

# Operation Manual

*Voclarion 2.3*



**voclarion**

# Contact Sheet

In case of a malfunction, contact the persons below.

## First Contact:

Name:

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Company:

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Phone:

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Mobile Phone:

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Fax:

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E-mail:

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Notes:

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## Second Contact:

Name:

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Company:

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Phone:

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Mobile Phone:

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Fax:

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E-mail:

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Notes:

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Make sure you follow the safety instructions at all times!

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## Publication Date and Software Version

Published on 19. september 2014. Based on Voclarion 2.3 (revision 30201). Manuals are updated continually. For the latest version check the Download Center on the Voclarion PBX Manager on a regular basis. For information about manual orders and manual invoices please contact LuLu support.

## **Errata**

We make every effort to ensure that there are no errors in the text or in the code of this book. However, no one is perfect, and mistakes do occur. If you find an error like a spelling mistake or a faulty piece of code, we would be very grateful for your feedback. By sending errata you may save another reader hours of frustration and at the same time you are helping us.

## **Open Source**

The source code for the GPL'd parts of the Voclarion software is available on request for a nominal fee. Please contact us for more information.

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## Foreword

Thank you for choosing Voclarion, the full-featured PBX Voclarion is the world's most advanced PBX nowadays available. Please check our website on a regular basis for new features and updates. Or take a few seconds to subscribe to our mailing list or RSS-feed at [www.voclarion.com](http://www.voclarion.com)

This manual is useful to you if you have your own dedicated Voclarion and it covers the hardware installation and system configuration. The additional Operation Manual covers setting up your Voclarion, including the creation of companies, users and all parts of the dial plan.

If you have any suggestions involving the Voclarion, please let us know! You can reach us by sending an e-mail to [sales@voclarion.nl](mailto:sales@voclarion.nl). Your suggestions are important to us! We look forward to hearing from you. In the meantime we hope you enjoy Voclarion.

Thanks again,

Voclarion



## **Chapter 1. Introduction**

*This manual is designed to help you configure the PBX.*

*This chapter is an introduction to Voclarion.*

## 1.1. What is Voclarion?

---

The Voclarion you are about to use is a so called Private Branch eXchange (PBX), an advanced telephone exchange that serves a particular business or office. PBXs are also referred to as PABX (Private Automatic Branch eXchange) or EPABX (Electronic Private Automatic Branch eXchange). In this manual we prefer to use the term PBX.

PBXs make connections among the internal telephones of a private organization — usually a business — and also connect them to the public switched telephone network (PSTN) via trunk lines like analogue lines and ISDN, or to the Internet using a protocol like SIP. Because they incorporate telephones, fax machines and other hardware and software applications, the general term "extension" is used to refer to any point on the branch.

PBXs are differentiated from "key systems" in that users of key systems manually select their own outgoing lines, while PBXs select the outgoing line automatically. This is called least cost routing.

### 1.1.1. Asterisk

Voclarion is based on Asterisk™<sup>1</sup>, the world's leading open source telephony engine and tool kit. Offering flexibility

unheard of in the world of proprietary communications, Asterisk empowers developers and integrators to create advanced communication solutions.



### 1.1.2. Voclarion

Voclarion is specialized in open source projects, especially Asterisk. However we really think Asterisk is great software, it is missing (in our point of view) some important business features. So, we created a telephony system named Voclarion, based on Asterisk and we added a lot of new features. Nowadays we add new improvements almost weekly, most of the time based on user suggestions, making Voclarion one of the most advanced systems available!

Some great Voclarion Features:

<sup>1</sup> More about Asterisk at <http://www.asterisk.org>

## **What is Voclarion?**

- ✓ Graphical role based user interface
- ✓ Out-of-the-box provision system
- ✓ Very advanced and flexible call distribution system
- ✓ Real time reports
- ✓ Desktop call software
- ✓ 24/7 support

## 1.2. Feature Overview

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Below you see a short overview of all features. All features are available unlimited and without restrictions<sup>2</sup>. Please visit [www.voclarion.com](http://www.voclarion.com) for more information about new features and the road map to the new software edition.

### 1.2.1. Telephony Services

- Advanced Voice Mail System
  - PIN protected
  - Separate away and unavailable messages
  - Default or custom messages
  - Multiple mail folders
  - E-mail / SMS notifications<sup>3</sup>
  - Voice mail forwarding
  - Message waiting indicator support
  - Message waiting stutter dial tone support
  - Rewind and fast forward within a message
  - Listen remotely to your voice mail (with DISA)
- Auto Attendant / Automatic Call Distribution (ACD)
  - Priority support
  - Welcome messages
  - Unlimited queues
  - Ring groups
- Interactive Voice Response (IVR) Menus
  - Unlimited amount of menus
  - Up to 99 entries / menu
  - IVR in IVR support
  - Define action on time out
  - Per menu default language
  - Dial extension while in IVR menu

---

<sup>2</sup> Hardware performance restrictions may apply. Hardware updates available on request.

<sup>3</sup> SMS optional

- Record and upload your own voice prompts by phone or external recorder
- Intro messages
- Overhead Paging<sup>4</sup>
- Flexible Extension Logic
  - Multi-layered access control
    - Role based
  - Multiple extensions per user
    - Hot desking
  - Multiple lines per extension
  - Outgoing caller ID
    - Per user / phone line and/or department
  - Configure routing of incoming calls
  - Redirect scheduling: redirect calls based on time (supporting flexible opening hours)
    - For incoming phone numbers
    - For extensions
    - Advanced recurring schedules
  - Switches: manual redirect calls
  - Define unlimited outbound trunks (Analog, SIP, IAX)
    - Least cost routing
- Phone Directory
  - Company wide phone book
  - Private phone book
  - Colleague phone book
  - Caller name lookup on incoming calls (shows name of caller on phone display)
  - Define fast dial numbers company wide and per user
  - Import from external database
- Fax
  - Unlimited number of software faxes
  - Fax to e-mail (PDF)

---

<sup>4</sup> Phone dependent

## Feature Overview

- Fax to printer<sup>5</sup>
- Print to fax
- Direct Inward System Access (DISA)
  - Obtain an internal dial tone from outside the company
  - PIN support
- Teleconferencing System
  - Unlimited conference boxes
  - User access control
  - Administrator access control
  - Created by web interface or by phone
- (Personal) Queues
  - Agent login / logout / queue pause
  - (variable) wrap-up time after each call
  - Multiple queues per agent
  - Real time queue reporting
  - Real time queue status page
  - Queue call priorities
  - Queue information access through ODBC
  - Local and remote agents support
  - Waiting time messages
  - Number in line messages
  - Play recorded messages while waiting
- ADSI Menu System
  - Advanced telephony functions support
  - PBX controlled visual menu system on analog phones
  - Visual voice mail notification
- Call Detail Records (POSTGRESQL, accessible via ODBC)
- Call Details History
  - Start – end time
  - Phone number information
  - Duration
  - Costs

---

<sup>5</sup> Additional network setup on your part is required

- Billing Information
  - Support rate plans for different providers, call types and time schedules<sup>6</sup>.
- Teleworkers / Road Warriors Support
- Digital Call and Conference Recording
  - Playback through the web interface
  - Downloadable
- Multi-Tenanting (run multiple virtual companies on one PBX)
- Protocol Bridging
  - Seamless integration of various technologies
  - Each technology offers a same features set
  - Interoperability between VoIP systems
- Click-to-Dial for MS Outlook<sup>7</sup> and HTML based interfaces
  - Call Me Now button (click to be called back)
- SIP Call Support
- Direct Media to save bandwidth

## 1.2.2. Call Functions

- Music on Hold
  - Flexible MP3-based system
    - File
    - Stream
  - Volume control
  - Random play
  - Linear play
  - Different set per queue
- Calls on Hold
- Wake Up Call
- Caller ID Features

---

<sup>6</sup> Special rate phone numbers vary in phone cost and will not show the exact phone costs. This will be fixed in the next release.

<sup>7</sup> Optional driver needed

## Feature Overview

- Caller ID blocking<sup>8</sup>
- Caller ID when on hold
- Name lookup
- PBX based ringtone support<sup>9</sup>
- Call Forward settings:
  - Call Forward when Busy
  - Call Forward when No Answer
    - variable timeout setting
  - Call Forward Unconditional
  - Call Forward when logged out
  - Call Forward when phone unreachable
  - Call Forward when DND (do not disturb)
  - Different call forward settings for internal calls and other self defined contact types
  - Can be activated by phone, from GUI or by desktop software.
  - Manager-Secretary forwarding
  - Call forward override (caller permissions needed)
  - Call forwarding based on incoming number
- Call Transfer
  - Callee transfer
- Call Parking and Retrieval
  - variable and predefined parking spaces
  - predefined parking spaces can be monitored by BLF on the phone.
- Call Back When Busy
- Call Return
- Remote Call Pickup
  - Own department
  - Specific phone
  - Specific pickup group
- Do Not Disturb

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<sup>8</sup> Not implemented yet

<sup>9</sup> Currently SNOM phones only

- Can be disabled
- Dial By Name
- Three-Way Calls
- Callee Transfer (call transfer using #)
  - Supported also on cell phones
- Pairing
  - Connect two phones to one extension.
- Intercom and Paging<sup>10</sup>
  - With activation signal on the phones\
- Channel Spying (listen with your phone to another conversation)
  - Spying: you can listen to the conversation, colleague and caller cannot hear you.
  - Speak: you can listen to the conversation and speak with colleague. Caller cannot hear you.
  - Barge-in: you can listen to the conversation and speak with both.
  - Key to switch to next caller.
  - Can only be used when both parties have the correct permissions.

### 1.2.3. Scalability

- Voice over IP
  - Integration of systems on different locations
  - Use of existing data connections
  - One dial plan across multiple offices
  - Quality of Service (ToS/DiffServ)
- Built-in Provisioning System using DHCP
  - Supporting all common phones, ATAs, soft clients and other SIP based hardware
  - Auto firmware update
  - Support for over 100 devices, including all major phone brands.
- VPN Functionality

---

<sup>10</sup> Not supported by all phones

## Feature Overview

- Multi-location support for phones (teleworkers)
- Software switches to switch traffic between networks
- Built-in Firewall
  - DMZ
- Mass-Import of Settings by Uploading CSV Files
- Call Routing and Discovery
  - DUNDi
  - ENUM

### 1.2.4. Message plan

- ESPA is the European standard for exchanging information between various alarm systems and Paging Systems (PS or "Beepers"), personal security systems and fire panels. Voclarion reads the ESPA message flow and takes action on predefined warnings. For example, a fire alarm can make Voclarion call certain people and connect them to a conference.

### 1.2.5. Management

- Installation
  - Stand alone on location
  - Hosted, ASP application
- Graphic User Interface
  - Role based
  - Multi language
  - Organized:
    - Locations
    - Companies
    - Departments
    - Employees
    - Phones
- Services Management
- Logging of all Changes in the Web Interface
- Call Reports

- Over 15 real time reports
- Real time queue reporting
- Real time queue status
- Real time agent reports
- Missed calls report
- Call details (see exactly how a call is processed by the PBX)
- Authorize Outgoing Calls
  - Based on type of phone number and phone costs
    - Allow / Disallow / PIN protected
  - Prepay support
- Upgrade Phones Through the Web Interface
- Personal Phone Directory with Fast Dial Numbers
- Call History with Search Option
- Call Recording
  - enable / disable for all
  - internal call recording (enable/disable)
  - manual (part) recording
  - select recorded party (you, other, both, none)
  - receive recording by e-mail
  - download as WAV file format
- Call Playback when Recorded
- Change Key User Settings
- Change Call Forwarding
- Enable/Disable Voice Mail
- Phone Help
- ODBC Support
- SOAP Support
- Fail Over
  - Define fall back extensions and trunks
  - RAID disk configuration with hot spare disks, dual power supply<sup>11</sup>
  - Master / Slave configuration
  - Fail over support

---

<sup>11</sup> Depending on configuration.

## Feature Overview

- Documentation
  - Installation Manual
  - Operation Manual
    - Programmer's Guide
    - Basic end user phone manuals
      - Basic manual and quick card
- Support contains<sup>12</sup>
  - Nightly backups of settings and voice messages
  - Upgrades to new versions of Voclarion
  - On site support
  - Support by phone

### 1.2.6. Additional Software

- SOAP connection for Windows7
  - Perform telephone actions from scripts.
  - Voclarion Switchboard Desktop
  - Application for call monitoring and distribution
  - Presence information
  - Live queue status
  - Live trunk status
  - Drag and drop call distribution
  - Connection with intranet for lookup customer details on incoming call
  - Connection with intranet for lookup customer details on accepted call
  - Multi company support
  - Drag and drop calls between a queue and employee list
  - Put a call behind a busy user. When calls are not answered, they will return to you with a notification.
  - Call history for all calls
- Calvi support
- Plugin Support
  - Third-party software support

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<sup>12</sup> As part of your support agreement, may vary. Please consult your official Voclarion dealer.

## 1.2.7. Voice over IP

Voclarion offers transparent connectivity between Voice over IP protocols and traditional telephony equipment.

- Native (open) Interconnection Protocol (IAX)
- Session Initiation Protocol (SIP)

## 1.2.8. Telephony Types

- Robbed Bit Signaling Types
  - FXS and FXO
  - Loop Start
  - Ground Start
  - Startle
  - E&M
  - E&M Wink
  - Feature Group D
- ISDN PRI Protocols
  - 4ESS
  - Lucent 5E
  - DMS100
  - National ISDN2
  - Euro ISDN
  - BRI
- GSM<sup>13</sup>
- Private GSM<sup>13</sup>
- DECT
- WiFi
- Fax<sup>14</sup>

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<sup>13</sup> Optional hardware needed.

<sup>14</sup> Unlimited number of software faxes

## 1.2.9. Codec / protocol support

- GSM
- G.729A<sup>15</sup>
- G.726
- G.722
- G.723.1 (pass through)
- G.711-alaw
- G.711-ulaw
- Linear
- Mu-Law
- A-Law
- ADPCM
- iLBC
- LPC-10
- Speex
- ESPA 4.4.4
- MP3 (decode only)

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<sup>15</sup> Optional licenses needed

## 1.3. How to read this manual

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This manual guides you – the system administrator - through the process of creating companies, departments, users and phones, combining them to an advanced dial plan. You will find that there is often more than one way to reach a certain goal. Multiple methods have been described only in cases where each clearly offers different advantages under different circumstances.

If you are using Voclarion for the very first time, it is highly recommended you read the manual from start to finish. We will guide you through all settings of the PBX. If you don't want to use a specific feature, just skip the paragraph.

If you are a more experienced user, you probably will use this manual only for reference. At the start of the manual you'll find an extensive table of contents and at the end of the manual you'll find an index to help you find what you are looking for. Please also note the chapter about PBX features (page 351) and the Quick Card for easy reference (page 383).

### 1.3.1. Topics of This Manual

This manual describes the process of setting up your telephony system. Or in other words: what has to happen when someone is calling your company and what has to happen if you place a phone call? We assume you already have a PBX installed and connected to several operational phone or Internet lines. If not, please start with the Installation Manual.

If you used a previous version of Voclarion, you probably want to start reading the next chapter about new features we added and important changes we made to Voclarion 2.3.

To use the telephony system, the PBX needs to have some basic information about your company. Which departments are there? How many employees does your company have? What are the numbers of the inbound phone lines? All this information is required to setup a dial plan. Chapter 2 attends to your company's internal structure: departments, extensions, users and phones are created.

## **How to read this manual**

The next chapter deals with all aspects of inbound telephony traffic. It describes how to set up IVR menus, define queues and setup scheduling issues. The Direct Inward System Access (DISA) feature is described here as well.

Outbound traffic concerns such as least cost routing, are discussed in chapter 3.16

Your PBX is equipped with several management features, like reporting. This functionality is described in chapter 6.

The next chapters will guide you through some less important Advanced Settings that might be helpful to you. In most cases you don't have to change these settings.

Finally, an overview of most used phone features is included in this manual. The Quick Card might be helpful as reference. We included an index at the end of this manual.

## **1.3.2. Other Manuals**

A wide variety of (phone) manuals is available. Go to the Download Center (chapter 8.1) to download manuals in PDF for free or visit [www.lulu.com](http://www.lulu.com) to order manuals in print.

## 1.4. Conventions

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### 1.4.1. Conventions

Conventions that are used in this document are listed below.

Icon	Description
	Info, hint or or example. It contains information you can use to your advantage.
	This is a note with additional information.
	This is a caution sign. Read this very carefully.
	This is a warning sign. Read this very carefully.

*Table 1: Icons used in this document*

### 1.4.2. User Input

#### 1.4.2.1. Mouse

Mouse actions assume a right-handed mouse configuration. The terms “click” and “double-click” refer to using the left mouse button. The term “right-click” refers to using the right mouse button. The term “middle-click” refers to using the middle mouse button, pressing down on the scroll wheel, or pressing both the left and right buttons simultaneously, based on the design of your mouse.

## Conventions

### 1.4.2.2. Keyboard

Keyboard shortcut combinations will be displayed as follows: **Ctrl-N**. Where the conventions for “Control”, “Shift,” and “Alternate” keys will be Ctrl, Shift and Alt, respectively. The first key is to be held down while pressing the second key.

### 1.4.2.3. Special signs

Text between braces “<>” has to be replaced with actual information. For example, if you see <Your IP Address> you have to replace this by your IP address, like 192.168.1.100.

### 1.4.2.4. Time and dates

When we refer to a time or duration we use the following notation to specify the time format:

- y: year(s)
- m: month(s)
- d: day(s)
- h: hour(s)
- m: minute(s)
- s: second(s)

The number of characters indicates the number of digits, so *m:ss* indicates a time notation like 0:43 or 0:04.

### 1.4.2.5. Caller ID

The Caller ID (or more properly calling number identification) is a telephone service, available on POTS lines, that transmits a caller's number to the called party's telephone equipment during the ringing signal. Where available, the Caller ID can also provide a name associated with it (Caller Name). The information available to the called party may be made visible on a telephone's display or on a separate attached device.



A Caller ID consist of a number of digits without any other characters. The format of a Caller ID is determined by your provider and can differ. It's very important to use the correct format, otherwise the Caller ID will not or incorrect be send with your calls and SIP registration and trunk forwarding will fail. Contact your provider for more information. You can use trunks to reformat Caller ID's.

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### Caller ID priority

On different places within the Voclarion GUI you can set a Caller ID. You can add a Caller ID to a company, but also to a user. When a user makes a phone call, which Caller ID will be sent with the call?

This is a case of priorities. Each Caller ID has its own priority. The trunk Caller ID has the highest priority, the company Caller ID the lowest. To determine which Caller ID must be sent, the system uses the first Caller ID it finds (top to bottom).

1. Trunk: Caller ID
  2. Extension: Caller ID
  3. User: Caller ID
  4. Department: Caller ID
  5. Company: Caller ID
- If you don't want the callee to see your phone number when making a phone call, enter a "0" for *one of the Caller IDs*. The Caller ID is now hidden, according to RFCi-3325.
  - Leave the field blank to not change the Caller ID.

