phonalisa®

Manual version 6.2.1

New features, new case studies, extensive section of questions and many more.

Preface

Dear administrators and users,

thank you for purchasing the Phonalisa telecommunications system. You will notice after a short time that this decision will pay off. On the following pages we will show you how to configure your phone system from scratch and most important - how you can use all the functions effectively. The telephone system has been completely redesigned and expanded to include many new and interesting features like the new user interface, a much more flexible callforwarding system, a new group system a more flexible queuing.

At this point I would like to thank the authors Franziska Sprenger, Sebastian Kauder, Stefan Zipfel and Wolfgang Held for their contributions to this manual. Due to their commitment and valuable contributions they have helped decisively designing this manual.

We work constantly to improve our products an their quality which also includes this manual. If you have any questions, suggestions or review after reading this manual, please do not hesitate to contact me (soeren.sprenger@phonalisa.de). I wish you great success in working with your Phonalisa IP phone system.

Yours sincerely,

Sören Sprenger - CTO-Phonalisa Europe GmbH -

Contents

Part I
The installation and basic setup

1 Quickstart

Phonalisa can be installed on Windows, Mac OSX and Debian Linux. Please install Phonalisa on your preferred platform by following these steps.

1.1 Windows installation

1. Please open the Phonalisa installer from the file system.

2. Read and accept the Terms of Use, enter the installation path (eg: C:\phonalisa) and click "Install".

3. After finishing the installation open the "Phonalisa Startpanel", which you will find in the Start menu in Programs.

4. When the program is running, press the "Start WebUI" button.

5. The setup wizard opens and helps you with the first steps after installation.

Warning: Please do NOT use the Internet Explorer (IE10), the system was designed for modern web standards like WebSockets and WebRTC. These will not work properly with any version of the Internet Explorer.

1.2 Mac OSX installation

1. Please open the DMG file and drag the app icon to the application folder to install the Phonalisa telephone system.

2. After the installation, you can run Phonalisa from the application folder.

3. Now press the "Start WebUI" button.

4. Smart wizard will guide you through all setup steps .

1.3 The Debian Linux installation
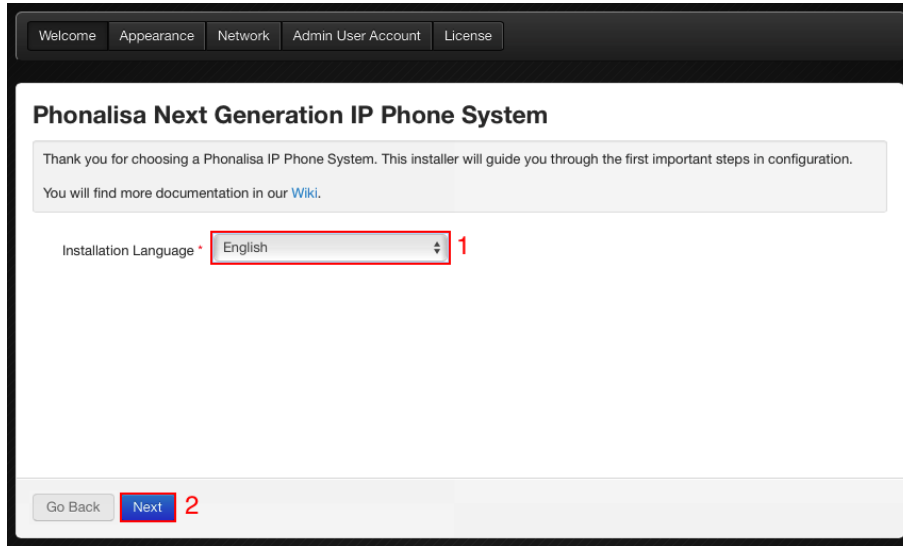
Please download the Phonalisa ISO from the Wiki (Link) .Install the Linux version of Phonalisa on a virtual machine or on a dedicated computer. The installation image guides you through installation, therefore it is possible to install the system without Linux know-how. It is based on a Debian Jessie 8.1 installation CD, which has been adapted for the installation of the Phonalisa system.

1. After executing the ISO you will see the installation screen for a short time and the wizard will automatically start the installation.

2. In the following window you will be asked to set a root password and to confirm it.

3. Please use „full disk use" in the next menu, confirm your choice on the next screen by pressing „Enter". Then select the point "all files on a partition". Finally acknowledge your changes again and agree writing these on the disk.

4. Then you will be asked to enter your HTTP proxy data. If you do not use a proxy, skip the screen with „Next".

5. Finally, please select /dev/sda as boot loader and press „Enter".

6. After the installation there are no additional steps are necessary, the system is installed and can be set up immediately. The IP of your Phonalisa which allows you to sign in with your browser on the Web Interface will be found in the section "URL of the web interface" on the home screen.

2 The Phonalisa Setup Assistent

The Setup Assistent guides you through the basic steps of the configuration and has to be used once.

1. Select the language

2. and press "Next".
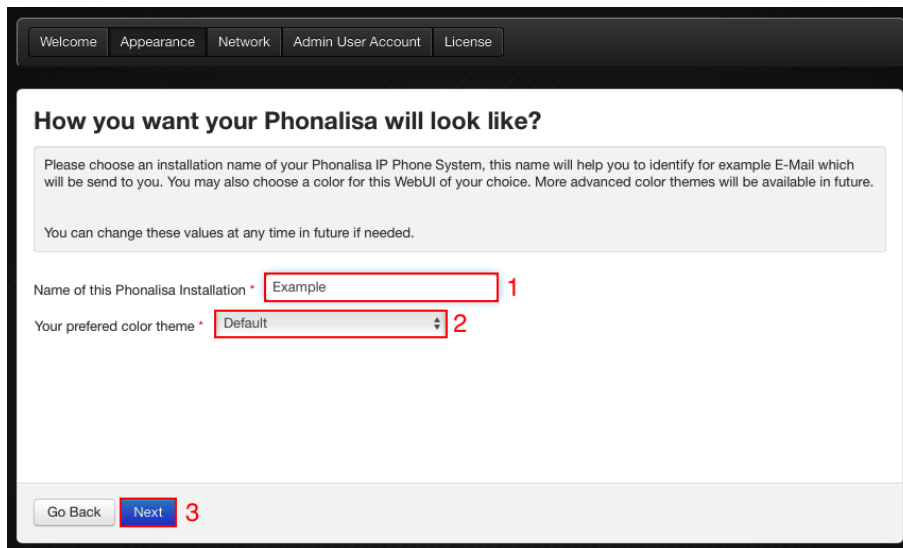
1. After that you give your Phonalisa installation a name. This will help you to distinguish systems if you use Phonalisa in other locations or companies.

2. Choose a color-scheme for your user interface. You can change it at any time in "System" -> "Settings".

3. Press "Next".

1. Enter the IPv4 address of your Phonalisa PBX

2. and enter the subnet mask of your network

3. Here you can configurate your proxy settings.

4. Press "Next".

1. Next add the first user by entering the username,

2. the password,

3. the firstname and the lastname (optional),

4. the internal extension,

5. the E-Mail address,

6. and a PIN-Number.

7. Then press "Next". You will automatically be logged in with your first user on the web interface of your PBX.

1. Enter your license key here.

2. Now you have to enter your first name and surname for which you acquired the license,

3. as well as your address details,

4. and the country.

5. Pease enter your e-mail address in this box.

6. Specifying the branch is optional.

7. Click "Next" to finish.

8. If you have not yet purchased a license, then you can skip the step at this point.

3   Brief Overview of the web interface

Here you will get a brief overview of the Phonalisa web interface and its structure.

1. This is the main menu, which contains the following items:
   - Dashboard: Here you have an list of your phone calls, voicemails, faxes.
   - Addresses: You can create as many address books as you need and find them listed there. The CSV-Import/-Export is also available here.
   - Calendar: Allows you to list and schedule your appointments.
   - Statistics: Find the call-evaluation of your groups and queues.
   - Settings: In this menu you will find all the settings of your own telephone such as the speed dial list, audio files, and call forwarding.
   - Admin: Contains the user and group management, permissions, billing and much more.
   - Provisioning: All provisioning-related settings, such as the key profiles and firmware management.
   - Routing: You can set your incoming and outgoing routes. You will also find the gateways, voice menus, queues and agents.
   - System: Here you can change the basic behavior of your system, for example, the time profiles and internet connection data.

2. This is a submenu, the entries depends on the chosen main menu item.

3. This is the content area of the selected module.

4. With the language selector you can easily change the current language of your user interface.

5. This dropdown panel is used for the "sudo" (substitute user do) function which allows you to switch the user quickly if you have administrator privileges.

6. The Logout button.

7. This button opens and closes the realtime peer monitor.
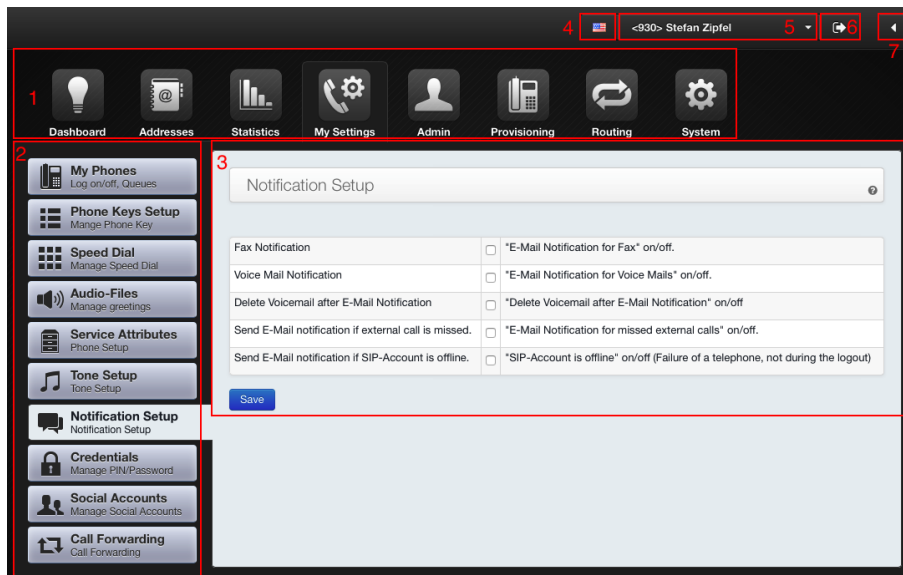
4 First steps after the installation

<u>Summary:</u>

1. Login to the web interface

2. Add a user

3. Configure a phone

4. Login the user at a phone

5. Register a gateway

6. Configure the routing

7. Make some phone calls to test the system

8. Setup your email account to allow the system to send emails

4.1 Logging on to the Web interface
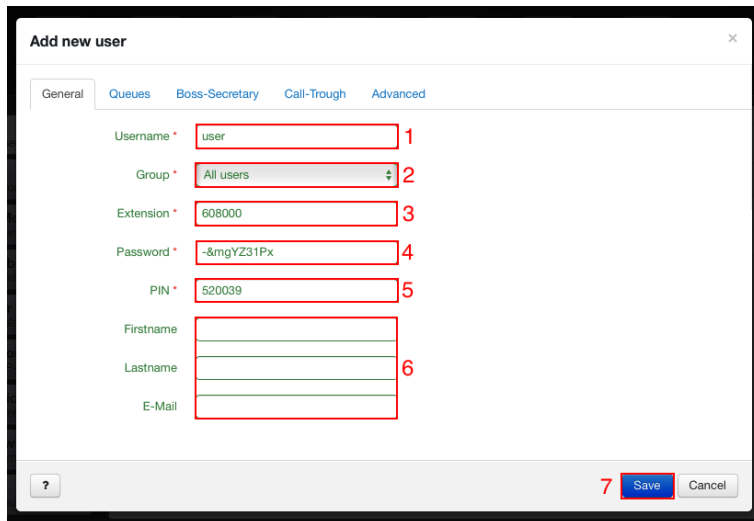
When you have finished the setup, you will be logged in automatically. If you want to sign in later, use the username and password you have chosen in the setup wizard before.

4.2 Add a User

When adding a new user only the basic settings will be set for now. You should create two users which will be sufficient for initial testing. You can make further adjustments later (take a look at 15.1).

Select "Admin" -> "User" and click "Add New User".

1. Enter a username. The username must not contain any special characters. Only lowercase letters and numbers are allowed because the username is used in URL's such as E-mail addresses / Jabber ID's . Example: john.doe, agent.smith

2. As long as you have no other groups defined every user will be added to the "All Users" group.

3. Enter the extension of this user. The extension is used for internal calling and has to be unique.

4. You can change or accept the system-generated password.

5. Also the PIN is automatic generated, you can change it at any time. The PIN is used to assign a phone to a user.

6. Enter anything fitting your needs.

7. Click "Save".

4.3   Setup the Phones

There are 2 ways to setup phones with the Phonalisa telephone system. Preferred is the automatic configuration of the phones.

1. Automatic configuration (autoprovisioning):
Phones which are supported by the Phonalisa provisioning can be configured automatically. Here you will find a step by step tutorial for configuring a Snom phone. The provisioning-URL's for all other supported phones will be found here 4.3.1. If the phone is automatically configured, it will log in to the system with a temporary (dummy) phone number.

2. Manual setup:
Sometimes it's necessary to manualy configure a phone, for example when:

  • a phone is not supported by autoprovisioning

  • you need more than one SIP-Account on this particular phone

You need to enter the following informations into the Phone:

  • Proxy/Server/Registrar: this is the IPv4 address of your Phonalisa PBX

  • Username: this is the username which you find in "Admin" -> "Sipaccounts" (per default this is the extension of the user assigned to that account)

  • Password: this is normally automatic generated and has nothing to do with the users web interface password, you can see the password in the edit dialog of an SIP-Account.

4.3.1   Registration of IP phones with a supported provisioning: Example Snom phones

Look for the IP of each phone in its menu and paste it in your browser url panel. Log in with the username *snom* and the password *snom*. Select the "Advanced" in the menu on the left side and click the "Update". Please select "Update automatically" and enter the Settings-URL http://hostip:8080/{mac}.xml. Then click "Reboot" and confirm the operation with *"Yes"*. The phone reboots and gets assigned a dummy user.

  • Windows/OSX: http://hostip:8080/{mac}.xml

  • Linux-Appliance: http://hostip/{mac}.xml

There are some changes in this version, for example we simplified the provisioning URL's. In addition, there are differences in the provisioning URL's between the platforms (this is because of the different HTTP and HTTPS port). This setting is preconfigured by the setup wizard.

* It is important that you enter the IP of your system in the menu items "Provisioning-Server: Host" and "Provisioning-Server: SIP-Host" on the Phonalisa web interface under "System" -> "Configuration" -> "Provisioning" **BEFORE PROVISIONING**, otherwise provisioning is not available.

Siemens Openstage

  • Specify the IP of the Phonalisa system as DLS server. The DLS-default port (18443) is also used by Phonalisa and is correct.

tiptel

  • Windows/OSX: http://hostip:8080/

  • Linux-Appliances: http://hostip/

4.3.2 Registration of IP phones without provisioning. Example: Zoiper softphone

Download the Zoiper softphone client from *http://www.zoiper.com/* and follow the installation instructions. After the installation, open the settings (Settings) of the softphone and go to the SIP settings. Open the Phonalisa web interface and go "Admin" -> "SIP-Accounts". Edit the user's SIP account which is destined for the Softphone.

1. The "user" corresponds to the SIP username.

2. The "Password" is the SIP password of the user.

3. The "Domain" corresponds to the Phonalisa host.

4. Click "Next".
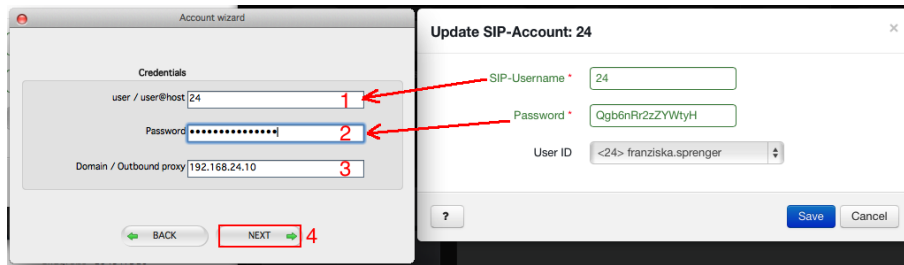
Click "Next" one more time to complete the configuration. Now you will be able to make calls.

Dial a phone number on the dialpad: An active call:
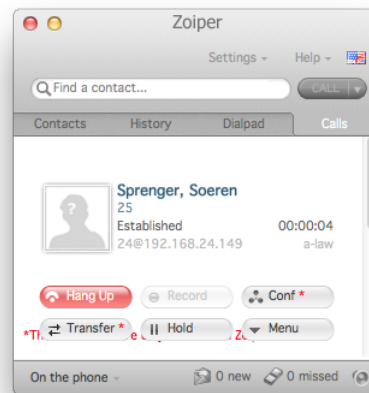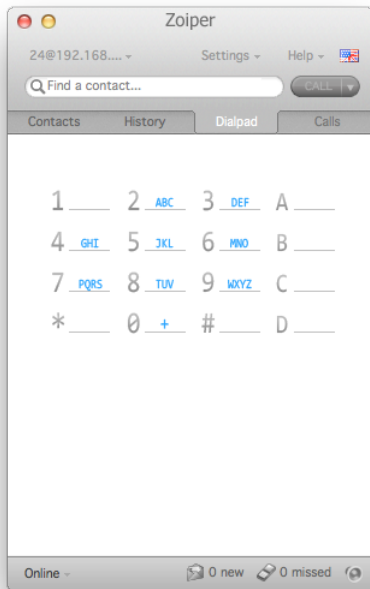
* Optional: License upload: Upload a valid license for your system in "System"-> "License" . You can also proceed without a license but the system is then limited to one simultaneous external conversation (1 external channel).

4.4 Register a user on a phone

There are two ways to register your user on the phone.

Option 1: Directly on the phone
Dial *0-user-extension (for example, *025), then enter the PIN followed by the # key.

Option 2: By web interface
Go to "Provisioning"-> "Provisioning".

1. Every registered phone has a dummy number.

2. Click the edit button.

| Tiptel IP 280 | 192.168.24.157 | 00156522c887 | | 09.70.13.016 | Nobody 9575640 | 1 | | 2 |
| Tiptel IP 282 | 192.168.24.191 | 0015651faeb3 | | 07.61.13.010 | Nobody 9594183 | | | |

3. Select a user who should be assigned to the phone from the dropdown box. Optionally: You can also enter a description of the phone.

4. Save the changes.

Depending on which phone is used either the provisioning has to be retriggered or it has to be restarted manually . The button which is marked red below starts the "triggering" :

| Tiptel IP 280 | 192.168.24.157 | 00156522c887 | | 09.70.13.016 | 931 (Muster, Max <931>) | | |

4.5 Setting up the gateway

The Phonalisa routing system determines how incoming and outgoing calls handled. It is one of the most important components. Gateways are used by the system to receive and send calls. A gateway can be a built-in ISDN interface, a media gateway (e.g. beroNet or Patton) or a SIP provider on the Internet. Incoming and outgoing routes are determined by tables with patterns for each gateway.

For external calls from the system, you need a SIP gateway. This can be an external service provider such as Sipgate or an gateway such as beroNet. Go to "Routing" -> "Gateways" and click "Add new gateway".

1. Enter a name which is freely selectable and only helps differentiate your gateways.

2. Specify the host of the gateway.

3. Set the username and

4. a password.

5. If you want to access an external line, check this box. Then set your desired external line (dialing prefix) in the box below. It is not necessary to set an external line.

6. Save your entries.

4.6 Configure the inbound and outbound routing

To make a calls, you have to define incoming and an outgoing lines for the gateway you just created. Go to "Routing" -> "Routing Inbound". For testing purposes just set a star *. This rule would apply to all the called extensions. Alternatively you can specify the extension directly, e.g. 760823 or 969. In the dropdown box "Target", you can choose an extension the call is placed to.

The configuration of the outgoing routes is analogous to the incoming, in this case the pattern matches the dialed number, e.g. * or 0*.

4.7 Perform the telephony tests

To make sure all settings are correct, you need to perform a phone test. While your devices are successfully signed in and show the user and the extension on the screen you can perform an internal telephony test. Therefor dial the extension of another user on your phone. Now try to make an external call. Do this by dialing an external number (e.g. cell phone number) on your phone.

Tip: If a test does not bring the expected result, you can check the log messages of the system (details see 39)

4.8 The e-mail settings

After having successfully installed Phonalisa, you need to set your email adjustments. This for example is necessary to get recorded voicemails sent by e-mail.

Go to "System"-> "Configuration" -> "Messaging" for set up.

Part II

How to work with your Phonalisa communication system

In the following part, we will explain the functions and their applications of the Phonalisa web interface .
The Phonalisa phone system is available in different versions. So it might be possible that your Phonalisa is missing some points or features listed in this manual.

5   The Dashboard - An overview

On the dashboard, you have a complete overview of all incoming and outgoing calls (user and group), faxes, voicemails and call records.

1. This is the dial field. Input a phone number and press enter, to confirm dialing this number.

2. By clicking on the "Calls" button the list is filtered only by the calls.

3. You get an overview of your voicemails when you click "Voicemail" button.

4. By clicking on this button you can see your call recordings. These can also be played directly.

5. Click "Fax" to view a list of your received and sent faxes.

6. A fax can be sent directly here or by using the address book.

7. The call log entry shows all the important connection and contact details (if available). For an explanation of the symbols, see point 5.1.

8. With this buttons you can add notes, delete notes, or download a message.

9. To play a voice message, click the play button.

10. By clicking on the garbage bin you can delete the entry. The stars displays current message rating, this for example can be used to evaluate how successful a phone call has been.

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5.1 The dashboard symbols

Successful incoming call	Successful outgoing call	Missed incoming call	Unsuccessful outgoing call	Forwarded call
Incoming active call	Outgoing active call	Successful call to group / call group	Unsuccessful call to group / call group	Voicemail
Recording a call	Successful incoming fax	Successful outgoing fax	Unsuccessful incoming fax	Unsuccessful outgoing fax
Successful call to a queue	Unsuccessful call to a queue			

5.2   The call list

The incoming and outgoing calls (successful and unsuccessful) are marked with symbols here. By clicking on the phone number you can either dial the number or create a new contact in the address book. If the contact is already stored in your address book then the contact name will be displayed automatically .
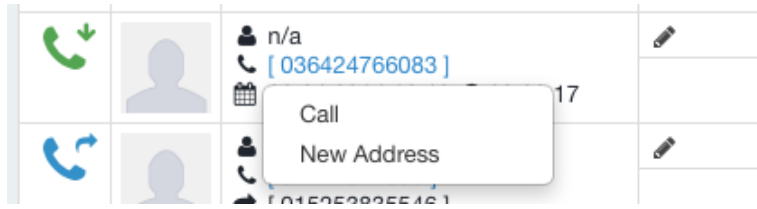
If you want to save the contact in one of your address books - this dialogue appears :

1. Select the address book in which you would like to save the new contact. Do you need more address books? In point 6.3 you will find more information about it.

2. Here you can select the category of the phone number. The choices are, e.g. mobile, fax, mobile (private) and further.

3. In this area you are able to edit the phone number .

4. Enter the firstname, lastname, company name.

5. Save the contact. Now, the new entry is located in the address book.

5.3 The voicemails

In the voicemail tab you will find a list of all received messages. You can play them directly with the play button, pause with the pause button (1) or download with the download button (2).
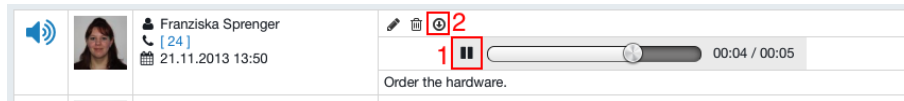
5.4 The call records

Important calls can be recorded. Currently this feature is supported only by the Siemens Openstage 60 SIP. Please note that there are legal provisions for call recording. These must be observed, for example, the notification of the caller before recording.

5.5 The fax list

All incoming and outgoing faxes are listed here. You can see the status (1), display it in your web browser (2) or download the fax as a PDF file (3). All relevant information (page number, ...) will be shown if you click the button (4).

5.5.1 Sending and receiving faxes

Click the panel "Send new Fax" to send a fax.

1. Enter the destination fax number here.

2. Select local PDF document.

3. Select the text of the fax header from the dropdown box. Information about the text of the fax header, you will find in 15.5 and 16.9.3.

4. Click "Send".
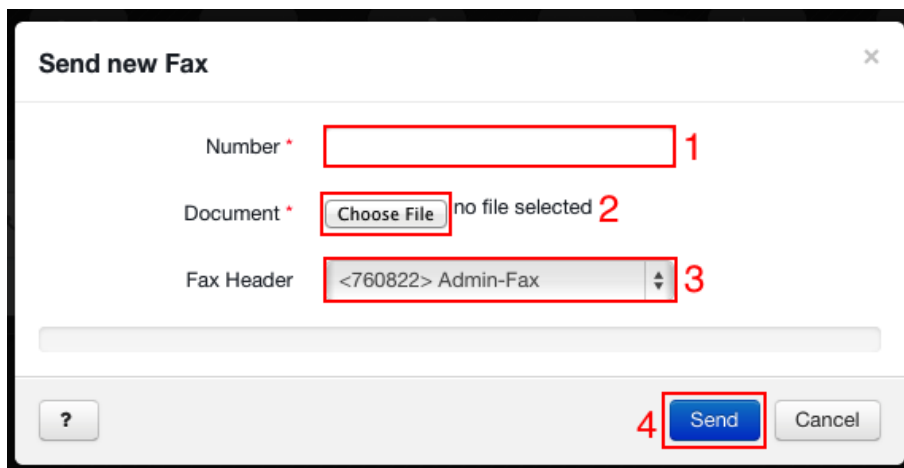
5.6 Adding a note to a event

To capture all the important information and have a better view on the dashboard, you can add a note to any event (call, fax, ...). So you remember, what you have discussed last time. This note will be displayed in the address book with the corresponding contact in the contact history. Thanks to the built-in editor, you can highlight important notes, use quotes and add links.

1. Click the pen in the corresponding line which will disappear when the editor opens.

2. In the editor you can display the text bold and italic, strikethrough, use quotes and much more. If you want to enter a link or an e-mail address, a small window will open in which you can make additional settings. The link and the e-mail address are then clickable.
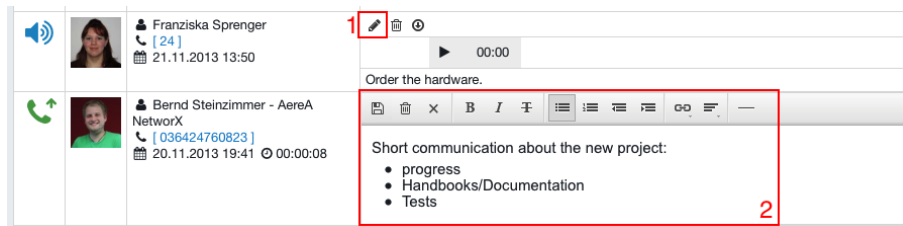
5.7 The real-time presence monitor

In the real-time presence monitor you see all members of your group and the groups you have "Monitor-Permissions". Permissions can be changed in group settings. You can see which colleague is on the phone, respectively is available and you can also see the presence status. Green means online, red means busy.
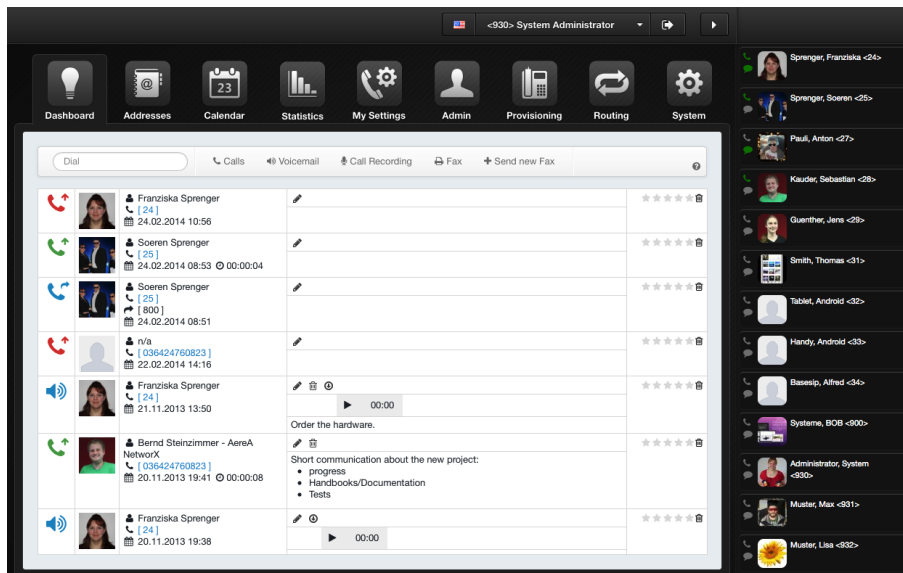
6 The Address Book - An overview

The address book offers a lot of features. Here you will get a brief overview of the main elements.

1. At the top you have the internal address book which contains all users of your telephone system. If you have created other address books you can switch between them with these panels.

2. Here you find the contacts of the selected address book.

3. If you have selected a contact (in 2), the details will appear at the right side .
   - General: Here you will find the corporate and private contact data and the contact type.
   - Addresses: Here you can find the business and private address data.
   - Business: In this tab, you are able to deposit information about the business, for example website, room number, etc.
   - Social Networks: Here you can deposit the links, for example, for a Facebook profile or a LinkedIn profile etc.
   - Last contact: Here you can see when you had the last contact with this person (incoming and outgoing).

4. Click the pen to edit the contact and on the garbage bin to delete it.

5. Click the "+" to add a new contact, input fields in area (3) becomes visible.

6. With this button you have special functions such as CSV import.

7. To select all addresses in the address book at once, select the checkbox.

8. In the dropdown box "Group Action" you can delete, copy or move the selected contacts (all, one, a selection) to another address book or perform a CSV export.

9. This point will take you directly into the admin area where you can create a new address book or to edit existing ones.

6.1 The address book types

There are private and group-based address books, as well as the internal address book. Entries of a "Private" address book type can only be seen and edited by you but will be assigned to a group. This allows users of this group, to create and view private entries in the address book. An address book of the type "Group" is designed to allow all group members to see, create, and edit entries from the address book.

All internal users of the phone system are listed automatically in the "Internal" address book . Additional information, such as address, date of birth and much more can be added to each item. Informations such as the user's extension or the e-mail address can not be changed. This is only possible via the user administration. If you delete a user, it will be automatically removed from the internal address book. Actions like a CSV import, copy or move entries from other address books into the internal address book are not allowed. These actions are only available in the private and group address books.

6.2 The address book permissions

There are read, write and CSV import permissions which can be assigned to existing groups. The inheritance ability of groups plays a crucial role for the assignment of permissions . Since a "Child-Group" inherits from its "Parent-Group", the child group automatically inherits the appropriate permissions (information about the heritage feature, see point 16.1). The read permission ensures that a group only has read access to an address book and will not be able to write, delete or make a CSV import. If a group has read and write access to an address book, then they may create, edit, delete, move and copy any entries. The permission of synchronization is, for example, only useful for Xing address books because the entries change frequently. If the CSV import is permitted the group may import data into the selected address book.

6.3 Creating a new address book

To Create a new address book go to "Admin" -> "Addressbook".

1. Enter a name and

2. select the type.

3. Select an icon which will be shown in front of the address book's name in the menu "Addresses" .

4. Now you can assign the permissions. After saving the new address book it is available in the menu "Addresses".

6.4 Edit a existing address book

Select the address book you want to edit. Make sure that you have read and write permissions.

Add, edit or remove address

1. To add a new address, click the "+".

2. Now the address book is in edit mode and you can make entries on the right side . It is possible to make changes in all tabs and switch between them without saving before.

3. To upload a photo, simply click the avatar image. You can select an image from your own files. The photo will be shown in your call log, presence monitor (only for internal users), in the instant messenger, and on the phone (depending of the support) .

4. Finally, click save. To delete an address, you can use the delete button or delete via the group action.

5. As administrator, you can click this button to create additional address books or edit an existing one quickly.
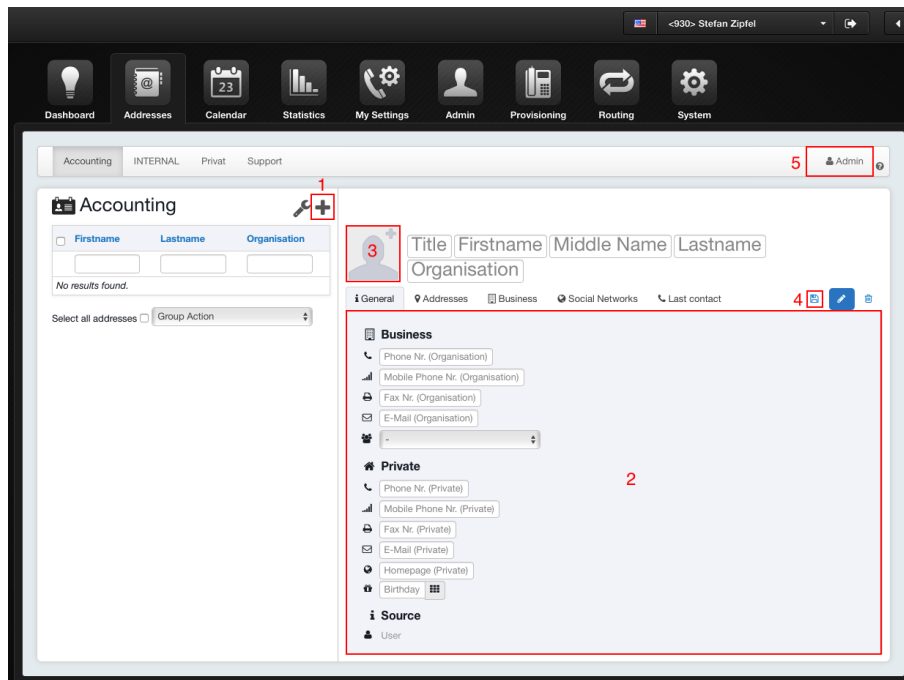
Call, emails and fax sending from your address book:
To do this, click the icon in front of the phone number, the email address or fax number.

