions in the queue are busy

##### timeout

Call divert when no agent picks up within the time specified ▶ In the field next to **after (s)**, specify the time (in seconds) after which call divert is to apply

##### empty

Call divert when no agent is registered with the queue



To where is the call diverted?: ▶ Select in each case the required destination from the selection menu

**Default number** Divert to **Default number**

**Temporary number** Divert to **Temporary number**

**AM with announcement** Divert to the answering machine of the extension specified under **VM number (internal user)**. The caller hears the selected announcement and is able to then leave a message.

**Announcement** The caller only hears the announcement selected and cannot leave a message

## Recording announcements for the answering machine

You can record multiple announcements, which are then available for selection for destinations **AM with announcement** and **Announcement**.

- ▶ In field **Comment**, enter a name for the announcement ▶ Click **+** ... The announcement is entered in the list
- ▶ Click **record** ... The phone with the extensions of user name **Admin** rings ▶ Pick it up ▶ Say the announcement ▶ Press

## Saving settings / enabling call divert


- ▶ Click **Save** ... The call divert set for the queue is enabled

## Call divert for hunt groups

**Prerequisite:** Hunt groups must be set up (→ p. 23)

- ▶  **Administration** ▶ **Routes** ▶ **Call forward** ▶ **Hunt groups**

The screen shows all hunt groups set up with extensions and names.

- ▶ Click  next to the hunt group for which you want to set up a divert

## Destination numbers for call forwards

- ▶ Enter the number ▶ **+**  
Specify as many destination numbers as you require. These numbers can then be used in the rule.

## Specify rules for call divert


Forward ...

	always	busy	no answer
internal	<input type="text"/>	Number 2	Number 3
external	Number 1	<input type="text"/>	<input type="text"/>

- ▶ Setting up rules separately for internal and external calls:  
When is call divert to apply?: ▶ Configure the settings in the required column. You can define call diverts in one, multiple or all columns.  
**always**            All calls are diverted  
**busy**              Call divert when all the hunt group extensions are busy  
**no answer**        Call divert when no hunt group party can be reached  
To where is the call diverted?: ▶ Select in each case the required destination number from the selection menu
- ▶ Click **Save** ... The call divert for the hunt group is enabled

# System

## Licensing

- ▶  Administration ▶ System ▶ License

Once you have successfully performed the licensing process for the PABX (→ p. 13), you can see the license code on this screen.

## Firmware updates

Gigaset makes available new firmware versions as required on <http://wiki.gigasetpro.com>. They can be loaded and installed from this screen. You receive email notifications as required about new firmware versions.




The configuration of your PABX is not affected by firmware updates. To be on the safe side however, perform a backup before loading new firmware (→ p. 72). Always read the release notes for new firmware.

- ▶ Download the latest firmware from the Gigaset server and save it to the PC

- ▶  Administration ▶ System ▶ Update

**Installed version** shows the version number of the current firmware

- ▶ Click  next to **Choose update file** ▶ Select the file from the file system of the computer ... The file name is entered in the text field ▶ Click **Upload & install** ... The firmware file is loaded and installed

**Upload progress** shows the progress of the installation.

The Login screen is shown once the installation is complete. All the configuration settings are kept as they were.



Do not close the browser window during the firmware update (it takes 3 to 4 minutes).

## CDRs

All outgoing calls are listed on the **CDRs** (Call Data Records) screen.

▶  **Administration** ▶ **System** ▶ **CDRs**

Information on all calls currently being displayed:

**Calls:** Number of calls displayed

**Total call duration:** Total duration of calls displayed

**Average call duration:** Average duration of calls displayed


**Duration:** Total duration of calls, including ring time

The following information is displayed for every call:

<b>Time</b>	Date and time of call
<b>Caller</b>	Caller's extension
<b>Target</b>	Number called
<b>Duration (s)</b>	Call duration
<b>Type</b>	Shows whether a call took place, and if not why not: ● <b>answered</b> , ● <b>no answer</b> , ● <b>busy</b> , ● <b>failed</b>

### Filtering lists

To evaluate CDRs by certain criteria, you can filter the list by **Time**, **Caller**, **Target**, **Duration (s)** and **Type**. It is possible to specify multiple filters. For call numbers, subsets can also be selected using \* (e.g. 043\*, i.e. all numbers starting with 043).