Take a look at the following examples:

## Conventions

### Examples

Trunk Caller ID	<i>none</i>
Extension Caller ID	2125554510
User Caller ID	2125554599
Department Caller ID	2125554510
Company Caller ID	2125554500
<b>Outgoing Caller ID</b>	<b>2125554510</b>

Trunk Caller ID	<i>none</i>
Extension Caller ID	2125554510
User Caller ID	2125554599
Department Caller ID	<i>0</i>
Company Caller ID	2125554500
<b>Outgoing Caller ID</b>	<b>no Caller ID</b>

Trunk Caller ID	<i>none</i>
Extension Caller ID	<i>none</i>
User Caller ID	<i>none</i>
Department Caller ID	2125554510
Company Caller ID	2125554500
<b>Outgoing Caller ID</b>	<b>2125554510</b>

### 1.4.2.6. Caller Name

A Caller Name is a name which is send with the outgoing call, like a Caller ID. The Caller Name is not supported by all telecom providers. The Caller name priority behaves in the same way as the Caller ID (see chapter 1.4.2.5).

### 1.4.2.7. Wildcards

Wildcards are used to select a group of items. For example 1234X selects all 5 digit numbers starting with 1234. If needed more wildcards can be used within one number. The manual indicates on which information wildcards can be used. The following wildcards are available:

- . (dot) selects one or more digits
- [xyz] selects one of the digits x,y,z.
- X selects one digit 0-9
- N selects one digit 2-9
- Z selects one digit 1-9

Priority of wildcard numbers (first is the most important):

1. Numbers without wildcards
2. Numbers with the least wildcards
3. Other numbers

**Example Wildcards on Inbound Numbers**

A company has phone numbers in the range 2125554500  
– 2125554599.

- Number 2125554500 has to connect to an IVR menu (extension 100).
- Numbers 2125554501 – 2125554509 have to connect to the Sales queue (extension 200).
- All other numbers have to connect to the operator (extension 400).

Configuration of Inbound Numbers:

<b>Inbound number</b>	<b>1<sup>st</sup> extension</b>
2125554500	100
212555450Z	200
21255545XX	400

### 1.4.3. References

#### 1.4.3.1. Referring to the PBX Manager

This manual refers to the PBX Manager on many locations. The PBX Manager is the graphical user interface of your phone system.

The next frame shows how we refer to a specific page within the PBX Manager. The “>” indicates a mouse click to a next page. An italic word indicates you have to replace this with the option you would like to see, for example: “*user*” means you have to select the user you would like to see.

In this example there are two different ways to go to a “Locations”-settings:

<b>Web interface menu</b>	Company > Locations > <i>Location</i> Quick Setup > Locations > <i>Location</i>
---------------------------	--

- I. Login to the PBX Manager, choose COMPANY from the menu, choose LOCATIONS from the sub menu and select the location you would like to see.
- II. Login to the PBX Manager, choose QUICK SETUP from the menu, choose LOCATIONS from the sub menu and select the location you would like to see.

#### 1.4.3.2. Notes

On some occasions we use numbered footnotes (¹) to clarify the text. Notes are in depth additions to the main text and can in most cases be ignored without further consequences. Footnotes refer to a text on the bottom of the same page.

#### 1.4.3.3. Bibliography

The main text contains references to the bibliography at the end of the manual. All references are numbered and placed between braces, like [1]. You may lookup these sources for more information about the given subject.

#### 1.4.3.4. Illustrations

On some places we refer to illustrations. All illustrations are numbered and we will refer to it with “see Illustration 4”. Sometimes you'll see something like “see Illustration 5/3”. The second number (3) refers to a number within the image.

## 1.5. Important Safety Information

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At all times follow the general precautions below.



### 1.5.1. General Instructions

- Please inspect the box the PBX was shipped in carefully and note if it was damaged in any way before installing it. If the PBX itself shows damage you should contact the supplier before installing, and file a written damage claim.
- This equipment has been tested and found to comply with CE requirements.
- Read all information contained in this manual. Follow all product warnings, cautions, and instructions.
- Keep the unit free of dust, moisture, condensation, high temperature exposures of more than 40° C (104° F) and vibration. Do not expose the unit to direct sunlight.
- Handle the unit carefully. Do not drop or otherwise expose the unit to physical shock.

### 1.5.2. Installation and Operations

- Install the unit so that the power cord is not obstructed in any way. Do not connect this unit to an extension cord.
- This unit is designed to operate at one specific voltage and current setting. The proper voltage and current required are listed on the product label.
- Mount the unit on a stable wall surface. Do not mount inside of a separate enclosure unless it is properly ventilated.
- This unit is equipped with a 3-wire grounding plug. The plug will only fit into a grounded power outlet. Do not modify this plug in any way. If it cannot be inserted into the outlet, have the outlet replaced by a licensed electrician.

## **Important Safety Information**

- Do not overload wall outlets. Overloaded outlets could result in fire and/or electrical shock.
- We recommend the use of Cat 5 cables for all external signal connections.
- Do not use solvents, liquid cleaners, water or abrasive powders to clean this unit. Only use a damp soft cloth for cleaning.
- Do not block the vent slots and opening.
- Do not insert wires, pins, or any other material into the unit's vent slots or access points. This could result in electrical shock and serious unit malfunctions.
- Cut off the unit from its power source before cleaning.
- Do not install the unit near water or moisture, heating appliances, or electrical noise generating devices such as televisions, monitors, fluorescent lamps or electric motors.

### **1.5.3. Modifications**

- Changes or modifications to this equipment not approved by Voclarion may cause this equipment to not comply with regulations, and void the user's authority to operate this equipment.
- Products contain no user-serviceable parts inside. Refer servicing to qualified service personnel.
- Do not disassemble this product. Dangerous electrical shock could result. The unit must only be disassembled and repaired by qualified technicians.
- When you have to add or remove hardware, always shut down the system in a proper way and disconnect power before adding or removing hardware.
- Never install telephone wiring during a lightning storm.
- Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.

## Important Safety Information

- Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephony equipment.

### 1.5.4. Malfunctions

- If the unit malfunctions, disconnect the unit from all (telephone) lines and check the line. If the line operates properly, have the unit repaired by a qualified technician.
- Unplug and transport the unit to a service technician if the power supply cord is frayed or damaged, if the cabinet is cracked or broken, or when the unit has been exposed to moisture, has been dropped, or is otherwise not operating properly.

## 1.6. Voclarion PBX Manager

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An important part of the Voclarion is the Graphical User Interface (GUI). The GUI is a user friendly interface that allows you to manage your PBX. Voclarion's GUI is called the PBX Manager.

### 1.6.1. Roles

The Voclarion PBX Manager is a role based GUI. This means the PBX Manager supports different *types* of users, like 'employee' or 'manager'. Each of them has his own rights and limitations. For example: a *regular* employee can change his own settings, but not the settings of his colleagues. A department manager can change the settings of all members of his department and can see reports. Because of role-based login it is safe for all employees to use the PBX Manager. Information available will change according to the given role.

### 1.6.2. Accessing the PBX Manager

The PBX Manager can be accessed from a web browser from any location<sup>16</sup>. Simply surf to the IP address of the PBX by entering the address in your browser. The login screen will now appear in your browser's default language. You can change the language by clicking on the flag of the desired language (illustration 1/2). It is wise to bookmark the URL of the PBX Manager for quick reference.

The user name and password are set when the account is created. A lost password cannot be restored. It can only be set again by the department manager, company manager or system administrator or by the user when he is already logged in.

#### 1.6.2.1. Browser Support and Security

To access the PBX Manager you need to have a browser installed on your computer. Currently we support all official browser releases from Chrome<sup>17</sup>,

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<sup>16</sup> A firewall may restrict access to the PBX Manager.

<sup>17</sup> Google Chrome, download the latest version for free at <https://www.google.com/chrome/>

Firefox<sup>18</sup>, Apple Safari<sup>19</sup> and Internet Explorer<sup>20</sup>. Other browsers will probably work, but are not supported yet.



Some browsers can save passwords. We do not encourage this without setting at least a master password first. See your browser manual for more details on security.

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### 1.6.2.2. Errors and Warnings

Make sure Javascript and cookies are enabled in your browser. If not, the PBX Manager will display a warning on your screen and errors will occur. See your browser's help pages for more information about Javascript and cookies. Some graphics may need Flash to work properly.

In some cases the PBX Manager will display a message "Too busy". The first priority of Voclarion is handling phone calls. If the system is very busy or low on memory, it will disable access to the PBX Manager to make sure phone calls are not interrupted. Try again in a few minutes by pressing the TRY AGAIN button or press the BACK button of your browser. Do not use the reload button of your browser! If you cannot access the PBX Manager for an hour or longer, please contact support.

### 1.6.2.3. Locked User Accounts

As a precaution each user account will be blocked if the username-password combination is entered incorrectly for three consecutive times. When trying to login again, the web interface will display: *"This account is locked. Contact your administrator"*.

On the user's page in the Voclarion web interface you will see that field Account Locked is set to "Yes". The account can be unlocked by the Company Manager or System Administrator. If the System Administrator's account has been locked, contact the system manager.

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18 Firefox 3.0 or higher, download the latest version for free at <http://www.mozilla.com>

19 Safari 4.0 or higher, download the latest version for free at <http://www.apple.com/nl/safari/>

20 Internet Explorer 8.0 or higher, download the latest version for free at <http://www.microsoft.com>

### 1.6.3. Overview

The web interface contains some basic elements that makes it easy to navigate, see illustration 1.

1. LOG OUT: Here you can see as which user you are logged in. This is also the place to log out. When you are done, do not forget to logout. Because



Illustration 1: Web interface

of security reasons, your session expires after a certain amount of idle time and the PBX will logout automatically. When you close all browser windows you will automatically log out<sup>21</sup>.

2. COLLAPSIBLE STATUS BAR: The status bar contains some information that might be useful to you. You can collapse this menu by clicking the arrow, to save desktop space.
- LANGUAGE SELECTOR: The PBX Manager comes with different language modules. Choose a different language by clicking on the flag. If you do not make a choice, the software will select your browser's default language

21 Your session will expire.

setting. If the desired language is not available, the PBX Manager will choose English.

- MESSAGE INDICATOR: If it blinks, you have new messages about Voclarion. Click on the icon to see new and old messages. Hover the icon to see the date of the last check.
  - PRINT PAGE: Click the printer icon to print the current page. A printer dialog will open.
  - ABOUT: See version numbers and terms of use of this PBX. Here you can also see the number of activated users and the release notes.
3. MAIN MENU: The main menu is used to navigate throughout the website. You can expand as many sub menus you like, which makes it easy to navigate and to jump between different companies. The options shown in the main menu depend on the type of user you are (see more information about users on page 47).
- Click on a menu item to go to the corresponding page.
  - Click [+] to expand the menu. It will reveal a submenu.
  - Click [-] to contract the menu.
- The company menu is limited up to 10 companies. If the Voclarion contains more companies, a menu item MORE will be shown, showing all companies. In this case, the menu will always show the current company and the first 9 other companies.
- The first menu item is called 'last visited' and will show the last 5 unique pages you visited (if available). Hover the title to see more information.
4. MODIFICATION BUTTONS: Buttons to add (+), remove (X) or edit (✎) an item are located on top of the page, right below the title.
5. HELP: The PBX has an advanced help system. Virtually each single field has extensive information about how to use it. Just click on the help (?) symbol.

6. The current selected company.
7. On this location warnings are shown, for example when your Internet connection is down. Hover with your mouse to see more information about the warning.

### **1.6.4. Conventions Used in the Online PBX Manager**

Conventions that are used in the PBX Manager, the user interface of the PBX are listed below. If you are not sure about what an icon represents, hover over it with your mouse to see more information.

<b>Icon</b>	<b>Description</b>
	Info, tip or example. If you need help while using the PBX Manager, you can click on this icon for more information. The PBX Manager has an advanced help system which provides information for virtually every form field.
	This is an OK sign. The action you performed was successful.
	This is an error sign. The action you performed was not successful. Read the message careful before continuing and try to correct the error.
	This is a caution sign. Read the message very carefully before continuing.
	This is a warning sign. Something went wrong. Read the message very carefully and try to correct the error before continuing.
	This is a status icon. The color of the icon indicates the status of all kinds of hardware. Hover your mouse over this icon to see status information. <ul style="list-style-type: none"><li>● green: OK, hardware functions normally</li><li>● blue: status unknown</li><li>● red: error</li><li>● yellow: hardware reachable, but not at optimal efficiency.</li><li>● grey: hardware is unregistered or disabled.</li></ul>

Icon	Description
	This is an add sign. Click on it to add a new item. This sign can normally be found on the right top of the screen.
	Remove an item. This sign can normally be found on the right top of the screen.
	Edit an item. This sign can normally be found on the right top of the screen.
	Save a setting, an item or submit a form.
	Assign an item. This is used, for example, to assign a phone from a list to a user.
	Cancel the current action.
	Sort this column. Click once for ascending and once again for descending. This icon can be found next to the column title of some columns.
	Search for an item or look up information from a list.
	Select a date.

Icon	Description
	Start a ping. Ping is a computer network administration utility used to test the reachability of a host on an Internet Protocol (IP) network and to measure the round-trip time for messages sent from the originating host to a destination computer.

*Table 2: Icons used in web interface*

#### 1.6.4.1. Tabs

Several dial plan items contain too much settings to place on one page. We grouped these settings orderly into tabs. Tabs appear at the top of the screen. To select a tab, simply click on it. Commonly used tabs are:

- List: This tab contains a list of items, for example all incoming numbers of the company. It makes it much easier to switch between items.
- General: basic information about the current selected item.
- History: a record of changes made to the selected item.

#### 1.6.4.2. Fields

Each tab contains a number of labels and fields. The label is the name of the field. The field contains the information (settings) provided by you. Some information is required and you cannot continue without providing it. These fields are marked with an asterisk (\*).

## 1.7. Choosing a Password

---

On several occasions you have to create or change a password. For security reasons it is important that you choose a password with great care. Therefore, do not use passwords that are easy to guess. Write your password down and store it in a safe place. If you lose a password, it cannot be retrieved, only be set again.

When choosing a password, please keep in mind the following rules for selecting a safe password:

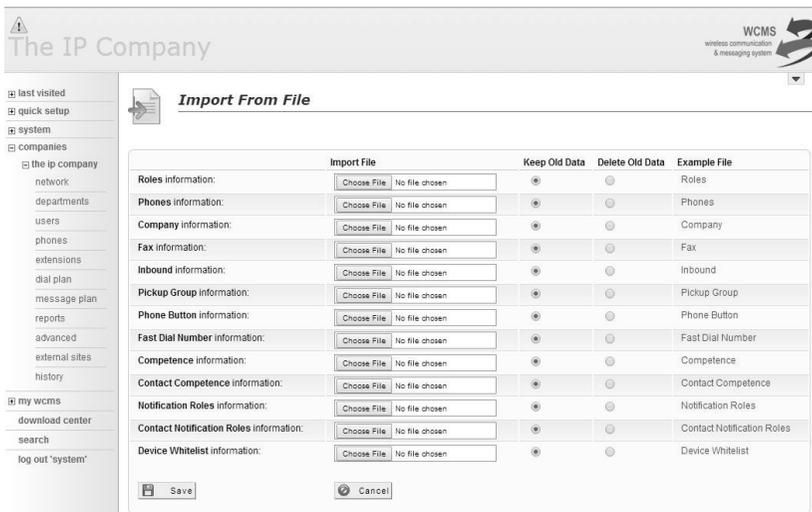
- Do not use passwords that are easy to guess. Do not use names like the name of your wife, your child or your goldfish.
- Do not use common words like “table” or “football”.
- Do not use keys in the same row on your keyboard like “qwerty” or even “1qa2ws3ed”.
- Passwords are case sensitive. You can use upper and lower case characters, like “PilakaVa”
- You can use digits and special characters like -\_=\$!/?&+.,%@.
- Easy to remember are sounds, like “Do.LotaMa” or passwords derived from a sentence; “This PBX is a great tool!” would create a password like “Tpiagt!”

## 1.8. Importing from spreadsheet

All information can be added manually, one by one. Some information – users for example – can be imported by spreadsheet. You can add all users at once.

**Web interface menu**      Companies > Company > Advanced settings > File import

To import a spreadsheet go to the tab IMPORT. Here you can find a link to an example spreadsheet. You can download the file by right-click on the link and choosing “Save link as...”<sup>22</sup>. Save the file on a location that is easy to find, your desktop for example.



Afbeelding 2: import

The downloaded file is in CSV-format (comma separated value). If you open the file in a text editor you will see it is *human readable*. This makes it very easy to edit

<sup>22</sup> The exact description may vary in some browsers.

and therefore CSV is supported by almost all spreadsheet software. In a CSV-file all fields are saved as text, separated by a comma. The first line is the header, the next lines are content.

The spreadsheet contains some comments (start with #) which will be ignored by the PBX. The comments explain how the user should fill the spreadsheet. Below the comments you will find the headers of the columns, as described below. Each row is a set of information.

When finished editing the file, save the spreadsheet as CSV or text file, not as the software's own format. Choose SAVE AS... or EXPORT, depending on the software you use. For more information check the manual of your spreadsheet software.

Click BROWSE and select the spreadsheet.

1. To keep the data already in the system, check the radio button KEEP OLD DATA. To overwrite all data already on the system, check the radio button DELETE OLD DATA.
2. The spreadsheet will now be imported and processed. When an error occurs, a warning message will be displayed. In that case no data will be imported and you can try again.



## **Chapter 2. Company Setup**

*This chapter describes how to create a new company, department, users and phones.*

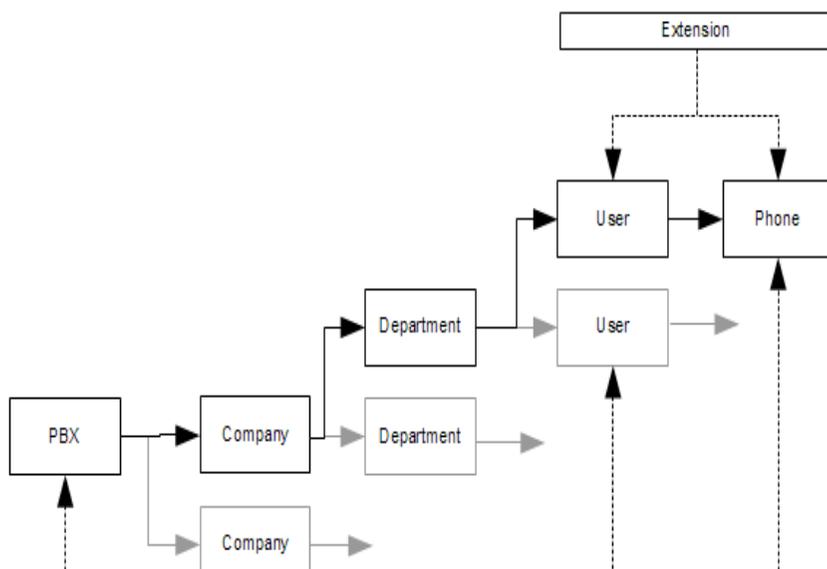
## 2.1. A new Company

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### 2.1.1. Introduction

After the registration of the Voclarion, the system administrator can add one or more companies. Each company can contain one or more departments. After adding companies and departments you can add users and phones.

Because phones can be on different (parts of) networks, you also have to assign a location to both the Voclarion PBX and the phones. A location defines a physical location and a location within a network.