6.5   The contact type

The contact type is used to organize certain addresses or people in a group. The function, for example, is required for user-defined call forwarding and if you want to be reached only by contacts with the type "VIP" after closing time. You can determine contact types and define even more for all address books together.
Select "Add new Contact Type" in the dropdown menu for this . The self created contact type is added to the list.
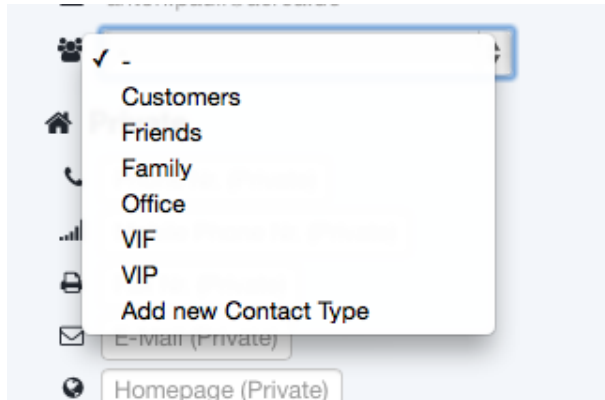
6.6   The group actions

With the group actions, you can copy or move multiple addresses to another address book. Nothing can be copied or moved to the internal address book. The group actions are used to perform actions in stack, e.g. deleting multiple addresses simultaneously or export multiple addresses to CSV file. You can select individual addresses by clicking the checkbox to the left of each address or by "Select all addresses" at the bottom.
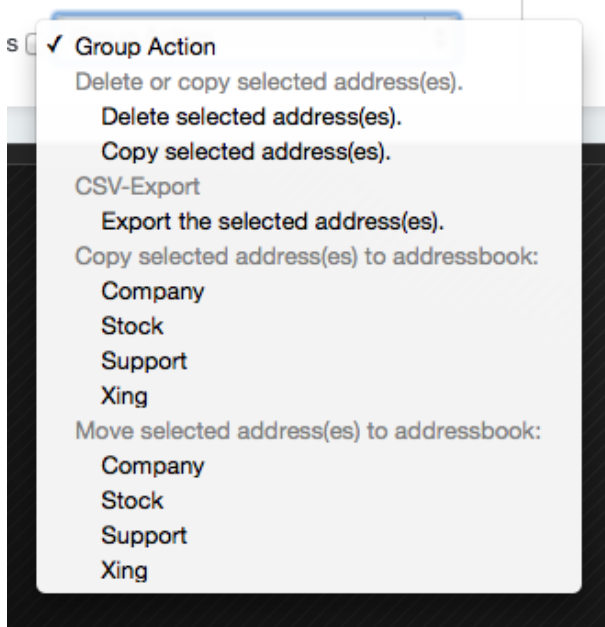
6.7   Selecting personal ringtones in private address book

In a private address book you can select unique ringtone for each contact. This allows you to distinguish the caller from the ringtone in addition to the picture on the display. The select a ringtone update a contact. Now you can choose a ringtone. Please be aware, this option depends on your device.

7 The CSV export of address data

The acronym CSV stands for "Comma Separated Values" and describes the structure of a text files which is used to save or exchang simple structured data . The file name extension is .csv.

7.1 Perform an CSV export

To export data from an address book, you must have read and write permissions.

1. a) Now select the records you want to export (only the visible entries on the active page will be selected).
   b) You can also check "Select all addresses".

2. Select "CSV Export" via the group action. The csv-file containing the exported data will be saved on your computer.
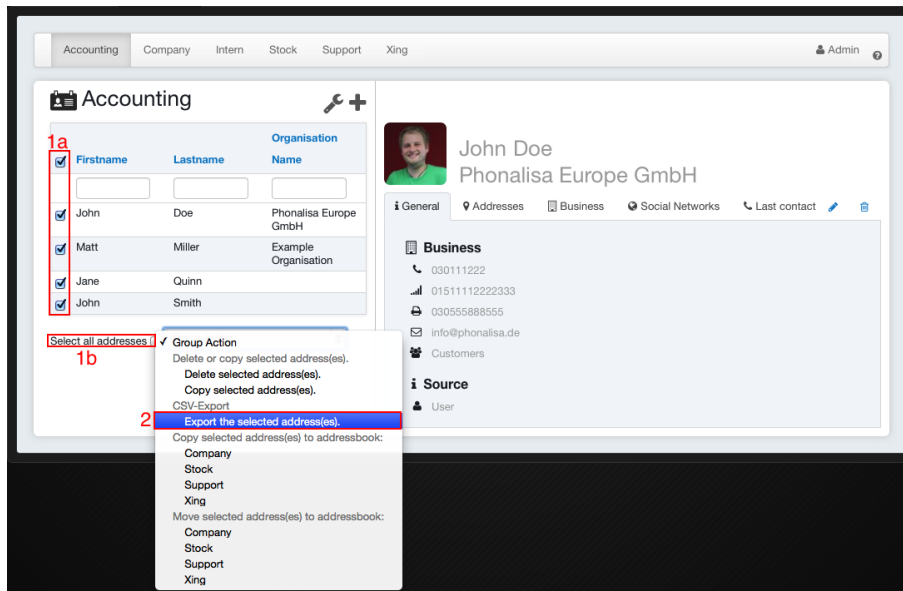
8 The CSV import of address data

At this point, you get a brief overview of the structure of a csv-file. The file can be opened in spreadsheet programs or in text editors, for example.

1. This is the header of your table. You can skip the first line, to prevent the import of header data.

2. Here you can see the entries. The table is very wide and shows only this small region. Further to the right there are still the information for the addresses and phone numbers.

8.1 Perform an CSV import

For the CSV import you need the permissions of the address book which you want to import data to (see item 6.2). Then go to the address book and click the wrench next to the plus button. The "CSV-Import" button will appear. Click it.
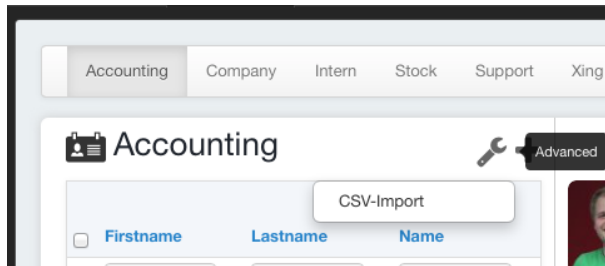
1. Select the file you want to import from your local files.

2. Click "CSV-Import".
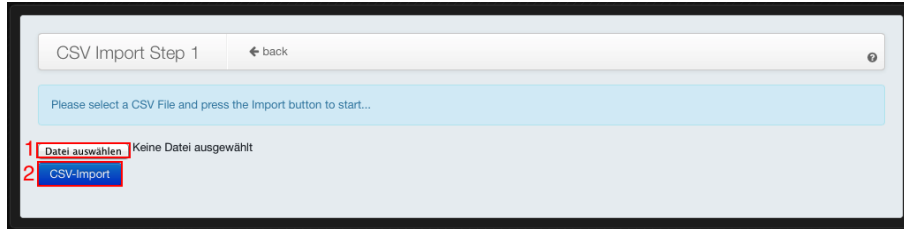
Now you can see a small preview of your data and can make some adjustments.

1. The assignment of the data can be changed using the dropdown boxes.

2. This is the header of the table, it is advisable to hide them during import.

3. Here you have a few examples from your CSV file and how they are assigned.
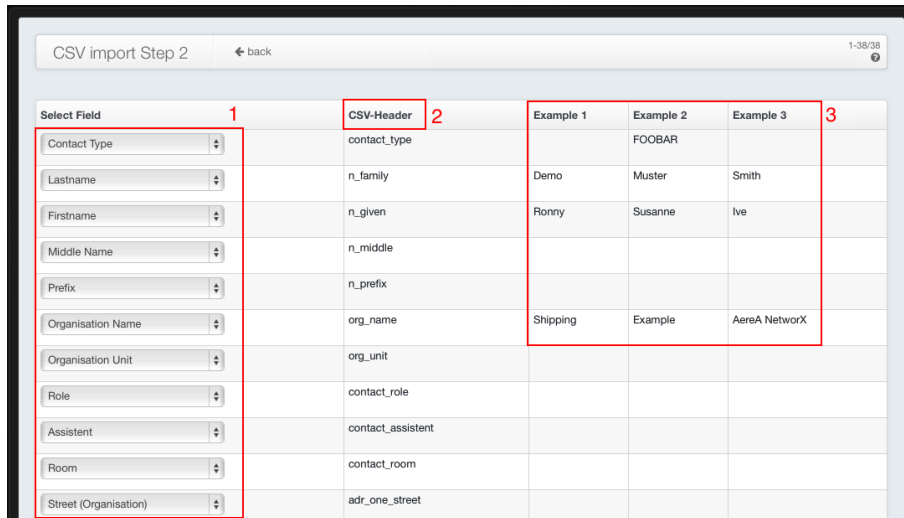
Use the dropdown box to select the encoding used in your CSV file (but UTF-8 is the most common). You can scroll through the various pages in the preview and adjust assignments if necessary. Finally, click "Run import!".
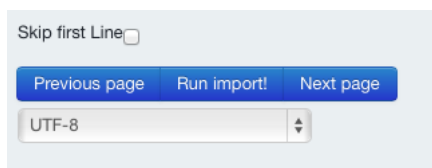
9 The social accounts

In the section "Social Accounts" in the "Settings" menu Phonalisa offers the possibility to import contacts from XING or iCloud into your Phonalisa address book. To use this feature simply click on the button "Add new account" button.

9.1 The XING Import

1. Select the Social Network.

2. Enter a description.

3. Select the address book for the import.

4. Click "Add".
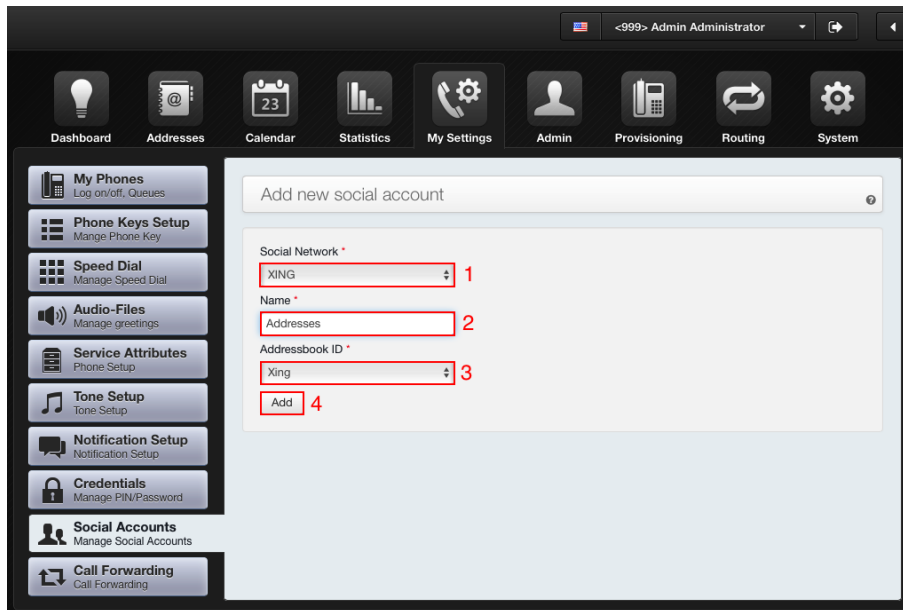
After clicking the "Add" button you will be redirected to the XING login. There you will be asked to sign in with your account information.
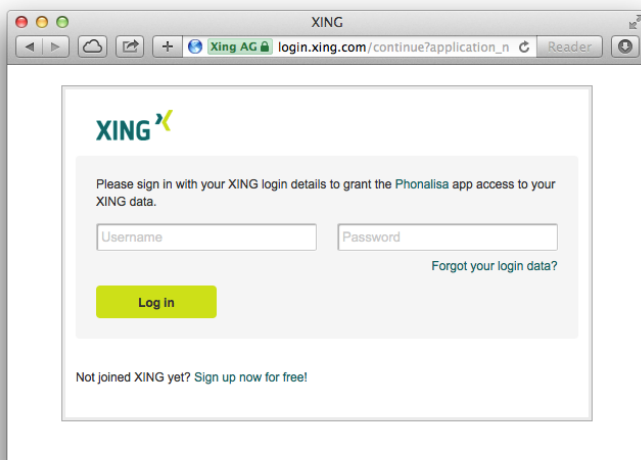
You will then be directed back to Phonalisa where you get an overview of your accounts in the chart. Click the "Import Addresses" button and wait for a moment, because this may take some time to complete (depending on the number of your contacts). Then you will find the imported contacts in your address book.

Verified	Name	Token	Secret	Addressbook	Token expires at	Updated	20 ⬍
Yes	My iCloud contacts			Privat	-1456223722 seconds		⇄ 🗑
Yes	✗ My XING contacts	37fe489d0757ccba01d1	ea86f2e0758cecdb4d83	Privat	-1456223722 seconds		⇄ 🗑

9.2   The iCloud import

1. Select the Social Network.

2. Enter a description.

3. Select the address book for the import.

4. Click "Add".
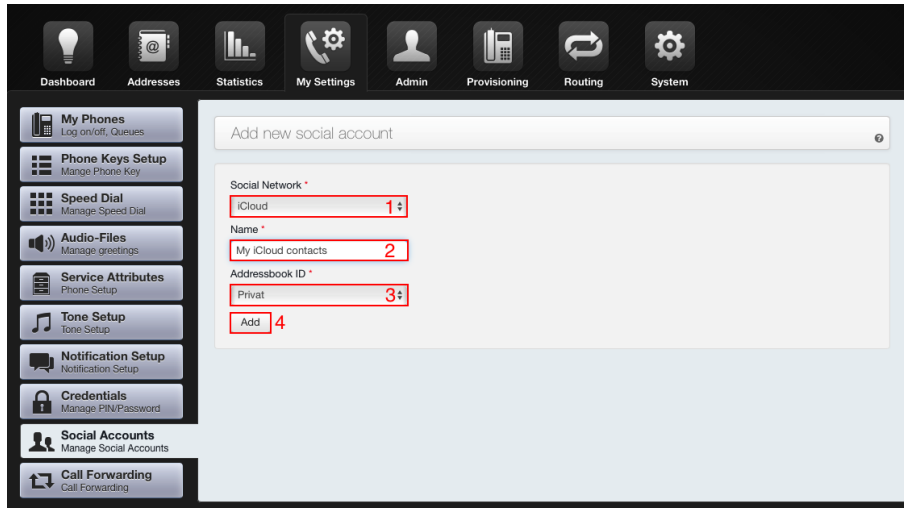
In case of an iCloud import, you will be asked to enter your Apple-ID (1) and password (2) in the following window.
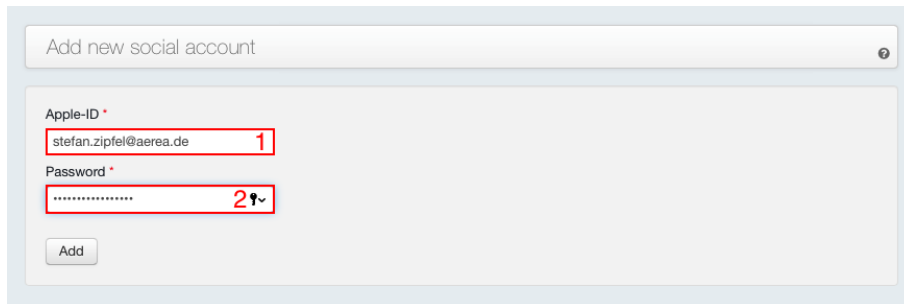
You will then be directed back to the Phonalisa where you get an overview of your accounts in the chart. Click the "Import Addresses" button and wait for a moment, because this may take some time to complete (depending on the number of your contacts). Then you will find the imported contacts in your address book.

Verified	Name	Token	Secret	Addressbook	Token expires at	Updated	20 ⬍
Yes	My iCloud contacts			Privat	-1456223722 seconds		⇄ 🗑
Yes	✗ My XING contacts	37fe489d0757ccba01d1	ea86f2e0758cecdb4d83	Privat	-1456223722 seconds		⇄ 🗑

10 The Calendar

Here you will get an overview of the structure of the Phonalisa calendars.

1. In this line you will find the calendars which you created. These can be shown and hidden. If you hide an calendar, only the events from other calendars will be visible.

2. With the button you can change the view of the calendar.

3. Depending on the representation of the calendar (2) , it is possible to scroll through the month, week or day with the arrows.
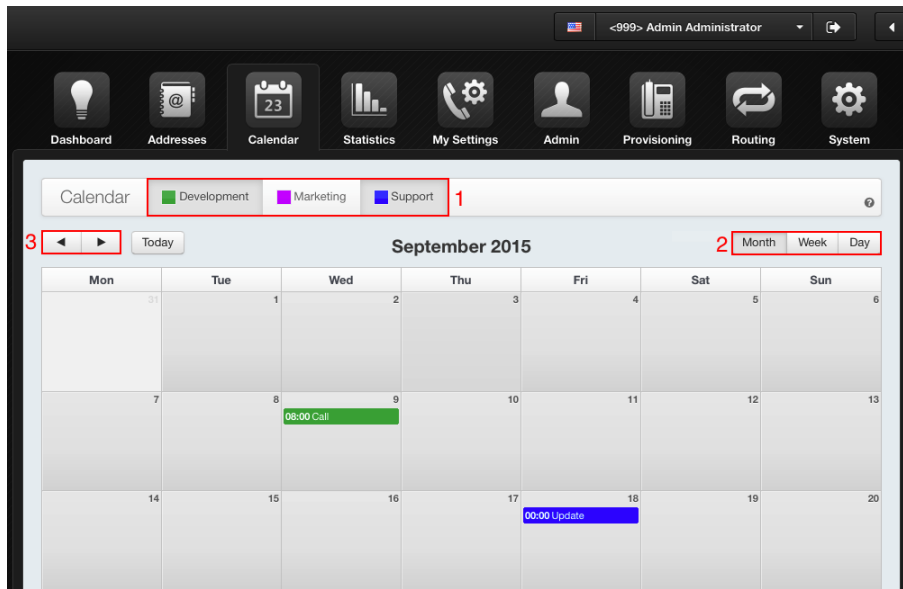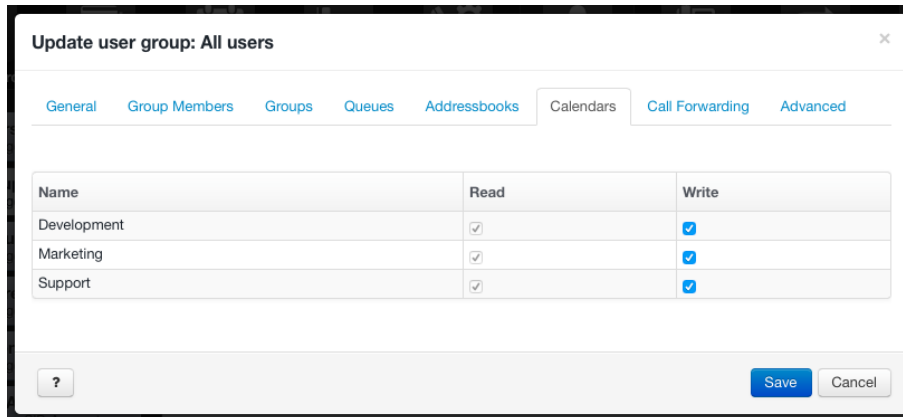
10.1 The calendar types

There are "Private" and "Group" types. Entries of a "Private" type calendar of can only be seen and edited by yourself but will be assigned to a group. It will allow users of this group, to create and view private entries in their calendars. A calendar of type "Group" is designed to allow all entires to be seen, created, and edited by the group's members in the calendar.

10.2 The calendar permissions

There are read, write permissions which can be assigned to existing groups. The inheritance ability of groups plays a crucial role for the assignment of permissions . Since a "Child-Group" inherits from its "Parent-Group", the child group automatically inherits the appropriate permissions. The read permission ensures that a group only has read access to a calendar and will not be able to write or delete entries. If a group has read and write access to a calendar, they may create, edit, delete, move and copy any entries. The permission of synchronization is, for example, only useful for Xing calendars because the entries change frequently.
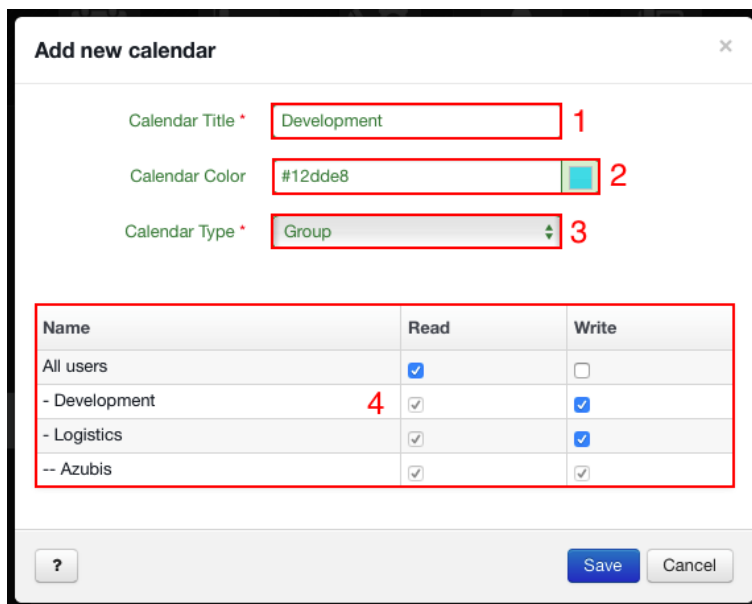
10.3 Creating a new calendar

Go to "Admin" -> "Calendar" for this:

Click "Add new calendar".

1. Enter a name.

2. Choose a color by using the color picker which helps you to differentiate the entries.

3. Select a type.

4. Now you can assign the permissions. After saving, the new calendar is available.
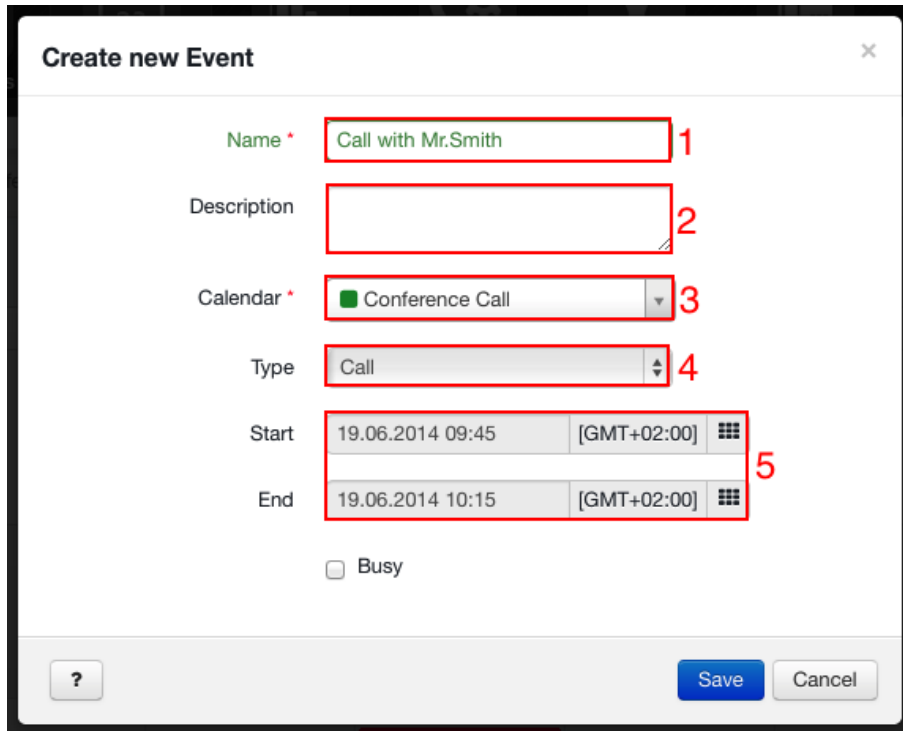
10.4 Create calendar entries

Go to "Calendar". Select the day. You can change the view using the buttons month, week, day. Click the day or the time at which the event starts. If you hold down the mouse button, you can drag the entry from the start time to the end time. When you release the mouse button, the entry appears in the dialog box:

1. Enter a name for the appointment.

2. Optionally, you can add a description, such as discussion topics, objectives of the appointment or an address.

3. Select the calendar.

4. Select a type.

5. Here you will find the selected start and end dates again and the corresponding times listed.

You can move the created entry to a different day and different time by "Drag & Drop".

10.5   Edit the calendar entries

To modify the date or time subsequently:

To edit calendar entries click the entry. In the opened editor window, you can edit the date and time.

Step-1: Change date (day) or retain it.

Step-2: Select hours.

Step-3: Select minutes.

You can select the date step by step.
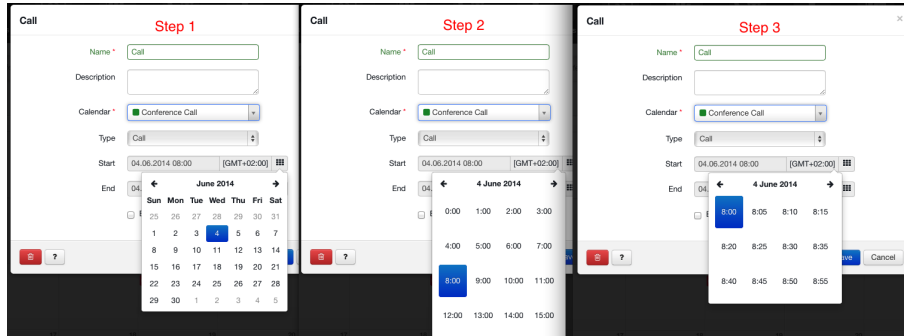
Change calendar type afterwards:

To edit an appointment click on it. You can select a pre-set type or create one by yourself. Just click "Add new Event Type" in the dropdown list "Type". Enter a new type and save. The new type will now appear in all other entries you create or edit. Only when you delete an appointment in which the user-created type was used and there is no further appointment with this type it is automatically deleted.
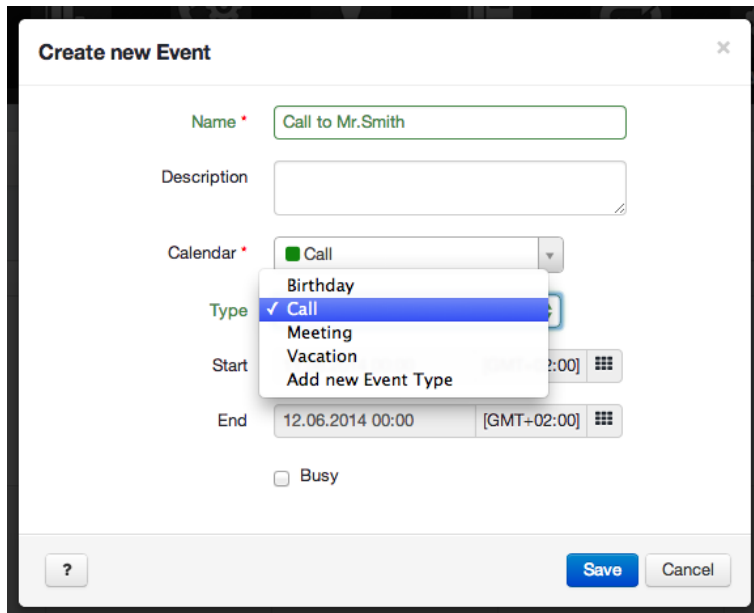
11 The Statistics

The statistics are used to monitor and evaluate group calls and queue calls from extern. Based on the statistics you can schedule employment and the pause time of your staff more efficient. You now see exactly when and how your personnel is reached best. With Phonalisa you can evaluate individual groups and individual queues separately.

11.1 The permission for evaluating the statistics

Before you can view your call statistics, you have to distribute the permission by proceeding to "Admin" -> "Groups".

Go to the tab "Queues". Here you can select the queue you would like to evaluate.

In the tab "Groups", you can select the groups you want to evaluate.

The selection you made will be displayed in the dropdown boxes of the statistics.

11.2 The daily analysis

You can see exactly how many external calls were successfully answered and missed a day. Please note that internal calls are not included in the daily analysis.

1. Choose the group or queue that you want to evaluate here. Important: You must have the permission to view the statistics (see point 11.1).

2. You can select whether the report should display a day, a week, a month or a year.

3. Depending on the selection in point 2, the display changes. You can then choose between weekdays, calendar week, months and years. After that, the selected statistics are displayed.

4. Here you can make fine adjustments to the data display.
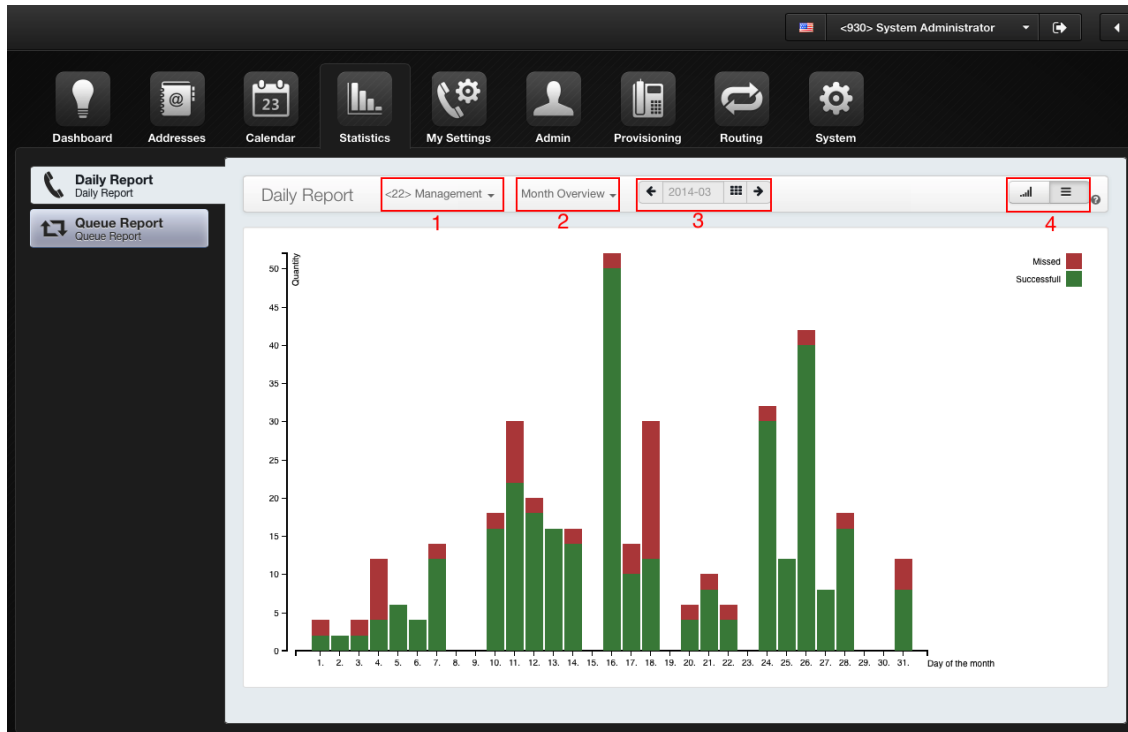
11.3 The queue analysis

The queue analysis allows you to see all your queues compared to each other in a glimpse.

1. You can choose between day, week, month or year view.

2. Depending on the previous selection the day picker changes. Then you can choose between weekdays, calendar week, months and years. After that, the selected statistics are displayed.
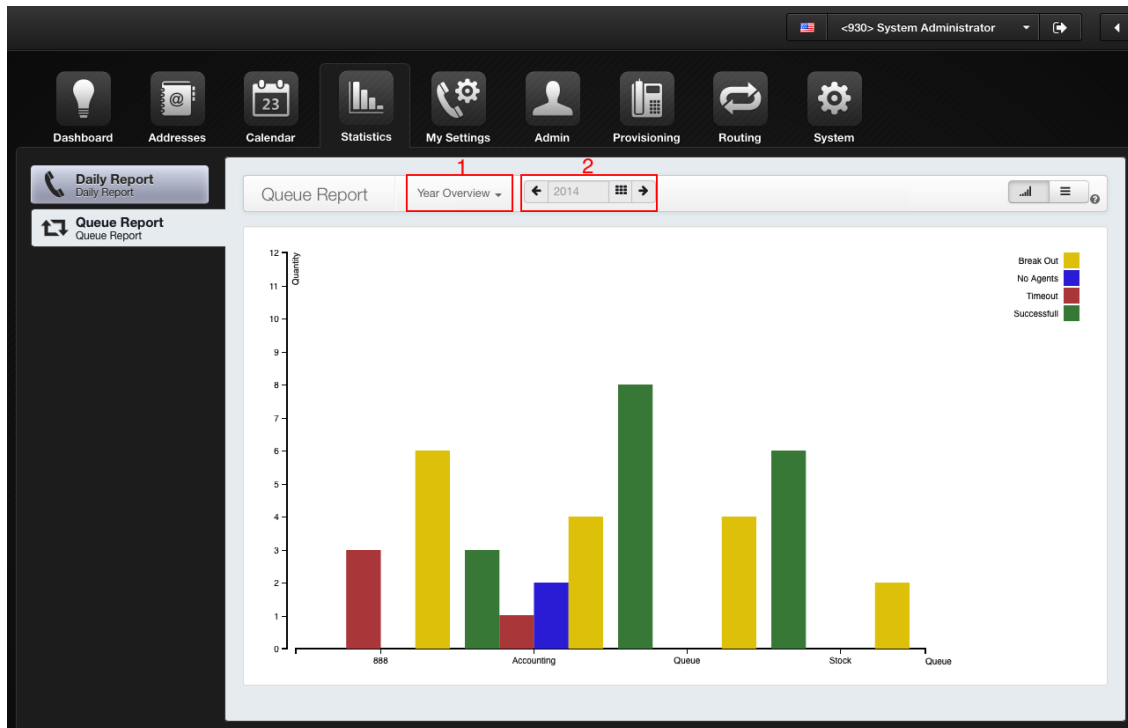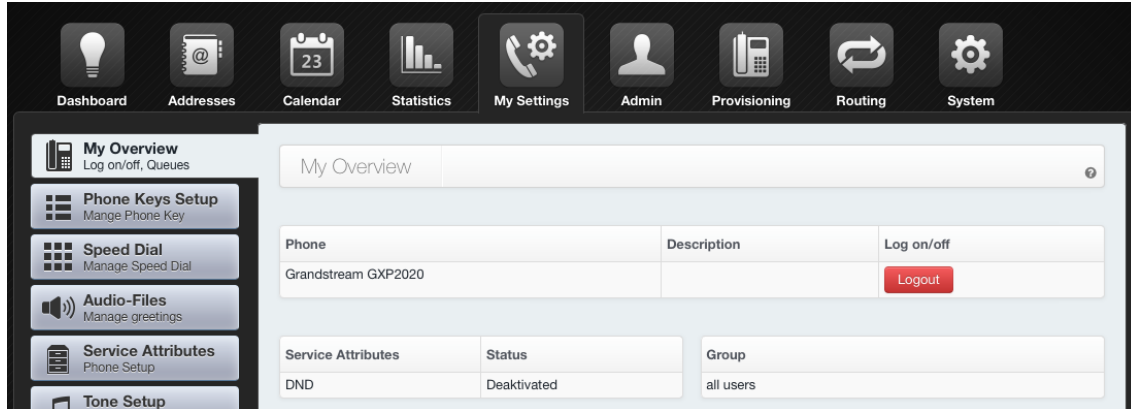
12 My Settings

In the following points you will find all setting options of your own phone.