- ▶ Select filter criterion from the options lists / enter it in the fields ▶ Click  ... Only the entries appropriate for the filter are displayed

### Exporting the list

To process further CDRs, you can export the list displayed to your PC or storage medium as a CSV file. The specifications are as those for the contact list export (→ p. 31).

- ▶ Select the type of list formatting: **Encoding** (UTF or ISO) and **Seperator** (**Semicolon** or **Comma**) must be selected, and enable/disable **Header**
- ▶ Click **Download CSV** ▶ Specify the destination and save

## Network

If you ran the Installation Wizard all the way through when setting up your PABX, network configuration is already complete. You can make changes here if you wish.

The following values are taken as defaults if you have not yet performed a network configuration:

IP address	192.168.0.50
Network mask	255.255.255.0
Default gateway	None
DHCP server	Enabled
DHCP address range	192.168.0.100 - 192.168.0.150

Your PABX must be integrated into your local network for it to work properly.




Changes to the network configuration can trigger a restart.

Changing the network configuration may result in the link between your computer and the PABX going down. You will then not be able to access the user interface. In this case, you must restore access using the new IP address.

If you make a mistake and the PABX can no longer be accessed from the network, connect your computer directly to the PABX (→ p. 6).

---

## IP configuration

▶  Administration ▶ System ▶ Network ▶ IP configuration

Parameter:

---

<b>IP address</b>	<p>IP address for your PABX. It must be assigned a fixed (static) value.</p> <p>The following must be observed:</p> <ul style="list-style-type: none"><li>◆ The IP address must be from the address range used on the router/gateway for the local network. The valid address range is determined by the IP address of the router/gateway and the network mask (see example).</li><li>◆ An IP address must be unique across the network, meaning it may not be used by another device connected to the router/gateway.</li><li>◆ The fixed IP address may not belong to the address range reserved for the DHCP server of the router/gateway.</li></ul> <p>Check the settings on the router or ask your network administrator.</p> <p>Example:</p> <p>Router IP address: 192.168.2.1 Network mask in network: 255.255.255.0 DHCP server address range: 192.168.2.101 – 192.168.2.255 Possible IP addresses for the phone: 192.168.2.2 – 192.168.2.100</p>
<b>Netmask</b>	<p>The subnet mask specifies how many parts of an IP address the network prefix is made up of.</p> <p>255.255.255.0 for example means that the first three parts of an IP address must be the same for all devices in the network, whilst the last part is specific to each device. For subnet mask 255.255.0.0, only the first two parts are reserved for the network prefix. Enter the subnet mask used by your network.</p>
<b>DNS server</b>	<p>IP address of the DNS server preferred.</p> <p>DNS (Domain Name System) allows you to assign IP addresses to symbolic names. The DNS server is required to convert the DNS name into the IP address when a connection is being established to a server.</p> <p>You can specify your router/gateway's IP address here. This forwards PABX address queries to its DNS server.</p>
<b>DNS Server (optional)</b>	<p>IP address of an alternative DNS server. This is used when the preferred DNS server cannot be reached.</p>
<b>NTP Server</b>	<p>IP address of your preferred time server (NTP = Network Time Protocol). NTP ensures reliable time information across a network.</p>
<b>NTP Server (optional) IP</b>	<p>IP address of an alternative time server. This is used when the preferred time server cannot be reached.</p>

---

## DHCP server

The PABX has an integrated DHCP server, which is enabled on delivery and has an address range appropriate for the network mask set. If you want to use another DHCP server in your network, you must disable the one in the PABX. Running two DHCP servers in parallel in a network is not permitted.

▶  Administration ▶ System ▶ Network ▶ DHCP server

When you do not use the integrated DHCP server:

- ▶ Disable the DHCP server from the **ON/OFF** switch
- ▶ Enter the IP address of the DHCP server on the **IP configuration** screen. Define this IP address on the DHCP server as a static address for the PABX.
- ▶ On the DHCP server, set Option 114 (URL) as follows:

`http://<IP address of PABX>/gigaset-prov/`

This option ensures that the PABX is set as a provisioning server on all phones. How this option is set depends on the system on which the DHCP server runs.