*Illustration 3: Company overview*

Users have to log in to the Voclarion and have different roles. A role defines the user permissions. A user can be a operator, a sales employee, a system administrator etc. The system administrator can assign one or more other system administrators and for each company he can assign one or more company managers. Although not required, it is recommended to assign administrator rights to at least one user.

The company manager manages all settings of his own company, while the administrator is able to manage *all* settings in the PBX, including system settings.

## 2.1.2. Multi Tenanting

The Voclarion supports multi tenanting, which means that more than one company can be added to the PBX. The company settings can be entered manually or by mass import from a spreadsheet. Please note that not all information can be imported, which means that some information must be added manually. If more than one company is registered to the same PBX, you import the desired spreadsheets separately for each company.

Before you can import any company information, you'll have to create at least one company. Departments, users and extensions can also be imported from spreadsheet. In this chapter we discuss the manual setup. See the installation manual for more information about mass import from a spreadsheet.

If you want to add two or more companies, you can create separate companies on the PBX (recommended). You can also combine the companies to one company with different departments dividing the companies.

When to create only **one** company:

- If you have employees working for both companies and you want to supply only one phone to each employee.

It is not possible to receive calls from two separate registered companies on one single phone. Receiving calls for two different companies is only possible if the two companies are registered as one on the PBX.

When to create **two** separate companies:

- If you have two or more phone lines like ISDN connected to the PBX and want to set a different caller ID (identification number) for both companies. Do not forget to add separate trunks to set the right caller ID.
- Employees working for two companies at the same time need two phones.

## **A new Company**

It is possible to call extensions of another company on the PBX. Therefore it is useful to create system unique extensions. To dial extensions from another company, you must make one of the two companies Super Company (see page 63).

### **2.1.3. Receptionist**

The receptionist is a special role. For both companies and departments you can specify a receptionist. On many occasions you can connect the call to this role. For example, if a caller reaches a voice mail box, the caller can press "0" to be connected to the receptionist. The receptionist is also part of the Dial by Name feature and the Call Forwarding settings. Calls which for any reason cannot be connected to another extension end up with the receptionist. So the receptionist is also a fall back. The Voclarion will choose based on priorities, which receptionist the call is send to.

#### **2.1.3.1. Receptionist Priority**

In the process of deciding which receptionist to select, the Voclarion will proceed as follow:

1. The receptionist of the called employee's department has higher priority and will be tried first.
2. If no department receptionist is assigned or the call is bounced<sup>23</sup>, or the user is member of 2 or more departments, the employee's company receptionist will be dialed.

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<sup>23</sup> The person is busy, not logged on or does not answer the phone and has no call forwarding or voice mail set.

## 2.1.4. Adding a New Company

### Web interface menu

Companies

This page shows a list of all companies on this Voclarion. Besides the selected rate plan, billing and prepay status you also see the number of activated users per company. You need one activated user for every simultaneous logged in person. More information about activating users can be found on chapter 2.8.2.3 (page 115).

Click on a company name to see the detailed settings, or click [ADD \(+\)](#) create a new one.

1. Click [ADD \(+\)](#) and fill out all required fields marked with a “\*\*”.

Field	Description
<u>COMPANY NAME:</u>	The full name of the company, like “My Company LTD”.
<u>LICENSES:</u>	The number of activated users for this company. Only available when Licenses per company is activated (see Installation Manual).
<u>LICENSES IN USE:</u>	The number of activated users for this company in use.
<u>BILLING:</u>	Enable billing. This activates the rate plan to show costs information in reports and Call Detail Records. Billing is also needed for using Prepay calls. See the Installation Manual for more information about Billing.
<u>RATE PLAN:</u>	Select a rate plan suitable for this trunk from the list. Only available when Billing is enabled. See the Installation Manual for more information about how to activate Billing.

## A new Company

<b>Field</b>	<b>Description</b>
<u>PREPAY:</u>	Set a prepay amount for this company. Use a dot (.) as separator, like “25.50”. Before making a phone call, the user credits are checked. After the call the account is credited. Only available when Billing is enabled. See the Installation Manual for more information.
<u>COMPANY SHORT NAME:</u>	Name displayed on phone displays when not logged in and lists, like “mycomp”. Do not use any spaces, capitals or special characters like &%\$#@”.,
<u>PHONE DISPLAY NAME:</u>	The name that must be shown on the phone display when no one is logged in, instead of the name/extension of the user. If this field is empty, the PBX will display the name entered for <u>COMPANY SHORT NAME</u> .
<u>ADDRESS, ZIP CODE, CITY,</u>	Address information.
<u>COUNTRY:</u>	
<u>TIME ZONE</u>	The default time zone for this company (can be changed for department or phone).
<u>DEFAULT LANGUAGE:</u>	The default language used by the PBX for external voice messages (like IVR menus), internal voice messages (like e-mail menu) and text messages (phone display). Each user can change his own language settings afterward.
<u>DEFAULT MUSIC ON HOLD:</u>	The music that is played when a caller is put on hold. Queues can have an alternative music on hold setting. However, when no music on hold is selected the PBX will use this setting. For more information on music on hold, see chapter 3.7.
<u>SUPER COMPANY:</u>	Is this company a Super Company? Users of a Super Company are able to dial extensions from other companies within this PBX. Other companies are allowed to dial extensions of this Super Company.

Field	Description
<u>EXTENSION NUMBER LENGTH:</u>	The number of digits of each extension (e.g.: enter "4" to use extensions like 1000, 1001, 2022, 3333). Choose between three, four or five digit extensions. <b>Warning:</b> You cannot change this setting without removing the entire dial plan for this company. We recommend to use at least 4 digit extensions. Note that you need extensions for users, phones, conference boxes, queues and IVR menus.
<u>TELEPHONE NUMBER:</u>	A phone number contains an area code and a subscriber number
<u>CALLER ID:</u>	The identification number sent with each outgoing call. If you don't want to send a caller ID of this company fill out a "0". The format of a caller ID is specified by your telecom provider. A typical caller ID has only digits, no spaces and has no country code <sup>24</sup> . More information about caller IDs see chapter 1.4.2.5.
<u>CALLER NAME:</u>	Is sent with each outgoing call. See chapter 3.2.3. When the caller ID is set to "0" nothing will be send.
<u>FAX NUMBER:</u>	A fax number contains an area code and a subscriber number
<u>DOMAIN NAME:</u>	Provide the company's domain name, used for e-mail services. A domain name contains a top level domain and a second level domain. Examples: mycomp.com. Do not enter a third level domain, like "www".
<u>SIP DOMAIN:</u>	Domain name used to accept SIP calls. Do not start with "www" or a protocol indicator like "http://". Examples: mycomp.com, sip.mycomp.com.
<u>USE RDNIS AS DDI:</u>	If present, use the RDNIS value as DDI number. <b>Note:</b> If you have set this field to "yes", and you have forwarded all your calls from your mobile phone to the office number, the RDNIS will be your mobile phone number (see chapter 3.2.2.4).

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<sup>24</sup> Your telecom provider will add a country code if necessary.

## A new Company

<b>Field</b>	<b>Description</b>
<u>USE DEPARTMENTS AS PICKUP GROUPS:</u>	To use this department as pickup group, set this field to "Yes".
<u>E-MAIL ADDRESS:</u>	Company's e-mail address
<u>ENTIRE FIRST NAME ON PHONE:</u>	This shows the entire first name on the phone display. If set to "No", only the initial is shown, in order to make room for the last name.
<u>ALLOW RECORDING CALLS BETWEEN EMPLOYEES:</u>	By default calls between employees are not recorded. Note: If an employee's mobile number is set, the call from and to this number is marked as internal call.
<u>DIRECTORY SEARCH METHOD:</u>	On what would you like to search when using Dial by name (*35)? <ul style="list-style-type: none"><li>• Last name</li><li>• First name</li><li>• Both</li></ul>
<u>PREFIX FOR OUTGOING CALLS:</u>	This (set of) digits is added in front of the Outgoing Number and can be used for trunking. A trunk can remove the digits before placing the call. This feature serves to emulate old PBX systems where you need to dial '0' or '9' to get an outside line. Note that the emulation is not perfect. Convincing end-users that 'VoIP PBX systems just work different', is the recommended action to take.
<u>RECEPTIONIST:</u>	When you have added users to the company, you can select a receptionist from the user list.
<u>ADMINISTRATOR:</u>	Assign an administrator for this PBX (administrator only).
<u>LOCATION FOR PBX:</u>	If the company is not situated in the same location as the PBX, select the right location for the company.

2. Click SAVE to save this information. The company will now be created.



Although the Voclarion does allow users in different companies having the same extension, this is not recommended when you use Super Companies.

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## 2.2. Departments

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When a company is created, departments can be added<sup>25</sup>. If you have imported information by spreadsheet, departments have already been created. You can use this chapter to fine tune department settings.

Without departments you cannot add users. Each department contains one or more users and a user can be a member of one of more departments. For each department you can assign a manager, who is allowed to manage settings of only his own department. If you like you can assign more than one manager to one department and one person can be manager of more than one department. Note that it is not allowed to move one department to another company.

Default users of the same department can use the pickup feature to pick up a colleague's ringing phone. This is by default enabled and can be disabled if you like. If people of different departments have to pick up each ringing phones, a custom pickup group can be created. (see page 360 for more information about how to use Call Pickups and page 161 on how to create a Pickup Group).

Users can be part of one or more departments and/or pickup groups.

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<sup>25</sup> Due to technical reasons, you can create up to 256 departments on one PBX.

## 2.2.1. Creating a new Department

To create a new department, go to the location below.

<b>Web interface menu</b>	Companies > <i>Company</i> > Departments
---------------------------	--

1. An overview of all departments is shown. Click ADD (+) to create a new department.
2. Enter the name and short name for the department. The system will suggest a short name. The name is used for your own reference, the short name is used for displaying on phones etc.
3. Click SAVE to create the department.
4. After saving, you'll see the overview of all departments, including the new one.
5. If you would like to fine tune the department settings, you can edit the department you just created by clicking on the name and the EDIT BUTTON ()
6. Provide the requested additional information (see below) and click SAVE to exit.

<b>Field</b>	<b>Description</b>
<u>RATE PLAN:</u>	The rate plan applied to the users of this department. Only available when Billing is enabled. For more information about rate plans, see the Installation Manual.
<u>PREPAY:</u>	Set a prepay amount. Use a dot (.) as separator, like "25.50". Before making a phone call, the user credits are checked. After the call the account is credited. Only available when Billing is enabled. See the Installation Manual for more information.

## Departments

<b>Field</b>	<b>Description</b>
<u>CALLER ID:</u>	Add an alternative caller ID. This caller ID will overwrite the company caller ID and will be used for phones within this department. The format of a caller ID is specified by your telecom provider. Commonly this is a range of numbers, similar to a phone number. A typical caller ID contains only digits and no spaces and has no country code <sup>26</sup> . More information about caller IDs see chapter 1.4.2.5.
<u>CALLER NAME:</u>	Is sent with each outgoing call. See chapter 3.2.3. When the caller ID is set to "0" nothing will be send.
<u>PREFIX FOR OUTBOUND CALLS</u>	Place these digits in front of the Outgoing Number. The feature can be used to force all calls from this department to a specific trunk. All outbound calls are prefix with the given digits, the outbound dialplan can detect this prefix, and route all calls to a specific trunk, removing the prefix in the process.
<u>COMPANY:</u>	Choose the company from the list to add this department to.
<u>RECEPTIONIST:</u>	Choose a receptionist for this department (see chapter 2.1.3).
<u>MANAGER</u>	Choose a manager for this department.

### 2.2.2. Assign Users to a Departments

When looking at the details of a department, you'll notice a few tabs at the top of the page. These make it easy for you to manage users and queues for the selected department. At this point we will not describe how to create new users. For more information see chapter 2.8.

If you already have a bunch of users you can assign them to this department, from the users-tab. Click on the USERS tab and click EDIT (). You'll see a list of all

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<sup>26</sup> Your telecom provider will add a country code if necessary.

users for this company. Select the desired users and press the SAVE button to assign the users to this department.

### 2.2.3. Assign Queues to a Departments

When looking at the details of a department, you'll notice a few tabs at the top of the page. These make it easy for you to manage users and queues for the selected department. At this point we will not describe how to create queues. For more information see chapter 3.10. To see the queues for this department, go to the QUEUES tab.

### 2.2.4. Report

When looking at the details of a department, you'll notice a few tabs at the top of the page. The Report tab shows the call information for this department. Select the time window and press GENERATE to see the information. For other reports go to COMPANIES > COMPANY > REPORTS.

## 2.3. Extensions

---

Extension are short unique internal phone numbers, assigned to all items of your dial plan. A user has an extension, an IVR-menu has an extension and even sound prompts have an extension. Extensions are used to connect a caller to a dial plan item.

For example: an incoming call is connected to extension 6001, and IVR-menu. This menu has a few options, like press 1 for support, 2 for ... When the caller presses 1, he will be connected to extension 501: the support queue. When an employee comes available, the caller is connected to the extension of the employee.

Before adding the dial plan items it is recommended to consider a logical plan for numbering extensions. This will help you organize your dial plan. We recommend to define an wide enough extension range for each extension type. Note the difference in amount of available extensions between 3 and 4 digit extensions.

Dial Plan Item	3-digit Extensions	4-digit Extensions
Regular Users	200-299	2000-2999
Physical Phones	300-399	3000-3999
Software Faxes	400-499	4000-4999
Queues	500-599	5000-5999
IVR Menus	600-699	6000-6999
Page/Intercom Groups	700-750	7000-7500

*Table 3: Example extension ranges*

Conference box ranges are not specified in this screen. See chapter for more information about conference box ranges.



- All extensions can be called from a phone. This makes it easy to check your dial plan items.
- The duration of a call is limited to 4 hours. After this time, the line will be disconnected automatically.
- If you dial a non-existing extension, the PBX will return a message: “The number you dialed is not in use.”

### 2.3.1. Extensions and Emergency Numbers

When assigning extensions, be aware that special numbers such as emergency numbers cannot be used as an extension. Which numbers you cannot use is country dependent. In the USA 911 cannot be used and in Europa 112 is blocked. In Belgium special information numbers start with 12XX – 18XX. Therefore it is recommended not to use extensions of the 1XXX range.

### 2.3.2. Extension Ranges

**Web interface menu**      Companies > *Company* > Dial Plan > Extension Ranges

Before creating a dial plan you can define extension ranges. When you create a new extension, the PBX will take an extension from a predefined range.

There are ranges for:

- Extensions: extensions are assigned to employees using a phone.
- Phones: a phone can have his own extension. You cannot place a call with this phone, but the phone is reachable for others.
- IVR menus
- Queues

## Extensions

- Page groups: each page group has an extension, called a page group number.
- Conference boxes: each conference box has an extension, called a box number<sup>27</sup>.
- Trunks
- Parking lots
- Plugins
- Ring groups
- Sound prompts
- Switches

To define extension ranges (recommended) go to the location above and:

1. Click ADD (+).
2. Select a type of range (function) .
3. Enter the range (from - to).
4. Click SAVE.



Once created, you cannot change extension ranges without removing the complete dial plan first! Make sure you make the extension ranges wide enough. We recommend 4 or 5 digit extensions.

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<sup>27</sup> For each type of extension you can create only one range. To create a range for conference boxes, see chapter .

### 2.3.3. Adding Extensions

Now we have defined the extension ranges, we finally can create our extensions. Extensions can be created separately, like we describe here. In some cases extensions can be created automatically while creating a queue, for example. The benefit of creating extensions manually is that you can set some properties.

**Web interface menu**      Companies > *Company* > Extensions

1. Click ADD (+).
2. Enter a number for this extension, according to the extension range.  
Furthermore you can set the following options:

<b>Field</b>	<b>Description</b>
<u>CALL RECORDING</u> :	<p>Would you like to record all conversations automatically from and to this extension?<sup>28</sup>. Not available for all extensions.</p> <ul style="list-style-type: none"> <li>• <u>NOT AT ALL</u>: no recording<sup>29</sup></li> <li>• <u>MY VOICE AND THE OTHER PARTY</u>: both sides will be recorded.</li> <li>• <u>MY VOICE ONLY</u>: only the user's voice, other voice is not recorded</li> <li>• <u>EXTERNAL PARTY ONLY</u>: the other party, the user's own voice is not recorded.</li> </ul>

<sup>28</sup> Files are saved in WAV. For more information see chapter Error: Reference source not found.

<sup>29</sup> Can still be activated manually.

## Extensions

Field	Description
<u>MAIL RECORDED CALL:</u>	When enabled, the Voclarion will e-mail each recorded call to: <ul style="list-style-type: none"><li>• The person that setup an external call</li><li>• The person that is the last spoken with an external caller.</li></ul> <p>A valid email address must be available from the user settings.</p>
<u>VOICE MAIL BOX NUMBER:</u>	The number of the mail box. The voice mail box number and password are optional. The default value is the extension number.
<u>VOICE MAIL PASSWORD:</u>	The password for this mail box. The voice mail box number and password are optional. The default value is the extension number.
<u>VOICE MAIL RETENTION TIME:</u>	The time in days voice mail is kept by the PBX.
<u>TIME_ZONE:</u>	The time zone for this extension.
<u>CALL FORWARD OVERRIDE:</u>	When activated the Call Forward Override feature can override this person's call forwarding settings. A user must be actually logged in on his phone. Example: if the user has forwarded all calls to his mobile number, the colleague using Call Forward Override will ring his phone.

### 3. Click SAVE.

The top of the page contains a few tabs with related features which can be used in combination with this extension. We discuss these features in the upcoming chapters.

- Switches, see chapter 3.6.
- Redirect Schedules can be used to redirect calls to another extension on specific times and dates. Per extension you can assign one or more redirect schedules, see chapter 3.5.
- CWFD; Call Forwarding.



If an extension is not assigned to a user, the voice mail cannot be used.  
Assign the extension to a user to activate the voice mail.

### 2.3.3.1. Extension assigned to a user

Once the extension is assigned to a user specific settings become available. You can change them by editing the extension after assignment.

Field	Description
<u>CALLER ID:</u>	You can assign a specific caller ID for this user. A caller ID is send with call to identify the caller and contains only digits and has no country code <sup>30</sup> . More information about caller IDs see chapter 1.4.2.5.
<u>CALLER NAME:</u>	You can assign a specific Caller Name for this user. A Caller Name is send with call to identify the caller and contains characters, digits and spaces. More information about Caller Names see chapter 3.2.3.
<u>LINE:</u>	The line (button) on the phone to assign this extension to.
<u>RING MOBILE AND THIS EXTENSION TOGETHER:</u>	Also ring the mobile phone of this person until one of the phones is picked up, or the call is forwarded (to voice mail)
<u>MAX CONCURRENT CALLS:</u>	How many concurrent calls can this extension accept? Set this to >1 if you want to use the Call Waiting feature <sup>31</sup> .

### 2.3.4. Automatic Call Forwarding based on availability

Every dial plan item has an Extension assigned, an internal phone number. Extensions are used to connect a caller to a dial plan item. Depending on the item the extension is assigned to, an action will take place. For example if you connect

<sup>30</sup> Your telecom provider will add a country code if necessary.

<sup>31</sup> Call Waiting: tour phone accepts new calls when you are calling. New calls are put on hold. To use this feature, set Call Waiting to "Yes" in your phone settings (see chapter 2.4). To use Call Waiting your phone must support more than one line.