12.1 My Phones

Here you can see all your phones listed. You can directly log these on and off via the web interface. You also see in which queue you are a member and whether you are logged in. If you are not logged in to a queue, you can do this here.

12.2 The Phone Keys Setup

1. Choose the phone you want to make the key assignment for. Depending on your phone model, there are several phone key functions.

2. Select the function.

3. Enter the target number for the function.

4. Here you can enter the label. This column is missing on devices without a graphical representation of the key assignment, .

5. In the column you will see if a button is disabled (i.e. you have an active key profile).

6. Click "Save & Restart Phone" to apply the changes. Depending on your phone model, it will either completely restarts or it take the changes without rebooting.

12.3   The Speed Dial

Dialing phone numbers via the speed dial function, you can save a lot of time. The speed dial feature code can be set in "Configuration" -> "Dialplan".

How to create the speed dial list:

1. Specify the speed dial in this field.

2. Here you enter the complete phone number, including an external line (if available).

3. Enter here the description.

4. Click the plus button to add additional row.

5. Finally, save everything.

12.3.1 The import of speed dial lists

You are also able to import complete speed dial lists to save yourself a lot of time. The imported file must be a .json file. This file needs to be exported from a Phonalisa telephone system.

Procedure for applying:

1. Select the "Import" button.

2. Here you can define the behavior in casse of already-used numbers. You may overwrite or skip (so they are not present twice).

3. Select a .json file from your files.

4. Click "Upload" to complete the process.
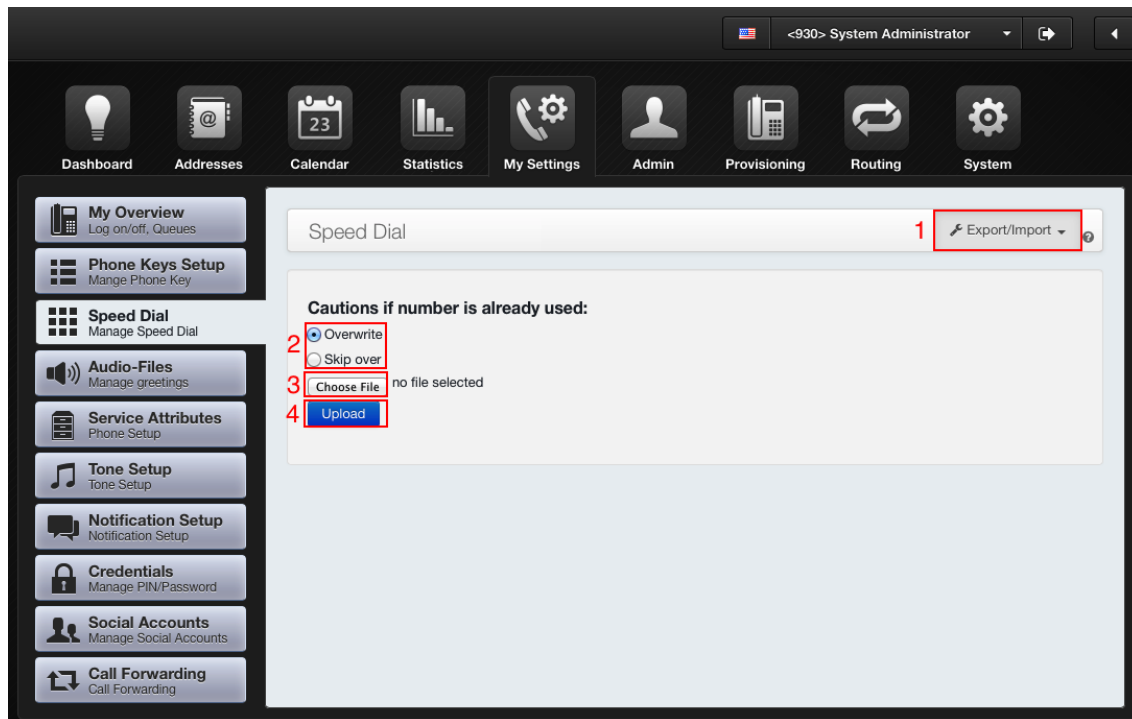
12.3.2 The export of speed dial lists

Select the button "Export". The file is then downloaded into your download directory.

12.3.3 The export of speed dial lists as PDF

To create a PDF document including the speed dial list, click the button and select "PDF-Export". The document is opened in a separate browser window and can be printed there.
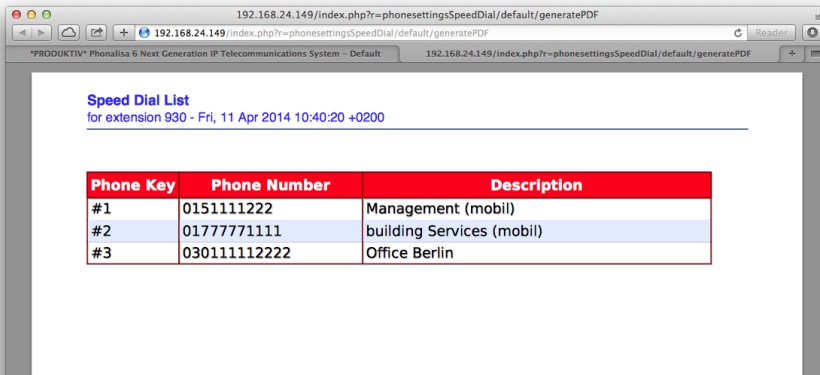
12.4 The Audio-Files

There are audio files and system audio files in Phonalisa. To learn more about the system audio files, see point 34. In the audio files you can set own announcements for call forwarding. Go to "My Settings" -> "Audio-Files" and click "Add new Audio-File" button.

1. Enter a name for the announcement.

2. For better distinction of the various messages, it is recommended to enter a description.

3. Select a message file (.wav format) from your own files or

4. Click the "Start Record" button. Then your phone will ring and you can record your personal announcement.

5. Save your announcement.

Here you have a list of your audio files. You can also edit (pen button), delete (garbage bin button) or download each audio file.
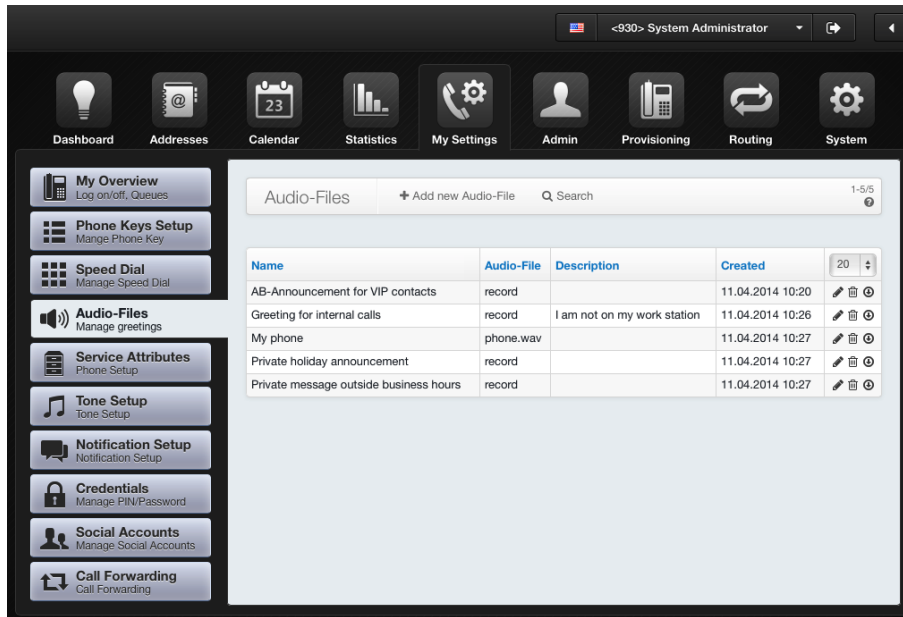
12.5 The Service Features

In this area you can make various settings for your phone. But notice that, not all functions are supported on all devices.

Service attributes:
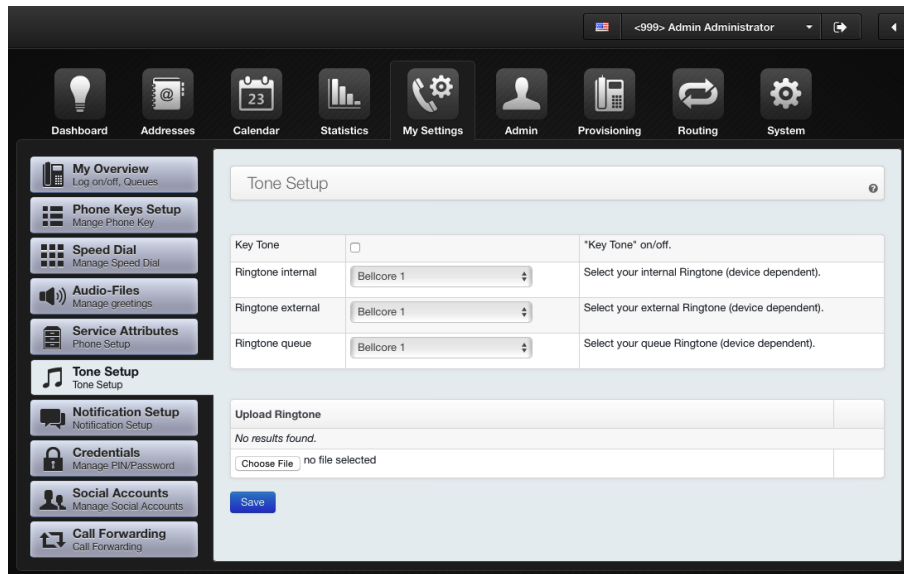
- CLIR for external calls: CLIR stands for "Calling Line Identity Restriction". When this feature is enabled, your number will not be transferred for external calls.

- CLIP for internal calls: CLIP stands for "Calling Line Identification Presentation". Here you can set a phone number which is transmitted for internal calls instead of your normal extension number.

- CLIP for external calls: CLIP stands for "Calling Line Identification Presentation". Here you can set a phone number which is transmitted for external calls instead of your normal extension number.

- Call waiting: If this function is enabled, in busy case you will hear a call waiting signal in your receiver. The caller gets signaled a dial tone. If this function is disabled, the caller gets a busy signal. Note: The second caller will only get a busy signal indicates - if you do not have "call forwarding on busy case" enabled.

- DND: The function means "Do Not Disturb". If it is activated, your phone is permanently occupied. Calls will not reach you - until you turn it off again.

12.6   The Tone Setup

In this menu you can adjust all sound settings of your phone. The support of some function are depended on the phone model.

The key tones can be enabled or disabled. You can set the ring tone for internal calls separated from the external. For this, you can use an uploaded ringtone or select one from the dropdown lists. The phones of snom and Tiptel also allow the sounds of the microphone and speaker to be customized. In the section bellow you can upload an MP3 file if this function has been activated by the administrator. This is then also listed in the dropdown lists.

Ringtone to your queue calls, can also be configured here.

12.7 The Notification Setup

If you would like, your Phonalisa telephone system will send you an email, if you missed an external call, received a fax or a voicemail. The individual notifications can be enabled or disabled. You can also specify whether the original voicemail will be deleted after sending.
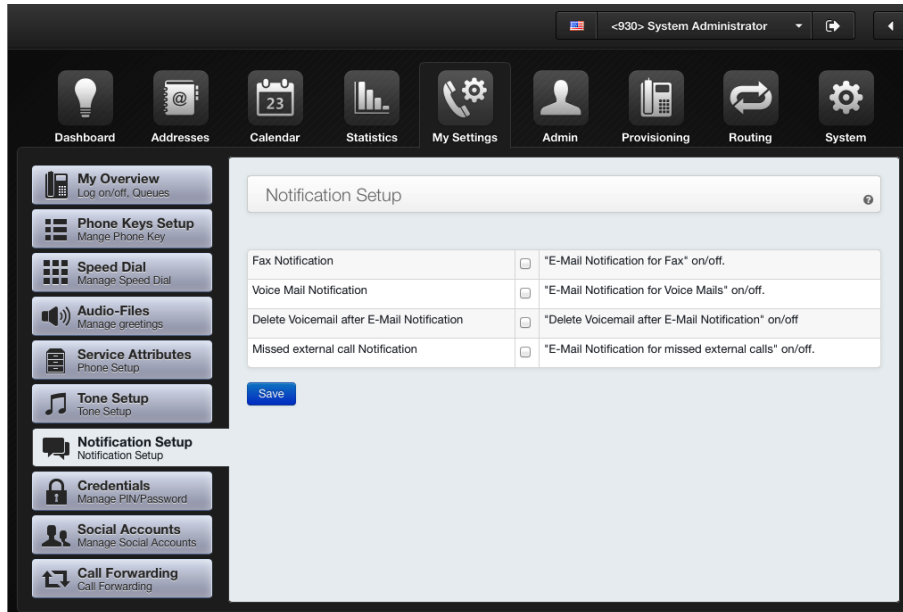
Note: The e-mails will contain the fax as a PDF file and the voicemail as a wav file.

12.8 The Credentials

Here you can change your password and your PIN. The password is used, to log in to the Phonalisa web interface. The PIN, for example, is required to listen to your voicemails or to log in on the phone.
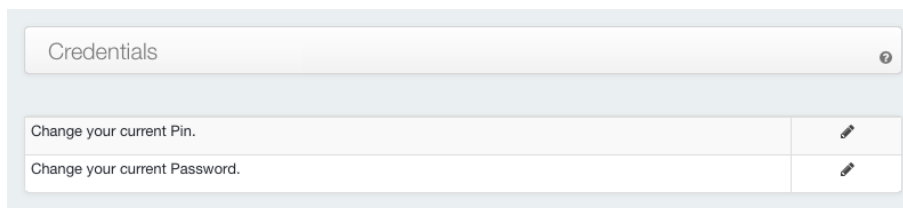
13   The Call Forwarding

The call forwarding system in Phonalisa is so flexible that, with a well thought configuration no calls go unanswered. It can define call forwarding scenarios for each user (user-defined). In adition to that an immediate forwarding to voicemail, an announcement, a phone number or no call diversion is also selectable. The customized call forwarding allows the user to choose from a variety of filters or combine them.

13.1   The call forwarding rules

The following filters are available:

1. Phone status: The phone has been switched off, busy or no answer after x seconds.

2. Jabber status: This rule refers to an instant messenger. The cases can be online, away, do not disturb or offline.

3. Time rules: Use date/time profiles and redirect a variety of cases like outside of business hours, within the holidays, for example.

4. Caller has the contact types: For each contact in your address book you can create a contact type. Then you can select this as a filter in your call forwarding.

5. Call source: Calls from internal and external can be redirected separately.

6. Night circuit: With this filter you can redirect calls depending on the night circuit. The night circuit can be enabled and disabled for each group.

In the next few points you will learn, how you can select individual filters and combine them.

13.2   Call forwarding disabled

This scenario can be created by using the System Call Forwarding templates and will then be selectable for different groups. Details can be found in point 20.

13.3   Immediately call forwarding to voicemail

Click "Add new call forwarding scenario" (1).
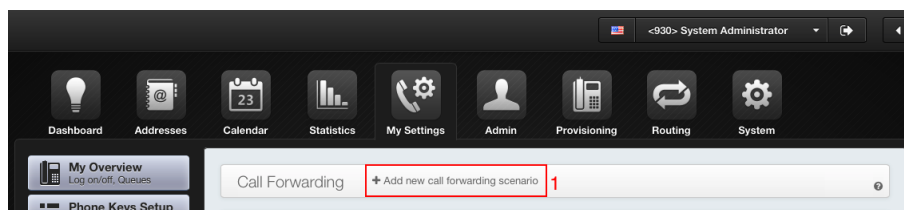
1. Enter a name and optionally a description.

2. Select "Immediately" as call forwarding type.

3. Click "Create Template".
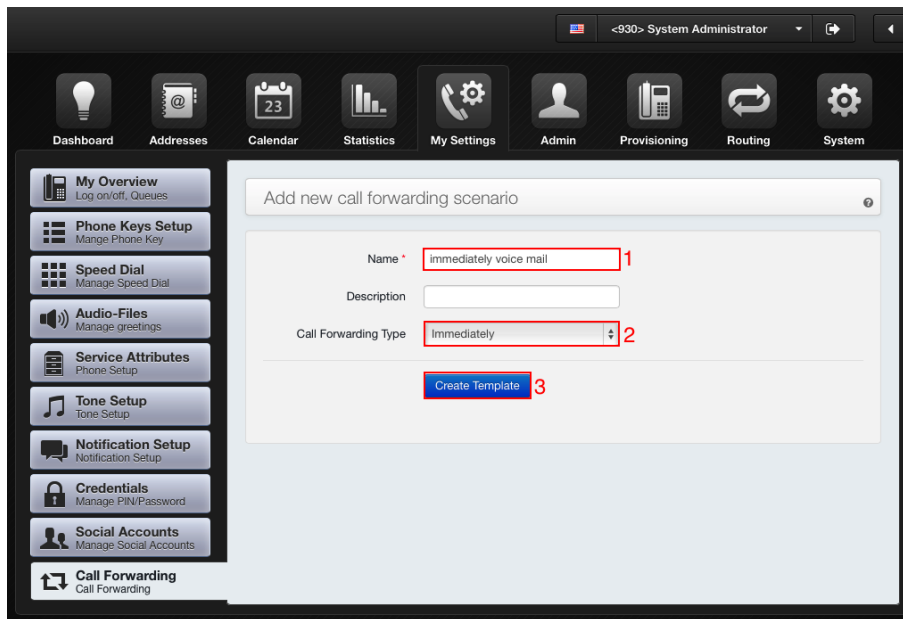
1. Select the voicemail in the dropdown.

2. Determine the appropriate announcement.

3. Save your entry.
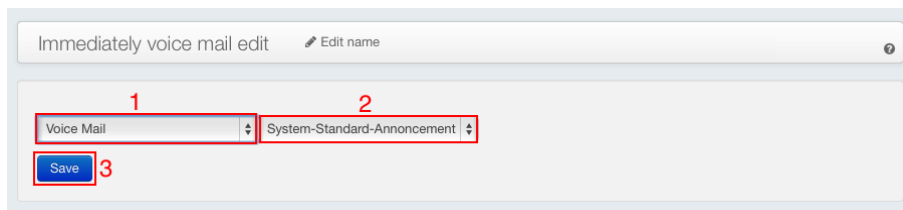
Warning: **You can add more personal announcements in "My Settings" -> "Audio-Files" (see point 12.4) or use the system default announcement (extension number not available).**

13.4   Immediately call forwarding to a phone number

Create a new scenario and select "Immediately" in the dropdown after entering the name. Select from the dropdown list "Phone number" and enter the phone number or extension number. You can also forward to a phone number of a conference room, an queue, an group, an IVR-menu, an external phone numbers or extension number of an external host. Please note that if you enabled the outside line in your gateway, the outside line must be preceded at the external phone numbers / mobile numbers, e.g. 00151... .

13.5   Immediately call forwarding to a announcement

Proceed described equally as in the previous point before. Select "Announcement" in the checkbox and determine the appropriate announcement.

Warning: You can add more personal announcements in "My Settings" ->"Audio-Files".

13.6   User-defined call forwarding

In the custom call forwarding, the individual filters are used. You can create a separate rule for each filter or combine multiple filters together.

13.6.1   Redirecting by using the phone status

Create a new scenario and select the "Custom (Rule based)" call forwarding. Select the phone status "Busy", "Call Time Out after" and/or "Signed Out" from the dropdown and forward it to a phone number, voicemail or announcement.

13.6.2   Redirecting based on the presence status

The call forwarding system is equipped with an intelligent presence function. The Instant Messenger helps to improve the communication within the company. You write a short message instead of making long phone calls or even writting emails. Thanks to the presence function you will always know exactly whether a colleague is currently in place but better not be be disturbed (status busy) or ready for incoming calls (status online). Through the Instant Messenger feature the call forwarding system was extended enormously and is now much more flexible.

Add a new scenario with a custom call forwarding and edit the rules. Select the filter "Jabber status" and activate one or more cases (online, offline, busy, ...). Define a forwarding destination and save.
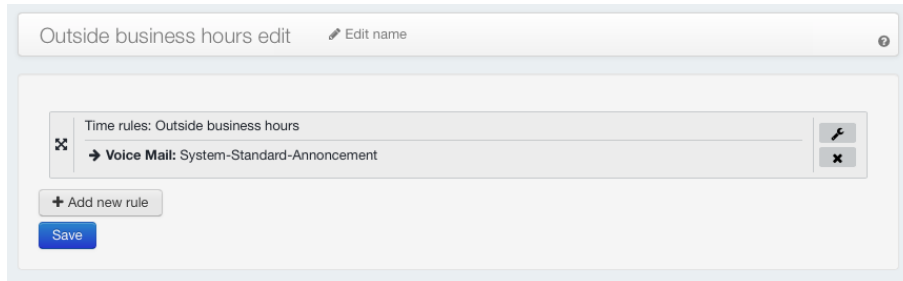
13.6.3 Redirecting with the date/time profile

The date/time profiles are versatile (details see 33). The following scenario explaines how the time rule *"Outside business hours"* works with the call forwarding.

Create a new scenario with a custom call forwarding. Activate the filter "Time rules" and determine a profile from the dropdown box. Define a destination and save the rule.
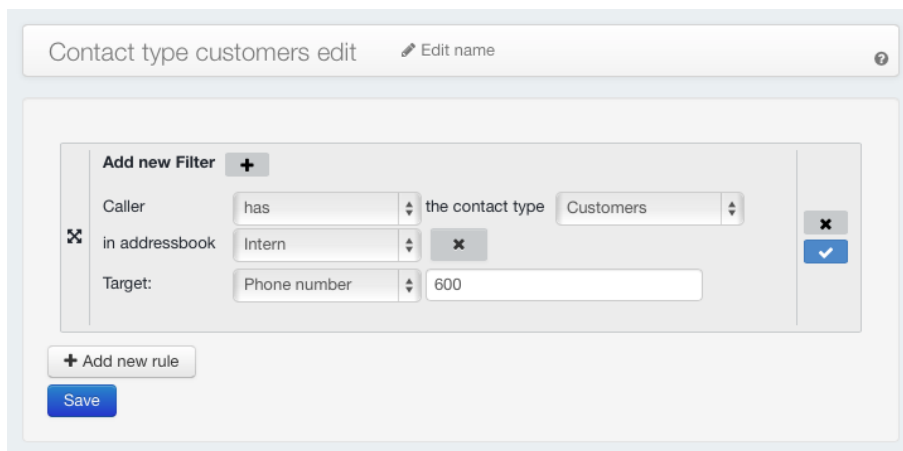
13.6.4 Redirecting based on the contact type

The usage of the filter "Caller has the contact type" gives the user new possibilities. You can select whether the caller is or is not of this contact type. Since a contact can also be stored in multiple address books, you also can select the address book.

Create a new scenario with a custom call forwarding. Activate the filter "Caller has the contact type". Now you can choose between "has" und "has not", then select the contact type and determine the address book in which the contact occurs. Define a destination and save the rule.

13.6.5   Redirecting based on the call source

If you want to redirect external calls only, you can do this with the filter "Call source". For example, you can redirect external calls to a IVR but the internal calls will still reach you.

Create a new scenario with a custom call forwarding. Activate the filter "Call source" and select internal or external. Determine a destination and save the rule.
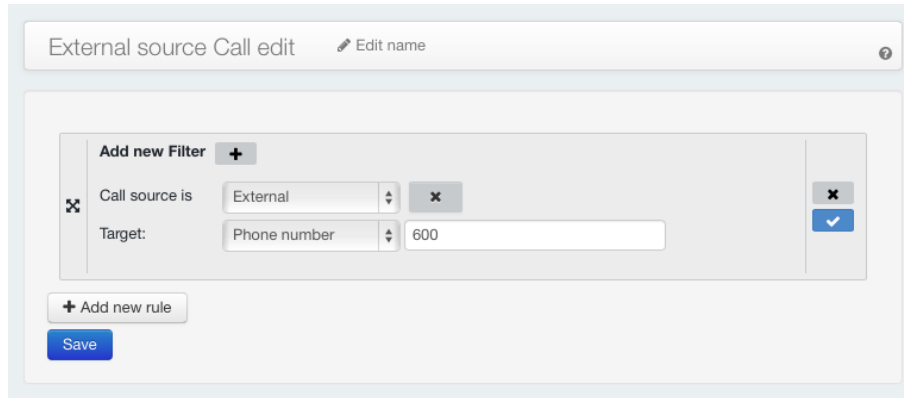
13.6.6   The combination of the filters

The combination of Jabber and contact type:
In this case you will see that calls are automatically redirected if the users status in Jabber shows away, busy or is offline . In addition, even the contact type is considered. If the caller's contact type is "VIP" calls will be forwarded to the user's cell phone. Since this rule has a higher priority than the jabber rule above the user should shift the second rule by drag-and-drop over the first rule.
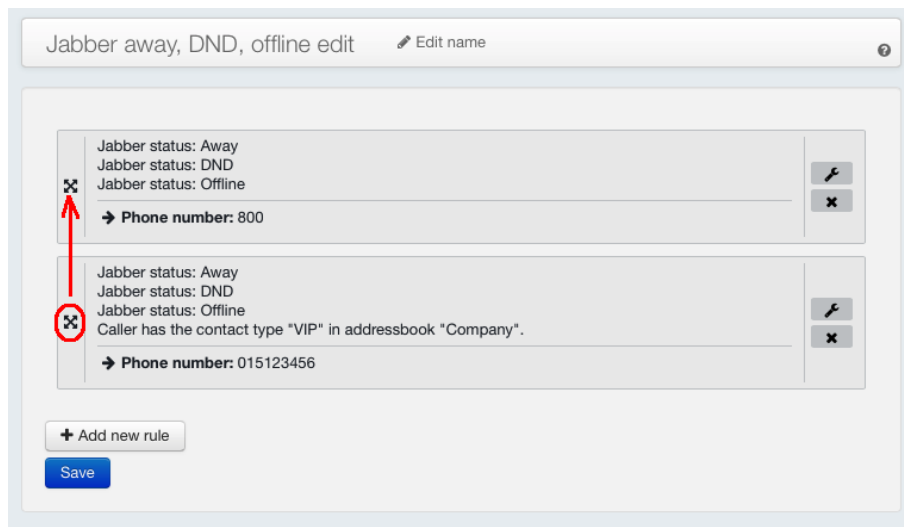
The combination of contact type and time rule:
Example: You have activated a time profile in your call forwarding and there are customers/partners who are allowed to contact you outside business hours on the main business number. For these "VIPs" you are able to create a contact type which is allowed to reach you even on weekends, at night or your cell phone.

Create a new scenario with a custom call forwarding and edit the rule. Add the filter "Caller has the contact type" , and choose "has". Then select "VIP" from the dropdown box and the corresponding address book. Now add the filter time rules, and select the profile "Outside business hours", put in your cell phone number and save the rule. If a "VIP" contact type is calling your cell phone rings. You can create another rule with the same time profile for all other (non-VIP) contacts and forward it, with an absence-announcement, to your voicemail.
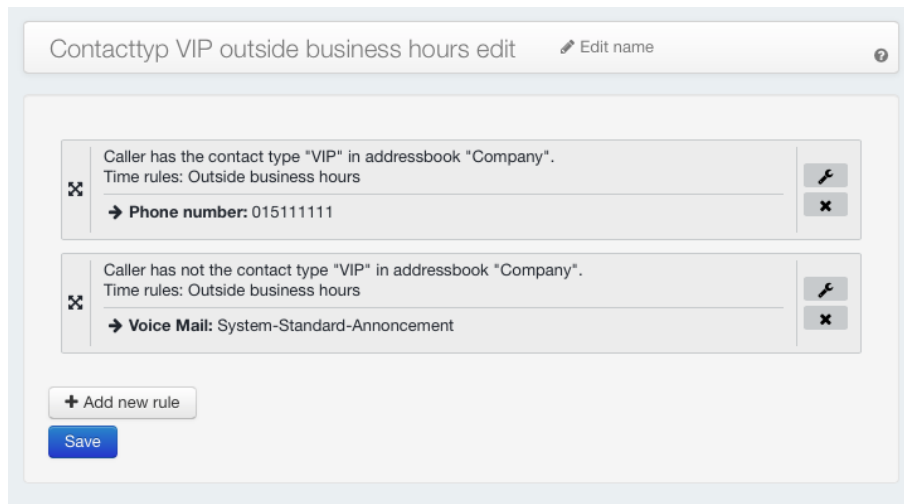
13.7   Additional function: toggle forwarding scenario

If you want to switch your call forwarding rules on or off via hotkey just add a speed dial in front of the corresponding scenarios. The definable function buttons are found in the call forwarding table in "System" -> "Configuration" -> "Dialplan". (In this case *2). Just add a number after the *key prefix. To activate the call forwarding you just need to dial this number with your phone, for example *22 for the "I'm in the office" forwarding.
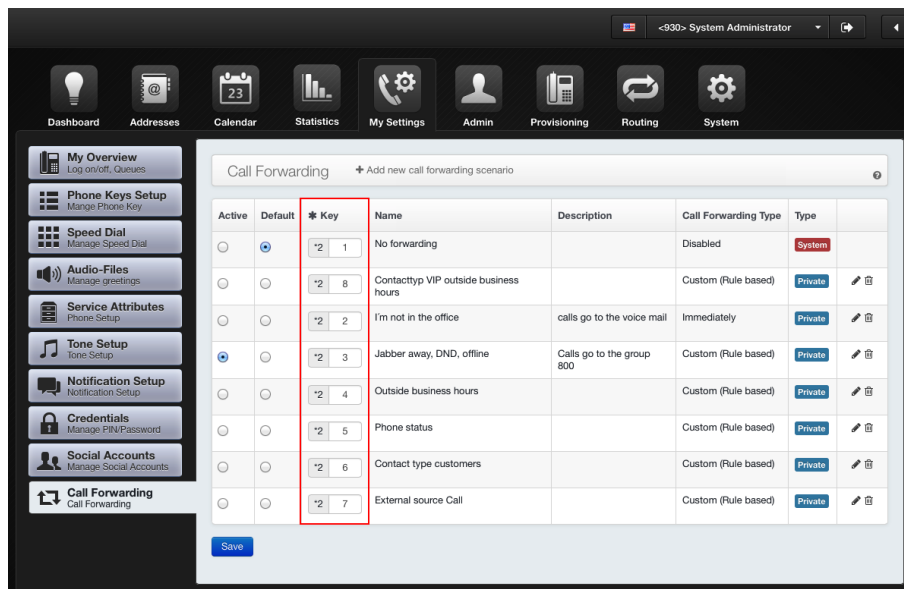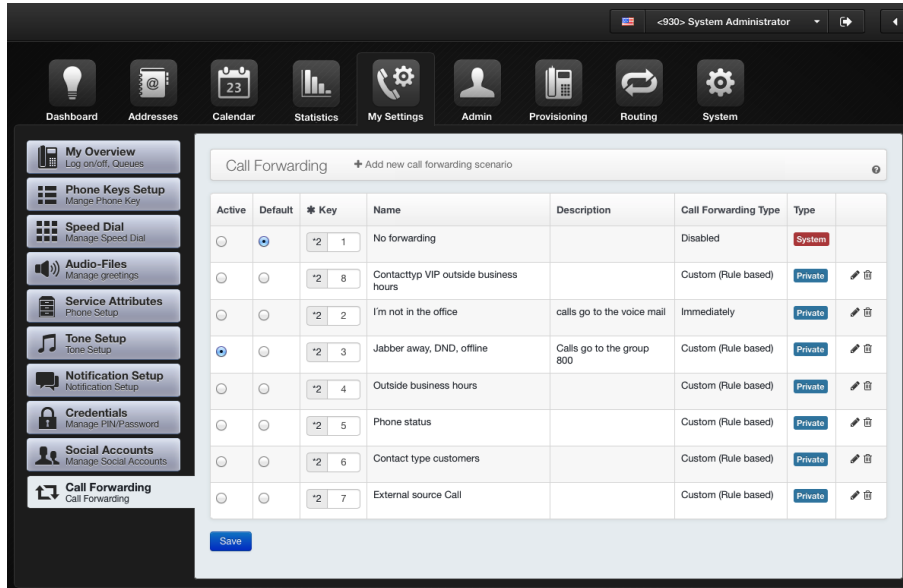
13.8 Additional function: Set call forwarding to active and standard

In your call forwarding scenarios, you can set one rule to "Active" and one to "Default". The "Active" state tells you that the selected call forwarding is active and the "Default" state occurs when call forwarding is disabled.