Example for a Linux DHCP server:

```
File dhcpd.conf: option dhcp_114_FW_URL "http://192.168.0.50/gigaset-prov/"
```

When you use the integrated DHCP server:

- ▶ Disable other DHCP servers in your network
- ▶ Enable the DHCP server from the **ON/OFF** switch
- ▶ Specify the address range such that it fits in with the other network settings, i.e. with the subnet mask and any IP addresses with a fixed assignment



If there are any devices in your network with a fixed IP address, you must ensure these addresses are outside the address range of the DHCP server.

---

## SMTP server

Using an integrated SMTP server, your Gigaset PABX emails to users voice mails, faxes, backup files, backup reports and other system messages (such as changes to user data by Administrators). You must configure the SMTP server of the system for this.

▶  Administration ▶ System ▶ Network ▶ Email delivery

### Parameter:


<b>Email delivery</b>	Enable/disable SMTP server
<b>Sender email address</b>	Email address of sender. Default is noreply@localhost
<b>Sender name</b>	Name of sender. Default is Galilei PBX
<b>Relay host IP</b>	IP address of SMTP server
<b>Relay host port</b>	Port over which the SMTP server communicates. Default setting: 25
<b>SMTP authentication</b>	Enable/disable SMTP authentication. Enabled: The SMTP server expects "login" of the sender before emails are sent. Entering of a user name and password is used for this.
<b>SMTP user name</b>	User name for SMTP authentication
<b>SMTP password</b>	Password for SMTP authentication
<b>SMTP authentication type</b>	The following methods are possible for authentication on the server: <b>PLAIN:</b> Standard RFC 4616. User name (for authorisation), user name (for authentication) and password are sent unencrypted. The three strings are merged into one and Base64-encoded. <b>LOGIN:</b> As PLAIN, but authorisation and authentication are in two steps. <b>CRAM-MD5:</b> Standard RFC 2195 <b>SCRAM-SHA-1:</b> Standard RFC 5802 <b>NTLM:</b> Secure authentication method with encryption using random number.
<b>SMTP transport type</b>	Determines the security standard for sending <b>plaintext:</b> no security <b>SSL/TSL:</b> Security standards with data encryption. TSL is based on SSL and is the standard with higher security.

---



## Fax

You can enable a Fax Service (Hylafax) for the PABX, and enter the relevant numbers. The PABX supports SIP protocol T.38, enabling the sending of faxes over (data) networks. The T.38 protocol must be enabled for the SIP gateway over which faxes are to be sent/received (➔ p. 51).

- ▶  **Administration** ▶ **System** ▶ **Fax**
- ▶ Enable/disable the fax service from the **ON/OFF** button
- ▶ **Fax prefix (incoming)** and **TSI prefix (outgoing)** must be entered
- ▶ **Additional TSIs**: Enter any other numbers, clicking **+** every time
- ▶ Save the settings with **Save**



Enabling the fax service automatically causes a system restart.

Users can use one of the TSIs specified here to send faxes. The TSI (Transmitting Subscriber Identification) identifies a fax device as a sender of a fax and is normally displayed at the top of a fax received.

## Date & time

The current system date and time are displayed and can be changed manually as required. They cannot be changed manually if you are using an NTP server (➔ p. 68).

- ▶  **Administration** ▶ **System** ▶ **Date & Time**
- ▶ Enter date (**Day, Month and Year (4 digits e.g. 2014)**) and time (**Hour, Minute**) in the relevant fields ▶ **Save**

## System settings

The **System settings** screen gives you many ways to change the basic system settings. The parameters are grouped in different tabs depending on function.

- ▶  **Administration** ▶ **System** ▶ **System settings**


Every parameter is displayed with name, value set and description.


**Example:**

Parameter	Value	Description
GS_FAX_ENABLED	<input type="checkbox"/>	<input checked="" type="checkbox"/>
GS_FAX_PREFIX	<input type="text"/>	<input checked="" type="checkbox"/>

## System

### Changing a parameter:

- ▶ Open the tab for the function whose parameter you want to change
- ▶ Change the parameter. ▶ Click .

The  symbol is displayed for all parameters you change.