## Extensions

a caller to an extension of a user, the user's phone accepts the call and will ring. Or if connected the extension of a menu, the menu will play.

### 2.3.4.1. Modes

The default action of an extension is accepting a call. However in some cases this is not possible, for example when a phone is busy. In this case you can send (forward) the call to an alternative extension. There are several reasons why a extension is not available: the phone is busy, the user is not logged in or the phone is not answered. For all reasons (or modes) you can set an alternative extension.

In the previous paragraph we talked about an extension connected to a phone. The same is valid for extensions connected to other dial plan items like a queue or a menu. A queue can be full (mode: busy) or all agents can have logged out (mode: logged out). Not all modes are applicable for all items. Table 4 defines how each mode applies to the dial plan items.

<b>Item vs. Mode</b>	<b>Busy</b>	<b>Logged Out</b>	<b>Not Answered</b>	<b>Malfunctioning</b>	<b>DND active</b>
<b>User</b>	All lines full.	Logged out	Time out answering phone passed	PBX cannot reach phone	User activated DND
<b>Queues</b>	Limit number of callers reached	There are no agents logged in	The maximum time in queue has passed	Does not exist	N/A
<b>Ring group</b>	N/A	Group is empty	N/A	Does not exist	N/A
<b>IVR Menu</b>	N/A	N/A	Time out. No menu option has been pressed.	Does not exist	N/A
<b>Conference Box</b>	Locked conference	N/A	N/A	Does not exist	N/A
<b>Page Group</b>	N/A	N/A	N/A	Does not exist	N/A
<b>Intercom Group</b>	N/A	Group is empty	N/A	Does not exist	N/A
<b>Trunk</b>	Max concurrent calls reached	N/A	No dial tone	Does not exist	N/A
<b>parking lot</b>	N/A	N/A	N/A <sup>32</sup>	Does not exist	N/A

*Table 4: Item vs. extension mode. The default mode is not shown. The default action for the default mode is Ringing Phone/accepting the call.*

<sup>32</sup> parkinglot has it's own fallback extension.

## Extensions

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- With Call Forwarding on an extension you can forward a call to another extension, based on availability. Another way of forwarding calls are Redirect Schedules (chapter 3.5). Schedules are call forwardings based on time. When both are assigned to an extension, redirect schedules take precedence.
  - A fast dial number and an extension can have the same number. When called the fast dial number takes precedence.
- 

### 2.3.4.2. Call Forwarding options

For all available modes you can set a call forwarding. For most dial plan items the default setting for the default mode is Ringing Phone / Accepting the call. The default setting for all other modes is *Bounce*, not accepting the call and sending it back to the dial plan.

You can change the call forwarding for all available modes. However keep in mind that if you change the default mode into something else than Ringing Phone, all other modes will be ignored! For example if you change the default setting for a user from Ringing Phone to Mobile, all calls will be send directly to the mobile phone, even if the user is logged off from his desk phone!

Call Forwarding Option	Description
Ringing phone	Phone will ring / call is accepted
Mobile	Connect the call to the users mobile phone. Only available when mobile phone number is provided in the user settings.
Receptionist	Connected the call to the department receptionist. When there is no department receptionist is set, the call is connected to the company receptionist. When the receptionist is unavailable, the Call Forwarding settings of the receptionist will be honored.
Voice mail	If voice mail is set for this extension, calls are connected to voice mail.
Forward to	Forward the call to the given phone number or extension.
Busy tone	Play a busy tone and end call.
Hangup	End call.
Bounce	Refuse the call and bounce it back to the dial plan. The dial plan decides now the next step to take.
Ask Caller	<p>Ask the caller what to do by playing an IVR-menu. The menu contains one or more of the selected options:</p> <ol style="list-style-type: none"> <li>1. Forward to receptionist</li> <li>2. Forward to mobile phone</li> <li>3. Forward to voice mail</li> <li>4. Forward to a colleague (uses the Colleague Search List)</li> </ol>

## Extensions

Call Forwarding Option	Description
Find Colleague	Connect to a colleague by using the Colleague Search List. This list contains up to 6 phone numbers which will be dialed one at a time. The PBX tries to contact a phone number for the given time. If unsuccessful, the PBX will call the next number from the list.

*Tabel 5: Call forwarding options*

### 2.3.4.3. Contact type

A contact type is a group of people with the same characteristics. Examples of contact types are Internal calls, External calls, VIP's, Customers, Management, etc. You can use contact types to group people and assign different call forwarding rules to them. When you're unavailable, you can now connect customers to a colleague and internal calls to your mobile phone. By default there are two contact types available:

- CFWD Internal: All internal calls. This contact type is available by default but not configured<sup>33</sup>.
- CFWD Default: Contains all contacts not included in another contact type. This contact type is available by default and active.

See chapter 10.8 for more information.

### 2.3.4.4. Making the Contact type visible

To make the contact type visible in the call forward page of an extension, press the ADD (+) icon in the tab of the extension page. Select the contact type from the pull down menu. To remove the contact type from the extension, press the CANCEL (X) icon. This will not delete the contact type from the system. Go to Contact Types to remove a contact type permanently.

---

<sup>33</sup> Note: a call from a colleague's mobile phone to the office is also an internal call.



If an extension is not assigned to a user, the voice mail cannot be used.  
Assign the extension to a user to activate the voice mail.

---

### 2.3.4.5. Setting Call Forwarding

To set Call Forwarding choose Extensions from the menu. Now click on the extension you would like to edit. On top of the screen you see one or more Call Forwarding (CFWD) tabs: one for each Contact Type.

1. Click on the CFWD tab to see the current settings.
2. Click EDIT () to change the settings.

## Extensions

Default Action	When Busy	When Not Answered	When Logged Out
Follow <sup>34</sup> *21	Follow *22	Follow *23	Follow *24
Ring Phone	-	(after X seconds) <sup>35</sup>	-
Mobile	Mobile	Mobile	Mobile
Receptionist	Receptionist	Receptionist	Receptionist
Voice mail	Voice mail	Voice mail	Voice mail
Forward <sup>36</sup> :	Forward:	Forward:	Forward:
Busy tone	Busy tone	Busy tone	Busy tone
Hangup	Hangup	Hangup	Hangup
Bounce	Bounce	Bounce	Bounce
Ask caller: <i>Receptionist</i> <i>Mobile</i> <i>Voice mail</i> <i>Find Colleague</i>			
Find Colleague	Find Colleague	Find Colleague	Find Colleague
1	1	1	1
2	2	2	2
3	3	3	3
4	4	4	4
5	5	5	5
6	6	6	6
Try for xx seconds			

Table 6: Call forwarding settings (part)

34 Call forwarding on your extension can also be changed by dialing a \*-code (see Quick Card). You can (de-)activate and change the forwarding. Would you like change this setting in this contact type when using the \*-codes? Select "Yes" if you do.

35 Number of seconds before call forwarding activates.

36 Can be used to connect all calls unconditional to an external phone number. Can be used within a redirect schedule or switch to connect to an external call center for example. In this case you create an extension called "CallCenter" which is unconditional forwarded. The redirect schedule or switch decides on when to forward calls.



---

By default there is no difference in response when calling a busy or a logged off phone. In both cases - when no call forwarding set - you'll hear the same busy tone.

CALL BACK WHEN BUSY (page 384) only works when your colleague is busy. To clarify if a colleague is logged out or not, you can forward internal calls to a sound prompt when logged out (see chapter 3.11 on sound prompts). To play an announcement *and* forward the call, you can use an IVR-menu or a queue.

---

### 2.3.5. Changing Call Forwarding by Phone

Call forwarding settings are also (de)activated by dialing a \*-code. If you often forward your phone calls you can program a function key on your phone to activate your call forwarding. See the Quick Card for more information.

If you use two extensions, it default forwards the calls for both lines. If you want to only forward calls for one specific extension, you cannot use a programmed function key, because using this key will forward calls of all lines. To forward calls from only one extension press the preferred line key to obtain a dial tone and enter the forwarding code and number.

### 2.3.6. Manager Secretary Forwarding

The Manager Secretary Forwarding has several varieties. In general all calls to a manager are forwarded to a secretary<sup>37</sup>. Only the secretary is able to call the manager directly. This way the manager won't be constantly disturbed by undesired phone calls. The secretary decides whether or not to transfer a call to the manager.

#### 2.3.6.1. Setting up the Manager Secretary Function

You can create a Manager Secretary Function by using Contact Types. Anyone who calls the manager, will be forwarded to the secretary, except for the secretary.

---

<sup>37</sup> This person doesn't have to be Company or Department Secretary. Also works on a group of secretaries, so the function is not limited to only one person.

## Extensions

We create this solution by creating Contact Types, see page Error: Reference source not found for more information about Contact Types. There are three important contact types in this case. Two are default available on your PBX: “Default” (external calls) and “Internal” (internal calls). The secretary is an exception on both and we have to create a new contact type called: “Secretary”.

In the next configuration we assume the following:

- Manager has extension 100
- Secretary has extension 200

## Configuration

1. Create a new Contact Type (Company > Advanced > Contact Type). Name it “Secretary”.
2. Go to the secretary's user setting and change the CONTACT TYPE to “Secretary”.
3. Now go to the Manager's extension to edit his Call Forwarding settings:

Call Forward (Contact) Type	Call Forwarding Setting
CFWD Default	Forward all calls to: 200
CFWD Internal	Forward all calls to: 200
CFWD Secretary – default action	Ring phone

This configuration forwards all calls of external and internal to the extension of the secretary. The secretary contact type is allowed to call the manager (ring phone).

## 2.4. Phones

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End points are devices developed for communication with people, for example a phone or a fax. Voclarion supports a wide variety of end points like all major SIP phones from all brands. We also support DECT phones and even Nurse calling systems.

### 2.4.1. Provisioning

The configuration of supported phones is done automatically by the Voclarion Provisioning System<sup>38</sup>. After supplying some basic information, the Voclarion generates all configuration files for your phones. You don't need the GUI provided with your phone. All configuration is done by Voclarion. The PBX also takes care of updating the phone's firmware. This means you can plug a phone into the LAN just out of the box. The phone will be configured automatically and the firmware will be updated to the latest supported version.

In short, the provisioning process goes like this:

1. Add the phones to the Voclarion Manager.
2. Install the phones and connect all wires.
3. At start up, the phone issues a DHCP request and waits for an DHCP-server to respond. If a DHCP server responds, the phone will ask permission to use an IP address or will ask for a new IP address.
4. The DHCP server performs a ping to check if the suggested IP address is available. If so, the server will grant the request. If not, a new IP address will be issued.
5. The DHCP server assigns an IP address to the phone. It also passes some additional information, like the IP address of the Voclarion.

---

<sup>38</sup> Except for soft phones and WiFi phones. See the installation manual or visit [www.voclarion.com](http://www.voclarion.com) for an overview of all supported hardware. If the provisioning of your phone is not supported by Voclarionvice hat to be configured manually.

## Phones

6. The phone configures its IP interface.
7. The phone connects to the given IP address and starts downloading firmware and configuration files. If you have connected several phones on an external location over DSL, this may take a while depending on the firmware file size and the available bandwidth. However the phone will state 'green' in the Voclarion Manager, you cannot make phone calls yet.
8. The phone registers to the Voclarion (with SIP).
9. The phone is fully functional and can be used.



Because of the automated process of configuring and updating phones we strongly discourage making changes to phone's configuration files and firmware.

---

Before a phone can be used, some basic information must be provided to the Voclarion. In short: we must tell the Voclarion which phones are part of which company. Each phone has its own serial number and MAC address. You can find both on the back of the phone. The MAC address identifies the phone. If you can't find any serial number, enter the MAC address in the serial number field (as well as in the MAC address field).

## 2.5. Adding Phones to the Voclarion Manager

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In this chapter we describe the installation of phones within the LAN. For installation of phone on external locations (VPN, IPtivity cloudbox), consult the installation manual.



You should not connect phones to the network until they are added correctly to the PBX. This is especially important for SNOM phones! If you do, phones will in some cases receive an IP address, but no configuration, which makes troubleshooting very hard.

<b>Web interface menu</b> Companies > <i>Company</i> > Phones
---

The page above shows an overview of all phones for this company. The Status icon shows the current status of the phone. Hoover for more information. To see the phone's settings, click on the Serial number.

1. To add a new phone, click on the button ADD(+).
2. Choose the brand and type of phone from the drop down list. If you would like to add an unsupported phone, select GENERIC SIP phone.
3. Click SAVE.
4. Then enter all information requested:

Field	Description
<u>DESCRIPTION:</u>	Enter a description for this phone. E.g. "John's office phone".
<u>SERIAL NUMBER:</u>	The serial number of the phone. This can be found on the back or bottom of the phone.

## Adding Phones to the Voclarion Manager

Field	Description
<u>MAC ADDRESS:</u>	The MAC address of the phone. This can be found on the back or bottom of the phone. It consist of 12 characters (digits and characters) in pairs usually separated by a colon. The first part of the MAC address will be filled in by the PBX.
<u>LOCATION:</u>	Choose the location where the phone will be installed <sup>39</sup> .



If your soft phone requires a MAC address, you can use the computer's MAC address. Make sure you use a MAC address only once.

---

After you added a phone, you will return to the overview screen. The status icon will show green and the phone is now fully functional and a user can login. If you like you can change some settings to tune the phone.

### 2.5.1. Changing Phone Settings

Go to the phone overview and select the phone you would like to change by clicking on the serial number. Depending on the type of phone, each page has one or more tabs at the top which include a variety of settings. The availability of settings on each tab depends on the type of phone.

Tabs	Settings
List	Overview of phones for this company.
How To	General information about the phone and a link to the manual for this phone.
General	General information about the phone. Like logged in user, extension, Pair status, Location settings, IP address and mac address.
SIP	Phone settings like NAT settings, qualify and codecs. Also included th user name and password for SIP lines, needed for the manual registration on Voclarion for DECT and smart phones.

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<sup>39</sup> See the Installation manual for more information about locations.

<b>Tabs</b>	<b>Settings</b>
Call <sup>40</sup>	Settings like Call Waiting , Volume Call Waiting, hash transfer and Message Waiting. Settings may differ from phone to phone.
Function Keys <sup>40</sup>	BLF buttons can be programmed for a specific function. See page 98 for more information about Function Keys. Only available when a user is logged in.
Extension Pad <sup>40</sup>	If you have an extension pad attached to your phone, you can add function keys. See page 98 for more information about Function Keys. Only available when a user is logged in.
Advanced <sup>40</sup>	Advanced settings like Audio Mode.
Video	Various video settings.
History	The history page showing modifications. Only visible to the administrator.

To make changes, go to the desired tab and click EDIT. The settings shown may vary depending on the type of phone. When you are done making changes, the phone needs to reboot to activate the changes. Some phones automatically reboot when saving the settings. Older phones have to reboot manually.

<b>Tab</b>	<b>Description</b>
<b>General Settings</b>	
<u>COMPANY:</u>	Move phone to new company.
<u>USER:</u>	Login user to this phone.
<u>DESCRIPTION:</u>	Description for this phone.
<u>SERIAL NUMBER:</u>	The phone's serial number. Can be found at the back or bottom of the phone.
<u>EXTENSION PADS</u>	The number of extension pads attached to the phone. After saving this information, a new tab Extension Pad will appear. Here you can configure the pad.
<u>INSTALLED:</u>	

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<sup>40</sup> Only available when logged in.

## Adding Phones to the Voclarion Manager

<b>Tab</b>	<b>Description</b>
<b>General Settings</b>	
<u>PHONE EXTENSION:</u>	Extension for this phone. With this extension the phone can only receive calls. To make and receive calls, a users has to login to the phone. If a phone extension is given and a user is logged in, calls can be received on both extensions.
<u>PAIRED WITH PHONE:</u>	When a phone is paired with another phone, both phones will ring on an incoming call. When answered the call, both phones will stop ringing and will be 'busy'. This is called Twinning. Note that paging, and intercom only address the phone that you actually logged in to.
<u>LOCATION:</u>	Select the location for this phone.
<u>THIN CLIENT ID:</u>	The identifier of the thin client this phone is associated with. This value could be used for Single Sign-on. It is a free field, where any type of value can be used. There are SOAP calls available that, when given a user name and a Thin Client ID, the user will log in or out from the phone.
<u>MAC ADDRESS:</u>	The Mac Address is used to identify the phone. The address has to be unique and can be found at the back or bottom of the phone.
<u>FIXED IP ADDRESS:</u>	When using DHCP leave the field bland. Otherwise enter an IP address if you decide to work with fixed IP addresses.
<u>NETMASK OVERRIDE:</u>	Specify a specific netmask for this phone to use instead of the netmask of the network. This feature is seldom used.
<u>CURRENT IP ADDRESS:</u>	Shows the current IP address.

<b>Tab SIP Settings</b>	<b>Description</b>
<u>NAT:</u>	Network Address Translation translates the internal IP address to an external IP address. Enable NAT only if the phone is behind a NAT router.

<b>Tab SIP Settings</b>	<b>Description</b>
<u>QUALIFY:</u>	Maximum response time of the phone in milliseconds (default 2000). Every 10 seconds the Voclarion checks if the phone can be reached. This can be compared with pinging. If no response is received, the phone will be marked as unreachable and cannot be reached. If a phone is located at a far distance from the PBX, you can increase the value.
<u>CODEC 1:</u>	The most preferred codec for this phone. Make sure the phone and the trunk codec are the same, otherwise the PBX has to translate all calls from one codec to another. This will decrease performance. If this codec is unavailable, the next is used.
<u>CODEC 2:</u>	The second most preferred codec. If this codec is unavailable, the next is used.
<u>CODEC 3:</u>	The third most preferred codec. If this codec is unavailable, the next is used.
<u>CODEC 4:</u>	The last preferred Codec

## Adding Phones to the Voclarion Manager

<b>Tab SIP Settings</b>	<b>Description</b>
<u>DTMF MODE:</u>	<p>Mode of sending DTMF tones. DTMF is used to send user input to the PBX, for example to select an option from a menu. Tones are generated when a user presses keys on the dial pad.</p> <ul style="list-style-type: none"><li>• Default</li><li>• Inband means the DTMF is transmitted within the audio of the phone conversation, it is audible to the conversation partners. Therefore only uncompressed codecs like g711 alaw or µlaw can carry inband DTMF reliably. Female voice are known to once in a while trigger the recognition of a DTMF tone. For analog lines inband is the only possible means to transmit DTMF.</li><li>• SIP INFO, out of band; DTMF tones are send outside the audio of the phone conversation. Only available with SIP channels, transmitted through a SIP message.</li><li>• RFC 2833, out of band; DTMF tones are send outside the audio of the phone conversation.</li></ul>

This page also shows the number of available lines for this phone and the connected extension. Click on the [EXTENSION](#) to see the extension settings (chapter 2.3) or click the [LINE NUMBER](#) to see and edit the settings for this line specifically:

<b>Field</b>	<b>Description</b>
<u>DESCRIPTION:</u>	Description for this line.