An example for this:
The rule "Jabber away, DND, offline" is activated and "No forwarding" is set to default in this case. Now dial *21 on your phone or click "no forwarding" in the web interface then the status is changed to "No forwarding". At a glance you see quickly which rule is currently active.

14   The Voice - Mailbox

Access via web interface or by phone
On the dashboard, there is a tab called "Voicemail". You can delete messages (by pushing the garbage bin),
download them via download button or play directly in your browser.
If you want to listen to your voicemails on the phone, dial 80 (the number can be changed). Depending on how
your system is configured, you have to enter your PIN to listen to the voicemails.
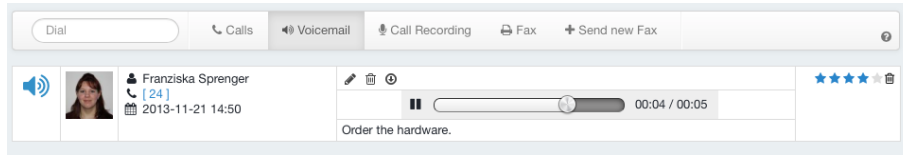
You can listen to voicemails not only on your own but from any other phone in the company too. In some
companies a PIN has to be entered for security reasons and to prevent abuse to use this function. To activate
this feature, go to "System" -> "Configuration" -> "Dialplan". Activate "Always ask for PIN to access voicemail".

Voicemail notification via email
If "Voice Mail Notification" is activated in "My Settings" -> "Notification Setup", you will be notified about incom-
ing voicemail by email. You can check these message on your dashboard.

Display on the phone (LED)
This depents on the particular type of phone.

15   The User management

The user administration offers a wide variety settings. The detailed overview shows you how to combine them. In advance you will find a small collection of links with information about used the terms:

- Create and edit groups, see 16.2

- Create and edit queues, see 28.1

15.1   The general settings

<u>Create, edit and delete users</u>
To create a new user click the button "Add new user"(1) but you can also import or export an user profil via CSV .
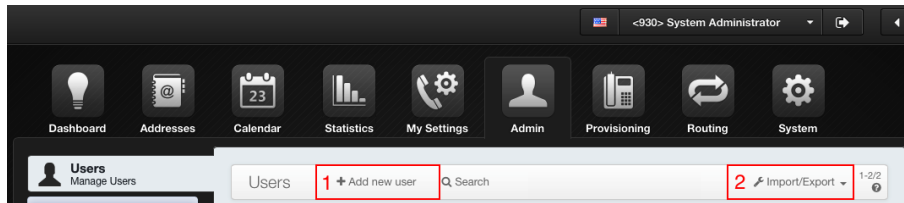
You can enter your values in the dialog window (see 4.2 for more informations). To edit or delete a user click the button at the end of the row. Once a new user is created, it will automatically be added to the internal address book and presence monitor. The password is required to log in to the Phonalisa web interface, and the PIN to verify on a phone or to check your voicemails. If an email address is entered you can also activate the notification to receive voice messages or faxes. You find Information about the configuration of notification in 12.7.

<u>Assign a user to a group</u>
Through the user management ("Admin" -> "Users" -> Update), you can select a group from the dropdown box.
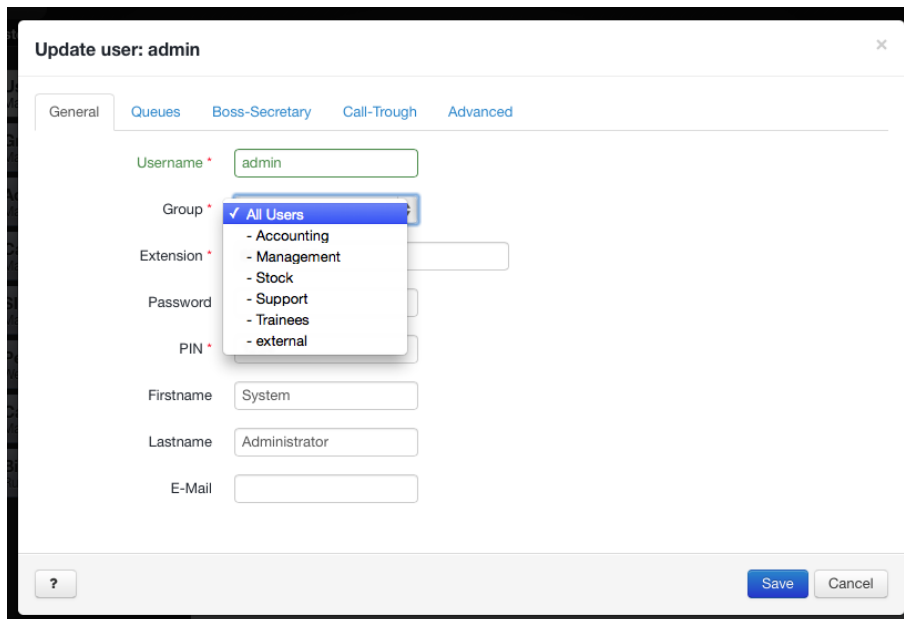
Also update the appropriate group through group management and select the corresponding user with the dropdown box (see point 16.3).

15.2   Assigning a user to a queue

Edit the user and select the tab queues to do this.

1. Select from the dropdown box the corresponding queue.

2. If "Static Member" is set, the user is immediately logged in as an agent and can take calls from the queue. If not, the user's phone does not ring until he logs in for queue calls.

3. The position and level to determine the order of incoming calls to this agent. See 28.5 for more information.

4. Here you can specify a timeout.

5. Click "+" button to assigned the user to the queue.

A user can be agent in to multiple queues.

15.3   The Boss-Secretary-Circuit

The boss-secretary circuit is designed for the secretaryto break through the call forwarding of the boss. If the boss has set a forwarding for himself, e.g. for external, internal, certain contact types or time based, the secretary can still reach him anyway. The administrator can assign any number of secretaries via the user administration. These can reach him on his internal extension without having to dial a special key combination.
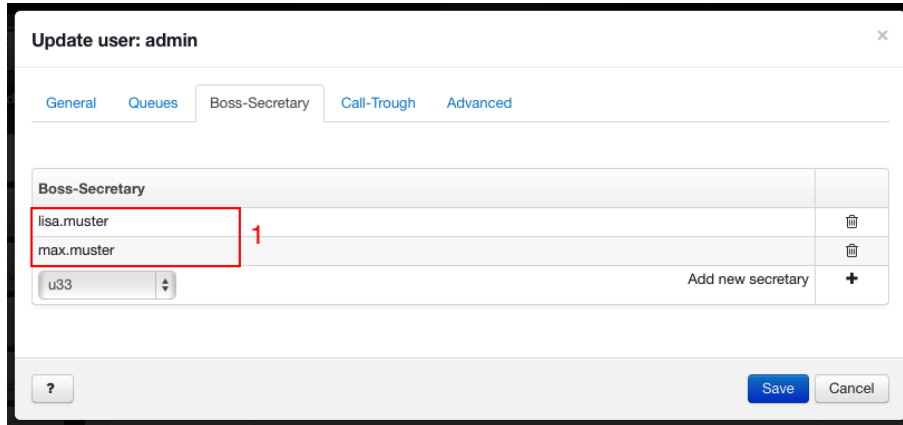
Edit the user. The settings are in the "Boss-Secretary-Circuit" tab.

1. Select the secretary from dropdown.

2. Click the "+" button to assign this user as a secretary.

1. You can also assign several secretaries to one boss or one secretary to several bosses.

The procedure of a call to the "Boss-Secretary-Circuit":
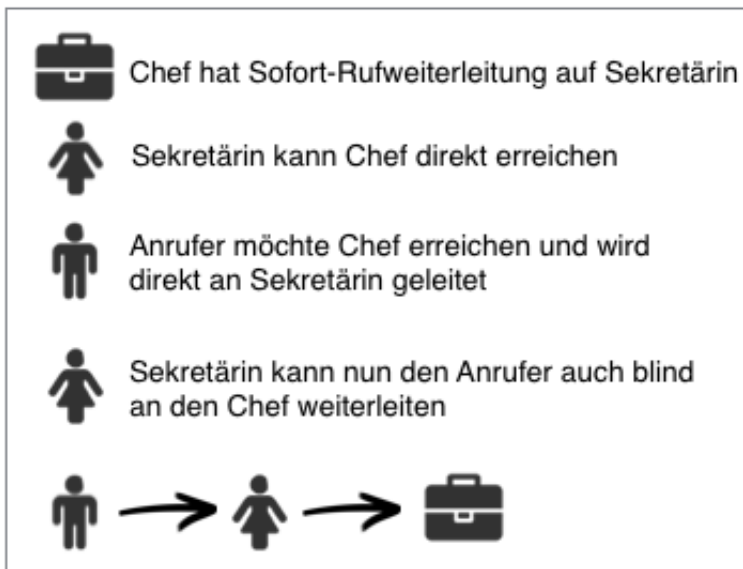
15.4   The configuration of external phone numbers for Call-Through

The user has to verify a phone number set with user administration (or several phone numbers) in order to use the telephone system external.

Example of usage: A user want to call a customer from his cell phone but the customer should not see his cell number. The Call-Through function was developed especially for this scenarios like this so the customer sees the number of the telephone system on his phone display. Procedure for implementation: The administrator determines in the administration of the user, one or more external phone numbers, e.g. a cell phone or number Home Office number. In addition, the Admin has to define a phone number from incoming routes with the aim of "Call-Through". Now the user can call the system and get a dial tone after which he can enter a phone number and will be connected.

Entry in the User Management:
Edit the user and select the tab "External numbers". Click to enter the "+" to a number.

1. Enter the phone number you want to authenticate to the system.

2. Activate the checkbox for the Call-Through feature.

3. If enabled, the selected phone number will be called parallel for incoming calls to your extension with.

4. Click on the "+" button to add another number.

Configuration in the user management:
Edit the user and select "Call-Trough". Click the "+" button to enter a phone number.

1. Enter the phone number to which you want to authenticate with the system.

2. Activate the checkbox to verify the phone number.

3. Click the "+" button to add another phone number.

Entering in the routing:

1. Enter the phone number or pattern from which you want to call the external line.

2. Select "Call-Through" as a target and save the routing rule.

Sequence of an example call:

1. A User has authorized his cell phone number 0171-111222 in the system.

2. He now calls the 760825 with his cell phone from an external line and gets a dial tone.

3. Now he can choose any internal or external number and send the extension from his office.

15.5 The advanced user settings

Summary of the advanced setting options (Part 1):

Language, time zone, time format, date format
You can adjust the language and time separately for each user. The default time format is the 24-hour format. If a user want a 12 hour time format (am/pm) you can select it via dropdown. The changes you made will be visible in the calendar, the time profiles and on the dashboard. These are possible settings "year-month-day", "day.month.year" and "month / day / year" for the date format.

Parameter and Key Profiles
Their can be Parameter and Key Profiles assigned for each user. These profiles are the herschenden in Phonalisa hierarchy
These profiles behave according to the in the Phonalisa prevailing hierarchy. This means group profiles can be overridden by user profiles and they can by specially created profiles unless it was not locked.
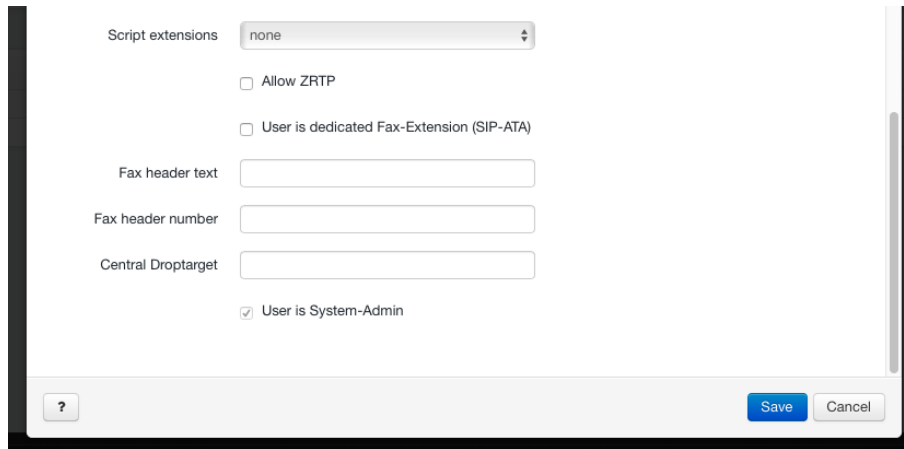
Announcement before answering internal / extern
You want to set an individual announcement before internal or external calls? Please select the appropriate announcement (only system announcement) in the dropdown box.

- Application example of an "announcement before answering":
  You work in a car dealership, then you can welcome your external caller with the following words, for example: "Welcome to xy, I will equally be there for you personally." So you can add a personal touch for the incoming calls and the caller knows that he has dialed the correct phone number.

Script Extension
Here you can select a Script Extension. It can be used to customize the system according to your own wishes. They are small pieces numbering plan that can be executed at certain points. The Script Extension modul can be found under "Routing".
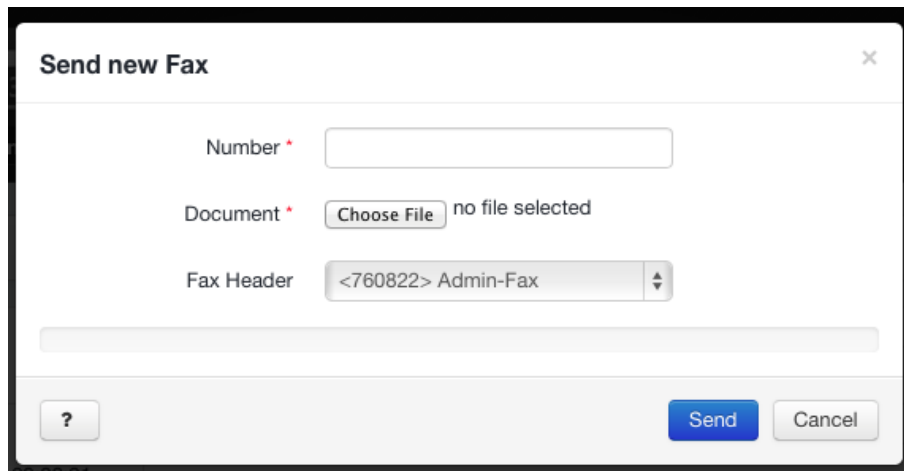
Allow ZRTP
It is used for call encryption within the telephone system.
ZRTP composed of „Z" (Phil Zimmermann) and „Real-Time Transport Protocol") is a cryptographic key exchange protocol for negotiating encryption of keys between two points of an IP phone call (VoIP). It uses the Diffie-Hellman key exchange, and the Secure Real-Time Transport Protocol (SRTP) for encryption.

Fax-Header-Text and Fax-Header-Number
For each user, a personal fax header text and number can be set. Once you have made your entries, you can send a fax from the Dashboard and select your fax header text and number with the dropdown.

Central intercept position
If the user missed a call and he had no corresponding call forwarding defined, the call is forwarded to the Central intercept position. Here you enter a phone number (e.g., from an group, extension number of another user, queue, ...) which will pickup the call.

User is System-Admin
By activating this option if you want to give a user all admin rights and he will become system admin. He will not be able to give up the privileges himself after that. Another admin is rquired for this. The system must always have at least one Admin, therefore it is not possible to delete the last admin.

15.6   Log in a user on an phone (Hot Desking)

To upload the appropriate user with all his configurations dial *0-user-extension on a phone. Therefore it is possible to change the workplace of an employees without moving the phone. His "own" phone will not be logged off at the moment and the additional "new" phone now contains all the address books, ringtones, phone call lists, etc. of the user. For example an employe can log in with his extension at the home office and continue to work with all the phones configurations.

16 The Groups management

The Phonalisa group system is used to groupe multiple users together. This group then can get its own extension and is reachable via routing. A lot of features for the group can be laid down and then be transferred via the "permission inheritance" to child groups.

The structure:

1. Here you can create a new user group.

2. In this area you will see your groups with the corresponding hierarchy. The arrow in front of a group indicates that it is a parent group.

3. If the group has its own extension, it will be displayed here.

4. The moon indicates an active night circuit of the group.

5. In this column you see how many users are assigned to a group.

6. If you click this button, the "Ringmap"-menu is opened.

7. This button opens an list of users in the group.

8. This button opens the dialog box for editing the group.

9. You can delete a group, by clicking on the garbage bin. The users in this group are then automatically moved to the parent group. The group "All Users" is the only group that can not be deleted.

16.1   The parent and child groups

Each created group can be a parent group. The rights of a parent group are inherited by the child group. This includes, for example, the read and write permissions for the calendars and address books.

Here you see your groups listed. The arrow in front of the name indicates a parent class. It is possible to change the group hierarchy by using "Drag & Drop". Just click on the group name and drag it on another group.
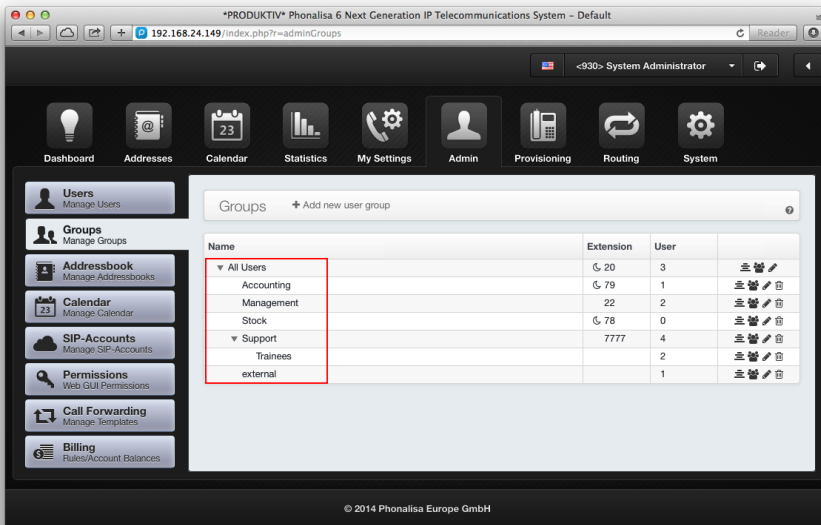
Alternatively, you can also edit the group and select the parent group in the dropdown box.
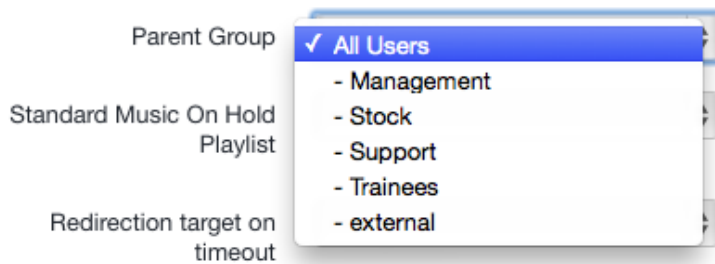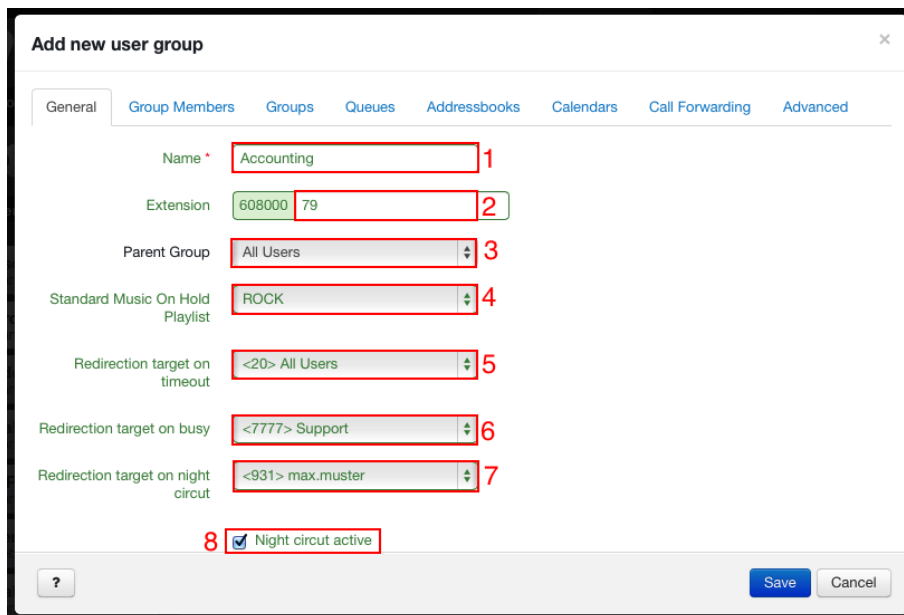
16.2   Add new user group - General settings

Go to "Admin" -> "Groups". There you see that, the group "All Users" is already exists. Click "Add new user group" (1) and make your preferences. If you delete a group, all users will automatically be moved to the "All Users" group.
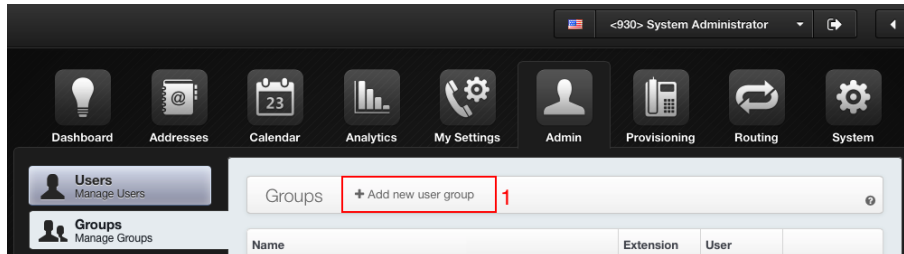
1. Enter the group name here.

2. Here you can specify a phone number for the group. IA group can also be used as a forwarding destination, incoming routing destination or as IVR-menu destination if this is set.

3. Select the parent group.

4. You can select "—" for ringing instead of music or a "Standard Music On Hold Playlist". Each group can have its own music on hold playlist. Which is not to be confused with the hold music. The caller get the "music on hold" while he is on hold but the "hold music" before a group member accepts the call.

5. Here you can specify the "Redirection target on timeout". More informations can be found in point 16.2.1.

6. If all group members are busy, the call will be forwarded to the specified number.

7. If the group uses the night circuit, the call will be forwarded to this number.

8. Here you can activate and deactivate the night circuit.

16.2.1 Forwarding of calls

Phonalisa offers call forwarding in case of "Timeout", "Busy" and night circuit. Calls can then be redirected to a user, another group, queue or an IVR-menu.

When Timeout is reached?
You can set the call timeout in the "Ringmap" of groups and the global dial timeout in the system settings "System" -> "Configuration" -> "Dialplan". Here you enter the global dial timeout in seconds (1). In this panel you can specify the Central intercept position. If a call can not be redirected to a user or group, it will be forwarded to this number.

16.2.2 The night circuit

If the last group member leaves his workplace, the night service can be activated. All incoming calls to the group are now redirected, e.g. to a user, group, queue or IVR-menu. It is necessary you check the "Night circut active" checkbox to use this function. Assigning the function to a button on your phone possible too. Go to "My Settings" -> "Phone Keys Setup" for this. Select the key function night circuit and set your group as target extension. Please note, the "Night Circuit" button currently will be displayed only on the OpenStage. On other phones it can be put on a BLF button.

16.3 Add/Remove a user from a group

<u>Edit a group:</u>

1. Select the user from the dropdown box.

2. Click "+" to add the user. If you remove a user from a group (by clicking on the garbage bin), the user will be moved into the "All Users" group.

16.4   The Permissions

Following permissions can be assigned:

Sudo-Permission:
Has a group sudo permissions to another group, users may become members of the other group. Simply choose another user in the Sudo-bar for this. As a Sudo user, you have access to all data, address books, calendars and voicemails of the selected user. To prevent abuse, you should assign these privileges carefully.

Monitor-Permission:
This permission controls the visibility of colleagues in the presence monitor. If a group has only monitor permission on itselves, they can only see their own group members in the presence monitor. Depending which groups you have monitor permissions on users will be visible in the presence monitor.

Pick-Up-Permission:
This permission is needed, if you want to pick-up calls from other users or a complete group. You just need to dial *81*collegues-number on your phone to pick-up the incoming call.

Statistics-Permission:
With this function, you can access the statistic of your own group or another group.

Intercom-Permission:
The intercom feature allows you to perform direct announcements to user or group phones. Dial *85*user-number or dial *85*group-number to perform an announcement to a user or group.
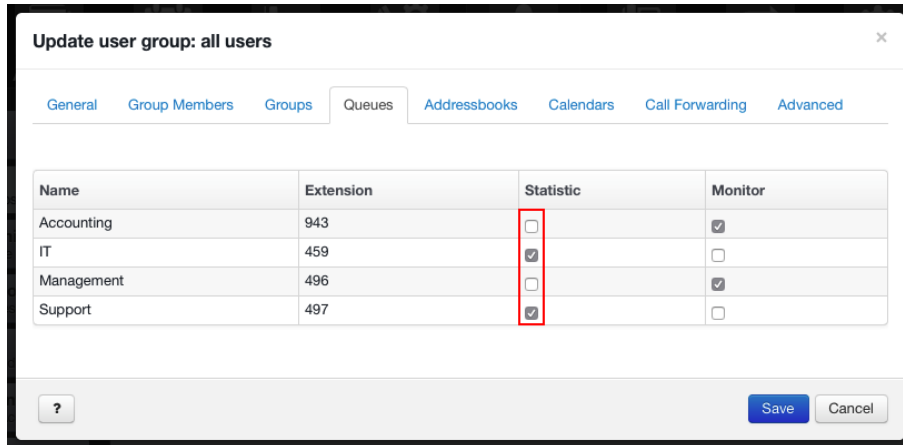
Eavesdrop-Permission:
This feature is very popular with staff training (e.g. call centers). Trainers can listen to conversations between agent and customer. The trainer can speak with the agent but the customer will not hear the trainer. For listening to a conversation, dial *86*agent-number. Now you can press the "1" on your phone to speak with the agent. "2" allows you to talk to the customer, but the agent can not hear you. By dialing "3" you are in conference with the agent and the customer, so both can hear you. If you press "0" you are switched back to mute and nobody can hear you.

Update user group: All Users

General   Group Members   Groups   Queues   Addressbooks   Calendars   Call Forwarding   Advanced

Name	Sudo	Monitor	Pick-Up	Statistic	Intercom	Eavesdrop
All Users	☐	☑	☑	☑	☑	☐
- Accounting	☐	☑	☑	☑	☑	☐
- Management	☐	☑	☑	☑	☑	☐
- Stock	☐	☑	☑	☑	☑	☐
- Support	☐	☑	☑	☑	☑	☐
-- Trainees	☐	☑	☑	☑	☑	☐
- external	☐	☑	☑	☑	☑	☐

?   Save   Cancel

16.5   The queue statistics

In this area you can grant group permissions to see the statistics of a queue. If a queue is selected via the checkbox, you can see the statistics in the menu "Statistics" -> "Daily Report" respectively "Queue Report". For more details, see point 11.2.

16.6   The Monitor Options for Queues

In this area you can define which queues are displayed to the members of a groups in the Agent-Monitor. Activate the box behind the queue. Now you will find all agents of the selected queue appear with their current status in the section "Admin" -> "Agent Monitor". For details see chapter17.

16.7 The address book and the calendar

Here are all available address books listed. You can assign the read (1) and write (2) and even the CSV import permissions (3) for each address book. It is the same for the calendar. You find all available calendars listed and can assign the read and write permissions.

16.8 Using the System Call Forwarding

In "Admin" -> "Call Forwarding" , you can configure call forwarding templates which can be used in the entire system (see point 20). All call forwarding templates are listed in the group settings tab "Call Forwarding". If you select the proper rules (1), the members of the group can use this call forwarding template for their own call forwarding.

16.9 The advanced group settings

In the advanced tab you can make a wide variety of settings, for example the key profiles, the fax settings or the blacklist settings.

16.9.1 The parameter profiles and the key profiles

With the key profile, it is possible for all group members to have the same key layouts. In the parameter profiles you can set additional settings for your phone. See point 22.3 and 22.4 for more details.

1. Select the profile in the dropdown box of the "Advanced" tab.

To apply the settings, the phones need to reload. Go to "Provisioning" -> "Provisioning" and click "Trigger phone to request new settings" for each group member.

16.9.2   The blacklist for incoming and outgoing calls

For each group blacklists for incoming and outgoing calls can be set. So it is possible to prohibit certain groups, to make calls to cell phones or expensive service numbers.

16.9.3   The fax header text and the fax header number

You can set custom fax header texts and numbers not only for users, it is also possible to set it for groups. After you have made your settings, you can use them in the "Send new Fax"-dialog box on the dashboard.

16.9.4   The Central intercept position for groups

The Central intercept position is used, if all group members are busy and no call forwarding is set for the case "Timeout". You can enter any number, e.g. a queue, group number or an external number.

16.9.5   The billing system

The billing system feature has to be enabled with "Enable billing system for outbound calls" in advanced group settings. After that you can see all listed users in "Admin" -> "Billing". Now you can make additional settings (see point 21 for more details).

16.9.6 The Ringmap

Click the button (1) to perform the ringmap settings.
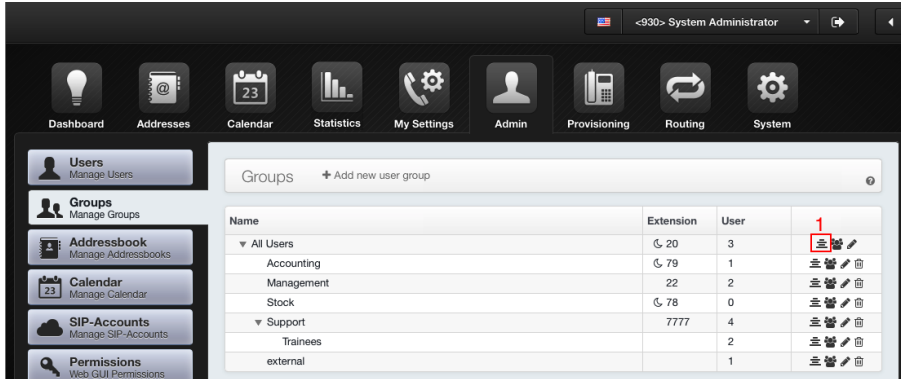
1. Here you can see the group members.

2. The phones with the extensions 930 and 32 ring first. The ringing time is 60 seconds.

3. The phone of the third user (extension 33) rings after 10 seconds. Its ringing time is also 60 seconds, so it will ring for 10 more seconds than the others in point 2. (The bars can be moved by Drag & Drop.)

4. In this area you see the "Child-Groups" with theirs corresponding group phone number. The groups that have no group phone number are listed below without an ringmap bar.
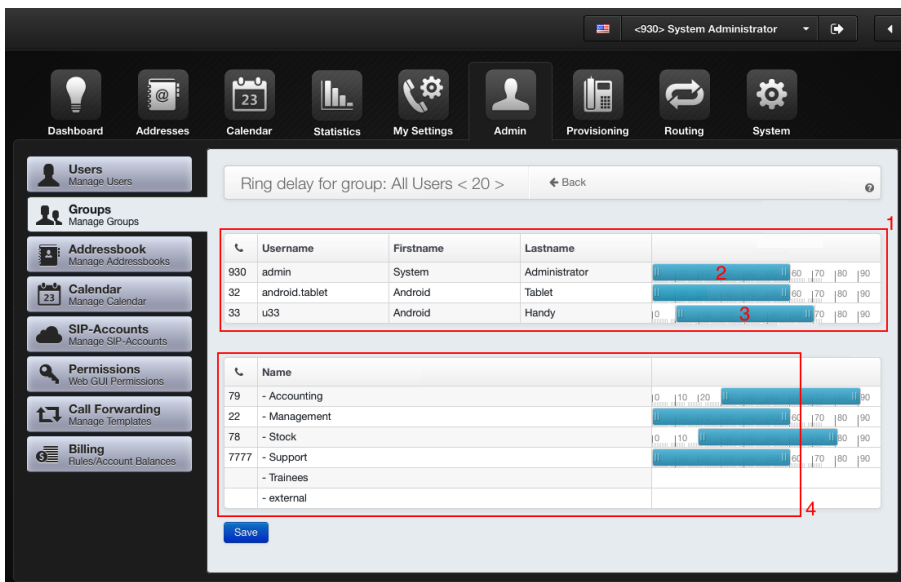
17 The Queue Monitor

With the queue monitor you get the ability to monitor all agents and their assigned queues. You can see at a glimpse which agent is assigned to which queue and its current state. Furthermore its helps to watch which agent currently processes wich call. So you will always have brief overview of your agents.
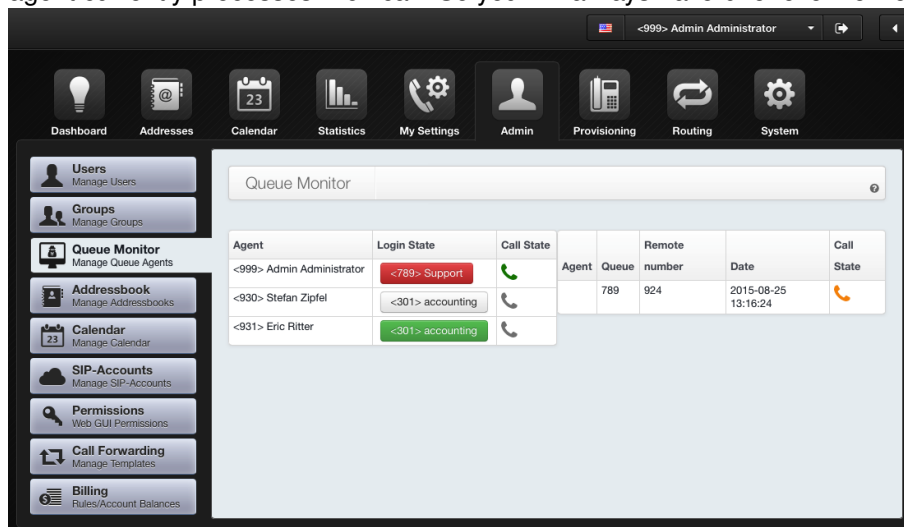
17.1 List of all agents and their queues

1. In this column you see all agents who are assigned to a queue.

2. The login state shows you which queue has static member (grey), if an agent is logged out (red) or is logged in(green).