### Resetting a change

- ▶ Click 

### Saving changes:

A bar is shown above the list provided there are changed parameters on a screen:

There are uncommitted values. [Click to view](#)

- ▶ Click **Click to view**. The changed parameters are listed with old and new values.
- ▶ **Confirming a change:** ▶ Click **Commit**
- ▶ **Rejecting a change:** ▶ Click **Revert**

---

## Backing up and restoring the system

Perform regular backups of the data and system settings of your PABX. You can backup the data manually at any time or create a schedule for regular, automatic backups. Backup files are kept in the file system of the PABX. They can also be backed up to external storage. The backup files contain all the data and system settings of the PABX.

---

### Automatic backup

- ▶  Administration ▶ System ▶ Backup ▶ Automatic backup

The **Automatic backup** screen enables you to plan automatic system backups. You can have backups made every week on certain weekdays, or every month on certain days.

- ▶ Enable/disable automatic backup with the **Automatic backup enabled** switch ▶ Select **Weekly Schedule** or **Monthly Schedule**

#### When Monthly Schedule is selected:

- ▶ Specify time: ▶ In field **Days (comma seperated)**, specify the days for the backup  
**Example: Time = 00:30, Days (comma seperated) = 1,10,20**  
A backup is made on the 1st, 10th and 20th of every month at 0.30.

#### When Weekly Schedule is selected:

- ▶ Specify time. ▶ Use the switches to activate the days on which a backup is performed

---

## Manual backup

▶  Administration ▶ System ▶ Backup ▶ Manual backup

▶ Click **Trigger backup**

The backup file is created as an archive file and stored in the file system of the PABX. The file name contains the time and date of the backup (example: galilei-backup-20141118-144301.tar.gz).

### Saving a file to an external location

▶ Click **Download backup file** ▶ Select the destination and save




All backup files are available to download, including those created automatically.

---

## Restore


Backup files created manually or with regular backups can be loaded back onto the system if required.

▶  Administration ▶ System ▶ Backup ▶ Restore

All backup files on the phone system are made available. The most recent file is at the bottom.

▶ Click **Upload** next to the file name

### Restoring the system from a backup file stored externally:

▶ Click  next to **Choose backup file** ▶ Select the file from the file system of the computer or network ... The file name is entered in the text field ▶ **Upload**



The process can take a while.

Settings and information changed since this backup are lost.

The Login screen is shown after the system is restored successfully.

## Status and diagnostic information

The **System status** menu has detailed information on the status of the PABX, the connections and interfaces, and diagnostic information. You can restart or shut down the system if required.

### General information

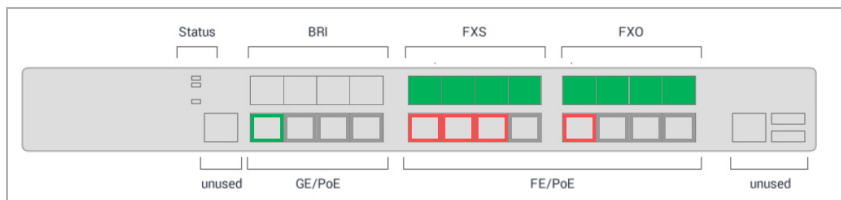
- Administration ▶ System status ▶ General information

General information: **Software version, Licence state, System date and time, System uptime, CPU load, Memory state, Disk state, Number of users, Number of phones**








### Interfaces

- Administration ▶ System status ▶ Interfaces





The screen shows on a diagram the status of the PABX interfaces (connections).



#### Information on ISDN connections: BRI (digital modules)

	Disabled	Not connected.
	Active - OK	Connected and ready.
	RAI alarm	Remote Alarm Indicator: is sent by a terminal when the input signal is lost.
	LOS/LOF alarm	LOS (loss of signal): Fault message when a network component ascertains the loss of the input signal. LOF (loss of framing): Fault message when an ATM receiver station loses the frame description. It is used to monitor the performance of the bit transmission layer in frame-oriented networks.
	AIS alarm	AIS (Alarm Indication Signal): Fault message when a transmission error is detected in the transmission channel, or a fault message is received by another unit on the transmission path.
	D-channel alarm	Fault on the D-channel (used to transmit control information).
	NFAS alarm	NFAS (Non-Facility Associated Signalling): Signalling protocols sent over a link completely independent of the carrier channel.





## Information on the analogue connections: FXS/FXO (analog modules)

	Not connected	Not connected
	Idle	Connected, currently no connection
	Handset offhook	Receiver picked up
	Call connected	Connection established



FXO ports are only available for the Gigaset T640 PRO.