<b>Field</b>	<b>Description</b>
<u>EXTENSION:</u>	The extension connected to this line. Calls to this extension will come in over this line and -if available- the BLF connected to this line will blink. Outgoing calls over this line have the caller ID as set on this extension <sup>41</sup> . Click on the <u>EXTENSION</u> to see the extension settings (chapter 2.3).
Active:	The actual state of this line (active/not active).
Protocol:	The protocol used for this line.
Status:	The state of the line, see chapter 1.6.4.
User name:	The user name and password for SIP lines, needed for the manual registration on Voclarion for DECT and smart phones.
Password:	
<u>CODEC 1:</u>	The most preferred Codec for this phone. Make sure the phone and the trunk Codec are the same, otherwise the Voclarion has to translate all calls from one Codec to another. This will decrease performance.
<u>CODEC 2:</u>	The second most preferred Codec, will be used when the first codec is unavailable.
<u>CODEC 3:</u>	The third most preferred Codec
<u>CODEC 4:</u>	The last preferred Codec

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<sup>41</sup> If not overruled by other caller ID settings, see chapter 1.4.2.5 on caller ID priority.

## Adding Phones to the Voclarion Manager

Field	Description
<u>DTMF MODE:</u>	<p>Mode of sending DTMF tones. DTMF is used to send user input to the Voclarion, for example to select an option from a menu. Tones are generated when a user presses keys on the dial pad.</p> <ul style="list-style-type: none"><li>• Default</li><li>• Inband means the DTMF is transmitted within the audio of the phone conversation, it is audible to the conversation partners. Therefore only uncompressed codecs like g711 alaw or µlaw can carry inband DTMF reliably. Female voice are known to once in a while trigger the recognition of a DTMF tone. For analog lines inband is the only possible means to transmit DTMF.</li><li>• SIP INFO, out of band; DTMF tones are send outside the audio of the phone conversation. Only available with SIP channels, transmitted through a SIP message.</li><li>• RFC 2833, out of band; DTMF tones are send outside the audio of the phone conversation.</li><li>• None, accept RFC2833, but send nothing.</li></ul>

Tab	Description
<b>Call Settings</b>	
<u>CALL WAITING:</u>	<p>Make this phone accept concurrent calls. New calls will be put on hold. When you enable Call Waiting, make sure <u>MAXIMUM CONCURRENT CALLS</u> is set to at least 2 (see extension settings, chapter 2.3.3.1).</p>
<u>VOLUME CALL WAITINGTONE:</u>	<p>Set the volume of the call waitingtone (0-15). When a call is on hold, the user hears two short beep tones.</p>

<b>Tab</b>	<b>Description</b>
<b>Call Settings</b>	
<u>MESSAGE WAITING NOTIFICATION:</u>	Choose whether you want to receive a notification on new voice mail. Choose between “Silent” , “Beep” , “Reminder” , “Blinking LED”.
<u>MESSAGE WAITING DIAL TONE:</u>	On new voice mail the dial tone will stutter. To deactivate this, choose “Normal”.
<u>DIAL TONE DURING HOLD:</u>	After you put a caller on hold, you can choose to hear a dial tone. If you choose “No”, then there is silence instead.
<u>TRANSFER ON HANGUP:</u>	When set to “Yes” the waiting call will be transferred to the current call when you hang up.
<u>DO NOT DISTURB (DND) ALLOWED:</u>	Enable/disable the Do Not Disturb feature.
<u>ENABLE CMC (CLIENT MATTER CODE):</u>	Enable Client Matter Code by entering “Yes”. Makes it possible to supply a CMC while in a call.

<b>Tab</b>	<b>Description</b>
<b>Function Keys</b>	<b><i>Only available when a user is logged in.</i></b> Function keys are programmable buttons on a phone, often with a BLF to indicate the status of the button. Function keys can be present on the device itself, but can also be added as an optional extension pad. We discuss function keys in chapter 2.5.2.

<b>Tab</b>	<b>Description</b>
<b>Extension Pad</b>	<b><i>Only available when a user is logged in.</i></b> An extension pad contains a set of programmable buttons, often with a BLF to indicate the status of the button.

## Adding Phones to the Voclarion Manager

<b>Tab</b>	<b>Description</b>
<b>Video</b>	
<u>FRAME RATE:</u>	Amount of video frames per second, sent to the receiver.
<u>BIT RATE:</u>	The speed with which frames are sent.
<u>PACKET SIZE:</u>	The size of video packets.

<b>Tab</b>	<b>Description</b>
<b>Advanced Settings</b>	
<u>AUDIO MODE:</u>	Choose the audio mode when a call comes in: <ul style="list-style-type: none"><li>• <b>Speaker</b> Calls can be made or received using the handset or handsfree speakerphone and can be switched between the two modes. When on speaker, you can return to handset mode by picking up het handset.</li><li>• <b>Headset</b> Calls can be made or received using the headset. Calls can be switched between the headset and handset</li><li>• <b>Speaker/Headset</b> Incoming calls are sent to the speakerphone. By pressing a key you can switch between the handsfree speakerphone, the headset, and the handset.</li><li>• <b>Headset/Speaker</b> Incoming calls are sent to the headset. By pressing a key, you can switch between the headset, the handsfree speakerphone, and the handset.</li></ul>
<u>HEADSET DEVICE:</u>	Choose the type of headset device: RJ connector, Cinch connector or None.
<u>SHOW MISSED CALLS:</u>	Shows the number of missed calls in the display of the phone.

Tab	Description
<b>Advanced Settings</b>	
<u>VOLUME CALL</u>	When on the phone and there's another incoming call, the PBX will give a repeating beep tone. The volume of this tone can be adjusted.
<u>WAITINGTONE:</u>	
<u>HASH TRANSFERS</u>	(De-)activate # to transfer an incoming call. Callee transfer is often used by DECT or mobile phones which don't have a hold/transfer button (see chapter 10.10).
<u>INCOMING CALLS:</u>	
<u>HASH TRANSFERS</u>	(De-)activate # to transfer an outgoing call. Callee transfer is often used by DECT or mobile phones which don't have a hold/transfer button (see chapter 10.10).
<u>OUTGOING CALLS:</u>	
<u>HEADSET MICROPHONE</u>	The default volume of the microphone of the headset on the phone.
<u>VOLUME:</u>	
<u>HANDSET MICROPHONE</u>	The default volume of the microphone if the handset on the phone.
<u>VOLUME:</u>	
<u>SPEAKER MICROPHONE</u>	The default volume of the microphone of the speaker of the phone.
<u>VOLUME:</u>	

## 2.5.2. Function Keys

If your phone has programmable Function Keys (on the phone itself or on an optional extension pad), you can use the PBX Manager to assign a function to these keys. Sometimes a function key is equipped with a Busy Lamp Field (BLF) which shows the status of the key.

In most cases, a company has nearly all of the same phones. And the desired function keys are the same for almost everyone. That's why we created Function Key Templates, so that you no longer need to manually set all keys on every phone. With Function Key Templates you create a template containing a set of Function Keys. Once done, you can assign this template to several phones.



If you like to use Function Key Templates, create these first.

### 2.5.2.1. Manually Defining Function keys

**Web interface menu**      Companies > *Company* > Phones > *Phone*

Make sure you select an phone with a user logged in. At the top of the page look for the tab “Function keys” (to define keys on your phone) or the tab “Extension Pad” (to define the keys on an extension pad, when available).

1. Open the tab. An overview of all available keys is shown. By default all keys have “No function”.
2. Click EDIT ( ) to change the keys on the phone or on the extension selected extension pad.
3. Choose for each button a label, a function and provide a phone number / extension when needed. The label will be displayed on the phone screen (when available). An overview of all functions can be found on page Error: Reference source not found. Buttons defined by a template cannot be edited and are grayed out.
4. Click SAVE.
5. Reboot the phone before use.

## 2.6. Users

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After creating departments, you can add employees (we call them users) to the departments. In this chapter we show you how to create and edit users and assign roles to them.

### 2.6.1. Roles

This PBX has a role based user system, meaning there are several types of users (or “roles”). Each role has its own set of permissions: things a user can (not) do. In Voclarion 2.1 and before we had some predefined roles like System Administrator and Administrator which could be assigned to a user. These roles are still available and can be assigned to users. Other permissions had to be set per user. This has changed. Before continuing, it's advisable to create one or more roles. For more information see chapter Error: Reference source not found.

### 2.6.2. Login

The Voclarion needs to know who you are. A user has several ways of identifying himself. For example a user has a login for the graphical user interface and has to supply a PIN to listen to recorded voice mail messages. We describe the different ways to log in on this paragraph.

#### 2.6.2.1. User Name and Login Password

Every user has a user name and password to log in to the Voclarion PBX Manager; the graphical user interface. The password is set when creating a user and cannot be viewed, only changed by the department manager, company manager or the administrator. The password can contain digits, characters and special characters as described in chapter 1.7. As a precaution you have 3 consecutive chances to enter your user name and password correctly. After this the account is locked. See page 48 for more information about locked accounts.

#### 2.6.2.2. Phone Login and PIN

Before making phone calls, you have to log in at least once to a phone. Once logged in you can stay logged in, however it's a good idea to log out at the end of

## Users

the day and login at the start of the day. The Voclarion now knows if you are available or not and can handle calls more efficient. Also logging out at the end of your shift prevents abuse of your phone.

You identify yourself using your Phone Login and PIN (Personal Identification Number). A PIN is a combination of digits. The current Phone Login and PIN can be found at the user settings (see chapter 2.8.3.1). By default the Phone Login and the PIN are both similar to your extension. We urge you to change this, but keep in mind that the Phone Login needs to have the same amount of digits as your extensions. The login code and PIN may together contain up to 10 characters.

- To log in dial \*1 from your phone and follow the instructions.
- To log off dial \*1 again<sup>42</sup>.

The PIN is also used to authorize calls. The PBX can ask you to identify yourself to check if you have permissions to call certain types of numbers. Supply it when asked for.

### 2.6.2.3. Activated users

By default the system has 5 activated users, meaning five users can simultaneous log in. If you need more users activated, contact your reseller. The company overview (menu *Companies*) shows how many activated users are currently used by each company. To see the number of activated users for the PBX, go to the About page on the top menu.

### 2.6.2.4. Voice Mail Box Number and Voice Mail Password

The Voice Mail Password is used to identify yourself when accessing Voice Mail using a phone. The password is set automatically when creating a new extension and can be viewed and changed on the extension settings. The password contains four digits as described on page 74. By default the password and box number are similar to the extension. For more information about logging in to your Voice Mail see chapter 10.9.

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<sup>42</sup> Logging out can be blocked, see page 117.

Action / Role	SA	CM <sup>43</sup>	DM <sup>44</sup>	U <sup>45</sup>
Add/edit/remove all companies	<input type="checkbox"/>			
Add departments	<input type="checkbox"/>	<input type="checkbox"/>		
Edit department	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	
Add/edit/remove users	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	
Change user settings	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Manage conference boxes of own department	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Access company reports	<input type="checkbox"/>	<input type="checkbox"/>		
Access department reports	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	
Access personal reports	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Edit personal phone book	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Edit company phone book	<input type="checkbox"/>	<input type="checkbox"/>		
Edit PBX system settings	<input type="checkbox"/>			
View queue details	<input type="checkbox"/>	<input type="checkbox"/>		
Changing dial plan	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	

Roles:

SA = System Administrator

CM = Company Manager

DM = Department Manager

U = Regular user

*Table 7: Role based GUI*

### 2.6.3. Adding Users

There are two ways of adding new users to the PBX:

- You can add users by importing them from a spreadsheet. This is mostly used on new (empty) companies. All users will be automatically created with default settings. It's recommended to fine tune the user settings. If you add users by spreadsheet.

<sup>43</sup> Only for his 'own' company

<sup>44</sup> Only for his 'own' department

<sup>45</sup> Only personal settings

## Users

- You can also add new users manually one by one. This is mostly used to add a small number of users. The next paragraphs show you how to proceed.

### 2.6.3.1. Adding a New User

There are two ways to add a new user. If you first select a department, new users will be automatically assigned to this department. You can also select a company and while creating a user, assign a department.

#### Web interface menu

- Companies > *Company* > Departments > *Department* > *tab Users*
- Companies > *Company* > Users

1. Go to one of the pages described above to see a list of users for this department or company. Click ADD (+) in the top-right corner to add a new user.
2. Fill out all required information for the new user as described below.

Field	Description
<u>FIRST NAME:</u>	First name of the user
<u>LAST NAME:</u>	Last name of the user. First and last name will be shown on the phone display.
<u>ROLE:</u>	Select the role for this user.
<u>RATE PLAN:</u>	The rate plan applied to this user. Only visible if Billing is enabled, check the Installation Manual for more information.
<u>PREPAY:</u>	Set a prepay amount. Use a dot (.) as separator, like "25.50". Before making a phone call, the user credits are checked. After the call the account is credited. Only available when Billing is enabled. See the Installation Manual for more information.

Field	Description
<u>USER NAME:</u>	The user name is used to login to the PBX web interface. It can be up to 64 characters long and the first character must be a character (a-z). Each user name must be unique PBX wide. See table 7 on page 102 for unauthorized names. <b>Note:</b> Once created a user, the user name cannot be changed anymore.
<u>PASSWORD:</u>	The password is used to access the PBX web interface. Passwords have 4-40 characters.
<u>TIME_ZONE:</u>	Select the time zone for this user from the pull down menu.
<u>MOBILE_NUMBER:</u>	The mobile phone number of this user. It can be used in the call forwarding settings. Also the value for this field is used to identify employees within the dial plan and when using DISA (chapter 3.13).
<u>SEND_VOICE_MAIL_TO:</u>	If voice mail is used, you can receive the voice mail messages in your mailbox <sup>46</sup> . Supply one valid e-mail address <sup>47</sup> .
<u>LOCATION:</u>	Location of this user, for example "Office Amsterdam, first floor".
<u>LANGUAGE:</u>	The preferred language used for this user. Used for the PBX manager and instructions from Voclarion to the user on the phone.
<u>DEPARTMENT:</u>	The user's department. Only available if the user is added to a company and not to a department (see page 117). A user should be member of at least one department. Members of the same department can pick up calls for each other (Call Pickup, see chapter 10.7).
<u>EXTENSION:</u>	Select an extension for this user.
<u>PARKING_LOT:</u>	The parking lot this user can take calls from.

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<sup>46</sup> You can also listen to your voice mail messages by phone or from the reports page.

<sup>47</sup> If you would like to receive messages on more than one e-mail address, create an alias on your mail server, which forwards the message.

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Field	Description
<u>COPY PHONE BOOK FROM:</u>	Chose an existing user to copy the personal phone numbers from.
<u>USER FIELDS 1-5:</u>	Only visible if created first, see 2.9 chapter .These fields can store user specified information. For example the birthday or an ID referring to a database.
<u>NOTES:</u>	Information about this user.

Press the SAVE button to create the new user. After saving you'll return to the user list. You can fine tune the created user if you like. Click on the last name to see the user details and click EDIT () to edit the user. The edit screen contains a few new fields:

Field	Description
<u>CONTACT TYPE:</u>	Select a contact type for this user.
<u>CURRENT PHONE:</u>	This field indicates the phone the user is logged on to. Set it to empty to log out the user or select a new phone to log in. Phone will log in after saving the settings.
<u>FUNCTION KEY TEMPLATE:</u>	Apply a function key template for programmable function keys to this user (see chapter Error: Reference source not found).
<u>CALLER ID:</u>	You can supply an alternative caller ID for this user. If you don't want to set an alternative caller ID, simply leave this field blank. If the caller ID is set to "0", the caller ID is suppressed for outgoing calls for this user. Once set, this caller ID overwrites the company and department caller ID. A caller ID only contains only digits. For more information see chapter 1.4.2.5.

Field	Description
<u>CALLER NAME:</u>	You can supply an alternative Caller Name for this user. If you don't want to set an alternative Caller Name, simply leave this field blank. If the caller ID is set to "0", the Caller Name is suppressed. Once set, this Caller Name overwrites the company and department Caller Name. More information about Caller Names see chapter 3.2.3.
<u>PREFIX FOR OUTGOING CALLS:</u>	Add these digits at the beginning of the phone number. Can be used to select a specific outgoing trunk. The trunk can remove these digits before sending the call out. This setting supersedes the department prefix.
<u>VISIBLE IN COMPANY PHONE BOOK:</u>	Show this user in the Company Phone Book?
<u>PHONE LOGIN:</u>	This is the code for log in to a phone. By default this code is set the same as the extension of this user <sup>48</sup> .
<u>PIN:</u>	This is the password to login to a phone. By default the PIN is similar to the user's extension. It's very advisable to change this. Phone login and PIN must be unique and cannot be empty.
<u>CAN LOGOUT:</u>	Can the user log out? In most cases it is advisable to log out at the end of the shift. However in some cases (fax) <sup>4</sup> this is not desirable.
<u>QUEUE PAUSE:</u>	Can user take a queue pause? Can be used when the employee takes a break. When a queue pause is activated, calls from all queues will be held. Other calls will still be connected to the phone.
<u>FAILED LOGINS:</u>	The number of failed logins. After the 3 <sup>rd</sup> failed login, the account is locked for 24 hours. When a user logs in correctly after 24 hours, the number of failed logins will be reset to "0".
<u>LAST FAILED LOGIN:</u>	The time and date of the last failed login.

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<sup>48</sup> See chapter 10.2 for more information about how to login.

## Users

### Field

ACCOUNT LOCKED:

### Description

If the account is locked, this field is set to "yes". After 24 hours, when a user logs in correctly, his account will unlock automatically and this field will be set to "no". The system administrator can unlock an account.

all	foreign	table	rpcuser
analyse	from	then	nfsnobody
and	grant	to	pcap
any	group	trailing	mailnull
array	having	true	smmsp
as	in	union	apache
asc	initially	unique	asterisk
asymmetric	intersect	user	postgres
both	into	using	postfix
case	leading	when	amavis
cast	limit	root	named
check	localtime	bin	clamav
collate	localtimestamp	adm	ldap
column	new	lp	cyrus
constraint	not	daemon	upwatch
create	null	sync	tomcat
current_date	off	shutdown	plcmspip
current_role	offset	halt	ntp
current_time	old	mail	jive
current_timestamp	on	news	sophos
current_user	only	uucp	astium
default	or	operator	openexchange
deferrable	order	games	administrator
desc	placing	gopher	
distinct	primary	ftp	
do	references	nobody	
else	returning	rpm	
end	select	vcsa	
except	session_user	nscd	
false	some	sshd	
for	symmetric	rpc	

Table 8: Blocked user names

## 2.6.4. Changing Departments

A user must be a member of at least one department and can be member of several departments. If you want to *change* the user's department, first assign a new department and then remove the current department.

**Web interface menu**      companies > *company* > users

This page shows a list of all users for the selected company. Click on the last name of the user you would like to change.

1. A menu item with the user's name will appear. Unfold the menu.
2. Click on “departments” from this menu. This page shows all departments for this user.
3. Click [ASSIGN DEPARTMENT](#) to assign another department. Or (if the user is member of more than one department), click [REMOVE DEPARTMENT](#) to remove a user from a department.



Note that users cannot be active without at least one department. If you remove all departments linked to a user, the user will be put in a temporary department named “[LOST + FOUND](#)”. This temporary department will automatically disappear from the department list when the user is assigned a new department.

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## 2.6.5. Add additional Extension

Sometimes it's useful to add more extensions to one user. For example if you would like to call out with different caller ID's. You can assign extensions (with different caller ID's) to different line buttons on your phone. When pressing a line button first, your call goes out with the corresponding caller ID. Also incoming calls are directed to the corresponding line button.