3. The call state shows you the reachability of agents grey (logged out), green (available), yellow (calling) and red (call busy).

17.2 List of incoming calls

1. This panel shows you which queue agent is assigned to the call.

2. Here you see which queues is called by the caller.

3. The phone numbers of incoming calls will be shown here.

4. Date and the time of the call is shown in this column.

5. The call state shows if a call is processed(green), incoming(yellow) or terminated(red).

17.3   Manuell call assignment

To assign calls manually just "Drag & Drop" it on an agent you wanted to take the call. The call will then auto-matically be forwarded to the assigned agent. This also gives you the ability to forward calls of one queue to another queue.
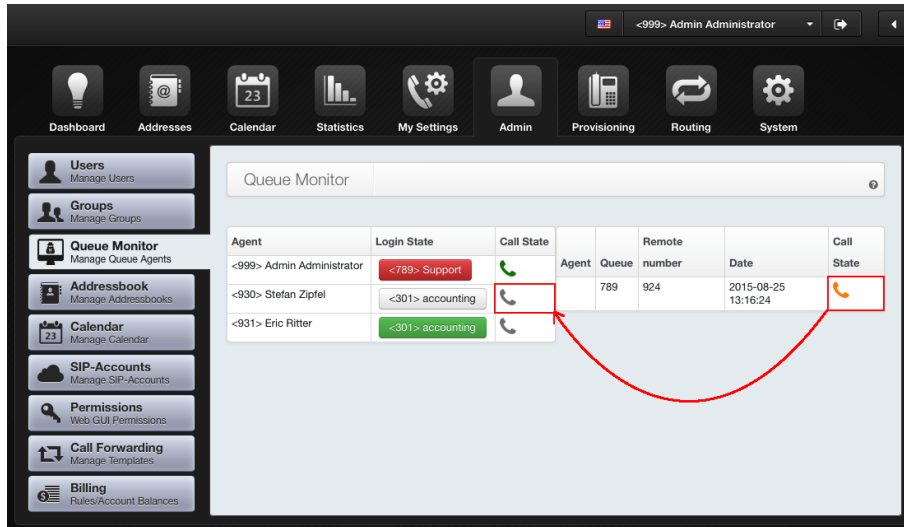
Warning: Please do not assign a call to an agent who is already busy with another call. If you do assign a call to a busy agent it will be terminated.

17.4   Grant queue monitor permissions

To give a group queue monitor permissions go to "Admin"->"Groups" and edit the desired group. In the tab queues of the update user groups window you can grant monitor permissions for the queues.
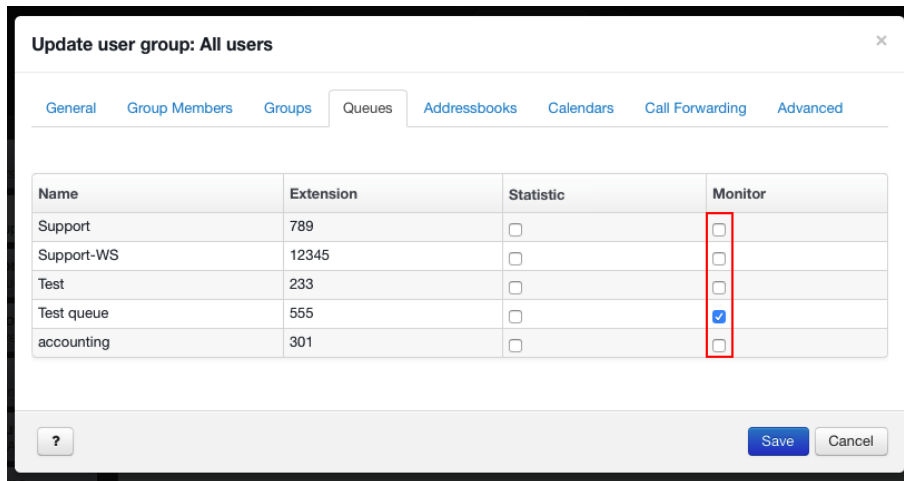
18 The SIP-Accounts

The SIP-Accounts are used from the phones to log on to the system. They are generated when a user is created. Multiple phones can register on one SIP-Account (multiple SIP-Registration), so a user can have multiple phones.

Overview of SIP-Accounts:

1. Here you find the status of each account.

2. In this column you can see the SIP user name (usually the extension number of the user).

3. Here you can see, when the SIP-Account has registered the last time.

4. This column shows when the SIP-Account will expire.

5. Here is the IP address of the phone on which the account is registered.

6. In this column you can see the device name of the phone.

7. Here you can see the user.

8. To edit or delete a SIP-Account, click on the corresponding buttons.

18.1   Creating a new SIP-Account

Click "Add new SIP-Account" .

1. Enter here the SIP-Username.

2. Enter the password for the new SIP account here.

3. Select a user ID from the dropdown box.

4. Save your settings.
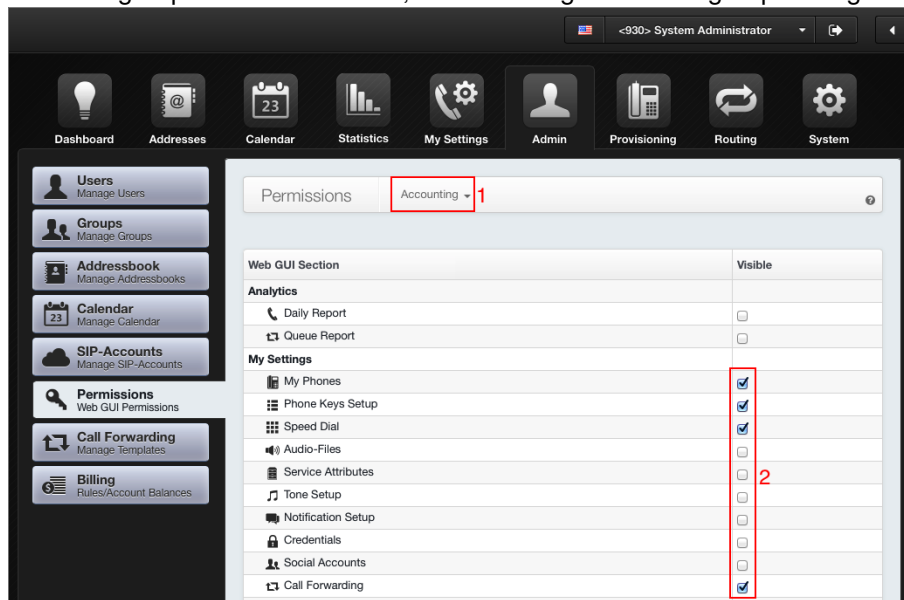
19 The Permissions for the GUI modules

You can show and hide GUI-modules individually for each group. This will ensure users only have access to the GUI-modules they need.

1. Select a group from the dropdown box.

2. Select the GUI-modules for the group. Example: the group "Accounting" only has permission to see only the items "My Phones", "Phone Key Setup", "Speed Dial" and "Call Forwarding" of "My Settings".

The dashboard is the only GUI-Module which is always visible and can not be hide. Which address book, calendar groups or users can see, can be configured in the group settings.

20 The System Call Forwarding

In a company several call forwarding scenarios are used many times, e.g. a rerouting in case of busy or call forwarding outside of business hours. So it is obvious that members of a group use the same call forwarding template. So a call forwarding template is created once and can be assigned to all groups. This saves a lot of time and the admin does not have to configure the same rule for each user. Of course, users can still create specific forwarding rule.

20.1 Creating a System Call Forwarding template

Creating and editing the System Call Forwarding templates is the same as the user's call forwarding. The only difference is that only system audio files can be selected for announcements. Go to "Admin" -> "Call Forwarding". Click the button "Add new call forwarding template". Now you are able to create a template. Choose between "Immediately", "Custom (Rule based)" and "Disabled". In point 13 you can read how to combine the different cases.

Create a "Disabled" call forwarding template:
Add a new call forwarding template. Insert the name, select the call forwarding type "Disabled" and click "Create Template". It is very important to have this kind of call forwarding template. Because if you have set a call forwarding on a phone key and would like to toggle (on/off) then, you need the counterpart.

20.2 Assign the templates to a group

Now the system call forwarding template can be assign to a group. Proceed to "Admin" -> "Groups" and edit the corresponding group.

1. Go to "Call Forwarding". Select the call forwarding template which shall be used with the group.

2. Save the changes.

Now you can see the call forwarding template in the "My Settings" -> "Call Forwarding". The "System" in the column "Type" means that this call forwarding is a template.
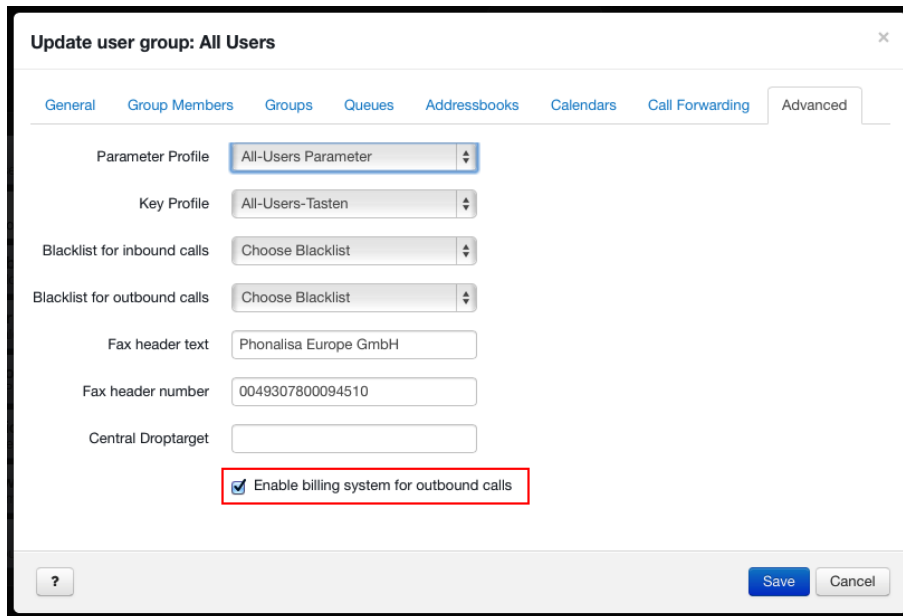
21 The Billing-System

The Billing-module is only available in the Phonalisa Medical Edition.

In this section you can manage your phone billing (call records). To use the billing, you have to enable it in the group management. Go to "Admin" -> "Groups" and select the group that should use the billing module. Then go to the "Advanced" tab and check "Enable billing system for outbound calls".

You can see all users of the group in "Admin" -> "Billing" . Now add the first appropriate price list.

21.1 Create a price list

For correct billing, you have to create a price list first. For example landline numbers, mobile numbers or international numbers. Click "Prices" und dann "Add new price rule" for this.

1. Specify a name for the price rule.

2. Enter the **gross** (incl. VAT) price per call minute.

3. Here you can enter the pattern for the price rule. In this example, all german primaries are set.

4. Save your entries.

21.2 Assign credit

Now you have to assign credits for the individual users (patients, visitors, ...). You will find all users listed here.

1. Click the lock to enable the edit mode of this panel.

2. Now you can enter the balance. In this way you can process several entries simultaneously.

3. Finally, save the changes.

21.3 Querying the account balance

To query the account balance, dial *88* on your phone. Current balance can also be read anytime via the web interface in the section "Admin" -> "Billing".
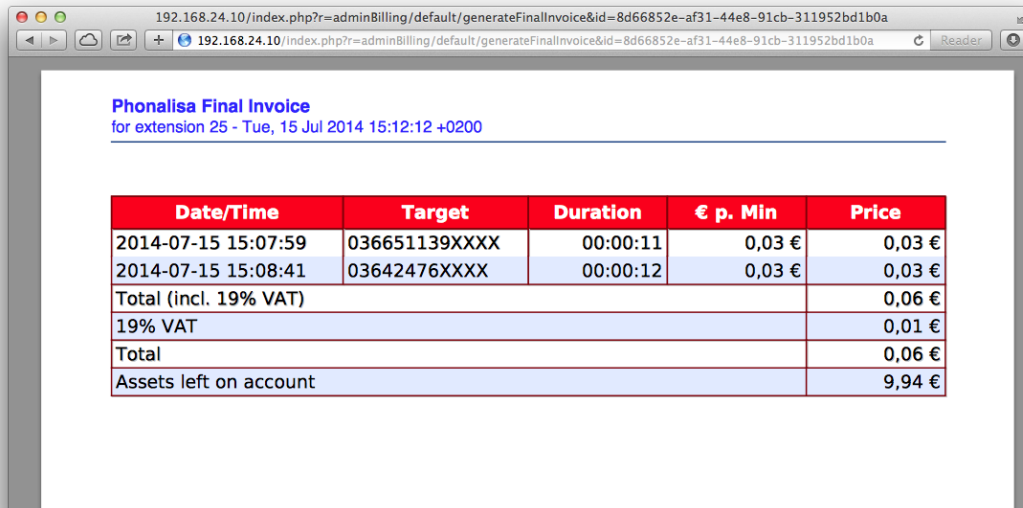
21.4 Generate the final invoice

For example, if the patient leaves the hospital, you can generate the final invoice for their extension.

Click the button.

The final invoice is created for the extension and displayed in the browser. In the example, 0.06 € were used and listed to the remaining balance.

After a final invoice has been created, the balance will be reseted to 0.00 €.

22 The Provisioning

22.1 Manage the phones

The provisioning overview:
Here you will find a list of all your registered phones supported by the Phonalisa provisioning.

1. Here you see the type designation, for example, snom 760, tiptel IP 280, ...

2. This column shows the IPv4 addresses of the listed phones.

3. Here are the MAC addresses of the phones listed.

4. Here is an optional description of the phone, e.g. Location of the phone, ...

5. This column shows the firmware of the individual phones.

6. In this column you can see the SIP-Account ID that is registered on the phone.

7. Click this button to reload a configuration in case you have changed something on the phone (service features, ringtones, ...).

8. If you click the pen you can edit the phone. So it is possible to assign a different SIP-Accountor change the description text. If you click on the garbage bin the phone will be deleted.

9. You access the network scanner via the "Search for Phones" button.

To Assign the phone to another SIP-Account:

1. Select a SIP-Account from the dropdown box.

2. Here you can enter a description text.

3. From the dropdown box you can select a firmware or use standard.

4. Finally, save when you are ready.

22.1.1 Search for phones

The network scanner allows you the opportunity to search for phones which are connected to your network. You may, upon completion of the search, connect supported phones to your Phonalisa by triggering their Auto-provisioning.

1. With the "Start Search" - Button you can activate the network scanner, which scans your network for devices.

2. This button removes the results of already conducted searches from the table below.

3. The "Search"-Button allows you to search for specific devices, IPs or manufacturers.

4. Click this button to hide or show by the autoprovisioning unsupported devices.

5. With this button you can hide or show all devices which are already provisioned on your Phonalisa.

6. Using the "Back"-Button to return to the screen "Manage devices".

1. This list shows you when exactly a device has been found by the network scanner.

2. Here you can find the MAC addresses of individual devices.

3. The column "IP" contains the IPs of all devices connected to your network.

4. The manufacturer of a device can be found here.

5. With the "Initiate provisioning" - Button you can trigger the autoprovisioning for all supported devices.

Found	1	MAC address	2	IP	3	Vendor	4	
2015-10-14 08:36:45		00:04:13:41:85:90		192.168.24.232		snom		
2015-10-14 08:36:23		00:15:65:45:70:15		192.168.24.53		tiptel		5
2015-10-14 08:36:41		00:15:65:4D:49:99		192.168.24.198		tiptel		
2015-10-14 08:36:23		00:15:65:13:ED:BD		192.168.24.57		tiptel		
2015-10-14 08:36:45		00:15:65:4D:49:81		192.168.24.233		tiptel		

Once you triggered the provisioning, you have to wait a moment for the phone (blue) to been registered on your system. If you click the "Back" button, you will return to the "Provisioning". There you will find the phone in your device list assigned with a dummy SIP account and can now be assigned to a user.

Provisioning	Search for Phones	Q Search					1-6/6
Type	**IPv4-Adress**	**MAC-Adress**	**Description**	**Firmware**	**SIP Account ID**		20
Siemens OpenStage 60	192.168.24.230	001ae80316b8		03.03.40.01	927 (Muster, Hans <927>)		
snom 360	192.168.24.229	000413290F4D		08.07.03.25	997 (Meyer, Lisa <997>)		
snom 870	192.168.24.232	000413418590		08.07.03.25	925 (Sprenger, Soeren <925>)		
Tiptel IP 282	192.168.24.197	0015651faeb3		07.61.13.010	991 (Jansen, Jenny <991>)		
Yealink SIP T22P	192.168.24.198	0015654d4999		07.73.00.050	999 (Administrator, System <999>)		
Yealink SIP T22P	192.168.24.233	0015654d4981		07.73.00.050	Nobody 9548370		

22.2 Manage the firmware

Go to "Provisioning" -> "Firmware". To upload a new firmware, click "Add new firmware".

1. Here you can select the phone model, which should receive new firmware.

2. Select the new firmware in your files.

3. If you check this box, the firmware is set as default firmware.

4. Finally, save everything.

If your phone is not supported by the Phonalisa provisioning, you can change the firmware on the web interface of the phone.

22.3 The Provisioning Profiles

The provisioning profiles are used to configure values on a phone or phone keys. This is commonly used for time settings (summertime/wintertime) or to assign a door intercom system to a key. The big advantage of a provisioning profiles is to be configured multiple phone types via a single profile. However it is important to ensure the values are adapted to the respective types.

22.3.1 Create a new Provisioning Profiles

In this section it is shown how to create a new provisioning profile by the example of Siemens OpenStage and the CTI functionality. Select "Provisioning Profiles" in the menu "Provisioning", and click "Add new provisioning profile".

1. Select the phone type.

2. Specify a name for your profile.

3. Fill the table with parameter values.

4. Click the "+" to add values.

5. Save your entries.

To use an existing profile for other phone types, edit it and repeat the steps for the new types. Please note, the values have to be adapted to the particular type and save the profile before you switch to a different phone type.

22.3.2 The variables of the Provisioning Profiles
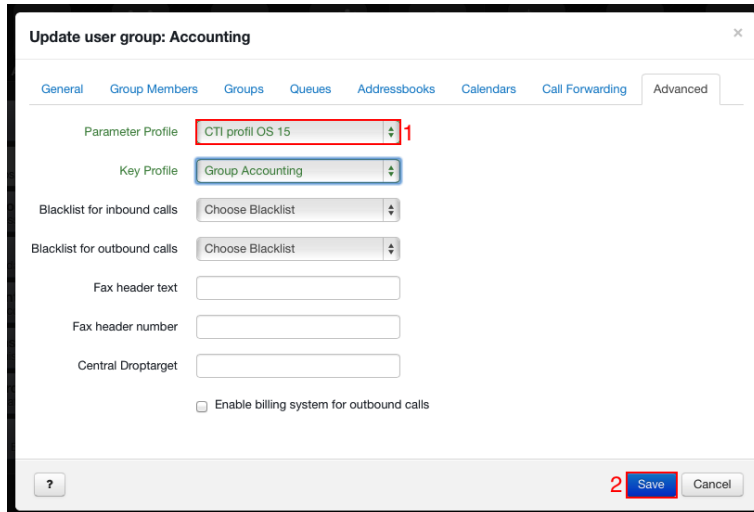
In addition to the preset values, variables can also be used in the provisioning profiles of the Phonlisa, e.g. entries from the system settings or SIP settings. These variables must always include the % sign (e.g. %USER_NAME%). In the table below you will find a list of possible keys.

Key	Value
%PROV_PROTOCOL%	Protocol setting of the provisioning server in system settings
%PROV_HOST%	IP of the provisioning server
%PROV_PORT%	Port of the provisioning server
%PROV_PORT_SECURE%	HTTPS-Port of the provisioning server
%PROV_BASEPATH%	path to the provisioning host where the files for provisioning are located
%PROV_URL%	returns the URL of Phonalisa, e.g. http://192.168.24.59:8080/
%MAC%	refers to the MAC address of the phone
%USER_EXTENSION%	extension of the user
%USER_NAME%	username of the user
%USER_FIRSTNAME%	first name of the user
%USER_LASTNAME%	last name of the user
%PHONE_LABEL%	Combination of last name, first name and extension
%SIP_AUTHNAME%	current SIP account ID
%SIP_HOST%	SIP-Host of the provisioning server in system settings
%INTERNAL_SIP_PORT%	SIP-Port for the internal SIP profile
%EXTERNAL_SIP_PORT%	SIP-Port for the external SIP profile
%INTERNAL_TLS_PORT%	TLS-SIP-Port for the internal SIP profile
%EXTERNAL_TLS_PORT%	TLS-SIP-Port for the external SIP profile
%API_URL_INTERN%	internal API-Url
%API_URL_EXTERN%	external API-Url

22.3.3 Assigning a provisioning profile

The created profile can now be assigned to the groups. Proceed to "Admin" -> "Groups". Select the group and edit it. Go to "Advanced".

1. Select a profile from the dropdown menu.

2. Save your entries.

22.4 The Key Profiles

These are used to give all members of a group the same key profiles, without assigning it to every single phone.

22.4.1 Create a new Key Profile

In the example, a key profile (Tiptel IP 282) is created for the group of accounting staff. 3 buttons on the employees phone shall be assigned identical and not be deleted or renamed by the users. To create the key profile, go to "Provisioning" -> "Key Profiles" and click "Add new key profile".

Procedure:

1. Select the typ of phone you want to create the profile for.

2. Specify the profile name.

3. Here you can choose the keys function (functions are depending from the phones).

4. Enter the destination number or the data here.

5. Here you can enter the label.

6. If you check this box, users can not make any changes to the key.

7. Save your settings.

22.4.2 Assigning a phone key profile

The created profile can now be assigned to the groups. Proceed to "Admin" -> "Groups". Select the group and edit it. Then go to "Advanced".

1. Select the profile from the dropdown box.

2. Save the changes.

Now the group members have the button settings on their phones. In "My Settings" -> "Phone Keys Setup" they can see their key profiles. For example you can see the key profile is locked for the user.

22.4.3 Assigning multiple phone types to a phone key profile

Edit the created key profile and select another phone type from the dropdown box and specify the keys. It is possible to assign group members with different phone models to the same key profile.

22.5 The Ringtones

The Ringtones modul in the "Provisioning" allows you to specify systemwide ringtones for phones. The uploaded ringtones can then be used by any user in "Tone Setup" under "My Settings".

1. After clicking the "Add a new ringtone " button a dialog window opens in which you can now upload a ringtone from your own files.

2. Here you receive an overview of system-wide ringtones and see the name and the format of each ring tone.

3. With the garbage bin, you can delete old or unneeded ringtones.
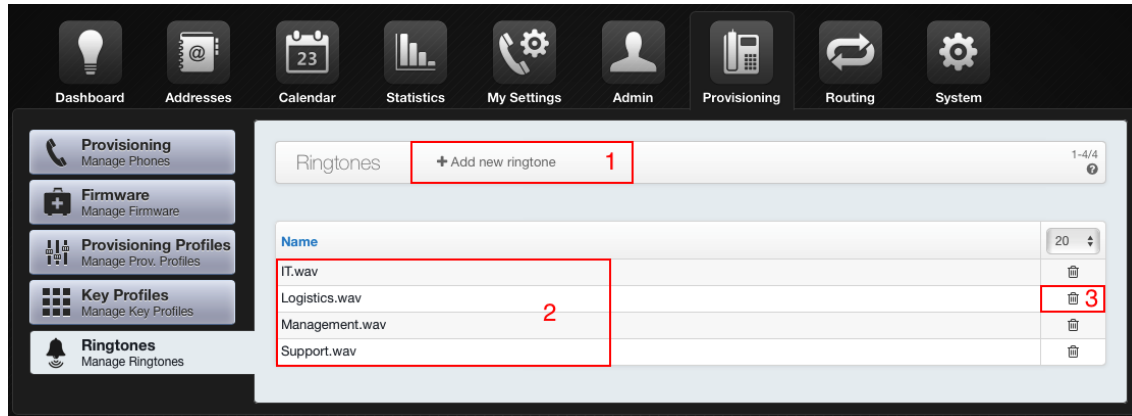
23 The Routing-System

The Phonalisa routing system determines how incoming and outgoing calls handled. It is one of the most important components. Gateways are used by the system to receive and send calls. A gateway can be a built-in ISDN interface, a media gateway (e.g. beroNet or Patton) or a SIP provider on the Internet. Incoming and outgoing routes are determined by tables with patterns for each gateway. More about this in point 23.2.

23.1 The Gateways

To make an external call, you need a SIP gateway. This can be an external service provider such as Sipgate or a gateway such as beroNet. Go to "Routing" -> "Gateways" and click "Add new gateway".

23.1.1 The gateway access data

1. Enter a name which is freely selectable and only helps differentiate your gateways.

2. Specify the host of the gateway.

3. Set the username and

4. a password.

5. If you want to access an external line, check this box. Then set your desired external line (dialing prefix) in the box below. It is not necessary to set an external line.

6. Save your entries.

23.1.2   The Gateway Channels

1. Here you can specify the number of simultaneous telephone calls.

2. Blacklists are also can be set for incoming and outgoing phone calls for each gateway.

23.1.3 The SIP-Advanced settings

These panels should only be processed if you are completely sure what you are doing.

Explanation of the points:

1. Here you can enable the "Caller ID in From"-header.

2. Enter the expire seconds here.

3. Here, you enter the realm.

4. In this panel, enter the "From User".

5. Here you enter the "From Domain".

6. Enter here the proxy.

7. Here you can enter the register proxy.

8. In this panel, enter the contact params.

9. Here you can activate the "Use TCP Transport".

10. Enter the ping here.

11. Here you can activate the "Gateway supports calldeflection (Berofix)".

23.1.4 Example: Connect Sipgate phone number

The Phonalisa telephone system is able to signal phone numbers as a local number outside. Sipgate is very well suited as provider because it has a free deal offer. You can choose a number at the desired location and let documents be send there. A postbox, which is operated there, is required! If the documents are filled out, you send them back to Sipgate and after a short time, your number is callable.

Warning: In the free tariff, the phone number can only be called. Once you establish a call forwarding (e.g. to a cell phone) or would like to make external calls, you have to first charge money on the Sipgate account. For a small fee per call or call forwarding, you can use the phone number completely.

Instruction:

Step 1: Register at Sipgate Portal

- The documents were reviewed by Sipgate and you have received your login information by email.

- Enter the code on sipgate.de on the login screen.

Step 2: Check the General Settings

- Click on settings to open the data.

- Here you will now see your SIP-Account details and your SIP server data listed.

Step 3: Make settings in the Phonalisa web interface

- Go to "Routing" -> "Gateways".

- Create a new gateway.

- Enter *sipgate.de* as a Registrar/Server .

- As authname enter the SIP-ID(at)sipgate. The ID can be found in your SIP-Account data on sipgate.de.

- As secret enter the *SIP-Password*. You can also find that on sipgate.de in your SIP account data.

- Save the settings.

- After successful registration, you can use the Sipgate gateway in the routing.

23.2   Pattern (Pattern Matching)

Here we would like to explain a few basics. These are optional input options and no obligation!

The pattern (or expressions) which is used in Phonalisa for detecting and overriding phone numbers, are similar with the syntax of regular expressions. With the patterns, it is checked whether a certain phone number matches a concrete specification. Example: Is the dialed phone number a cell phone number? Patterns not only serve to check a certain condition but also to search and replace. So it is possible to replace incoming and outgoing extensions with the patterns.

The syntax of a pattern, for example is ^01[5-7] . This pattern checks if a phone number begins with 0 (the ^ means the search starts at the beginning) then carried out to check whether the 0 is followed by a 1. The next number has to be a 5, 6 or 7 for this is pattern to be valid.

Other Examples:
^0 - accesses to all numbers that begin with 0.
99$ - accesses for all numbers that ending with 99 ($ marks the end of the number)
^099$ - accesses exactly only for 099
^0[15] - accesses for all numbers that start with 0 and continue with 1 or 5

There are two patterns required for search and replace . One pattern that match the expression and another which composes the located information together. Therefore you can search for the head number, extract the direct dialing and provide it then with a prefix for incoming routes.

Example:
Our test-company has the test head number 0364247608-00, with double-digit extension numbers till 99. All internal extension numbers start with a prefix 6 - followed by the extension number (e.g. 612). To cut off the head number and the area code, so only the extension is routed, the following search pattern is required:
^0364247608([0-9][0-9])$

This pattern matches phone numbers beginning with 0364247608 and followed by 2 digits from 0-9. The brackets "(" and ")" means that the data is to be copied at this point. The copy will then be available in the second pattern. The data can then be used with the expression $1 there. In our example, the second pattern therefore is 6$1. An incoming call to 036424760823 is then rewritten to 623 and 036424760844 to 644, etc. .

23.3   The Routing Outbound

Phonalisa offers a very flexible routing system. Before you can create outbound rules, it is important that you have created at least one gateway.

23.3.1   Pattern / sequence and processing of rules

Also in the outgoing routing, you can find the patterns from point 23.2 again. However, these are much more simple to use than in the incoming routes. No search and replace is performed here. It is only checked whether number fits to the pattern. If a dialed phone number fits to a pattern its condition is used. The routing table is processed by the system from the top to the bottom. For example, if your pattern ^0 is on the top, then all phone numbers that starts with a 0 are concerned. So your next pattern ^01[5-7] will not be reached.

23.3.2   Least Cost Routing (LCR)

The "Least Cost Routing" allows you, to route your calls cost effective to various gateways. Therefore calls to a mobile networks can be routed through a GSM gateway or phone numbers abroad with a there resident SIP provider. Furthermore, you can set (via the call-by-call prefix) a low cost dialing code automatically in front of a phone number. Please note that this function should not be used with SIP providers on the Internet, because that could result in higher costs.

23.3.3   Route calls time-controlled

You can define on which times and weekdays it takes effect for each rule (you can set a corresponding date/time profile). Thus it is possible, e.g. to forward to an announcement, a queue or a IVR outside of business hours. You can combine multiple rules, because these are processed from top to bottom. Thus, even very complex scenarios are possible (e.g. break times / different team composition of a hotline at different weekdays, and much more).

23.3.4 Restrict routing for groups / group-based external Caller ID

Of course, certain external destinations can be made available only to certain groups of users. Simply use the "User Group" field and select the appropriate user group that is allowed to use this rule. It is also possible to give different user groups (for example an support team) a consistent sender phone number. For this, you only need to set the corresponding phone number, that should be transmitted, in the table column. Then save your settings.

23.3.5 Define failover gateways

If you want to make sure through redundancy that, for example, Emergency calls are more important then others, you can define multiple gateways in the column "Gateway ID". So if one gateway goes down or all lines are busy, the call is routed automatically to the next gateway. The list is processed from top to bottom.

23.3.6 Creating an outbound routing

To call the right external phone number and to be sure, that you use the right gateway, you must define an outbound route. For that, go to "Routing" -> "Routing Outbound".

Outgoing routing for an outside line 0 or *:

1. Activate the route (default it is already active).

2. Select a user or group (you can leave this field blank).

3. To allow outgoing calls with the outside line 0, enter the pattern ^0 or 0* in this field.

4. Here you can choose the gateway. If you have created several different gateways, you can select them as a fall-back if a gateway becomes unavailable.

5. You can optionally define a date/time profile, to limit the outgoing calls in time.

6. Here you can overwrite your outbound phone number. For this, select the checkbox and enter a new number.

7. Here you can additionally enter a LCR prefix.

8. Save the process.

9. Via "Drag & Drop" you can change the order of the rule.

23.4 The Routing Inbound

You have to adjust the inbound routes for all calls in order to reach their corresponding person. The extensions for your connection will be mapped directly to the internal users. In the pattern panel you have to enter a pattern which checks the incoming call (e.g. "^13$" to check exactly on the 13). In "Target" panel, you can define a fixed value or enter a search pattern (e.g. "(.*)"). The prefix "fax-", for example can be used to forward the rule directly to the faxbox of a user (e.g. "fax-13" for the faxbox of extension number 13).

Create an inbound route:

1. Here you can select the gateway you want to create an inbound routing for.

2. Enter a pattern the extension number (with head number) or a *.

3. Select a destination. This can be an individual user, fax of a user, group, queue, conference room, IVR, Call-Through or a manually entered destination number.

4. Here you can define a date/time profile, e.g. to receive calls only at a particular date/time.

5. In this panel you can define a CIP prefix. This prefix will be displayed on your phone screen during an incoming call. .

6. Save your entries.

7. Via "Drag & Drop" you can change the order of the rule. If you change the order of the rules, you must save the changes.
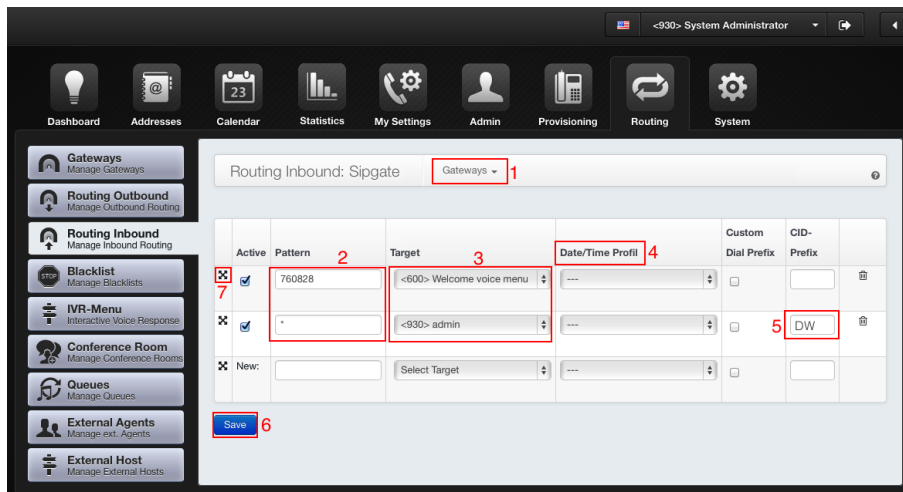
23.5 The emergency phone numbers

It is important to ensure that any Home office staff makes a call via the emergency center of his location. The same applies to any used SIP provider. If you have doubt about the handling of emergency phone numbers, ask the support of your SIP provider whether there are particular specifics.