## Information on Ethernet connections: Power over Ethernet (PoE)

	Not connected	Not connected
	Ethernet connected	Connection to Ethernet
	Power connected	Interface powering connected PoE device
	Ethernet and power delivered	Connection to Ethernet and power over PoE

## SIP-Status

The screen provides information on the registered SIP connections.

The following are displayed for every SIP connection registered: user name, refresh status, registration time and whether the DNS manager is enabled.

## Diagnostics

The diagnostic screens show the protocols created by the system. They may prove useful in the event service is required.

▶  Administration ▶ System status ▶ Diagnostics

The following protocols are available:

<b>System log</b>	Messages of the media server processes
<b>Telephony</b>	Messages on PABX activities
<b>Operating system</b>	Messages from the Linux operating system, e.g. boot information
<b>Intrusion detection</b>	The screen shows detected intrusion attempts into your system with the following information (where possible): <b>Date, Time, Jail, IP address, Description</b>

## Reboot & shutdown

- ▶  Administration ▶ System status ▶ Reboot & shutdown

Restarting the PABX:

- ▶ Click **Reboot** . . . The system is restarted. It takes about 2 minutes and phoning is not possible during this time. All the configuration settings are kept as they were.

Shutting down the PABX:

- ▶ Click **Shutdown** . . . The system is shut down. Phoning is no longer possible.



The PoE switch and fan remain ON.

Resetting the PABX to the default settings:

- ▶ Click **Factory reset** . . . The system is shut down and restarted with the default settings. All your configuration settings are lost, but can be restored from a backup file (➔ p. 73).

## Appendix

### Regular expressions

Regular expressions are used in the PABX configuration to:

- ◆ formulate search and replace patterns for call numbers. For incoming or outgoing calls, searches are performed on equivalents of the search pattern - places found are replaced by others. They define how numbers are sent, how sent numbers are displayed and how they are used for return calls.
- ◆ define patterns for call numbers for which certain rules are to apply, e.g. for which special authorisations are assigned or which are diverted at particular times.

There are different "languages" for regular expressions, for which comprehensive syntax rules are defined. The PABX uses Perl Compatible Syntax Expression (PCRE).



The syntax of regular expressions is very extensive and complex. Only a few elements are required for the PABX configuration. Only the main syntax elements used in the search and replace patterns for numbers are described here.

Please refer to the relevant Internet sites for more information.

.	Any character
\d	Any digit
\D	Any character (not digit)
*	The preceding element may occur any number of times
?	The preceding element may occur but need not
+	The preceding element must occur at least once
^(pattern)	Searches for the expression defined in the pattern from the start of the string Example: ^(.*) = Finds any call number ^ = Start of the string, . = Any character, * = Any number of times
	Sequence of alternatives Example: 0049 0 = Either 0049 or 0

## Appendix

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( ) Round brackets	Grouping of search patterns when multiple expressions are in a sequence or nested. Resolution is from the inside outwards. Example: (((0049 0)89)3450) stands for 0049893450 or 0893450 Matches of groupings found are stored and can be reused for substitution (backward reference). A grouping is referenced with \$n, where n is the position of the grouping within the whole expression. Example for outgoing calls: Search pattern <code>^(.*)</code> , replace pattern <code>0\$1 = "0"</code> is placed in front of any number
?(pattern)	Groupings not generating a backward reference Example: <code>?(?:0049 0)89)?3450</code> Pattern identical to the last pattern but no referencing possible
[ ] Square brackets	Alternatives; one of the characters in the brackets must be in the string Example: <code>[0-9][a-z]</code> stands for exactly 1 digit from 0 to 9, and one character from a-z, e.g. 3a, 5c, 9z... A difference is made between upper and lowercase. Note: <code>[0-9]</code> is identical to <code>\d</code>
{ } Curly brackets	{Minimum, maximum} number of characters/digits Example: <code>[1-9][0-9]{1,4}</code> stands for at least 1, and a maximum of 4 digits; the first digit may not be 0.
\ Special designation	If a string contains a regular expression also used as a meta character, it must be denoted specially using a preceding backslash \. This designation is required for: <code>^ \$ ( ) &lt; &gt;   { [ . * + ? \</code>

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