You cannot login on two phones at the same time. Only the first assigned extension can be used to login. For receiving calls on different phones see Twinning.

To add an additional extension to a user, go to the next page:

**Web interface menu**      Companies > *Company* > Users

1. Click on the user's last name to see the settings.
2. At the left menu, a new menu option with the name of the user is created. Unfold the menu and click on the menu option Extensions. The page shows the extensions for this user.
3. Click the ASSIGN EXTENSION button.
4. Select an extension from the selection box. If the selected extension is used by another person, his phone will log out.
5. The user's phone will reboot and the new extension is added to the user's extension list.

## 2.6.6. Remove Extensions

There are two ways to deactivate a user. Besides removing a user from a department, you can also remove all extensions from a user. If the user's last remaining extension is removed, the Phone Login and PIN of the user will automatically be set to default '0000'. After finishing his conversation, the user has no rights to make any calls or to log out. It is advisable to logout the user before removing the last extension. You always can log out a user by using the web interface (User settings).

## Users



- The removed extension will be removed from all queues, but the extension itself will not be deleted from the Voclarion.
- If the extension is used in the dial plan, e.g. in an IVR menu, this extension becomes inactive and will bounce all calls. Do not forget to replace this inactive extension by an active one.

### Web interface menu

Companies > *Company* > Users

1. Click on the user's last name to see the settings.
2. At the left menu, a new menu option with the name of the user is created. Unfold the menu and click on the menu option Extensions. The page shows the extensions for this user.
3. Click the REMOVE EXTENSION button.
4. You will receive a warning text: "Are you sure?". Click the YES button.
5. Click on the extension icon you want to remove.

## 2.7. User Fields

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With User Fields you can add your own information fields to the user settings and to the contacts in your phone book. You can add up to 5 field. Each field can be given a label, a field type and a help text. To create or edit user fields, go to:

**Web interface menu**      Companies > *Company* > Advanced > User Fields

1. Select the fields you would like to add or change by clicking on the tab at the top of the page.
  - User fields are visible on the user settings.
  - Phone book fields are visible on the phone books.
2. Click Add (+).
3. Enter the following information:

<b>Row</b>	<b>Description</b>
<u>TYPE:</u>	Type of accepted information, like text, date or email. The PBX will check if the input is valid. See chapter 2.9.1.
<u>ROW LENGTH:</u>	The maximum number characters per rows in edit mode. Only available for text field.
<u>NUMBER OF ROWS:</u>	The maximum number of rows in edit mode. Only available for text field.
<u>NAME:</u>	Label used for this field. When this field is blank, the field will not be shown.

## User Fields

### Row

### Description

HELP MESSAGE:

Text used as help text. An icon (❓) is placed next to the label. Click on the icon to see the text.

1. Click SAVE.

## 2.7.1. Field types

The following field types can be selected:

Type	Description
<b>Country</b>	Select a country from the list.
<b>Date</b>	Select a date by using the pull down menus, or the calendar icon.
<b>Date/Time</b>	Select a date and time by using the pull down menus.
<b>Email address</b>	Email address
<b>Floating point number</b>	A real number (that is, a number that can contain a fractional part). The following are floating-point numbers: 3.0, -111.5.
<b>Monetary amount</b>	Use a dot as separator.
<b>Number</b>	Number, no digits.
<b>Password</b>	Passwords are hidden when typing and are also not visible when viewing information. Type twice the same password, minimal 4 characters.
<b>Phone type</b>	Shows a pull down with all phone types. Select one.
<b>Sound prompt</b>	Select a sound prompt from a list of available sound prompts.
<b>Text</b>	Enter one line of text
<b>Text area</b>	Enter multiple lines of text
<b>Time</b>	Select a time by using the pull down menus.
<b>User</b>	Select a user from the list of employees.
<b>Yes / No</b>	You can select Yes or No.

## 2.8. Users

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After creating departments, you can add employees (we call them users) to the departments. In this chapter we show you how to create and edit users and assign roles to them.

### 2.8.1. Roles

This PBX has a role based user system, meaning there are several types of users (or “roles”). Each role has its own set of permissions: things a user can (not) do. Prior to Voclarion 2.2 and before we had some predefined roles like System Administrator and Administrator which could be assigned to a user. These roles are still available and can be assigned to users. Other permissions had to be set per user. This has changed. Before continuing, it's advisable to create one or more roles. For more information see chapter Error: Reference source not found.

### 2.8.2. Login

The Voclarion needs to know who you are. A user has several ways of identifying himself. For example a user has a login for the graphical user interface and has to supply a PIN to listen to recorded voice mail messages. We describe the different ways to log in on this paragraph.

#### 2.8.2.1. User Name and Login Password

Every user has a user name and password to log in to the Voclarion PBX Manager; the graphical user interface. The password is set when creating a user and cannot be viewed, only changed by the department manager, company manager or the administrator. The password can contain digits, characters and special characters as described in chapter 1.7. As a precaution you have 3 consecutive chances to enter your user name and password correctly. After this the account is locked. See page 48 for more information about locked accounts.

#### 2.8.2.2. Phone Login and PIN

Before making phone calls, you have to log in at least once to a phone. Once logged in you can stay logged in, however it's a good idea to log out at the end of

## Users

the day and login at the start of the day. The Voclarion now knows if you are available or not and can handle calls more efficient. Also logging out at the end of your shift prevents abuse of your phone.

You identify yourself using your Phone Login and PIN (Personal Identification Number). A PIN is a combination of digits. The current Phone Login and PIN can be found at the user settings (see chapter 2.8.3.1). By default the Phone Login and the PIN are both similar to your extension. We urge you to change this, but keep in mind that the Phone Login needs to have the same amount of digits as your extensions. The login code and PIN may together contain up to 10 characters.

- To log in dial \*1 from your phone and follow the instructions.
- To log off dial \*1 again<sup>49</sup>.

The PIN is also used to authorize calls. The PBX can ask you to identify yourself to check if you have permissions to call certain types of numbers. Supply it when asked for.

### 2.8.2.3. Activated users

By default the system has 5 activated users, meaning five users can simultaneous log in. If you need more users activated, contact your reseller. The company overview (menu *Companies*) shows how many activated users are currently used by each company. To see the number of activated users for the PBX, go to the About page on the top menu.

### 2.8.2.4. Voice Mail Box Number and Voice Mail Password

The Voice Mail Password is used to identify yourself when accessing Voice Mail using a phone. The password is set automatically when creating a new extension and can be viewed and changed on the extension settings. The password contains four digits as described on page 74. By default the password and box number are similar to the extension. For more information about logging in to your Voice Mail see chapter 10.9.

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<sup>49</sup> Logging out can be blocked, see page 117.

Action / Role	SA	CM <sup>50</sup>	DM <sup>51</sup>	U <sup>52</sup>
Add/edit/remove all companies	<input type="checkbox"/>			
Add departments	<input type="checkbox"/>	<input type="checkbox"/>		
Edit department	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	
Add/edit/remove users	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	
Change user settings	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Manage conference boxes of own department	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Access company reports	<input type="checkbox"/>	<input type="checkbox"/>		
Access department reports	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	
Access personal reports	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Edit personal phone book	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Edit company phone book	<input type="checkbox"/>	<input type="checkbox"/>		
Edit PBX system settings	<input type="checkbox"/>			
View queue details	<input type="checkbox"/>	<input type="checkbox"/>		
Changing dial plan	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	

Roles:

SA = System Administrator

CM = Company Manager

DM = Department Manager

U = Regular user

Table 9: Role based GUI

### 2.8.3. Adding Users

There are two ways of adding new users to the PBX:

- You can add users by importing them from a spreadsheet. This is mostly used on new (empty) companies. All users will be automatically created with default settings. It's recommended to fine tune the user settings. If you add users by spreadsheet.

<sup>50</sup> Only for his 'own' company

<sup>51</sup> Only for his 'own' department

<sup>52</sup> Only personal settings

## Users

- You can also add new users manually one by one. This is mostly used to add a small number of users. The next paragraphs show you how to proceed.

### 2.8.3.1. Adding a New User

There are two ways to add a new user. If you first select a department, new users will be automatically assigned to this department. You can also select a company and while creating a user, assign a department.

#### Web interface menu

- Companies > *Company* > Departments > *Department* > *tab Users*
- Companies > *Company* > Users

1. Go to one of the pages described above to see a list of users for this department or company. Click ADD (+) in the top-right corner to add a new user.
2. Fill out all required information for the new user as described below.

Field	Description
<u>FIRST NAME:</u>	First name of the user
<u>LAST NAME:</u>	Last name of the user. First and last name will be shown on the phone display.
<u>ROLE:</u>	Select the role for this user.
<u>RATE PLAN:</u>	The rate plan applied to this user. Only visible if Billing is enabled, check the Installation Manual for more information.
<u>PREPAY:</u>	Set a prepay amount. Use a dot (.) as separator, like "25.50". Before making a phone call, the user credits are checked. After the call the account is credited. Only available when Billing is enabled. See the Installation Manual for more information.

Field	Description
<u>USER NAME:</u>	The user name is used to login to the PBX web interface. It can be up to 64 characters long and the first character must be a character (a-z). Each user name must be unique PBX wide. See table 9 on page 116 for unauthorized names. <b>Note:</b> Once created a user, the user name cannot be changed anymore.
<u>PASSWORD:</u>	The password is used to access the PBX web interface. Passwords have 4-40 characters.
<u>TIME_ZONE:</u>	Select the time zone for this user from the pull down menu.
<u>MOBILE_NUMBER:</u>	The mobile phone number of this user. It can be used in the call forwarding settings. Also the value for this field is used to identify employees within the dial plan and when using DISA (chapter 3.13).
<u>SEND_VOICE_MAIL_TO:</u>	If voice mail is used, you can receive the voice mail messages in your mailbox <sup>53</sup> . Supply one valid e-mail address <sup>54</sup> .
<u>LOCATION:</u>	Location of this user, for example "Office Amsterdam, first floor".
<u>LANGUAGE:</u>	The preferred language used for this user. Used for the PBX manager and instructions from Voclarion to the user on the phone.
<u>DEPARTMENT:</u>	The user's department. Only available if the user is added to a company and not to a department (see page 117). A user should be member of at least one department. Members of the same department can pick up calls for each other (Call Pickup, see chapter 10.7).
<u>EXTENSION:</u>	Select an extension for this user.
<u>PARKING_LOT:</u>	The parking lot this user can take calls from.

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53 You can also listen to your voice mail messages by phone or from the reports page.

54 If you would like to receive messages on more than one e-mail address, create an alias on your mail server, which forwards the message.

## Users

Field	Description
<u>COPY PHONE BOOK FROM:</u>	Chose an existing user to copy the personal phone numbers from.
<u>USER FIELDS 1-5:</u>	Only visible if created first, see 2.9 chapter .These fields can store user specified information. For example the birthday or an ID referring to a database.
<u>NOTES:</u>	Information about this user.

Press the SAVE button to create the new user. After saving you'll return to the user list. You can fine tune the created user if you like. Click on the last name to see the user details and click EDIT () to edit the user. The edit screen contains a few new fields:

Field	Description
<u>CONTACT TYPE:</u>	Select a contact type for this user.
<u>CURRENT PHONE:</u>	This field indicates the phone the user is logged on to. Set it to empty to log out the user or select a new phone to log in. Phone will log in after saving the settings.
<u>FUNCTION KEY TEMPLATE:</u>	Apply a function key template for programmable function keys to this user.
<u>CALLER ID:</u>	You can supply an alternative caller ID for this user. If you don't want to set an alternative caller ID, simply leave this field blank. If the caller ID is set to "0", the caller ID is suppressed for outgoing calls for this user. Once set, this caller ID overwrites the company and department caller ID. A caller ID only contains only digits. For more information see chapter 1.4.2.5.
<u>CALLER NAME:</u>	You can supply an alternative Caller Name for this user. If you don't want to set an alternative Caller Name, simply leave this field blank. If the caller ID is set to "0", the Caller Name is suppressed. Once set, this Caller Name overwrites the company and department Caller Name. More information about Caller Names see chapter 3.2.3.

<b>Field</b>	<b>Description</b>
<u>PREFIX FOR OUTGOING CALLS:</u>	Add these digits at the beginning of the phone number. Can be used to select a specific outgoing trunk. The trunk can remove these digits before dialing the number. This setting supersedes the department prefix.
<u>VISIBLE IN COMPANY PHONE BOOK:</u>	Show this user in the Company Phone Book?
<u>PHONE LOGIN:</u>	This is the code for log in to a phone. By default this code is set the same as the extension of this user <sup>55</sup> .
<u>PIN:</u>	This is the password to login to a phone. By default the PIN is similar to the user's extension. It's very advisable to change this. Phone login and PIN must be unique and cannot be empty.
<u>CAN LOGOUT:</u>	Can the user log out? In most cases it is advisable to log out at the end of the shift. However in some cases (fax) <sup>4</sup> this is not desirable.
<u>QUEUE PAUSE:</u>	Can user take a queue pause? Can be used when the employee takes a break. When a queue pause is activated, calls from all queues will be held. Other calls will still be connected to the phone.
<u>FAILED LOGINS:</u>	The number of failed logins. After the 3 <sup>rd</sup> failed login, the account is locked for 24 hours. When a user logs in correctly after 24 hours, the number of failed logins will be reset to "0".
<u>LAST FAILED LOGIN:</u>	The time and date of the last failed login.
<u>ACCOUNT LOCKED:</u>	If the account is locked, this field is set to "yes". After 24 hours, when a user logs in correctly, his account will unlock automatically and this field will be set to "no". The system administrator can unlock an account.

---

<sup>55</sup> See chapter 10.2 for more information about how to login.

## Users

all	foreign	table	rpcuser
analyse	from	then	nfsnobody
and	grant	to	pcap
any	group	trailing	mailnull
array	having	true	smmsp
as	in	union	apache
asc	initially	unique	asterisk
asymmetric	intersect	user	postgres
both	into	using	postfix
case	leading	when	amavis
cast	limit	root	named
check	localtime	bin	clamav
collate	localtimestamp	adm	ldap
column	new	lp	cyrus
constraint	not	daemon	upwatch
create	null	sync	tomcat
current_date	off	shutdown	plcmspip
current_role	offset	halt	ntp
current_time	old	mail	jive
current_timestamp	on	news	sophos
current_user	only	uucp	astium
default	or	operator	openexchange
deferrable	order	games	administrator
desc	placing	gopher	
distinct	primary	ftp	
do	references	nobody	
else	returning	rpm	
end	select	vcsa	
except	session_user	nscd	
false	some	sshd	
for	symmetric	rpc	

Table 10: Blocked user names

## 2.8.4. Changing Departments

A user must be a member of at least one department and can be member of several departments. If you want to *change* the user's department, first assign a new department and then remove the current department.

**Web interface menu**      companies > *company* > users

This page shows a list of all users for the selected company. Click on the last name of the user you would like to change.

1. A menu item with the user's name will appear. Unfold the menu.
2. Click on "departments" from this menu. This page shows all departments for this user.
3. Click [ASSIGN DEPARTMENT](#) to assign another department. Or (if the user is member of more than one department), click [REMOVE DEPARTMENT](#) to remove a user from a department.



Note that users cannot be active without at least one department. If you remove all departments linked to a user, the user will be put in a temporary department named "LOST + FOUND". This temporary department will automatically disappear from the department list when the user is assigned a new department.

---

## 2.8.5. Add additional Extension

Sometimes it's useful to add more extensions to one user. For example if you would like to call out with different caller ID's. You can assign extensions (with different caller ID's) to different line buttons on your phone. When pressing a line button first, your call goes out with the corresponding caller ID. Also incoming calls are directed to the corresponding line button.

## Users



You cannot login on two phones at the same time. Only the first assigned extension can be used to login. For receiving calls on different phones see Twinning.

To add an additional extension to a user, go to the next page:

**Web interface menu**      Companies > *Company* > Users

1. Click on the user's last name to see the settings.
2. At the left menu, a new menu option with the name of the user is created. Unfold the menu and click on the menu option Extensions. The page shows the extensions for this user.
3. Click the ASSIGN EXTENSION button.
4. Select an extension from the selection box. If the selected extension is used by another person, his phone will log out.
5. The user's phone will reboot and the new extension is added to the user's extension list.

### 2.8.6. Remove Extensions

There are two ways to deactivate a user. Besides removing a user from a department, you can also remove all extensions from a user. If the user's last remaining extension is removed, the Phone Login and PIN of the user will automatically be set to default '0000'. After finishing his conversation, the user has no rights to make any calls or to log out. It is advisable to logout the user before removing the last extension. You always can log out a user by using the web interface (User settings).



- The removed extension will be removed from all queues, but the extension itself will not be deleted from the Voclarion.
- If the extension is used in the dial plan, e.g. in an IVR menu, this extension becomes inactive and will bounce all calls. Do not forget to replace this inactive extension by an active one.

**Web interface menu**Companies > *Company* > Users

1. Click on the user's last name to see the settings.
2. At the left menu, a new menu option with the name of the user is created. Unfold the menu and click on the menu option Extensions. The page shows the extensions for this user.
3. Click the REMOVE EXTENSION button.
4. You will receive a warning text: "Are you sure?". Click the YES button.
5. Click on the extension icon you want to remove.

## 2.9. User Fields

---

With User Fields you can add your own information fields to the user settings and to the contacts in your phone book. You can add up to 5 field. Each field can be given a label, a field type and a help text. To create or edit user fields, go to:

**Web interface menu**      Companies > *Company* > Advanced > User Fields

1. Select the fields you would like to add or change by clicking on the tab at the top of the page.
  - User fields are visible on the user settings.
  - Phone book fields are visible on the phone books.
2. Click ADD (+).
3. Enter the following information:

<b>Row</b>	<b>Description</b>
<u>TYPE:</u>	Type of accepted information, like text, date or email. The PBX will check if the input is valid. See chapter 2.9.1.
<u>ROW LENGTH:</u>	The maximum number characters per rows in edit mode. Only available for text field.
<u>NUMBER OF ROWS:</u>	The maximum number of rows in edit mode. Only available for text field.
<u>NAME:</u>	Label used for this field. When this field is blank, the field will not be shown.

Row	Description
<u>HELP MESSAGE:</u>	Text used as help text. An icon (🔍) is placed next to the label. Click on the icon to see the text.

1. Click SAVE.

## 2.9.1. Field types

The following field types can be selected:

Type	Description
<b>Country</b>	Select a country from the list.
<b>Date</b>	Select a date by using the pull down menus, or the calendar icon.
<b>Date/Time</b>	Select a date and time by using the pull down menus.
<b>Email address</b>	Email address
<b>Floating point number</b>	A real number (that is, a number that can contain a fractional part). The following are floating-point numbers: 3.0, -111.5.
<b>Monetary amount</b>	Use a dot as separator.
<b>Number</b>	Number, no digits.
<b>Password</b>	Passwords are hidden when typing and are also not visible when viewing information. Type twice the same password, minimal 4 characters.
<b>Phone type</b>	Shows a pull down with all phone types. Select one.
<b>Sound prompt</b>	Select a sound prompt from a list of available sound prompts.
<b>Text</b>	Enter one line of text
<b>Text area</b>	Enter multiple lines of text
<b>Time</b>	Select a time by using the pull down menus.
<b>User</b>	Select a user from the list of employees.
<b>Yes / No</b>	You can select Yes or No.