Warning: Please test the emergency calls after completing the installation and at the latest before productive use! It should be noted, that you should not just call the emergency number, if it is not actually an emergency. You can still test the emergency hotline if you call the landline number of the local emergency responders first and announcing a test call!

24 The Blacklists

A blacklist is used to block certain number ranges (e.g. 0900, ...) or explicit numbers. For this, you can enter in the input box a pattern or the complete phone number.
The blacklists can be used in group settings to prevent groups cell phone numbers but they also can be used on gateways. In each case the lists are to be applied for both incoming and outgoing calls.

24.1 Add a new blacklist

Proceed to "Routing" -> "Blacklist". Click "Add new blacklist".

1. Enter the name for the blacklist.

2. Enter here the number or pattern, that you want to block.

3. Click the "+" to enter more phone numbers or patterns.

4. Save your blacklist.
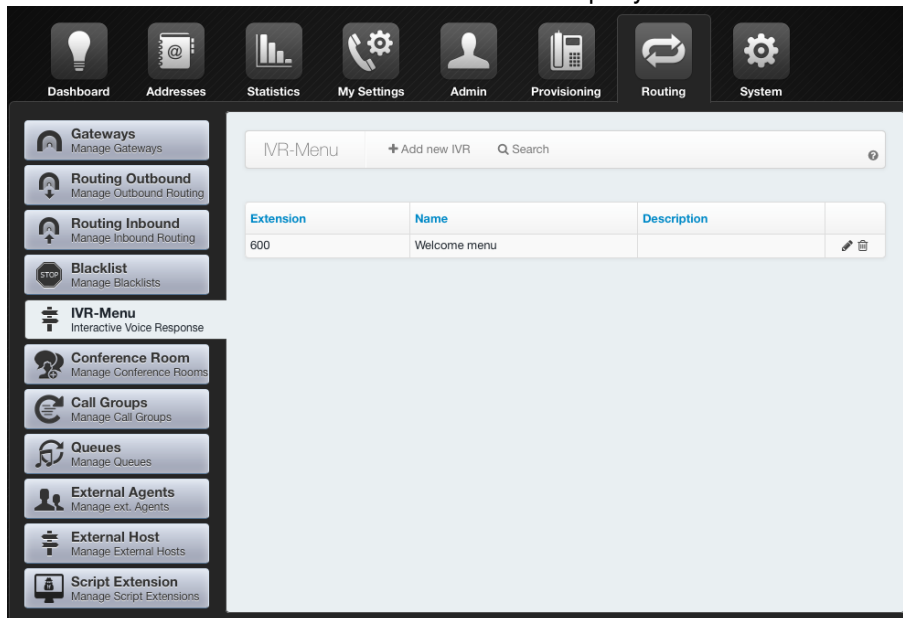
25 The IVR-Menu

Phonalisa gives you the ability to define single-stage and multi-stage IVR (Interactive Voice Response). Single-stage redirectes you to a call destination or an voicemail if you press the corresponding numbers 0-9, * or # on your phone. It is also possible to repeat a announcement or terminate the call. Multi-stage allows you to transfer not only to a call destination, but also to another IVR-menu.

The IVR-menu you can be used, for example, to filter your incoming calls. The will be greeted caller via the IVR-menu and asked questions or proposed targets which he confirmes by pressing a key. It is possible to build multi-stage IVR-menus and filter the calls even finer and so the caller can be connected to the optimal contact. This will save a lot of time within the company.

25.1 Creating and Editing single-stage and multi-stage voice menus

Before you begin to create the IVR-menu, you should think carefully what happens when pressing a phone key. If you want to pass a call to another IVR-menu (after button is pressed) you should consider the announcement texts before. Proceed to "System" –> "Audio-Files" (see point 34 for more informations). Record or upload (from your files) all required announcements and give them descriptive names.

Creating a multi-stage IVR-menu:
Click "Add new IVR". In the new window you can change the settings.

1. Enter an extension. The IVR-menu will be available with it.

2. Enter a title.

3. As announcement file, you can select a welcome message fotr example. Optionally, enter a description for the voice menu to distinguish different IVR menus better. The length of the response time determines how long the IVR-menu is waiting for keypress before the next event is triggered. You can also set the number of repetitions.

4. In the example, the call destination was set to the key-1 "<601> Second-stage IVR-menu". The message "You are now connected" indicates the caller to be forwarded.

5. Here you can select a Script Extension.

6. Click "Save" at the bottom of the window.

25.2 The Integrating of the IVR-menu in the routing

Go to "Routing" -> "Routing inbound". Select the appropriate gateway.

1. Enter pattern or phone number, which is reachable from outside.

2. Select the IVR-menu from the destinations.

3. Finally, save your settings.

If a date/time profile is defined additionally, you can activate the IVR-menu at certain hours or on a specific date. This function is optional.

26 The Conference Room

The integrated conference room function is ideal for phone calls with more than 3 participants. You can make the conference room available for external users via inbound routing and e.g. specify a DID number. A conference room can have any number of moderators, which are then automatically called as soon as the first participant "enters" the conference room.

26.1 Creating and editing a conference room

Click "Add new conference room".

1. Here you can specify a name.

2. Enter a phone number.

3. Here you can select the announcement, which will be played when "entering" of the conference room. Usually it is the system default announcement.

4. A PIN is used to "unlock" the conference room (optional setting) by participants. It helpls you avoid other participants randomly entering the same conference room.

5. The function "Use Conference-Member-Name" is optional. If activated, all new subscribers have to record there names before they can enter the conference room. The record then be plaied to all present participants (e.g. "John Doe has entered the conference").

6. If you check "Listen to Music", a dropdown box becomes visible. Here you can select a music on hold. This will be played until the moderator enters the conference. If there is no music selected the participants will speak to/hear each other.

7. Here you can select a Script Extension.

8. Finally, save the entries.

DID number:
also called direct dial-in (DDI) is a telecommunication service offered by telephone companies to subscribers who operate a private branch exchange (PBX) system. The feature provides service for multiple telephone numbers over one or more analog or digital physical circuits to the PBX and signals the dialed telephone number to the PBX for internal routing to a PBX extension.

This is the system default announcement when entering the conference room:
"You are currently the only person in this conference."

26.2   The Conference Room-Moderators

The moderator is used to open the conference room. Participants can enter the conference room before the moderator, but they will hear music on hold or silence until the moderator enters the conference. A conference room can be assigned any number of moderators. If multiple moderators are assigned, then also one alone can open the conference room.

Assign moderators of the conference room:
Edit the conference room and click on the "Moderator" tab.

1. In the drop-down box you can select the user.

2. If you check the "Automatic Moderator Call", the moderator will be called as soon as the first conference participant enters the conference room.

3. Click the "+" button to add the moderator.

26.3 Integration of the conference rooms in the routing

To make a conference room reached from the outside, you have to edit the incoming routing. Go to "Routing" -> "Routing Inbound". There you select the appropriate gateway.

1. Enter a phone number from your phone number block as a pattern.

2. As a target you specify a conference room.

3. If you define a date/time profile, you can make the conference room accessible only at certain hours or on a specific date.

4. Save the new routing rule.

27   The Call Groups

Call groups are similar to groups but much more flexible because they do not use the inheritance system.

27.1   Create a new call group

Go to "Routing"->"Call Groups" and click "Add new call group".

1. Type in the name of the call group.

2. Assign a number.

3. Here you can choose between an "On hold music" or a normal ringing.

4. Click to save your entries.

1. The targets for the different routing scenarios can be selected here. You can redirect a call to a user or her/his voicemail, a group, a different queue, an IVR-menu or a specially selected number for example the number a cell phone.

2. In this panel you can enter a maximum time a call is on hold before it is terminated, routed to the Central intercept position or to a target chosen in point 1 (Redirection targe on timeout).

3. The "Default Ring Time" refers to the maximum time span a caller is allowed to call a group. This time span well be shown in the ring map where you can adjust it with the bars.

4. The night circuit can be activated here.

5. Here you can select a Script Extension to be executed if the call endpoint is reached.

6. Click to save your entries.

Warning: Numbers like +4936 ..... or +43 ... are not allowed and must be in the format 04936 ... or 0043 ... because the rule with +49 etc is not supported by some gateways and providers at this point.

28 The Queues

A queue is used to strategicly distribut calls in the company. Callers can be conducted to certain queue levels or even skip levels (more on this in the following points). The queue can be equipped with internal and external agents and evaluated with detailed statistics. It has also a lot of additional function.

28.1 Creating and editing a queue

Go to "Routing" -> "Queues". Click "Add new queue". If you have already created queues, they will be listed below. You can see also playlist and which strategy the call distribution has.

28.2 Perform the general settings

Before the general settings are made, you will find some explanation of the terms here.

The CID-Prefix:
CID means Caller ID and the prefix is used to detect / assign to a queue. This is an optional setting. If you are an agent of multiple queues it signals directly to the phone from which queue the caller calls. Setting a 3-digit prefix of the queue's extension or abbreviation of the name was always positively accepted by users.

The strategies of call distribution:
Top Down: Agents will be called based on their level.
Agent With Least Talk Time: The agent with the shortest conversations will be called.
Agent With Fewest Calls: The agent with the fewest calls will be called.
Ring All: All logged agents of the queue will be called at the same time.
Sequentially By Agent Order: Selects agents in the order they were created, starting with the first one.
Longest Idle Agent: The agent, which had no phone calls for the longest time, will be called.
Round Robin: All agents phones will ring one after the other.
Random: Randomly ringing selection of the agents.

The timeout of calls:
You can enter a maximum time waiting in the "Advanced" tab if you edit a queue. It is the time a caller hase to wait until maximum timeout is used. You can set a "Redirection target on timeout" if you would like. If there is no timeout set in the queue, the system timeout will be used. The system timeout can be configure in "System" -> "Configuration" -> "Dialplan" at the point "Global dialtimeout (seconds)" definieren.
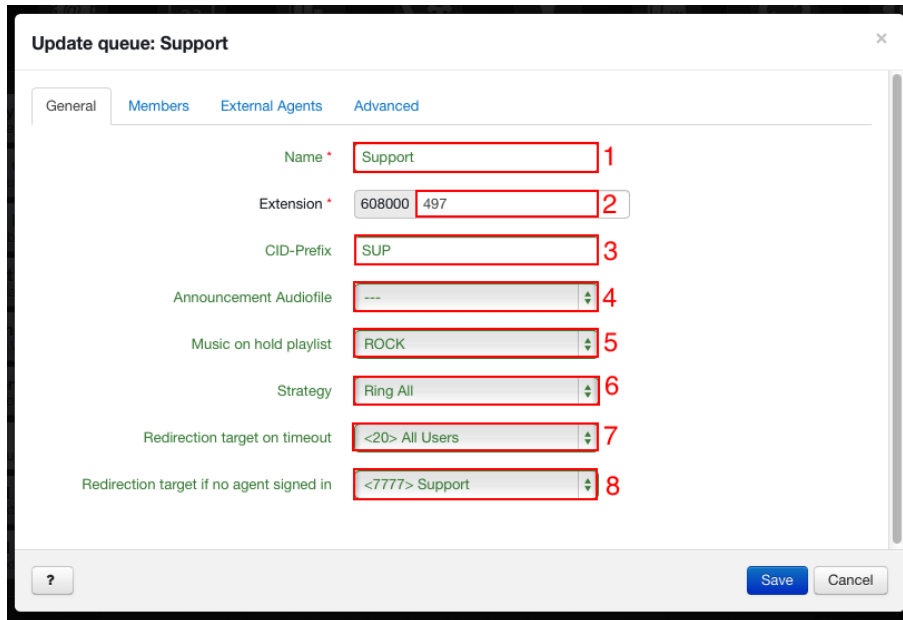
The queue state "empty":
A queue is empty when no agents are logged in.

General Settings:

1. In this box, enter the name of your queue.

2. Here you enter the phone number of your queue.

3. Here you can optionally enter the CID prefix.

4. The welcome announcement is also optional.

5. Select a music on hold playlist or select "Just Ring". The caller will get a simple dial tone.

6. Choose the strategy of the call distribution. In smaller companies, the strategy "Round Robin" or "Ring All" will often be applied. In call centers, mostly a distribution according to the "Sequentially By Agent Order" or "Longest Idle Agent" is better.

7. Here you can specify the "Redirection target on timeout". The call can be redirected to a user or her/his voicemail, a group, a different queue, an IVR-menu or a specially selected number for example the number a cell phone.

8. In "Redirection target if no agent signed in" select a user, a user's voicemail, a group, a different queue, an IVR-menu or a specially selected number for example the number a cell phone.

Warning: Numbers like +4936 ..... or +43 ... are not allowed and must be in the format 04936 ... or 0043 ... because the rule with +49 etc is not supported by some gateways and providers at this point.

28.3 Assign the user to the queue

Edit the queue and go to "Members".

1. Select the user from the dropdown box.

2. Check "Static Member" if the agent is automatically logged in as a static agent and can not log off the queue. If you leave it inactive a user can log in and log out of the queue as dynamic agent (via "My Settings" -> "My Phones"). A user's phone will not ring at queue calls if not logged in.

3. In this box you can choose a level between 1-10. This is then assigned to the user.

4. Enter the position a user gets assigned inside the level. These are optional settings.

5. Click "+" to add a user. A dropdown with all currently available users will appear in the next line.

28.4 Assign the external agents in the queue

In point 29.3 you will learn how to create an external agents. Edit the queue to add an external agent. Go to "External Agents". There are all external agents listed in a dropdown box. Select an agent and specify a level and fixed position (these settings are optional). Press "+" at the end. You can assign as many external agents to a queue as you want.

28.5   The advanced settings of the queues

The advanced settings of the queue configuration is slightly larger than the previous version. Most settings are optional but particularly well suited for call centers and enterprises with a high-volume telephony.

Advanced settings:

1. Here you can specify the maximum time limit for a caller to wait. If the timeout is reached the call is either directed to the objective of timeout queue (set in the queue configuration), to the systems central intercept position or it is terminated.

2. Like point 1 you can enter the time in seconds if no agent is reachable.

3. Put a caller on hold even no agent is logged in on the queue.

4. You can define the maximum waiting period for a caller to be put on hold if no agent is logged in.

5. This refers to the levels you previously entered in the associated users and external agents. If you enable this box, these levels can be used.

6. If this box is checked, the caller has to wait on every level, even if no agent is set. If the box is not checked, the caller only has to wait in levels with an agent.

7. Set the waiting time (in seconds) per level, before the caller is forwarded to the next level.

8. If the system shall save the position - then check this box and leave #9 blank.

9. Here you can enter after how many seconds caller's position will be deleted.

10. Time Base Score: Distinction between "System" and "Queue".

11. Here you can select a Script Extension.

29   The Agents of the Phonalisa telephone system
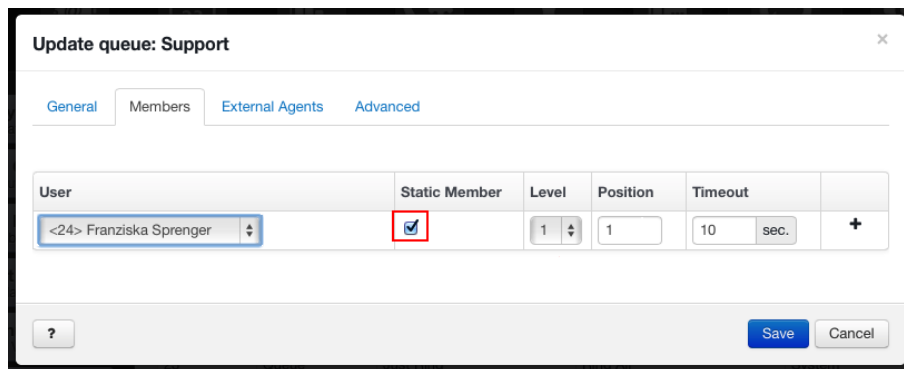
The Agents in Phonalisa are used to log certain terminal devices in queues. An agent who has logged into a queue, will receive calls from this point on. A phone can be agent in multiple queues at the same time. There are three types of agents: static, dynamic and external agents. The following points will discusse the advantages and disadvantages of each type.

29.1   The static agents

Static agents are permanent logged in one or more queues. You can creat a static member in "Queues" of the window "Update user" on the web interface in "Admin" -> "User". In this tab, you are now able to assign the appropriate queue to an agent. If you check "Static Member" the agent will be automatically logged into the queue. This means the agent is assigned to the queues permanent and can not unsubscribe. By logging in and out with the phone the agent automatically logs in and out of the queues. Static agents are always useful if you want certain users constantly assigned to certain queues. Another advantage is you can log into multiple queues with a single sign-up.

29.2   The dynamic agents

A dynamic agent can log in to a queue at anytime. Please note that the "Fixed logged" checkbox must not be set in the respective queues. For an example, see chapter 28.3. The membership in the queue is only temporary and will automatically be terminated as soon as logout occurs on the phone. This means the user has to log back into the queue after the logging back in on the phone.

The used function codes are:
*5Queue-number: Logging in to a queue
5Queue-number: Log out from a specific queue

You will not be able to use dynamic and static agent on a phone at the same time.
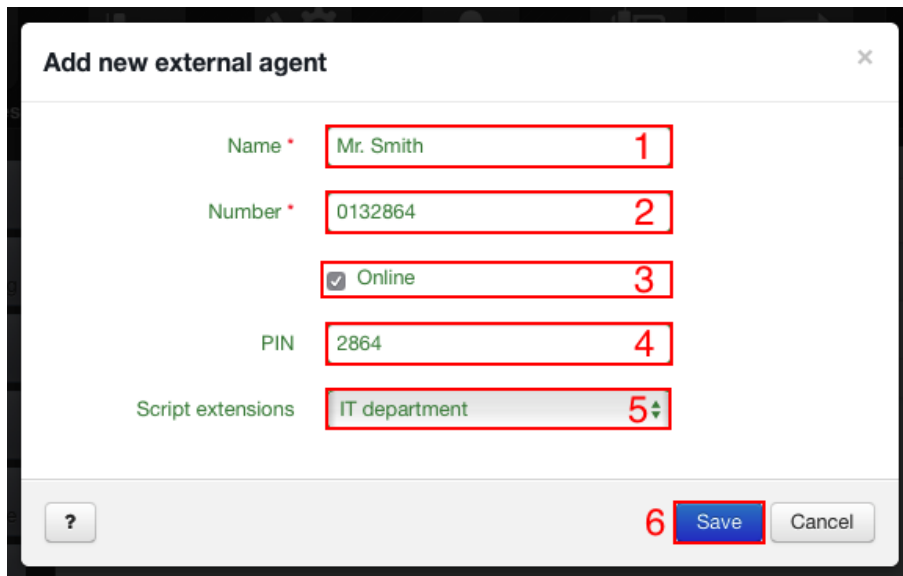
29.3 The External Agent

External agents are registered, e.g. with their home office number or mobile number, on the Phonalisa telephone system. They can be logged into a queue without being present in the office. The advantage of using external agents is an increased range of availability of a queue.

29.3.1 Add new external agent

Go to "Routing" -> "External Agents" and click "Add new external agent".

1. Enter the name of the external agent.

2. Enter the phone number (landline, cell phone or the internal extension).

3. Set the agent immediately online. Uncheck the box if you do not want this and they will be marked as "offline".

4. The PIN is an optional setting.

5. Here you can select a Script Extension.

6. Click to save your entries.

The point 28.4 refers to the assigning of an external agent in the queue .

30 The External Host

You can build your own telephone network by coupling several Phonalisa telephone systems. The participants can call each other with internal phone numbers. So it offers an advantage in terms of security against eavesdropping too.

30.1 Connecting several locations

Go to "Routing" -> "External Host" and click "Add new external Host".

1. Enter the name (e.g. location of the external host).

2. The foreign prefix is optional and serves as a "prefix" before the number of the other location.

3. Define your own prefix. This is cut off for incoming calls on the plant network.

4. Determine the type. You have the choice of "Static (no registration)", "External Host registered at this system" (B -> A) and "The system registered at the External Host" (A -> B). You do not need to enter a host if a system is connected to you the otherone will take care of this.

5. Enter the target host.

6. The username and

7. password have to match with the other side.

8. Save your entries.

The colleague of the external host performs the following configuration

1. He enters the name of the external host.

2. He sets the foreign prefix.

3. He selects the type "Static (no registration)".

4. Here he enters the external host.

5. The username and password have to match with the other system.

6. He saves the entries.

The Advanced settings:

1. Here you can activate the "Use TCP Transport".

2. Setup your enabled codecs for video and audio.

3. Here you can select a Script Extension.
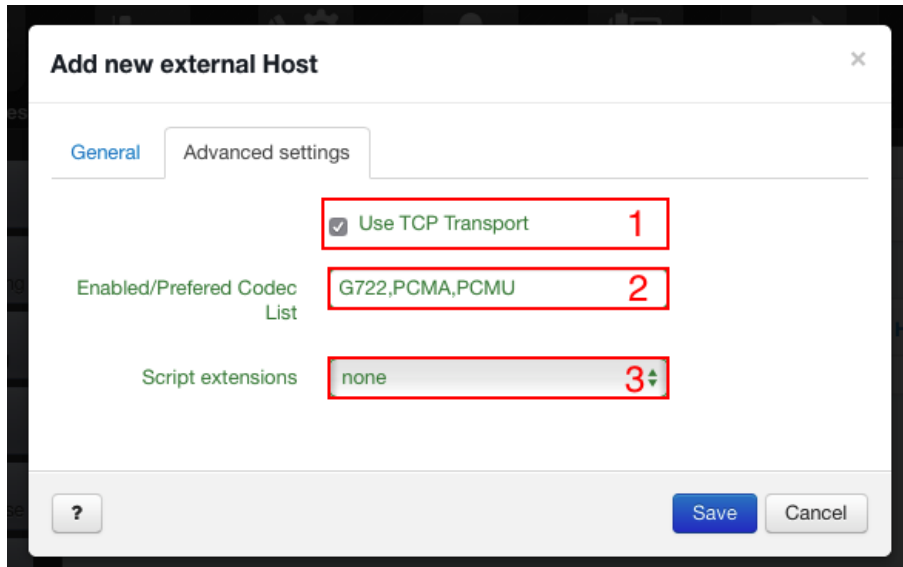
Setting up a call:
The system A with the prefix 608000 wants to call system B with prefix 608001. In A is a user with the extension 25, in B a user with the extension 40. Now, system A dials 608001-40 to talk to the user extension 40. The system B would dial 608000-25 to speak with the user extension 25 of system A.
Instead of the extension number they can also call a conference room, a IVR-menu, a group or a queue in the external host.
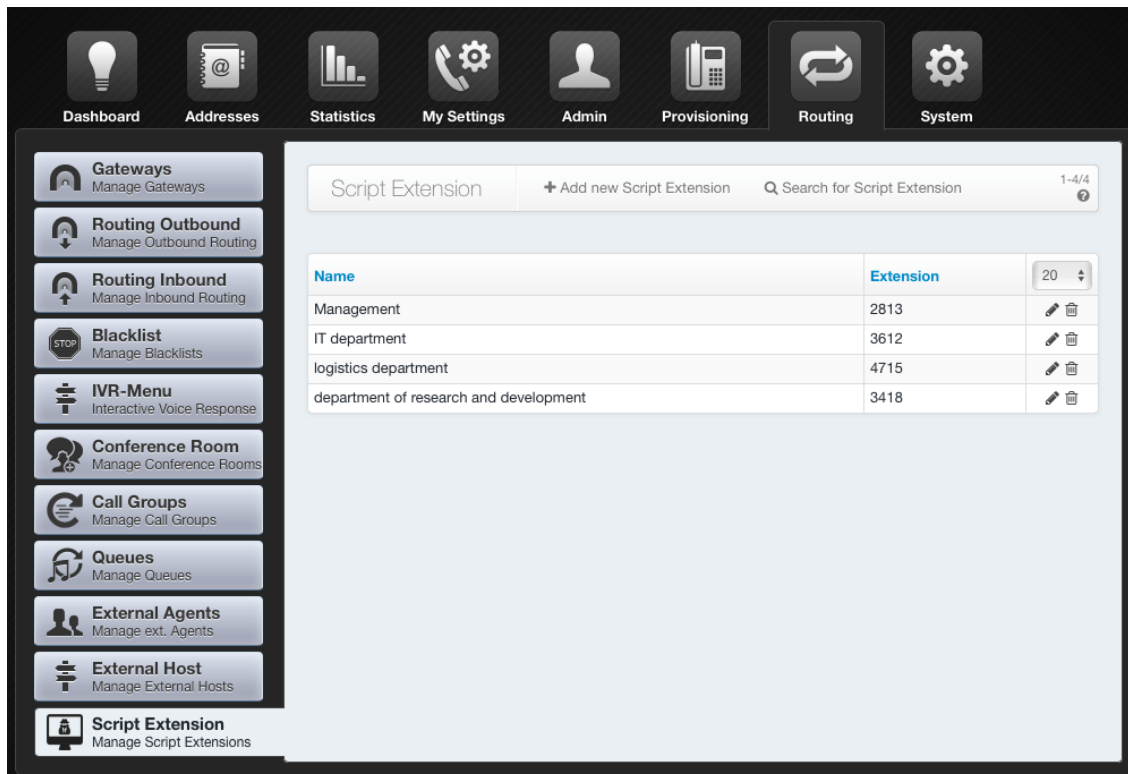
31 The Script Extension

The Script Extension allows you to use an editor for JavaScript, to integrate your own script into your Phonalisa.They can be used to customize the system according to your own wishes. The Script Extensions are small pieces numbering plan that can be executed at certain points. If you assign a number to the script extension and dial it on your device the java script code it contains is interpreted and executed by the system.

31.1 Using a Script Extensions

A possible scenarios is the querying names belonging to phone numbers of a CRM system, if it is not natively supported by Phonalisa - like a self-developed "in-house solutions". It is also possible to intervene in the call schedule of the Phonalisa call routing, by so-called callback functions which are implemented in the Javascript. In this way you can influence a certain behavior of a call going to a single user / group / queue / Gateway / etc, (answer a call, play sounds, overwrite numbers and many more).If you use script extensions with callback functions you can assign the end points (gateways / groups / users / Call groups / queues / etc ...).

Warning: Please be always careful in handling and using Script Extensions, if you do not know exactly what you are doing and don't test your scripts properly it can severely limit the function of the Phonalisa telephony.

31.2 Creating a Script Extension

Click "add New Script Extension" in menu "Script Extensions" Routing to create a new script extension.

1. Here you can specify a name for your Script Extension.

2. In this point you can assign an extension to your Script Extension.
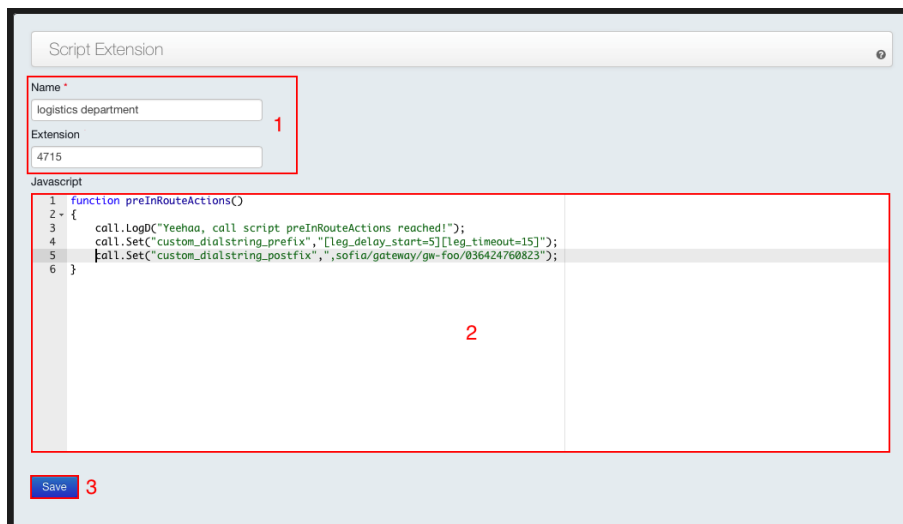
Warning: Please note the extension does behave like a phone's extension so it has to be unique. This means you can call your script from a user's phone or an IVR on this number.

31.3 Edit Script Extension

After creating a Script Extension, you can edit it here.

1. Here you can adjust the name and extension of your script extension subsequently.

2. In this section you will find the editor field in which you can enter Java script, which then executed later. The editor field has an autocomplete, which you can call with "Ctrl + Space".

3. Use the Save button to save your project or cache it in case of extensive scripts, because the screen is not automatically exited.

32 The System Configuration

In the following all relevant settings of the phone system are listed.

32.1 The System Overview

The System Overview provides a tabular summary of the most important data of your Phonalisa. It shows you information of the version, the current system status and a brief outline of the system statistics.

32.1.1   Version Overview

The first table provides a brief summary of the version data. These are particularly useful in support requests because they provide information about the current status of your Phonalisa and help to locate problems quickly.

1. The Phonalisa Version shows you the system stand of your Phonalisa.

2. Here you will find the release date of your version.

3. The Release Branch indicates whether your current repository is based on the stable, testing or experimental branch.

4. The Release Commit ID provides information about the last commit made. This will be especially useful for support queries because it helps developers to find the date and type of recent changes quickly.

Version Overview	
Phonalisa Version	6.2.0 **1**
Version Release Date	2016-03-14 **2**
Version Release Branch	stable **3**
Version Release Commit ID	795aad16a9c527c0fff29a75f48f4d5aeb248a64 **4**

32.1.2   The Current System Status

1. Here you will find the current number of users of your system.

2. In this line, you receive information about the number of active SIP accounts.

3. Here you can see the number of Gateway your Phonalisa.

4. In this area, you will find the number of registered gateways.

Current System Status	
Current number of users	18 **1**
Current number of active SIP-Accounts	4 **2**
Current number of Gateways	2 **3**
Current number of registered Gateways	2 **4**

32.1.3   The Statistics

Statistics give you a brief information about the number of received calls from users or queues per year on your Phonalisa. For a more precise breakdown of the data, see the chapter 11 The Statistics.

Statistics Overview	2016	2015	2014	2013	2012	2011	2010	2009
Received user calls	805	16	0	0	0	0	0	0
Received queue calls	36	0	0	0	0	0	0	0

32.1.4 The Repositories

Repositories are directories which serve the centralized version control. They include software packages and their associated metadata, such as package descriptions, information about dependencies and change logs. The installation or update of the Phonalisa is handled by the package manager. The advantage of it is to upgrade the entire system as well as individual applications of the Phonalisa therefore allowing a centralized access to current updates or the latest enhancements.

Phonalisa knows three different kinds of repositories: stable, testing and experimental.

experimental:

The experimental stand contains new features and bug fixes which are still in development. This state frequently receives new updates which fix errors occurring in the alpha test phase, containing new features or extending existing functionality. This version represents a development preview and therefore is only available for selected alpha testers.

testing:

The testing repository contains a first version which is passed on to beta testers for testing. All essential functions are implemented but not yet fully tested. Errors which usually only occur in practice can thus be detected and corrected before the release while still in the beta test phase.

stable:

The stable repository finally, contains a finished version of Phonalisa which is intended for publication to the customer.

32.2 The Local Settings

Here you can change settings that affect your Phonalisa telephone system on the local level.

1. Enter the name of your Phonalisa. You can see it in the top of your browser window. This function helps to distinguish systems if you are using multiple Phonalisa telephone systems.

2. In the drop-down box you find all available time zones.

3. You can choose from 3 different date formats. The formats are: "Day.Month.Year", "Month/Day/Year" or "Year-Month-Day". The selected setting can then be found, for example, on the dashboard or in the calendar.

4. The display of the time offers the 24h time format and 12h (am/pm) format. Depending on your habits, you can select the time format.

5. Here you can specify the default system language.

6. Here you can change the name pattern. In the empty boxes you can enter various punctuations to separate the information from each other. Possible entries: Title. Firstname Lastname - Company (or company title). Firstname Lastname (or V.) Lastname, Company. This setting is applied on the dashboard and in the presence monitor.

7. Here you can change the UI look of your Phonalisa-Version. You can currently choose from 4 different designs.

Note: For almost all changes you made you have to log out and back in completely in order to take effect.