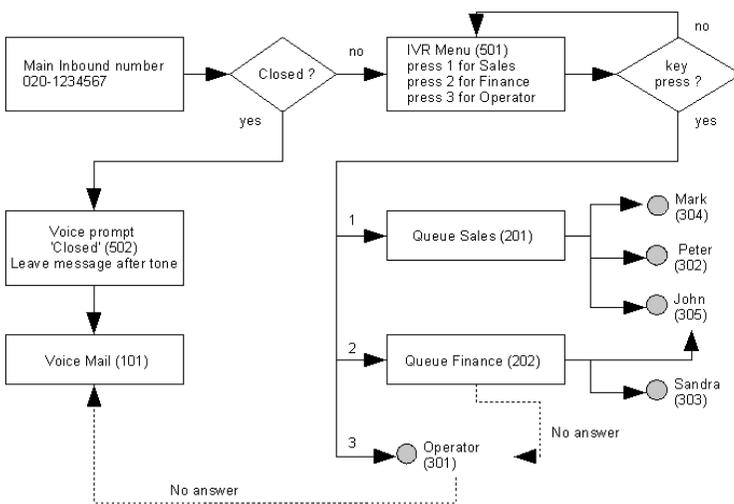
## **Chapter 3. The Dial Plan**

*This chapter describes how to (re)direct calls. You will learn how to configure queues, create Interactive Voice Response (IVR) menus and schedules. The basic elements of the previous chapter – departments, extensions, phones and users - are used to create the dial plan.*

## 3.1. Introduction

The dial plan is the most important part of your PBX. It describes how incoming and outgoing phone calls are handled. It includes inbound numbers, queues, redirect schedules, Interactive Voice Response (IVR) menus and many more.

All dial plan building blocks (items) have an extension<sup>56</sup> and can transfer phone calls to another extension. This way you can build a dial plan by connecting items to each other.



*Illustration 4: A simple dial plan with three digit extensions.*

Take a look at Illustration 4. Based on if our company is closed or not, a call is connected to a voice prompt with extension 502 (closed) or to an IVR menu with extension 501 (open). Based on the caller's choice, the call is connected to extensions 201 (sales queue), 202 (finance queue), 301 (operator phone) or 501 (itself if no choice is made).

A dial plan can contain many variables. It can be very helpful to draw a flow chart before continuing. Try answering the following questions:

<sup>56</sup> An extension is a three, four or five digit internal number, which is used to connect items to each other within a dial plan.

## Introduction

- Which inbound numbers do I want to use? What is the purpose of each of them?
- Do I need any IVR menus? Which IVR menus? What are the options a customer can choose from and what happens at every choice?
- Which departments need queues? Who are agent for which queues? What if all queue agents are on the phone? Do I want any music on hold? What kind of music?
- What should happen if an extension is busy or not answered? Should it be forwarded to another extension? Voice mail? Or must the call be rejected?
- At what times can customers call my company? What happens if the the company is closed? Are there any (recurring) days that we are closed?
- Are faxes processed by the PBX?

### 3.1.1. Everything has an Extension

Each dial plan item (menu, queue, phone, etc.) has an extension: a number defining the item. If you dial this extension, the item will be activated: a phone will ring or you will enter a queue. An IVR menu also has an extension. If you dial this extension you will hear the menu. When you press “1” the menu will connect you to the support queue with extension 201. And this queue will contact several phone extensions.

We start this chapter by creating all the items needed for the dial plan. Some items are already created by the Quick Setup. When we are done, all parts will fall into place. Now we have our incoming dial plan: the route of incoming phone calls.

At the end of the chapter (3.16) we create our outgoing dial plan. We tell the PBX how outgoing calls must be routed over which trunk.

## 3.2. Inbound Numbers

---

### 3.2.1. What is an Inbound Number?

When creating a new dial plan, inbound numbers is a good place to start. An inbound number is a phone number (provided by your telecom provider), which customers call. You cannot create inbound phone numbers yourself. For more information about your inbound numbers contact your telecom provider.

Examples of using inbound numbers:

1. Customers call the main inbound number: 020-1234567 and are connected to an IVR menu.
2. The support desk has an inbound number. Calls will be connected to a queue.
3. The CEO has his own inbound number, calls are directly connected to his phone.
4. A special inbound number (hot line) connects customers to the support queue with a higher priority. These calls will be placed on top of the queue.

Each inbound number is connected to an extension, meaning an inbound number can be connected to a phone, fax, queue, IVR menu, voice mail or any other dial plan item.

For each inbound number you can specify a first, second and third extension. When the first extension is not available, the Voclarion tries to connect the call to the second and so on.

#### 3.2.1.1. Inbound Number Priority

With priorities we assign a level of importance to an dial plan item like a call or schedule. More important calls are placed in front of less important calls in a queue.

## Inbound Numbers

On different places within the dial plan we can assign a priority to a call, for example on an incoming number. A call always contains the highest priority of all priorities assigned.

With priorities you can achieve some nice features like:

- Employees will be automatically placed in front of other callers in a queue.
- The maintenance crew on the road calls a special inbound number, which gives them a higher priority in the support desk queue.

Priority on inbound numbers can be used to give incoming calls on a specific number priority in queues. Several inbound numbers can connect to the same queue. Inbound number priority makes it possible to place calls from a selected inbound number (hotline) in front of other inbound calls. The default priority for all inbound numbers is '50'. Calls from an inbound number with a higher priority will be placed in front of other calls.

Priority range: 1 (highest priority) - 99 (lowest priority).

### 3.2.1.2. Adding an Inbound Number

#### Web interface menu

Quick Setup > Inbound Numbers

Companies > *Company* > Dial Plan > Inbound Numbers

On this page you can see an overview of all inbound numbers with their most important settings. Click on the number to see and change all settings. To add a new inbound number perform the following steps:

1. Click ADD (+) on the top right corner of your screen. The page to add a new incoming number will appear. Now provide the required information.

<b>Field</b>	<b>Description</b>
<u>PHONE NUMBER:</u>	Supply an inbound telephone number or a SIP URL. If the PBX detects a call from this number, it will connect the call to one of the given extensions. Wildcards "*" and "X" are allowed (see chapter 1.4.2.7) to group numbers.
<u>NAME:</u>	A name for this phone number. It can be something like "Main number" or "Support line".
<u>SHOW NAME ON DISPLAY:</u>	Show the name on the phone display on an incoming call. The outcome depends on the displays specifications. Besides the line name you can also display the company name <sup>57</sup> , queue name, a caller ID, caller name etc. If your display cannot hold enough characters, some information will be lost.
<u>QUEUE PRIORITY:</u>	Assign a priority (1 - 99) to callers. A priority can be useful when the call is connected to a queue. A call with priority "1" is more important than a call with priority "50" (default) and will be placed in front of a queue. The default priority is "0".
<u>LANGUAGE</u>	Set the language for calls from this number. Select a language from the list or select "Don't change language".
<u>GREETING PROMPT:</u>	Sound file played when entering. After this the caller will be connected to the first extension. Other items of the dialplan can have Greeting Prompts also.
<u>EXTENSION (1-3):</u>	Specify a first, second and third extension. If the first extension bounces the call <sup>58</sup> , the call will be connected to the second extension and so on <sup>59</sup> . You can pick an extension from the list of extensions or allocate one by pressing the <u>ADD</u> (+) button. If you would like the Direct Contact feature, use the first extension.

---

57 To save display space, you can add an agent sound prompt to a queue. This is a message played to the employee when answering a queue call. The message can be something like "Caller for MyComp". The voice prompt can not be heard by the caller. The prompt is set in the queue settings.

58 If a phone is not picked up and switches to voice mail, the PBX marks the call as "accepted", and will not bounce the call.

## Inbound Numbers

2. Click SAVE. You will return to the overview.

If you click on the inbound number you just created you'll see tabs appear on top of the screen, which provide additional settings:

1. LIST: List of all inbound numbers.
2. GENERAL INFORMATION: General information about this number, as described in this chapter.
3. SWITCHES: Add switches to this number.
4. REDIRECT SCHEDULES: Add schedules to this number.
5. HISTORY: See the history of changes for this line.

You will probably also notice that there are two new fields. You can set those fields to fine tune the inbound number.

Field	Description
<u>IDENTIFY CALLER BY ID</u> :	Identification by caller ID. <ul style="list-style-type: none"><li>• Activate this to use Direct Contact.</li><li>• For DISA; if set to "No" (default), a caller is obliged to enter a PIN when using DISA. If this field is set to "Yes", the PBX will compare the caller's ID with the mobile number as set in his personal information. If the caller ID matches, the caller will not be asked for a PIN and will be given access to DISA.</li></ul>
<u>INHIBIT CDR</u> :	If this is set to "Yes", all calls to this number will not be included in the Call Detail Records. Calls will not show up in reports and billing information (default is "No").

---

59 If your first extension is an IVR menu, it is not necessary to add a fall back extension. An IVR menu always accepts a call and never bounces it back to the dial plan.

## 3.2.2. Call IDs

When making a phone call, your phone company sends information about the call with it. In this paragraph we will discuss a few call IDs that might be important to you when setting up inbound phone numbers. We strongly suggest to read this chapter carefully when using non-geographical numbers.

### 3.2.2.1. Caller ID (CLID)

The caller ID (caller identification or more properly calling number identification) is a telephone service, available on POTS lines, that transmits a caller's number to the called party's telephone equipment during the ringing signal or when the call is being set up, but before the call is answered. Where available, the caller ID can also provide a name associated with it. The information available to the called party may be made visible on a telephone's display or on a separate attached device.

The concept behind caller ID is the value of informed consent [1].

### 3.2.2.2. Restrictions

When placing a phone call, the caller ID is sent to your phone company with each call. Your phone company now decides whether the given caller ID will be accepted or not, according to a white list and governmental rules. If the caller ID is rejected, the main phone number is used in most cases. Keep in mind that your PBX does not determine whether a caller ID is sent with a phone call or not, it only suggests a caller ID to your phone company.

In most cases a caller ID looks like an ordinary phone number, containing only digits. Your phone company will add a country code if needed. So you don't have to supply it. Contact your phone company for more information.

Most traditional phone companies only accept caller IDs provided by the phone company itself. Most SIP providers are less strict and will also accept other caller IDs like your mobile phone number for example. Contact your phone company for more information.

## Inbound Numbers

### 3.2.2.3. Direct Dial-In (DDI)

The Voclarion dial plan never uses the CLID, but always uses the DDI. Direct Dial-In (DDI), also called Direct Inward Dialing (DID), is a caller ID offered by phone companies for use with PBX systems. Thereby the phone company allocates a range of numbers all connected to their customer's PBX. As calls are presented to the PBX, the number the caller dialed is also given, so the PBX can route the call within the organization. This feature enables companies to have fewer lines than extensions, while still having a unique number for each extension, reachable from outside the company. The Voclarion dial plan always uses the DDI to forward calls.

Example: a company has an ISDN BRI connection with 2 phone numbers like 2125554500 and 21255545XX. With DDI the company can use more phone numbers, for example 2125554510 to 2125554525. Calls will be forwarded to one of the original phone numbers by the phone company, but because the DDI is set to the dialed phone number the PBX knows what number has been dialed and can forward the call accordingly.

### 3.2.2.4. Redirected Dialed Number Information Service (RDNIS)

Redirected Dialed Number Information Service (RDNIS<sup>60</sup>) contains the following services: Type of Number (TON), Presentation, and Redirecting Number (RGN), and Reason (busy, no answer, etc).

When redirecting calls, the Originally Called Number (OCN) contains, as its name suggests, the number that was originally called. The RGN identifies the telephone number redirecting a call.

### 3.2.2.5. Non-Geographical Numbers (NGN)

Phones and phone numbers are invented to be on one single geographical location. With Voice over IP, market demanded also non-geographical phone numbers. Non-geographical numbers are telephone numbers available for private sale which, rather than being assigned to a particular telephone line or circuit, provide callers with a contact number which gives no indication as to the

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60 A note about wireless phones: the 'forward all calls' feature does not necessarily send RDNIS, but 'forward when busy', 'forward when unavailable', and 'forward when unreachable' are supposed to send RDNIS. Functionality may depend on wireless provider more than PRI provider.

geographical location of the line being called. The owner of the number can re-target the NGN to any other telephone number including mobile, international and even other NGN numbers at any time. Thereby enabling them to take their calls on the move or at various locations at different times or simultaneously.

Non-geographical numbers are phone numbers that have no area code based on a physical location. For example in The Netherlands the code 088/085 is used. To function properly, NGNs are connected to a phone number with an area code, or in other words: an NGN is an alias for a location based phone number. However non-geographical numbers can be achieved using DDI, some telephony companies use RDNIS, which makes it a little more complex.

### 3.2.2.6. Adding Non-Geographical Numbers

With Voclarion you can use non-geographical phone numbers. However the dial plan will always use the DDI information. If your phony company sends RDNIS information instead of DDI, you might have to activate the feature USE RDNIS AS DDI to use the non-geographical number in your dial plan (see also example on page 141).

#### Web interface menu

Companies > *Company*

1. At the company settings page, set field USE RDNIS AS DDI to "Yes".

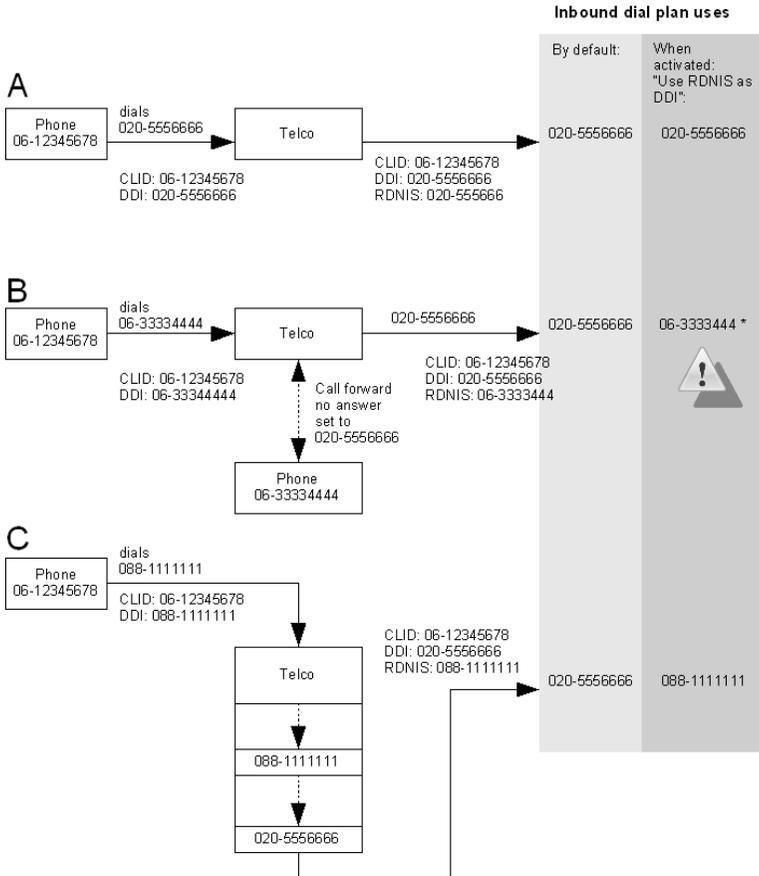
## **Inbound Numbers**



## Inbound Numbers



## Inbound Numbers



*Illustration 5: CID, DDI and RDNIS*

### 3.2.2.7. Examples of Call IDs

The illustration above shows three different calls (A – C). For each call we describe the IDs sent with this call.

- A. A phone with CLID 06-12345678 dials phone number 020-5556666. Logically both RDNIS and DDI are set to 020-5556666.
- B. A phone with CLID 06-12345678 dials phone number 06-33334444. This phone is busy and the call is forwarded to phone number 020-5556666 by

the telephony company. The telephony company sends the following information with the call: RDNIS: 06-33334444 and DDI: 020-5556666.

- I. When USE RDNIS AS DDI is disabled, the dial plan will use 020-5556666 as line number.
- II. When USE RDNIS AS DDI is enabled, the dial plan will use 06-3333444 as line number.



Please note that Voclarion will accept this call like it is originated from the second phone. This can lead to security issues.

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- C. A phone with caller ID 06-12345678 dials the non-geographical phone number 088-1111111. The telephony company uses RDNIS and sends RDNIS 088-1111111 to the original phone number 020-5556666.
  - I. When USE RDNIS AS DDI is disabled, the dial plan uses 020-5556666 as line number.
  - II. When USE RDNIS AS DDI is enabled, the dial plan uses 088-1111111 as line number.



Some phone companies allow you to block your caller ID by prefixing the dialed number with a special code. Contact your telephone company for more information.

---

### 3.2.2.8. Alarm caller ID Override

In the location settings you have to set an "Alarm caller ID". This caller ID is only used when calling the emergency number and will override all other caller ID settings. The Alarm caller ID is used to identify your location when calling emergency services.

### **3.2.3. caller name**

The caller name is a name you send with every outgoing call and is currently only supported in the USA. A caller name can have characters, digits and spaces.

### **3.2.4. caller name Priority**

Like a caller ID a caller name can be set on many locations within the dial plan. You can add a caller name to a company, but also to a user. When a user makes a phone call, which caller name will be used?

This is a case of priorities. Each caller name has its own priority. The trunk caller name has the highest priority, the company caller name the lowest. To determine which Name is send, the system uses the first Name it finds (top to bottom).

1. Trunk: caller name
2. Extension: caller name
3. User: caller name
4. Department: caller name
5. Company: caller name

When the caller ID is blocked, the caller name is also blocked. If you don't want to send a caller name with your outgoing calls, enter a "0" for the caller ID (see chapter 1.4.2.5).

## 3.3. Trunks

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Trunks are a group of phone lines which share the same characteristics and can carry multiple calls at the same time. Sometimes trunks are connected to hardware, like an ISDN telephony card. Trunks can have names like “ISDN lines”, “SIP provider X”, “Carrier select” or “GSM gateway”. Outgoing calls can be sent to a specified trunk by referring to it in the outbound dial plan. Trunks also accept incoming calls.

Trunks can be added to the Voclarion at two places: from the SYSTEM menu you can add System Trunks and from the COMPANY menu you can add Company trunks. System trunks can be used by all companies. Trunks that are configured within a company are only available to that particular company.

### 3.3.1. Adding Trunks

**Web interface menu**

*company specific trunks:*

Companies > *Company* > Dial Plan > VoIP Trunks or  
PSTN Trunks

*system wide trunks:*

System > Telephony > System VoIP Trunks or  
System PSTN Trunks

Quick Setup > Trunks

1. Go to one of the pages as listed above. The page shows a list of available system/company trunks.
2. Press ADD (+) to add a new trunk.
3. Select a type of trunk you wish to add:
  - Select PSTN trunks for a “DAHDI” legacy, 'old fashioned' telephony system hookup (usually ISDN).

- Select VoIP trunk for:
    - ✓ IAX to connect two or more Voclarions.
    - ✓ SIP to connect to SIP providers and Skype.
4. Click the Add (+) button to add the trunk.
  5. Depending on the trunk type, you will need to fill out several fields, as described in the next paragraphs.