32.2.1 The UI Themes

The following color themes are available:

Aqua Thunder
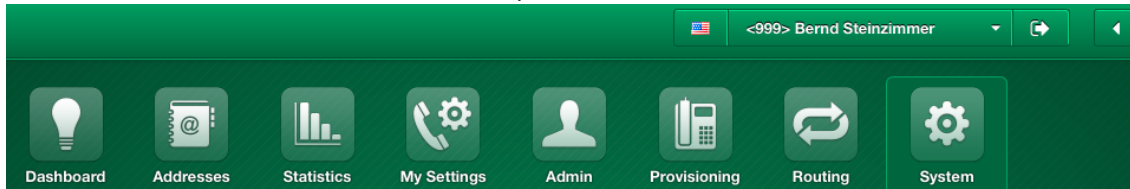
Bloody Thunder
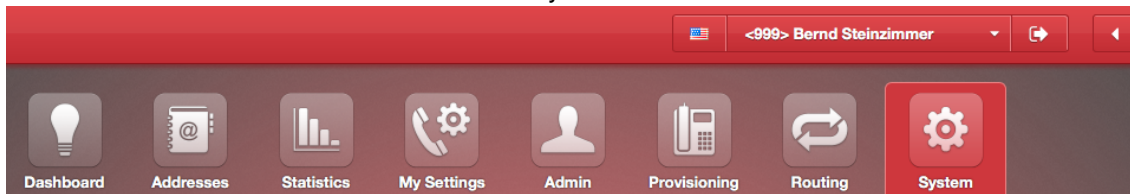
Ocean
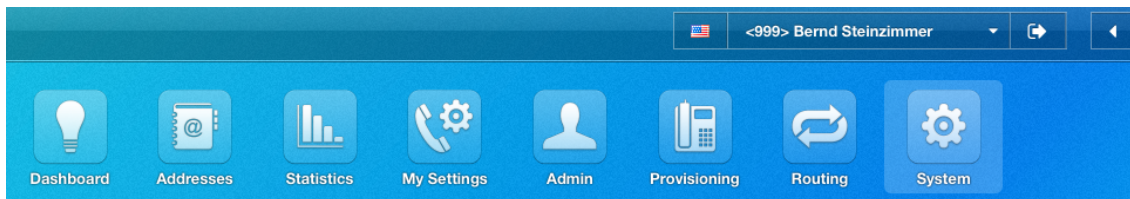
Blueberry Soda
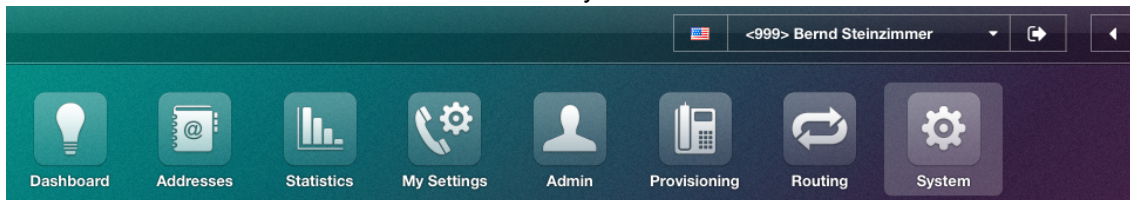
Default
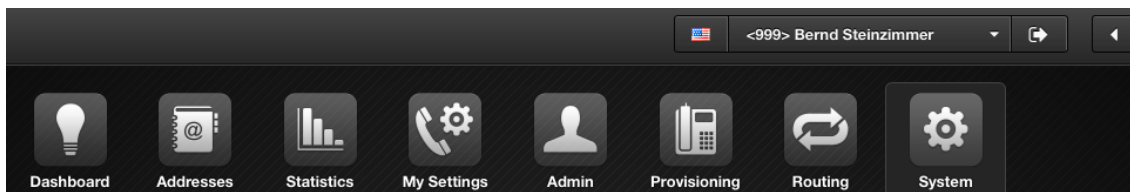
Green Rain
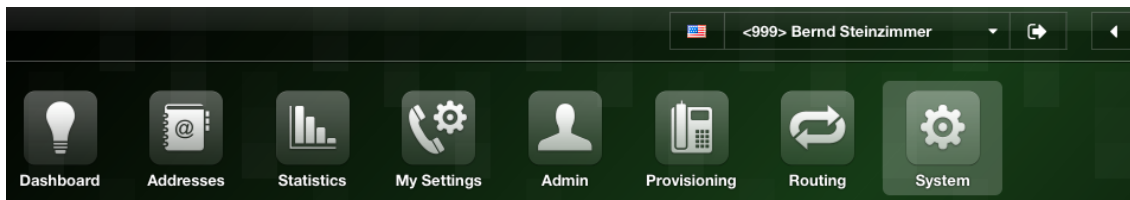
Bloody Rain
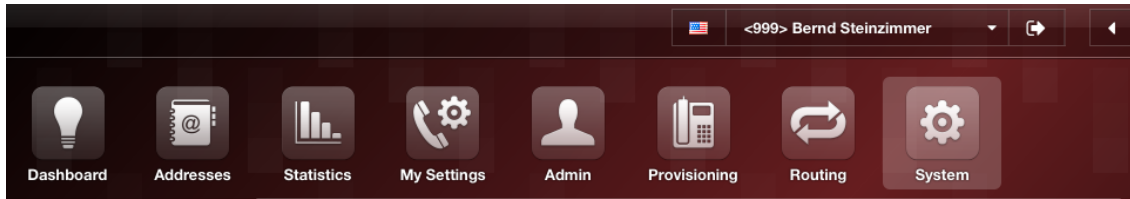
Earth Colors
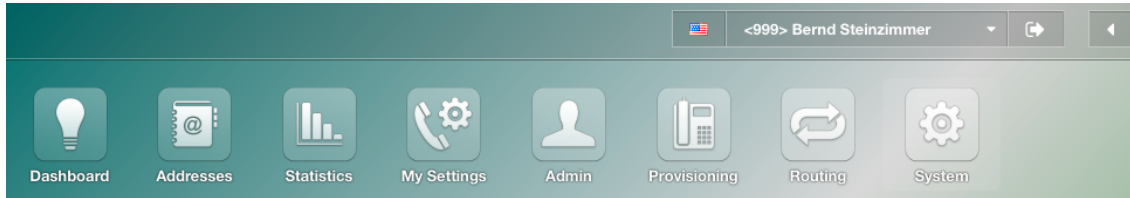
Pink Smile
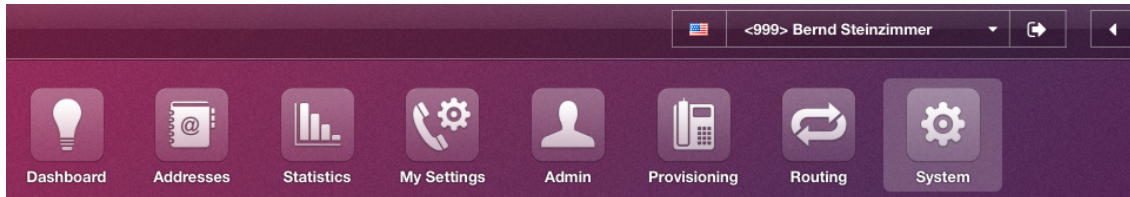
Phantom
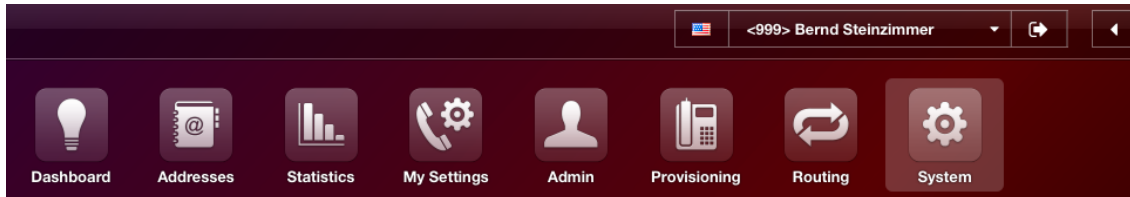
Jungle
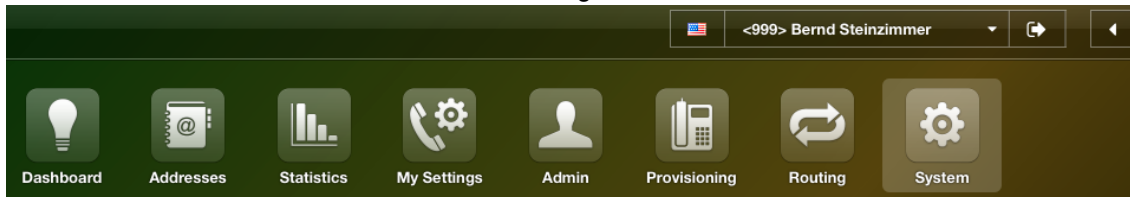
Blue Universe
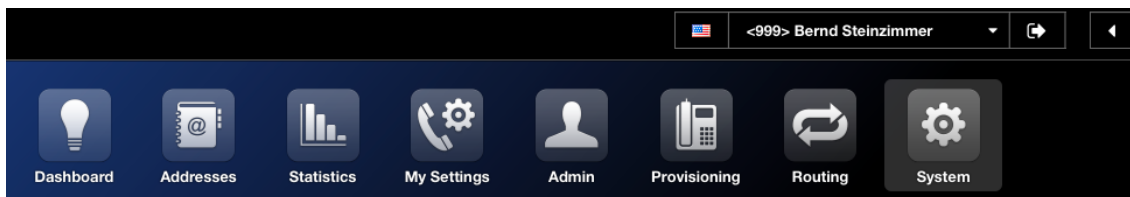
Colored Universe
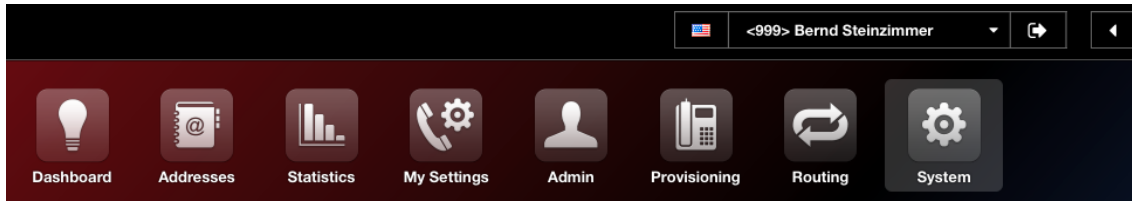
Red Universe
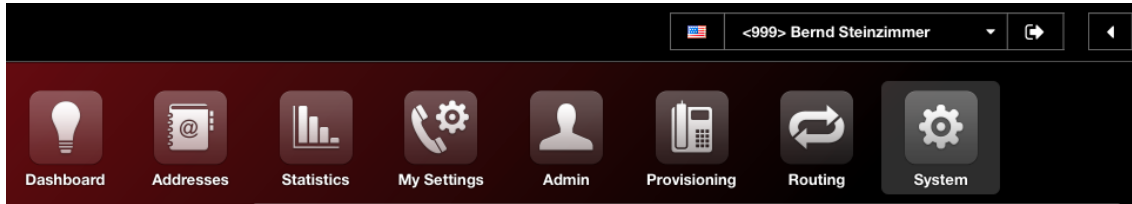
Sunrise
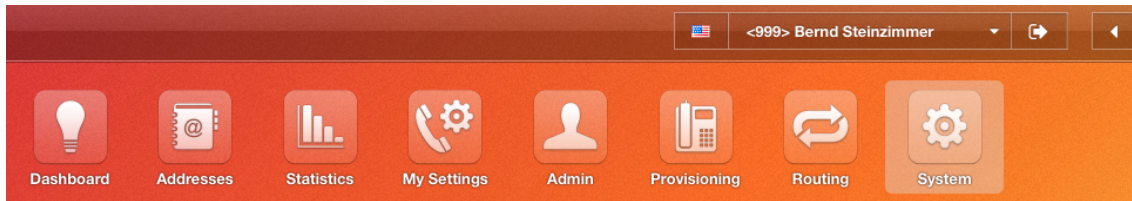
32.3 The Dialplan

Dialplan Part 1:

Option	Value	Default Value	Description
Always ask for PIN to access voice mail	[ ]	[ ]	If you activate this option every user will be asked to enter the PIN to listen to the Voicemails.
Always ask for PIN at user-login on a device	[X]	[X]	If you activate this option then every user will be asked to enter the PIN to login on a device.
Feature code for user-login on a device	*0	*0	Feature code for user-login on a device.
Feature code to activate a call forwarding	*2	*2	The call forwarding code is the key which can be assigned to a call forwarding (Settings -> Call forwarding).
Feature code for speed dial	#	#	Feature code for speed dial.
Feature code to activate and deactivate DND mode	*43	*43	Feature code to activate and deactivate the "do not disturb"-mode with a device.
Feature code for the "Eavesdrop"-Function	*86*	*86*	Feature code for Monitoring function. It is possible eavesdrop on calls, e.g. for the training of agents in call centers. Pressing button 1, you can talk to the agent without the callers notice. Pressing button 2, you can speak to the caller without the agents notice. Pressing the button 3, you can make a three-party conference with all participants.
Feature code for "Pick-Up"-Function	*81	*81	Feature code to make a "Pick-Up". So you can pick up a phone call.
Feature code for Voicemail-Menu on the phone	80	80	Feature code to manage the Voicemail on the device.
Feature code for conference	88	88	To enter or open a teleconference (type 88 and choose any 3 or 4-digit number).
Default Queue-Playlist	Standard		Systemdefault-Playlists for Queues.
International prefix of this PBX (e.g. the prefix of you country)	0049		This should be set to the international prefix of your country. This means the prefix will be automatically be replaced by "0" if you dial from the address book for example.
Automatic Dialout Prefix			This prefix will automatically be added to your dialed number if you dial from the addressbook. The best is you set this to your favorite dialout prefix.

Dialplan Part 2:

Option	Value	Default Value	Description
Global dialtimeout (seconds)	45	45	This timeout will be used as the ringing-timeout for incoming calls to users or groups. If the timeout is reached the call will be dropped or forwarded to the Central intercept position (if some). Of course, call forwardings can overwrite these settings.
Max. duration of voicemails in seconds	360	360	Max. duration of voicemails in seconds.
Central intercept position			Calls which can't be routed to users or groups e.g. in case of a user busy and no call forwarding defined this target will be used to complete the call.
Behaviour in case of "line is busy"			Enter here how a call is handled in case busy. This setting will not work if the user has activated a call through for busy.
Emergency Numbers	110, 112, 911	110, 112, 911	Here you can specify a comma-separated list of emergency numbers. These numbers cannot be selected as extension.

32.4   Set the provisioning values

To edit the provisioning values, go to "System" -> "Configuration" and to the "Provisioning" tab.

Option	Value	Default Value	Description
Provisioning-Server: Protocol	http:// _____	http://	Here you can configure the protocol that is to be used for the provisioning operation.
Provisioning-Server: Host	192.168.24.149	192.168.1. 130	With this the server for the phone provisioning is specified. Please be careful! This value should be set correctly in any case, because provisioning process would not be possible.
Provisioning-Server: Port (HTTP)	80		The port on the provisioning server which is used by default. With the Siemens OpenStage this value will be overwritten by your own configuration.
Provisioning-Server: Port (HTTPS)	443		The port on the provisioning server which is used by default. With the Siemens OpenStage this value will be overwritten by your own configuration.
Provisioning-Server: Path	/	/	The path on the provisioning host where the files for provisioning are located.
Provisioning-Server: SIP-Host	192.168.24.149		The address of the SIP-Host.
Provisioning-Server: NTP-Host	192.168.24.149		The address of the NTP-Host.
Extern API-Url		192.168.1. 130:28888	Here you can configure your extern API-Url.
Intern API-Url		192.168.1. 130:28888	Here you can configure the intern API-Url.
Intern Subnet list			Please enter a comma separated list of your internal subnets. eg. : 192.168.130.0/24, 192.168.130.5/24
Provisioning Profiles for logged out phones		Empty	Change here the provisioning profiles for logged out phones.
Key Profiles for logged out phones		Empty	Change here the key profiles for logged out phones.
Standard expire time		3000-3600	Enter here the default expiry time in seconds. You can specify a range (eg 3000-3600) or a fixed value (eg 3600).
Call initialization networks			Please enter a comma-separated list of call initialization is allowed networks.
Call service authentication-token			Enter here the call service authentication-token
Call service authentication-token			This token is used to authenticate the use of a call-service URL. It is automatically generated when running the Wizards.
Custom Ringtones	[X]	[X]	Here you can specify whether users are allowed to upload their own ringtones.

32.5   The System

Option	Value	Default Value	Description
Log-Level	VERBOSE	VERBOSE	You can select a log level from the dropdown box. "DEBUG" should only be used during a test phase for debugging, since the log file will be very large very quickly.
Automated Backups			Here you can specify in which interval automated backups will be made based on a CRON rule.
Archiving period of automatically generated backups in days			Here you can specify after which period automatically created backups will be deleted. If you do not want automatically created backups to be deleted enter "0".

32.6   Edit the Messaging settings

To edit the messaging settings, go to "System" -> "Configuration" and then on the "Messaging" tab.

Option	Value	Default Value	Description
XMPP / Jabber Domain	phonalisa.my-wan.de	phonalisa.local	Please enter your XMPP / Jabber Domain.
SMTP-Server for outgoing e-Mails	smtp.phonalisa.de	smtp.phonalisa.de	Please enter your Mailserver.
SMTP-Port	25	25	Please enter the Port for the SMTP-Server.
Use verification for sending email?	[X]	[X]	If you enable this option, the SMTP verification is enabled.
SMTP-Username	fax@phonalisa.de		Please enter the username for the SMTP-Server.
SMTP-Password			Please enter the password for the SMTP-Server.
Use TLS for emailing?	[X]	[X]	If you enable this option, the TLS encryption for SMTP is enabled.
Enable checking of TLS certificate?	[ ]	[ ]	If this option is activated, the TLS certificate check for the SMTP will be enabled.
Sender Name for system generated e-Mails	Phonalisa PRODUKTIV	Phonalisa Telefonanlage	Please enter the Sender SMTP-Name for e-Mails.
Sender Address for system generated e-Mails	noreply@phonalisa.de	noreply@phonalisa.de	Please enter the Sender SMTP-Address for e-Mails.
Send welcome e-Mail to new users.	[X]	[X]	Enabled/disable welcome e-Mail sending to new users. The e-Mail contains the login-data.
Fax Header Text	Musterfirma GmbH		This text will be displayed in the left column of your fax head.
Fax Header Number	+4936424760833		This number will be displayed in the right column of your fax head.

To check your e-Mail settings, you can send a test e-mail. Enter your email address in the box and click "Send".

If everything works fine, you will immediately receive an e-mail. The test e-mail looks like this:
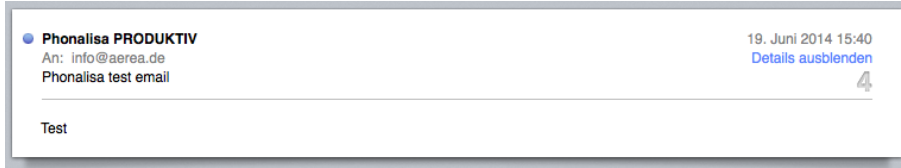
32.7   The Proxy settings

To edit the proxy values, go to "System" -> "Configuration" and then click the "Proxy" tab.

Option	Value	Default Value	Description
HTTP-Proxy: IP-Address	192.168.24.10		Please enter the IP-Address for your HTTP-Proxy.
HTTP-Proxy: Port	3128		Please enter the Port for your HTTP-Proxy.
HTTP-Proxy: Username			Please enter the Username for your HTTP-Proxy.
HTTP-Proxy: Password			Please enter the Password for your HTTP-Proxy.
HTTPS-Proxy: IP-Address	192.168.24.10		Please enter the IP-Address for your HTTPS-Proxy.
HTTPS-Proxy: Port	3128		Please enter the Port for your HTTPS-Proxy.
HTTPS-Proxy: Username			Please enter the Username for your HTTPS-Proxy.
HTTPS-Proxy: Password			Please enter the Password for your HTTPS-Proxy.

32.8 The SIP settings

To edit the SIP settings, go to "System" -> "Configuration" and to "SIP". As the settings are very extensive it is listed in two parts.

Option	Value	Default Value	Description
External SIP IPv4	auto-nat	auto-nat	Used as the public IP for SDP. Examples: ip address: "12.34.56.78" STUN-server : "stun:stun.server.com" DNS name: "host:host.server.com" where fs.mydomain.com is a DNS A record useful when fs is on a dynamic IP address, and uses a dynamic DNS updater.
Allow incoming calls from anonymous callers	[X]	[X]	Used to call your PBX from external via a public SIP-URI. For example someone could dial sip:admin@yourpbx.tld:5080 and would reach your user named "admin".
Use compact headers	[ ]	[ ]	Use compact headers to decrease the size of SIP-Headers. **WARNING: This can lead to compatibility issues with older SIP-phones and providers.**
IP's für the SIP-Server to listen on			Here you can define which IP Address(es) the SIP Server is listening on. The IP is obtained automatically. Leave blank if unsure, but you will have to set this if you have a multihomed server setup.
Internal SIP Port	5060	5060	Listen port for the internal SIP Profile.
Internal SIP Port (TLS Encrypted)	5061	5061	Listen port for the internal SIP-TLS Profile. TLS have to be enabled to take effect.

Explanation lower part:

Option	Wert	Standard	Beschreibung
Enable internal TLS encryption	[ ]	[ ]	Enables the internal TLS encryption. **Warning: A certificate is needed or your System will not work!**
IP Address for external SIP Profile (listen)			IP Address to listen on for the external SIP Profile
External SIP Port	5080	5080	Listen port for the external SIP Profile.
External SIP Port (TLS Encrypted)	5081	5081	Listen port for the external SIP-TLS Profile. TLS have to be enabled to take effect.
Enable external TLS encryption	[ ]	[ ]	Enables the external TLS encryption. **Warning: A certificate is needed or your System will not work!**
Enabled / Preferred Codec List*	PCMA, H263, H263-1998,H263-2000, H264	G722, PCMA, PCMU, GSM	Setup your enabled codecs for video and audio. You can set your codec priority, too. Possible codecs are: G722, PCMA, PCMU, G729, H261, H263, H263-1998, H263-2000, H264 The Codecs "G729, H261, H263, H263-1998, H263-2000, H264" are only available in passthrough mode, so they will only work between endpoints which both supporting the same codec.

* Important !!! If you make any changes to the codec setting the system has to be restarted to apply the changes.

32.9   The RTP settings

Go to "System" -> "Configuration" and to "RTP".

Option	Value	Default Value	Description
Globally enable ZRTP	[X]	[ ]	Globally enables or disables ZRTP support. It is also possible to enable it for a single user.
Allow ZRTP-Key Passthrough (Key Handshake)	[ ]	[ ]	Allowed ZRTP-Key Passthrough (Key Handshake).
Only forward RTP if ZRTP is active (no trusted Man-in-the-Middle)	[ ]	[ ]	If enabled, RTP is only forwarded when ZRTP is active (no thrusted Man-in-the-Middle).
Globally enable SRTP	[X]	[X]	Globally enables or disables SRTP support. It is also possible to enable it for a single user.
RTP port-range start	16384	16384	Defines from which port the RTP range starts.
RTP port-range end	32768	32768	Defines to which port the RTP range is used.
External RTP IPv4	auto-nat	auto-nat	Used as the public IP address for RTP. Can be an one of: ip address: "12.34.56.78" a stun server lookup: "stun:stun.server.com" a DNS name: "host:host.server.com" where fs.mydomain.com is a DNS A record-useful when fs is on a dynamic IP address, and uses a dynamic DNS updater.

33 The Date/Time-Profile

The Date/Time profiles can be used in several ways. They may create e.g. date/time profiles where you list all the holidays for the entire year, deposit special opening times or list the pause times. It it possibility to combine the date rules and the week rules in a profile together. The date/time profile then can be selected in a call forwarding system (by a user or directly with a System Call Forwarding) or, for example redirect to an voicemail with a holiday announcement.
You can use time profiles also in the routing system (inbound and outbound).

33.1 Create a Date/Time profile

For this, go to "System" -> "Date/Time-Profile" and click "Add new Date/Time-Profile".

1. Enter a name and optionally a description.

2. Click "+".

3. Select the date. Use the arrows to scroll backwards and forward in the months.

4. Now determine the time -> first the hour, then the minutes.
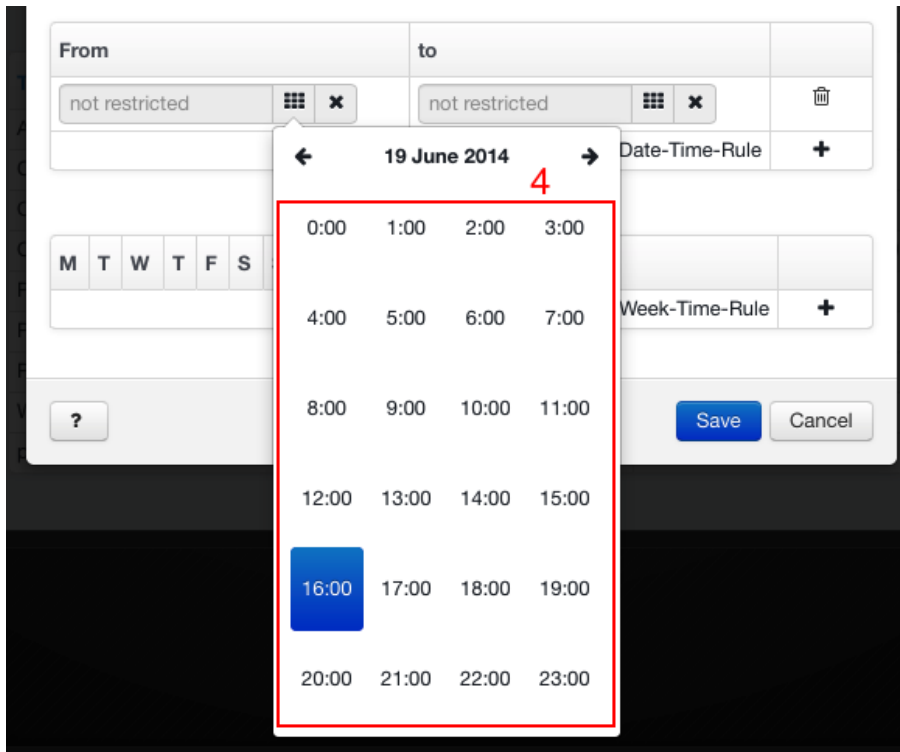
Proceeded with the end date alike. You can then simply click "+" and select another date (the number is not limited).

In the example, the final Date/Time-Profile "Holiday" is shown with date rules.

Update Date/Time-Profile: Holiday	✕

Title *	Holiday
Description	Summer

From			to			
19.06.2014 00:00	⊞	✖	20.06.2014 00:00	⊞	✖	🗑
22.06.2014 00:00	⊞	✖	28.06.2014 00:00	⊞	✖	🗑
				Add new Date-Time-Rule		✚

M	T	W	T	F	S	S	In the time from	to	
								Add new Week-Time-Rule	✚

?   Save   Cancel

33.2   Create a time profile with Week-Time-Rules

Go to "System" -> "Date/Time-Profile" and click "Add new Date/Time-Profile".

1. Enter a name and optionally a description.

2. Click "Add new Week-Time-Rule" and assign the weekdays

3. Click in the time panel or on the clock and enter the start time.

4. Determine the end time for this week rule.

Now simply click "+" and select new week days and times.

In the example, the time profile "outside business hours" are described. You can edit any time profile or add additional rules.

34 The Announcement

Similar to the user announcements is the system announcement. The only difference: The user announcements can only be used by a user, for his own call forwarding and the system announcement can be used system wide.

The system announcements are required for the following functions:

- Call forwarding (for voicemail texts, announcements).

- Welcome text in queues.

- Welcome message when entering the conference room.

- All announcements in the IVR-menu.

- Announcement before an internal / external call.

34.1 Create a system announcement

Go to "System" -> "Announcement" an click "Add new Audio-File".

1. Enter a name and optionally a description. If you do not enter a name, the name of the uploaded file will be used. If you record a self-spoken announcement the name will be "record".

2. To upload a file select "Choose File".

3. To record a self-spoken announcement, click "Start Record". **Important:** If you are logged in as admin, you have to switch to a user, which then can record the announcement. The phone, which is assigned to you as a user is then ringing.

4. Show announcement "Use Case".

Warning: You can not upload a file and record an announcement in one process.

35 Music On Hold

In the Phonalisa phone system you can use music on hold in several places, e.g. conference rooms, queues etc.

35.1 Add a new playlist

1. To create a new playlist, click "+Add new playlist". In the new window, enter the name of the list and an optional description. Then save the entries.
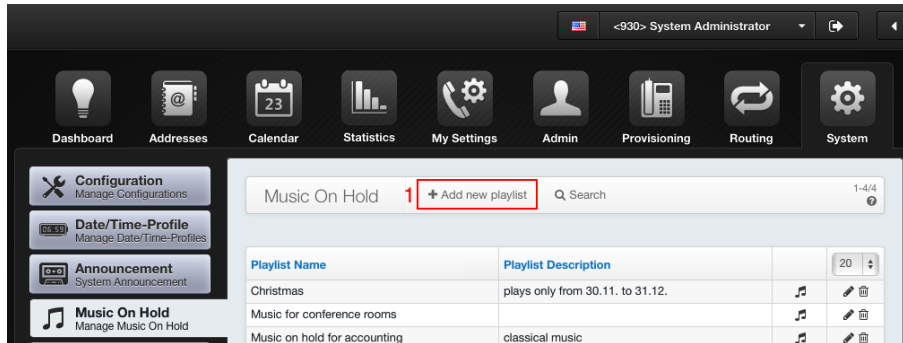
35.2 Assign audio-files to the Music on Hold

You will see the created playlists. Now click the button to assign audio-files to the list.
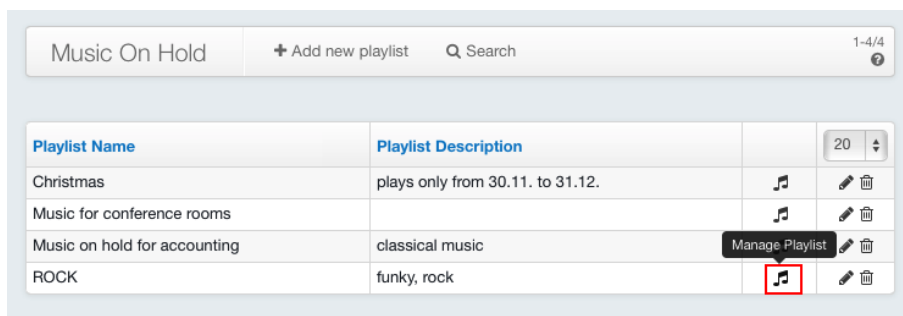
1. Click "Add new Audio-File" for upload. In the opened window you can enter a name and a description. Select a .wav-file from your files and click "Upload". If you have not entered a name and save it, the file name will be used.

2. Here you have an overview of which audio-files are assigned to the playlist.

3. By clicking on the edit button, you can change the name, description text or the file. To delete the song, you click delete button.
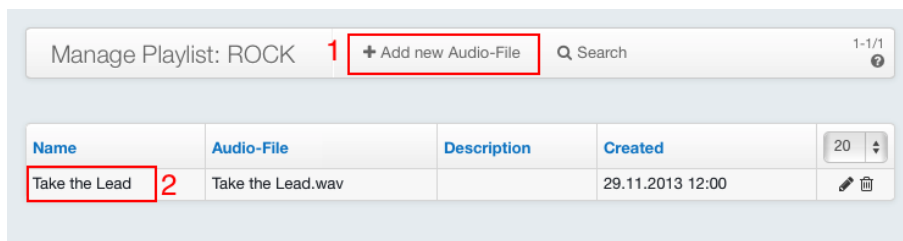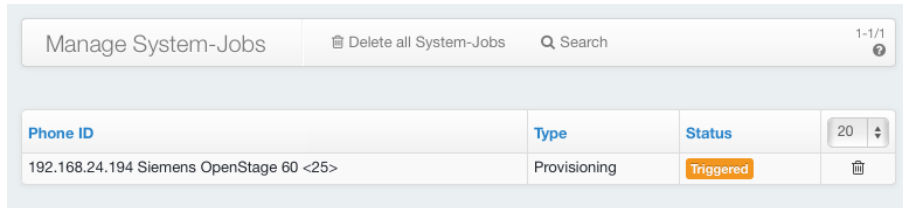
36   Manage the System Jobs

In the system jobs, you can see the phones, e.g. which are reloading there provisioning. Please note, there can be delays when many jobs have been triggered because jobs are only processed one after the other.

The example shows that, a tiptel has been triggered its provisioning.

Manage System-Jobs	🗑 Delete all System-Jobs	🔍 Search	1-1/1 ❓

Phone ID	Type	Status	20 ⇕
192.168.24.194 Siemens OpenStage 60 <25>	Provisioning	Triggered	🗑

The following states are possible:
"Running", "Finnished", "Triggered", "Inserted", "Need to trigger again" and "Failed".

37 The Backup- and Restore-Function

Phonalisa offers the opportunity to make a backup of your database in "System" -> "Backup / Restore". Furthermore you can upload a backup after updating or replacing hardware for example or you can return to a previous state after a factory reset.

1. Here you can upload a backup to your Phonalisa.

2. At this point you can create a backup of the current state of your database.

3. The "Search" button displays search fields in the table header which will allow you to find specific backups.

4. With the "Reset to factory settings" button you can reset your Phonalisa to a previous state of operation.

5. Using the "Restore" button, you can return to the previous system stands.

6. Allows you to download a particular backup.

7. If a backup is no longer needed, you can remove it using the dustbin.

37.1 Creat a new Backup

After clicking "Create new Backup" you can enter a short, descriptive comment for your backup. When pressing "Save" it can take a few minutes until it appears in the table depending on the size of your database.
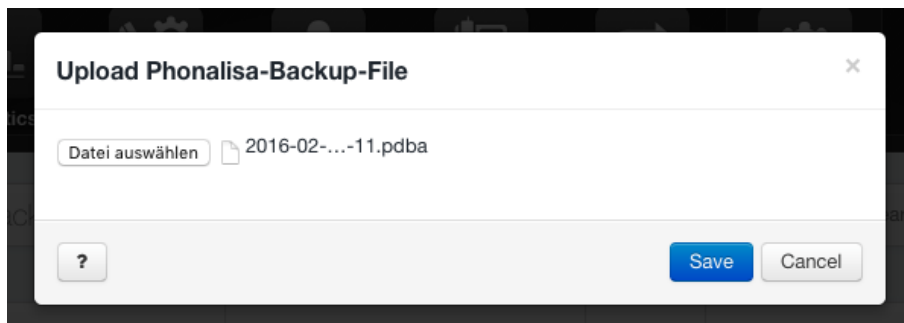
Warning: Please note the backup is an actual state of your database at the time you click "Save". Changes made later will not be considered.

37.2 Upload a Phonalisa-Backup-File

If you want to upload a backup file to your current Phonalisa, for example after updating or changing your hardware, you can do this by clicking "Upload Phonalisa backup file" and select a backup file to be uploaded.

37.3 Restore a Backup

You can easily restore a previous state of your system by using the "Restore" button, which you will find in the table.

Warning: Please read the warning carefully and make sure you create a backup of your current state of security before you perform a restore.
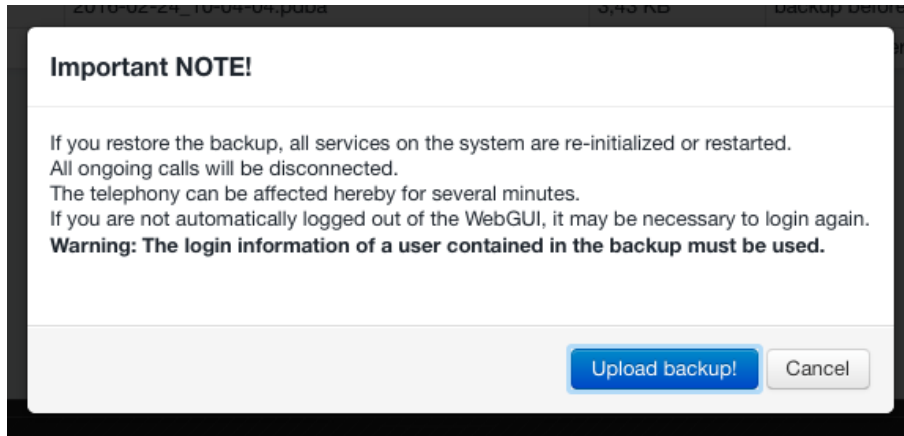
37.4 The factory reset

To reset Phonalisa to the factory settings click the button "Factory Reset" and confirm the security prompt. Please read the note in the window carefully. This function resets your system to the state before the execution of the wizard. This causes a complete deletion of the database which also contains loss of the license. Please consider doing a backup before the factory reset since these remain unaffected by the process.
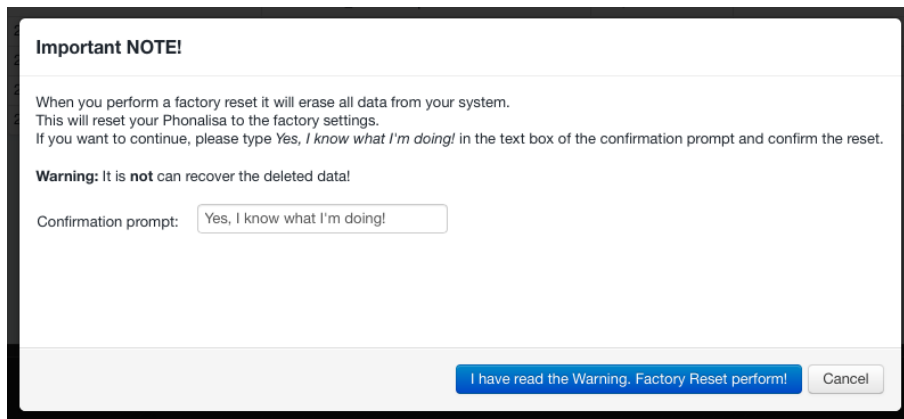
38 The License management

To upload a new license or to verify the license information, go to "System" -> "License". If you have not add any license, click "Add new License". Then select a valid Phonalisa license from your own files and click "Upload". After the license update you have to restart your system. In the table you will see all the information about the license: to whom it was issued, how many channels it has, which edition and it is valid. The "0" shows that the license has no expiration date.

39 The Call Detail Records

In the "Call Detail Records" you can see, e.g. why a user can not call to another user or why the call was terminated prematurely. All CDR-data are listed in the call history, go to "System" -> "Call Detail Records".

1. Here you can see the exact date and time of the call.

2. The source reveals where the call comes from.

3. Where the call should go or where he went, shows the target.

4. Here you can see the type, e.g. was the call "successful", was he forwarded etc.

5. The duration of the call. If the call e.g. was forwarded to a cell phone then you will not see a duration, because the call was forwarded successfully.

6. This is the internal extension which has been addressed internally.

7. If you click the magnifier, a detailed history will be shown. Here you can then analyze the case of an error, point by point - what has happened - where and when.

40   The Log Viewer

With the log viewer, you can search for errors and warnings on your Phonalisa telephone system. Proceed to "System" -> "Log Viewer".

Log construction:

1. Here you can view the log, in full-screen mode.

2. If there are values shown already in the information screen, you can clear the window with this button.

3. If you have many values in your log or you have changed the log level, you can clear the cache.

4. Here you can select the log level. You can choose between "Debug" (troubleshooting), "Verbose" (detailed display of the values), "Warning" (warning display) and "Error" (display of errors and failures).

5. Screen / display area which changed in relation of the choice (2-4).

41   The service numbers

Login on a device	*0 extension	
Logout on a device	*0*	
Activate call forwarding	*2 call forwarding feature code	The call forwarding code is the key which can be assigned to a call forwarding (Settings -> Call forwarding). This service number can be changed in the system settings.
Feature code prefix for speed dial	#speed dial	This prefix can be change in system settings.
Feature code to activate and deactivate DND mode	*43	Feature code to activate and deactivate "Do not disturb"-mode from the phone
Login and out from a queue	*5 queue	Log an agent out from a single queue, if the agent is not a static member.
Logout from all queues	*5*	logout an agent from all queue
Check voicemailbox	80	This prefix can be change in system settings.
"Pick-Up" a call	*81*extension	Prerequisite for the pickup of a call is the pickup authorization, it must be set of the groups settings.
Intercom to a group	*85*group-extension	Activates Intercom for a whole group.
Intercom to a user	*85*extension	Activates Intercom on a users phone.
Feature code for the "Eavesdrop"-Function	*86* extension	Feature code for Monitoring function. It is possible eavesdrop on calls, e.g. for the training of agents in call centers. Pressing button 1, you can talk to the agent without the callers notice. Pressing button 2, you can speak to the caller without the agents notice. Pressing the button 3, you can make a three-party conference with all participants.
Feature code for conference	88	To join or open a conference choose 88 and any 3- or 4-digit number on the phone. This service number can be changed in "System Settings".
Check account balance	*88*	Prerequisite: An account balance must have be set in "Admin" -> "Billing"
Activate and deactivate night circut	*89*group-extension	This will activate and deactivate night circut.

Part III

Frequently Ask Questions

42 Questions about the user management

42.1 I do not receive e-mails about incoming voicemails. What is wrong?

There can be several reasons:

1. You have not set an e-Mail address for the user. Help under 15.1

2. You have disabled the "Voice Mail Notification" in "My Settings" -> "Notification Setup". Help under 12.7

3. The configuration of the "Messaging" settings in the system are faulty. Help under 32.6

42.2 My voicemails are not displayed on my dashboard. What is wrong?

Please go to "My Settings" -> "Notification Setup". Here you have the "Delete Voicemail after E-Mail Notification" function. When this feature is enabled, you no longer get the messages displayed on your dashboard. Information can also be found under 12.7.

42.3 How do I know, who the system administrator is?

In "Admin"->"Users" there is the panel "User is System-Admin" in the "Advanced" tab of the corresponding user. If you enabled it, the user will be an system admin. It can only be disabeld by anothered system admin so it is guaranteed at least one system admin remains. Apart from that, there is no restriction on the number of system admins.

42.4 I want to move a user from the "All Users" group to another group. How do I do that?

There are several way to do that:

1. Via the user management. Help under 15.1

2. Via the group management. Help under 16.3

42.5 I have a immediate call forwarding to my voicemail box. Some users still reach me internally - Why?

In this case the boss-secretary-circuit is enabled. You are the boss and one or more secretaries are assigned to you. This secretaries can still reach you internally. More information in the item 15.3.

42.6 I want to configure the am/pm time format for me. Can I do that just for me or does the whole system get the setting?

You can also activate this function only for your own user. This can be set in "Admin" -> "Users". Edit the user and set the appropriate time format setting in the "Advanced" tab.

43 Questions about the group management

43.1 Why can I not pick-up conversations?

Most likely, you do not have the pick-up-permission in your group. You can find help in the point 16.4. Dial then *81*extension (- the user where you want to pick).

43.2 I do not see the statistics of my group. Why is that?

You need the statistics-permission to access the statistics of the corresponding group. For more information, see the point 16.4.

43.3 My presence monitor is empty. Where can I configure it?

To see the members of your group in the presence monitor, you need the "Monitor" permission. More information in the point 16.4.

43.4 I want my phone rinings with 5 seconds delay after the others. Is that possible?

Of course. You need to edit the ring delay in the group management. Select via "Drag & Drop" your desired time at the time bar. Then save your entries. For more information, see the point 16.9.6.

43.5 What happens to a group call if the timeout case occurs?

The timeout can be set in the system settings ("System" -> "Configuration" -> "Dialplan"). This is valid for group calls and calls to the user. You can set a "Redirection target on timeout" in the group settings so the call will not be completed or rejected. If not already done, the "Central intercept position" of the group will be used. If you have none defined the "Central intercept position" of the system will be used. If none is set at all the call will be terminated.

43.6 The group "Support" must not call on cell phones. Is that possible?

Of course, you need to assign a blacklist (for cell phone numbers) for outgoing calls for this group. Edit the group and go to "Advanced". There you can select a "Blacklist for outgoing calls". In the point 24 you can find more information about blacklists.

43.7 I want to set an announcement on a single phone. How can I do that?

For this you need the permission "Intercom". This has to be set to the group of the participant, on whose phone you want to make an announcement. Procedure: Dial *85*extension (- of the user whereby you want to make an announcement).

43.8 I want to make an announcement on the phones of a whole group. Is that possible?

Of course, just follow the same procedure as described in point 43.7 , but with one difference. You do not dial the user phone number but rather the phone number of the group (*85*group-extension-number).

43.9 The phones of an entire group should all have the same key assignment. Do I need to set it individually for each user?

You can assign a key profile to a group. The different device types are non-essential. For more information, see the item 22.4.3.

44 Questions about the address book and the calendar

44.1 I miss the CSV import icon in my address book. How can I activate it?

You can do this in several ways:

1. "Admin" -> "Addressbook". Here you can select the address book you want to import data to. Edit the address book and enable the CSV-import in the corresponding group row. The CSV-import is missing in the internal address book, because it is not allowed to import data to it.

2. In "Admin" -> "Groups" you can edit your group. In the "Addressbooks" tab you can enable the CSV-import permission for the corresponding address book.

44.2 Why I cannot change the name and e-Mail address in my internal address book?

These entries are adopted from the user administration and can only be changed there.

44.3 I have created a new address book - but do not see it. What is the cause?

You need to give your group the read or write access to the address book. Have you done this - the new address book will be visible in "Addresses".

44.4 In an address book I can see the entries indeed - but I can not edit them. Why is that?

You only have read access to the address book. Although you can also write e-Mails and dial phone numbers via click to dial, but you can not edit the entries.

44.5 I want to move entries from an address book to another. How does this work?

Select the entries you want to move. Go to the "Group-Action"-dropdown box and select "Move selected address(es) to addressbook: ...". In the dropdown box you will only see address books, you have write permissions to. Confirm the dialog. The entries disappear from the current address book, and now are in the selected address book.

44.6 Can I delete all entries in an address book at once?

Yes, this is possible. Select "Select all addresses", then "Delete selected address(es)" in the "Group Action"-dropdown box. This process may take a few seconds, depending on how many addresses are deleted.

44.7 I miss one of my self-created contact types.

The self-created contact types are only selectable if at least one entry has that type. If the last entry with this self-generated contact type will be deleted, then the contact type will be deleted too.

44.8 I created a new calendar - but I can not see him.

For this, you need read respectively write permissions. In the item 10.3 you can find more information.

44.9 Some events in the calendar are gone.

To look at your events in a given calendar, you must click the appropriate calendar top of the bar. In order to have a better overview, you can switch single calendar on and off. With the button month, week and day, you can also change the view.

44.10 I miss one of my self-created event types? Where did it gone?

It is the same like the self-created contact types. If the last event with of the self-generated event type will be deleted, then the event type is not longer listed in the event type selection.

44.11 Although I see the events on the calendar - but can not change them. Am I missing a permission?

Yes, you need the write permission for this calendar. Although you have the read permission to view the events - but it is not allowed to open or edit it.

45 Questions about the administration

45.1 Where can I find the SIP password for an user?

You will find passwords in the "Admin" section in "SIP Accounts". Click the "Edit" button there. Now you will see the password of the individual SIP account. You can see which user has been assigned to the account at the same time. Please consider that a user can have several SIP accounts.

46 Questions about the phone settings

46.1 I have wrote a song. Can I set it as a ringtone?

Yes, it is possible, go to "My Settings" -> "Tone Setup". Here you can upload your song (mp3, wav) and set it as a external, internal or queue ringtone for incoming calls. This function is dependent on the particular devices. For some devices, the ringtones have to be added with the phone web interface.

46.2 I have configured different phone keys - but my phone does not show this.

Some phones need to be reboot manually. Hereto, you have to unplug the power plug of the phone and reconnect it. The phone restarts and retrieves the new configuration.

46.3 Can I hide my own phone number?

You can do this via "My Settings" -> "Service Attributes". Just select the item "CLIR for external calls". This function is dependent on the particular devices and on the used gateway. In some gateways, this setting has to be set directly in the web interface of the gateway.

46.4 When I make a phone call, there is a constantly beeping from another call. Can I stop this somewhere?

For this, you have to go to "My Settings" -> "Service Attributes". When the "Call waiting" checkbox is activated, you need to deactivate it. Depending on the phone you have to reboot it manually. For some devices, this setting has to be set up via the Web interface of the phone.

46.5 My phone signaled an busy state although I am free. What causes that have?

Here it seems that the "DND" function in "My Settings" -> "Service Attributes" is activated.

46.6 I want to set a custom call forwarding with night circuit. Why is the group, I would like to adjust, not listed?

To select a group in the call forwarding system, you have to activate the "Night circuit active" setting in the corresponding group. Then the group is also "unlocked" for a custom call forwarding.

47 Questions about the queues

47.1 If I call a queue from internal one phone always just rings. However, they should all ringing. How do I set this?

This requires you to edit the queue ("Routing" -> "Queues"). In the first tab (General) you can determine the strategy of the call distribution. Select in the "Strategy"-dropdown box the point "Ring All". Further information can be found in the item 28.2.

47.2 When is a queue "empty" ?

A queue is empty if there is no agents (internal and external) logged in anymore.

47.3 Can I be agent in multiple queues?

Yes, this is possible. You simply have to register as a user or external agent in the queues. Information can also be found in the point 28.3.

48 Questions about the Routing/Telephony

48.1 Why the phones of some users are ringing, though they are not applied in the inbound routing?

The users are most likely members of a group the inbound routing is active on. Or the users are in a child-group whose parent-groups are listed in the incoming routing. Another possibility would be, that these users were deposited as an Central intercept position.

48.2 The telephone system has 16 channels. Why can I only run one simultaneous external calls?

Please check if you activated a valid license in "System" -> "License".

48.3 A call has failed. Where can I look why this is happened?

In the menu item"System" -> "Call Detail Records" you will find the call listed. You can also find help at the point 39.

48.4 I am bothered by some callers. Can I block them?

Yes, this is possible. you can create a blacklist for this callers (see point 24) and assign them to your group or gateway.

48.5 I want to allow calls to a conference room only for a certain time window. What do I do?

This configuration can be made in "Routing" -> "Routing Inbound". Here you can choose your conference room as a target and assign a date/time profile to it. So you can, for example, create a date/time profile with the time limitation from 09:00-11:00 o'clock and assign it to your conference room. How to create a time profile can be found in item 33.

48.6 I would like to use the ringmap in my call groups. Is that possible?

Of course. You need to edit the ring delay in the call groups in the menu "Routing". Select your desired time at the time bar via "Drag & Drop". Then save your entries. For more information, see the point 27.

49 Questions about the System Configuration

49.1 Why the system Central intercept position is not used?

The user is a member of a group in which a own Central intercept position is set. So this is used. If the group has no group Central intercept position, then it is possible that the user has defined its own Central intercept position.

49.2 I use the eavesdropping function for training purposes. What can I do if my colleague's voice is distorted?

This issue might be caused by some codecs. Go to "System" -> "SIP" and delete all codecs except the "PCMA-Codec". Please restart your system to apply the changes.

Part IV
The Glossary

A

AB:

AB is the german abbreviation for voicemail box. For more information please see voice mail box.

Admin:

The admin is the system's administrator. He configures all settings like new users, phones, network settings, etc.

Agenten:

Phones which are logged in to a queue are called agents.

Agents (static):

A static agent is a fixed member in one or more queues and can not log in or out of these at will.

Agents (dynamic):

Dynamic agent can log in or out of a queue using feature code .

Agent Monitor:

The module of the Phonalisa PBX which offers an overview of all agents and their queues is called Agent Monitor. Further information about the Agent Monitor, see the Chapter 17th.

Announcement before answering:

This feature is used to play an announcement (as a greeting) before taking the call. A distinction is made between internal and external. Procedure: The phone rings and the user takes off. First the "Announcement before answering" is played then the user is connected to the caller.

Audio Files:

In the Phonalisa telephone system audio files are used as announcements for voicemail, welcome messages for queues or announcement before answering. These audio files can be recorded over the phone or you can upload the file. There are 2 areas in the Phonalisa to do this the system audio files and the user audio files. Audio files which where uploaded in the system area are available to all users of the telephone system. If the audio files are uploaded in user aerea, they are only available for the user.

B

Backup:

With a backup all existing settings and data will be stored. A backup of the Phonalisa database may be restored after a factory reset of the system.

Billing:

This is Phonalisa's integrated billing system which is very useful to hotels or hospitals for example to account outgoing calls. In Billing you can create price lists and determine a price for a phone call on a landline number or a mobile number. To create the billing then clicking on the button behind the billable user. A PDF with the settlement will be created.

Blacklist:

In a blacklist phone numbers can be entered you do not want to be reachable for. The CRLs can be deposited per group or in the gateway directly. Callers who are on the blacklist - get a busy signal.

BLF:

BLF is Busy Lamp Field and refers to a key on the phone. These can (depending on the device) shine or flashing, when another user whose extension number you have set to this BLF key is called. Using BLF keys, you can also use the pickup function for a colleague's call.

Boss-Secretary-Circuit:

The boss secretary circuit is a function of Phonalisa which allows certain users to ignore diversions and reach an extension anyway.

C

Cache:

The cache is a fast memory buffer, which is used to store frequently used data to increase the operating speed of a PC.

Calendar Type:

The calendar-type displays whether it is a private calendar or a group calendar.

Call Forwarding Filter:

Based on specific criteria, the call forwarding rules are created. If you want to redirect calls from outside for example - then you choose the filter "Call external source". If only calls of a particular contact type (for example, VIP customers) should be forwarded, then choose "caller contact type X in the address Y". There are 6 different filters, which can all be combined. Note: The filters are only "custom call forwarding", but not with "immediate forwarding".

Call Forwarding Rule:

In a call forwarding rule various filters are listed. You can create as many rules per call forwarding scenario as you want.
Example "Of internal and external differently Handling Calls" :
The first rule, specifies that calls are routed from external to your mobile number. In the second rule, you can then filter all "Call source internally" which guides them to your voicemail. It is only important that the rules are listed according to their valence. They are "read" from top to bottom. Accesses the first rule in a scenario - will not be considered the bottom. but you can move the rules to drag and drop into the correct position.

Call Forwarding Scenario:

A scenario is the preamble of a call forwarding rule. Possible scenarios to be applied are as "Redirection for presence status", "Diversion out of schedule", "Forward on holidays", "VIP customers control," ... and so on. In one scenario, any number of rules can be applied. Example: You create a scenario with the label "Diversion terms of call source". In this Szerio then define the appropriate rules so it gives you the most suitable solution.

Call Groups:

The call groups are similar to groups in the group system, but they are much more flexible to use since they do not use the inheritance and authorization system.

Call Source:

This refers to the origin of the call - internal or external. The call source can be selected in routing as filter. So you can handle internal calls and redirect different than external, for example.

Call-Through:

Call-Through is a function which allows a user to call a designated number from external to use the telephone system as a bridge to a target number.

Call Waiting:

If you have the "Call Waiting" enabled, the caller gets a dial tone instead of a busy signal during an active phone call . The called hears his phone beeps. Note: If the function is activated a redirection in busy case is no longer effective because a dial tone is sent to the system from the telephone.

CDR:

The term CDR stands for Call Detail Record. In Phonalisa each call is listed in the CDR table which provides detailed information about it. For example, if a call has not reached the desired destination.

Checkbox:

There are several areas in the Phonalisa phone system where checkboxes are used to enable or disable a feature on the web interface.

Central intercept position:

A Central intercept position is used in cases where a call could not be accepted, for example when a user has no call forwarding.

Click2Call:

Click2Call describes the automatic establishment of a call by clicking a phone number in the address book. Depending on the terminal, the number is dialed immediately or you have to pick up the handset or push the speaker button before.

Click2Fax:

Click2Fax is a feature which will bring you automatically to the fax transmission area of Phonalisa after clicking on a fax number.

Click2Mail:

Click2Mail behaves similarly Click2Call. After clicking on an email address the mail client is opened automatically and you can start writting the mail.

CLIP:

CLIP is an abbreviation for "Calling Line Identification Presentation" and allows you to define a number, which is to be transmitted with a call.

CLIR:

CLIR stands for "Calling Line Identification Restriction Override". This function is used to hide your phone number during calls.

Conference:

The conference system in Phonalisa allows you to perform a conference of 3 or more (internal:unlimited, external:depending on the number of channels). You can set a personal welcome message for the conference room, specify a moderator and participants record their names. Selected as target in the incoming routing - you can make the conference room directly reachable from an external number.

Config. (Configuration):

Settings made by the user in functions and modules are called configuration.

Contact Type:

The contact-type specifies which group of contacts an address book entry belongs to.

CSV:

CSV is a non-standardized data format, which is used for exchanging simply structured data.

CTI :

CTI stands for "Computer Telephony Integration" and is as a general term for software which uses the integration and coordination of interactions over phone and computer.

Custom Call Forwarding:

This call forwarding type refers to a freely configurable scenario defined by user.

D

Daily Report:

This is an overview of the calls in the statistics module of a group or queue of a day. The evaluation can also be a week, a month or create one year.

Dashboard:

The dashboard is an overview on the web interface which displays all the phone calls (including those of the own group / queue / call group), voicemail, call recording and faxes a user. You can also trigger calls, send faxes and store phone numbers in an address book from there. For more information please see Chapter 5.

Date Format:

The date format represents the manner in which calendar data is displayed.

Date Rule:

In Phonalisa date rules are used in time profiles. It is possibile to enter one or more rules in a single time profile and diverte calls for vacation time, public holidays, etc.. Simply select the date or the time span for the profile.

Debug:

This describes the process to search and find errors in the program code.

Descending:

The agents will be called according to their selected level (queuing strategy).

DND:

The function means "do not disturb" (Do Not Disturb). When enabled, your phone is treated by the system as if it has an active call. Calls can not be reached as long as it is enabled.

Drag&Drop:

Drag and Drop, deutsch „Ziehen und Ablegen" ist eine Methode welche genutzt wird, um Aktionen auszuführen oder Beziehungen zwischen zwei Objekten auf einer grafischen Benutzeroberflächen herzustellen.

Dropdown-Box:

A drop box is a control element of a graphical user interface which is used to select a value from a predefined list. On the Phonalisa web interface many kinds of dropdown boxes can be found, for example, in the selection of the filters in the call forwarding, in the selection of targets in the incoming routing, etc.

E

Eavesdropping:

The eavesdropping function is a system designed for employee training, which allows instructors of a call center to listen in on a conversation added. The trainers can then talk to the agent without the callers notice.

Empty:

A queue is empty if no agents are registered in it. Possesses a queue "static agents" it can never be empty.

Expansion module:

This is a key attachment for phones. So it is possible for the user to have a lot more available keys (for example, for Besetzlampenfelder). However, it is dependent on the terminal, whether it supports an expansion module.

External agents:

Phones which are registered via an external connection to a queue are referred as external agents.

Event:

The event is related in Phonalisa to create a new calendar entry, for example, Telephone conference, trade show, customer visit, ...

Extension:

This is the internal telephone number of a user.

F

Factory Reset:

A factory reset is performed, for example, on a telephone. The function will reset the phone to the factory settings. In the process all configurations (ring tones, keys, etc.) will be set back to the state of the phone's first activation.

Fax2Mail:

The Fax2Mail feature allows you to receive a fax via a virtual fax number which is then forwarded via email. The fax is sent to a fax server which forwards the content as an image file via mail to the recipient.

Feature Code:

This is a feature code which can be entered on the phone, for example, * 2 to activate / deactivate the call forwarding, * 0 * for logging on the phone, etc.

Firmware:

Operating software which is embedded in electronic devices and required to load and run the operating system kernel of the actual operating system.

Forwarding cases:

The call forwarding cases in Phonalisa are manifold. In groups, or caller groups, there are the forwarding cases "timeout", "Busy" and "Night". For queues there are the cases "empty" and "Timeout". Based on these cases, you can enter destinations, therefore no call is lost.

G

Gateway:

In telecommunications a gateway can be the built-in ISDN interface, a media gateway (e.g. Beronet or Patton) or a SIP provider on the Internet. The Phonalisa routing system uses gateways to decide how incoming and outgoing calls are distributed.

Groups (children, parents):

A group is a combination of several users with the aim of restructuring and manage their permissions for the use of individual functions. The permissions of a parent group are inherited by the child grou). This includes, for example, the read and write permissions of the calendars and address books.

Group Hierarchy:

The order of the groups is calledgroup hierarchy the levels of parent groups are always higher than those of the child groups.

GUI:

The GUI short for graphical user interface is the web interface of the Phonalisa.

H

Hold:

A call can be placed in a waiting state by clicking the "Hold" key on the phone. During this time the "On Hold"-music is played to the caller.

Home-Office:

The workplace of an employee located in his own home is called home office. Which can also be connected to the internal telephone system.

I

Inheritance:

Inheritance refers to the group system in Phonalisa. Each created group can be a parent group. The permissions of a parent group will be inherited by the child group. This includes, for example, the read and write permissions for the calendars and address books.

Intercom: Using intercom functionality a phone is treated as an intercom therefore no telephone handset must be lifted to establish a conversation, the communication takes place via the speaker.

IVR:

A voice menu is a function which uses automated announcements to communicate with a callers. His or her answers will then be used for intelligent routing to another IVR menu or to an endpoint (users, groups, ...).

J

K

Key Profile:

A key profile can be created for a group or for individual users. In a profile, you can store the key assignments for multiple devices (from different manufacturers). Thus, it is possible to assign your group "secretaries" the same key assignments on their phones (even if they use different phone models). You can also assign your own keys to a user's profile but, this is then higher by level than the stored profile in a group.

Key Tones:

These are the sounds you hear when you press a phone key.

L

Last Contact:

The information of the last contact can be found in the address book. Here is displayed at each of your contacts when you last called him or her. The call notes are also included in the list.

LED:

As LED stands for light-emitting diode, for example, the lights on the phone are made of them. In the case of a new voice message, the LED lights up on the phone.

Login Status:

The login status in the Agent Monitor shows you if an agent is logged off (red), logged in (green) or is a static member(gray) of a queue. Red and green refers to dynamic agent, gray static.

Locked:

This refers to the key profiles in the "provisioning". The administrator can lock key bindings, so the user does not overwrite them while making key assignments.

M

Max Muster, John Doe,...:

These are sample entries, like Lieschen Müller.

Moderator:

In the Phonalisa phone system, there are moderators for conference rooms. You can set up your conference room so the participants hear each other only after the moderator joined the conference. You can also adjust the settings so the moderator will be called as soon as the first participant enters the conference room.

Monitor Permissions:

This allows group members to see the Agent Monitor. You must assign monitor permissions to the corresponding group . You can decide whether the group is allowed to see all the queues in the monitor or just a selection.

Music On Hold:

Music on hold is an audio file which is played to the caller during his call is held or forwarded. Phonalisa includes by default a free music on hold.

N

Name Scheme:

The name scheme determins how names are displayed in your Phonalisa. You can edit the dropdown box for the name scheme accordingly. In the smaller intermediate fields you can enter different punctuation to separate the information from each other.

Network Scanner:

The network scanner allows you to search phones that are connected to your network. These can be connected to your Phonalisa after the search by triggering the autoprovisioning .

Night Circuit:

The night circuit is a function which indicates the Phonalisa PBX that the last group member has left the workplace. With the night circuit activated all incoming calls to the group will be redirected, for example, to a queue or an IVR.

O

Offline:

The user is logged out (for example, from my phone).

Online:

The user is logged in (for example, chatting).

Outbound Dial:

This is an outgoing call - external telephony.

Outside Line:

The term refers to the exchange connection an outside line for calls to the public telephone networks. In telephone systems as the Phonalisa which distinguish between internal and external calls, it may be necessary to get an outside line, an access code (for example, zero) to select.

P

Parallel Call:

This feature allows you to get a call to one / several other extensions beside yours. You can add more extensions as parallel call numbers or your mobile phone number. Please note if you use your phone as parallel call number a further voice channel is used.

Password:

The password used to authenticate a user at login.

PDF:

PDF stands for "Portable Document Format" and designates a file format which allows independent of the application software and accurate representations of electronic documents.

Permissions:

Phonalisa has an extensive authorization system. You can restrict the access rights of user groups to GUI modules. So you can, for example, lock the Administrative area and prevent unwanted changes can be made.

Pickup:

Pickup is a function which allows you to tap and redirect calls to your own phone from other users using a function code.

Pin:

The pin is used to authenticate a user on the phone.

Presence monitor:

In the present monitor you can see which user is on the phone, which is free and as well as the instant messaging status in real-time.

Presence Status:

The presence status refers to a call forwarding filter. The presence status can be online, offline, busy or absent. You can create a call forwarding rule based on the status. For example, when a colleague leaves his workplace and the presence status changes to absent - then could call forwarding rule can redirect incoming calls to another colleague or the voice mail.

Q

Queue:

A queue is used to distribute incoming calls strategically to agents in a company.

R

Record:

This is a type of audio announcements. If an announcements is recorded over the phone, they get into the list of audio files of the type "record". Announcements which were uploaded instead have their file names as type.

Restore:

Phonalisa allows you to create a backup. To import the backup again you have to perform a "Restore". In this process, you load the backup back into the system.

Ring Delay:

The ring delay can be set for groups and call groups. It represents the time delay between reception a call on the telephone system and the ringing of the called telephone. Example: Employees 1 rings for 20 seconds, after 10 seconds employee 2 is switched on, after 15 seconds employee is 3 switched, etc ....

Ringing Time:

The ring time specifies the time in seconds for call a to be unanswered before a forwarding szenarion occurs.

Round-Robin:

This is a queue strategy. It means all agent's phones ring one after another. Other strategies are as "All ringing", "Longest free agent", "agent with minimum call duration", ...

S

Scenario:

A scenario is a plan of how something should be handled by the Phonalisa. An example of this can be found in point Call Forwarding Scenario.

Service Features:

The service features are related to the telephone functions such as CLIP, CLIR, Call Waiting and DND. Each those can be activated and disabled on the web interface .

Speed Dial:

With this function you can save a lot of time by dialing phone numbers only via speed dial. The speed dial prefix can be set on the "Settings" -> "Dialplan" in the systems menu. So you can dial a long phone number stored in your speed list very quickly by entering #4 on your phone. The speed dial list can also be exported / imported or saved as PDF.

Static Member:

Agents can be logged into a queue as a static member it is then not possible for the agent log in or out from the queue. He is then a static agent. See also "Agents (static)".

Statistics:

In the statistics Phonalisa offers an overview of the various group and queue calls.

Strategy:

The Strategies describe possibilities of distributing of queue calls in the Phonalisa. These are the strategies "Round Robin", "Down", "Longest free agent", "agent with the fewest calls" ringing "in number of agents List", "Agent with the lowest call time", "Zufälig" and "All ".

System-Job:

This is an overview in the Phonalisa which is showing you currently active jobs such as network scanner, address import, provisioning of devices (specify including extension), etc.

T

Timeout:

The timeout is used in many areas in Phonalisa telephone system. It is activated when group members, agents or users can not receive a call and the phone would continue to ring.

Time Format:

The time format specifies (as well as the date format), the way in which time information will be shown in writing.

Time Profile:

As time profile, a rule is called, which is created for a date or period of time, for example to create a scenario for the forwarding of calls outside of business hours.

Toggling:

Toggling is the alternately switching between two active calls.

Token:

A token is used in the IT to control parallel processes and to avoid collisions during execution.

Transfer:

In the case of a call transfer means forwarding. There are different types: "Blind Transfer" and "Transfer with Consultation". The blind transfer routes the call without asking directly to the final destination. In the transfer with consultation the participant will be asked if he wants to answer the call.

U

UI:

The UI is the user interface from the Phonalisa. In browser (Chrome, Safari, Firefox) the web interface can be opened and the phone system can be edited.

UI-Theme:

The UI Theme describes the appearance of your Phonalisa. Depending on your taste you can choose your favorite from a wide variety of stored color schemes.

User:

A user is a participant in the internal telephone system. Users can be assigned to a group, registered as agents to a queues and much more. The administrator can also lock the access to GUI modules for certain user groups.

V

Verbose:

Verbose refers to a possible log output and describes the "detailed display of values" in the log.

VIP:

VIP is the abbreviation of "Very Important Person" and refers to a person who has a special status or special privileges.

Voice-Mail (-Box):

The voicemail box stores all voice messages. You can check your messages directly via XML-app depending on your phone or dial "80" . This takes you to a menu where you can listen to the old and new messages and or delete them. Apart from that the feature code "80" to call the voicemail can be replaced by custom feature code.

W

WAV-File:

WAV files are uncompressed audio files which provide a high quality sound.

Week rule:

The week rules can be used in the time profiles. So you can, for example, create profiles in which the business hours are configured using the week rules. The time profiles can then be applied in the call forwarding or routing system.

Wizard: The wizard accompanies the first administrative user after the installation through the first important steps.

XYZ

Xing:

Xing is a social network on the Internet. Through the connection, it is possible to transfer the contacts from the Xink network in Phonalisa (with photo). So it is possible to display a Xing profile image of the caller is on the display, for example, of a Siemens OpenStage .

XMPP:

XMPP is an open standard of a communication protocol based on the XML standard. It allows the exchange of data and is used among others by instant messenger.

Zoiper:

Zoiper is a softphone client which can be configured easily with Phonalisa.

ZRTP:

ZRTP is a work based on the Real-time Transport Protocol key exchange protocol, which is used in the IP telephony (VoIP).