### 3.3.2. PSTN Trunk

The DAHDI (Digium Asterisk Hardware Device Interface) trunk provides an interface layer between Asterisk and the DAHDI interface card, It connects your Voclarion to traditional digital and analog telephone equipment (PSTN). When adding a trunk, the Voclarion will ask you for all basic information. In most cases these settings are sufficient to make the trunk function. You can however fine tune the trunk by editing it.

<u>NAME:</u>	A name that has meaning to you, for example the provider's name.
<u>STATUS:</u>	Status icon, shows the current trunk status (see page 51). Hover for more information. This field is visible after saving the trunk.
<u>EXTENSION:</u>	The extension for this trunk <sup>61</sup> . Can be used to connect calls to this trunk. Only on company trunks.
<u>COUNTRY</u>	The country in which this trunk is located.
<u>REMOVE LEADING</u>	The number of digits that should be removed from the start of the dialed phone number. These digits will not be sent over the trunk. (to strip "00" for certain international providers, enter "2"). Default is "0".
<u>DIGITS:</u>	

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61 When a trunk cannot be reached, the call forwarding Malfunction will be followed.

## Trunks

<u>PREFIX FOR OUTGOING NUMBER:</u>	The digits that should be added at the beginning of the dialed phone number. (to replace a leading "00" with "99", enter "2" at <u>REMOVE LEADING DIGITS</u> and "99" at <u>PREFIX FOR OUTGOING NUMBER</u> ). Default is "".
<u>MAXIMUM OUTGOING CALLS:</u>	The maximum number of simultaneous outgoing calls through this trunk. Enter "0" to allow an unlimited number <sup>62</sup> of calls (default).
<u>MAXIMUM INCOMING CALLS:</u>	The maximum number of simultaneous incoming calls through this trunk. Enter "0" to allow an unlimited number <sup>62</sup> of calls (default).
<u>CALLER ID:</u>	caller ID for calls send over this trunk. Overwrites other caller ID's.
<u>CALLER NAME:</u>	Caller Name for calls send over this trunk. Overwrites other settings.

After you added a trunk, you'll return to the overview page. On top you'll see tabs to allocate trunk channels and ports. Both will be explained next.

### 3.3.2.1. Channel Allocation

Once telephony cards are installed, channels are set and trunks are created, you can assign voice channels to trunks. The number and type of installed telephony cards determines the number of available channels. For example Digium's dual span digital interface cards has 60 channels<sup>63</sup> (see illustration 6). You can make one simultaneous call (inbound or outbound) per channel. By assigning channels to multiple trunks, you can limit the number of simultaneous calls per trunk as shown on illustration 6<sup>64</sup>. As you can see you can combine channels from multiple telephony cards to one single trunk.

To set the channels, select a telephony card and click on the tab PBX CHANNELS. The following information is required:

---

62 Not restricted by Voclarion. Your phone company may restrict the number of simultaneous calls.

63 In an E configuration. A T1/J1 configuration has 48 channels.

64 This is an E1 configuration, used in most countries. When using a T1/J1 configuration (USA) less channels are available.

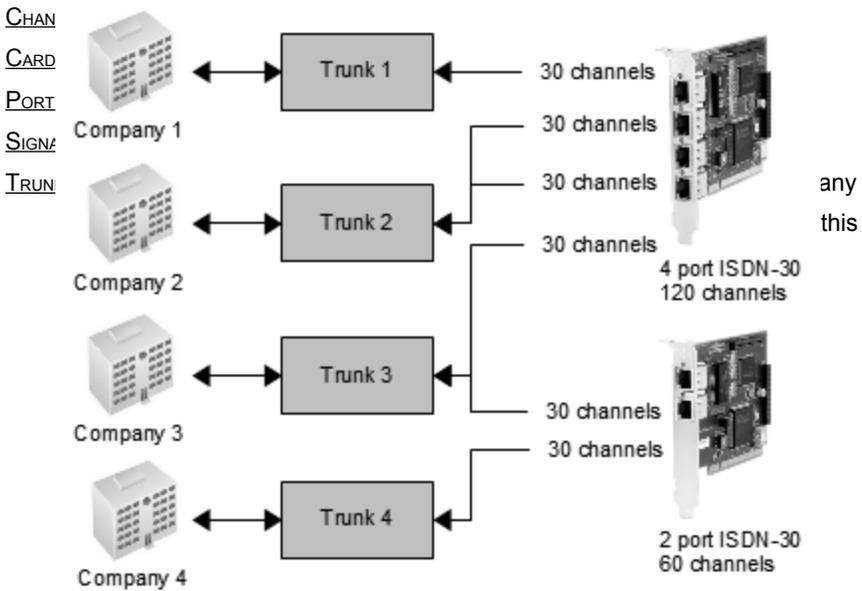


Illustration 6: Assigning channels to trunks

Press the SAVE button to save the settings.

You may have noticed one channel is missing from the list of available channels. This is the signaling channel (D-channel) used for sending data. On cards in an E1 configuration channel 16 is the signaling channel<sup>65</sup>, on a T1/J1 configuration it is the first channel.

### 3.3.2.2. Connected Ports

On the other side of the PSTN telephony card we have to connect phone lines. A telephony card can have one or more ports depending on the type of card. For example Digium's dual span digital interface cards support 48 (T1 / J1) or 60 (E1) channels over two ports (illustration 7).



Illustration 7:  
Dual span  
digital  
interface card

On the page mark each port/channel on which a phone line is connected. The Voclarion now knows which channels can be used.

<sup>65</sup> On an 30 channel ISDN telephony card channels will be numbered 1-31, with channel 16 missing.

## Trunks

<u>TELEPHONY CARD:</u>	The telephony card this channel is on.
<u>PORT:</u>	The port number of the card this channel is on.
<u>CHANNEL:</u>	Channel numbers (range).
<u>CONNECTED:</u>	Is this channel in connected to a phone line? Mark if the channel is connected.

Press the SAVE button to save the settings.

### 3.3.3. VoIP trunk: IAX

IAX is the Inter-Asterisk eXchange protocol native to Asterisk and supported by a number of other soft switches and PBXs. It is used to enable VoIP connections between servers as well as client-server communication. IAX is a VoIP protocol that carries both signaling and media on the same port, using a single UDP data stream (usually on port 4569) to communicate between endpoints, multiplexing signaling and media flow. IAX easily traverses firewalls and network address translators (NAT). This is in contrast to SIP, H.323 and MGCP which uses an out-of-band RTP stream to deliver information.

IAX also supports trunking, multiplexing channels over a single link. When trunking, data from multiple calls are merged into a single stream of packets between two endpoints, reducing the IP overhead without creating additional latency. This is advantageous in VoIP transmissions, in which IP headers use a large percentage of bandwidth. Both parties must support trunking.

When adding a IAX trunk, Voclarion will ask you for all basic information. In most cases these settings are sufficient to make the trunk function. You can however fine tune the trunk by editing it.

Field	Description
<u>NAME:</u>	Choose a name, for example your provider's name.
<u>STATUS:</u>	The current state of the trunk. Hoover for more information, see chapter 1.6.4. This field is visible after saving new trunk.
<u>EXTENSION:</u>	The extension for this trunk <sup>66</sup> . Can be used to connect calls to this trunk. Only on company trunks.
<u>COUNTRY:</u>	Location of the trunk.
<u>REMOVE LEADING DIGITS:</u>	The number of digits that should be removed from the start of the dialed phone number. These digits will not be sent over the trunk. (to strip "00" for certain international providers, enter "2"). Default is "0".

---

<sup>66</sup> When a trunk cannot be reached, the call forwarding Malfunction will be followed.

## Trunks

<b>Field</b>	<b>Description</b>
<u>PREFIX FOR OUTGOING NUMBER:</u>	The digits that should be added to the beginning of the dialed phone number (to replace a leading "00" by "99", enter "2" at <u>REMOVE LEADING DIGITS</u> and "99" at <u>PREFIX FOR OUTGOING NUMBER</u> ). Default is "".
<u>MAXIMUM OUTGOING CALLS:</u>	The maximum number of simultaneous outgoing calls through this trunk. Enter "0" to allow an unlimited <sup>67</sup> number of calls (default).
<u>MAXIMUM INCOMING CALLS:</u>	The maximum number of simultaneous incoming calls through this trunk. Enter "0" to allow an unlimited number <sup>67</sup> of calls (default).
<u>CALLER ID:</u>	Caller ID for calls send over this trunk. Overwrites other settings.
<u>CALLER NAME:</u>	Caller Name for calls send over this trunk. Overwrites other settings.
<u>TRUNKING:</u>	Use trunking on an outgoing call? Trunking saves bandwidth. Default is set to "No". Only change this if your provider supports trunking, otherwise the connection will fail. Contact your provider for more information.
<u>IP ADDRESS:</u>	The IP address of the other party.
<u>PORT:</u>	The port number to connect to the remote host (default: 4569)
<u>CODEC 1-4:</u>	When a call is setup, both sides will negotiate on which codec to use. The first codec known by both systems will be used <sup>68</sup> . It is recommended to select the same codec for both phones and trunk.

---

67 Not restricted by Voclarion. Your phone company may restrict the number of simultaneous calls.

68 G711-μlaw (default) has the best quality, but uses the most bandwidth. GSM has the worst quality, but uses far less bandwidth.

Field	Description
<u>QUALIFY:</u>	Trunk's maximum response time in milliseconds. Every few milliseconds Voclarion checks if the other side is reachable, comparable to 'pinging' in an IP network. If no response is received, the trunk is marked 'unreachable' and will be skipped for outgoing calls. You can increase the default response time if the other end is located far away. Set to "0" to disable the regular checking. This is used only for outgoing calls.
<u>AUTHENTICATION:</u>	Encryption method of authorization used for outgoing calls. Check <u>PLAIN TEXT</u> for no encryption, or choose one of the encryption methods. Default: "MD5".
<u>USER NAME:</u>	User name for authenticating outgoing connections.
<u>PASSWORD:</u>	Password or certificate for outgoing connections.
<u>PEER CONTEXT:</u>	If the remote host requires a context, supply it here.
<u>ENTRY NAME:</u>	The name of the configuration block in the Asterisk sip.conf-file. Will be provided by your SIP provider. Almost never used.
<u>REGISTER:</u>	Whether the PBX needs to register with the SIP provider. Set <u>REGISTER</u> only to "Yes" if the other site requires it or if your PBX has a variable public IP address (not supported).
<u>INCOMING AUTHENTICATION:</u>	Authorization method used for incoming calls. Select "plain text" for no encryption, or choose one of the encryption methods. Default: "MD5".
<u>INCOMING USER NAME:</u>	User name the other party must use for sending you calls.
<u>INCOMING PASSWORD:</u>	Password for the incoming user required for incoming calls.
<u>TRANSLATE INCOMING NUMBERS:</u>	Way of translating incoming numbers. If you translate numbers, you have to use the translated format within your dial plan. <ul style="list-style-type: none"> <li>• no conversion (default)</li> <li>• national format</li> <li>• ITU format</li> </ul>

## Trunks

Field	Description
<u>SEND INCOMING CALLS OUT:</u>	If a call comes in over this trunk and the dialed number is not recognized by the dial plan, send the call out over an outbound trunk. Can be used to sent out calls over another Voclarion.
<u>ALLOW CALLS TO EXTENSIONS:</u>	If a call comes in over this trunk and the dialed number is not recognized by the dial plan, try to call an internal extension directly?
<u>NOTES:</u>	Any notes that might be helpful to you.

### Example: Connecting Two Systems Using IAX

What we would like to achieve:

- When dialing an extension from 1000 till 1009 from Voclarion A, the call must be connected to the extension located on Voclarion B.

Configuration:

- Make sure there are no existing extensions in the range 1000-1009 present on Voclarion A; this range must only be configured on Voclarion B.
- To connect both Voclarions, create an IAX trunk on both Voclarions. For each trunk supply the IP address of the other Voclarion. To authenticate, also supply a user name and a password. These have to be similar for the trunk on both systems.

<b>Web interface menu</b>	Companies > <i>Company</i> > Dial Plan > Outbound Call Routing
	Companies > <i>Company</i> > Dial Plan > Trunks

1. Create an IAX trunk for both Voclarions and set the following information:
  - IP ADDRESS: The IP address of the Voclarion you want to connect to.
  - USERNAME AND PASSWORD: enter a user name and password, these must be the same for both systems.
2. Create an outbound call route (see chapter 3.16) on Voclarion **A**:

- TARGET: “100X” (any number from 1000 to 1009).
- TRUNK: select the IAX trunk you just created.

*Make sure there are no extensions present in the range 1000-1009 on Voclarion A.*

3. Configure for Voclarion **B** extensions 1000-1009.



Alternatively, you can dial the given extensions with a prefix like “91000”, so it is still possible to use the 1000-1009 extension range on Voclarion **A**. With outbound call routing (see chapter 3.16) you can send numbers starting with 9 to the other Voclarion.

---

### 3.3.4. VoIP trunk: SIP

A SIP trunk is a service offered by an ITSP (Internet Telephony Service Provider like Skype<sup>69</sup>) that permits to use Voice over IP (VoIP) calls by using a internet connection. When adding a trunk, Voclarion will ask you for all basic information. In most cases these settings are sufficient to make the trunk function. You can however fine tune the trunk by editing it.

Supply the following information:

<b>Field</b>	<b>Description</b>
<u>NAME</u> :	Choose a name, for example your provider.
<u>STATUS</u> :	The current state of the trunk. Hoover for more information, see chapter 1.6.4. This field is visible after saving the trunk.
<u>EXTENSION</u> :	The extension for this trunk <sup>70</sup> . Can be used to connect calls to this trunk. Only on company trunks.
<u>COUNTRY</u> :	Location of the trunk.
<u>RATE PLAN</u> :	Select a rate plan suitable for this trunk. Only available when Billing is enabled.

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<sup>69</sup> To use Skype you need a Skype Connect business account. For more information visit [www.skype.com](http://www.skype.com)

<sup>70</sup> When a trunk cannot be reached, the call forwarding Malfunction will be followed.

## Trunks

<b>Field</b>	<b>Description</b>
<u>REMOVE LEADING DIGITS:</u>	The number of digits that should be removed from the beginning of the dialed phone number. These digits will not be sent over the trunk. (to strip "00" for certain international providers, enter "2"). Default is "0".
<u>PREFIX FOR OUTGOING NUMBER:</u>	The digits that should be added at the beginning of the dialed phone number. (to replace a leading "00" by a leading "99", enter "2" at <u>REMOVE LEADING DIGITS</u> and "99" at <u>PREFIX FOR OUTGOING NUMBER</u> ). Default is "".
<u>MAXIMUM OUTGOING CALLS:</u>	The maximum number of simultaneous outgoing calls through this trunk. Enter "0" to allow an unlimited number <sup>71</sup> of calls (default).
<u>MAXIMUM INCOMING CALLS:</u>	The maximum number of simultaneous incoming calls through this trunk. Enter "0" to allow an unlimited number <sup>71</sup> of calls (default).
<u>CALLER ID:</u>	Caller ID for calls send over this trunk. Overwrites other settings.
<u>CALLER NAME:</u>	Caller Name for calls send over this trunk. Overwrites other settings.
<u>IP ADDRESS:</u>	The IP address of the other party.
<u>PORT:</u>	The port number to connect to the remote host (default: 5060)
<u>USE NAT:</u>	Should this trunk support Network Address Translation (NAT)? Default: No.
<u>CODEC 1-4:</u>	When a call is setup, both sides will negotiate on which codec to use. The first codec known by both systems will be used <sup>72</sup> . It is recommended to select the same codec for both phones and trunk.

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<sup>71</sup> Not restricted by Voclarion. Your phone company may restrict the number of simultaneous calls.

<sup>72</sup> G711-μlaw (default) has the best quality, but uses the most bandwidth. GSM has the worst quality, but uses far less bandwidth.

<b>Field</b>	<b>Description</b>
<u>QUALIFY:</u>	Trunk's maximum response time in milliseconds. Every few milliseconds Voclarion checks if the other side is reachable, comparable to 'pinging' in an IP network. If no response is received, the trunk is marked 'unreachable' and will be skipped for outgoing calls. You can increase the default response time if the other end is located far away. Set to "0" to disable the regular checking. This is used only for outgoing calls.
<u>AUTHENTICATION:</u>	Encryption method of authorization used for outgoing calls. Check <u>PLAIN TEXT</u> for no encryption, or choose one of the encryption methods. Default: "MD5".
<u>FROM USER:</u>	Occasionally required by a SIP provider. Contact your provider for more information.
<u>USER NAME:</u>	User name for authenticating outgoing connections.
<u>PASSWORD:</u>	Password or certificate for outgoing connections.

## Trunks

### Field

DMTF MODE ON

OUTGOING CALLS:

### Description

Mode of sending/receiving DTMF tones on outgoing calls. DTMF is used to send user input to/from the Voclarion and other systems, for example to make a choice from an IVR menu. Tones are generated when a user presses keys on the dial pad.

- Inband means the DTMF is transmitted within the audio of the phone conversation, it is audible to the conversation partners. Therefore only uncompressed codecs like G.711 alaw or µlaw can carry inband DTMF reliably. Female voice are known to once in a while trigger the recognition of a DTMF tone. For analog lines inband is the only possible means to transmit DTMF.
- SIP INFO, out of band; DTMF tones are send outside the audio of the phone conversation. Only available with SIP channels, transmitted through a SIP message.
- RFC 2833, out of band; DTMF tones are send outside the audio of the phone conversation (default).

REGISTER:

Whether the PBX needs to register with the SIP provider. Set REGISTER only to “Yes” if the other site requires it or if your PBX has a variable public IP address (not supported).

INCOMING EXTENSION:

Sometimes a SIP provider cannot send a phone number with a call. To identify this call, you can supply an Incoming Extension. Now the call is identified so you can route calls from this trunk trough your dial plan. The Voclarion tries to identify a call by the first value it finds (top to bottom):

1. Incoming extension (this trunk)
2. Caller ID (this trunk)
3. Company phone number

Field	Description
<u>INCOMING AUTHENTICATION:</u>	Authorization method used for incoming calls. Select “plain text” for no encryption, or choose one of the encryption methods. Default: “MD5”.
<u>INCOMING_USER_NAME:</u>	User name the other party must use for sending you calls.
<u>INCOMING_PASSWORD:</u>	Password for the incoming user required for incoming calls.
<u>DTMF_MODE_ON:</u>	Mode of sending/receiving DTMF tones on incoming calls.
<u>INCOMING_CALLS:</u>	<p>DTMF is used to send user input to/from the Voclarion and other systems, for example to make a choice from an IVR menu. Tones are generated when a user presses keys on the dial pad.</p> <ul style="list-style-type: none"> <li>• Inband means the DTMF is transmitted within the audio of the phone conversation, it is audible to the conversation partners. Therefore only uncompressed codecs like G.711 alaw or <math>\mu</math>law can carry inband DTMF reliably. Female voice are known to once in a while trigger the recognition of a DTMF tone. For analog lines inband is the only possible means to transmit DTMF.</li> <li>• SIP INFO, out of band; DTMF tones are send outside the audio of the phone conversation. Only available with SIP channels, transmitted through a SIP message.</li> <li>• RFC 2833, out of band; DTMF tones are send outside the audio of the phone conversation (default).</li> </ul>
<u>TRANSLATE_INCOMING_NUMBERS:</u>	<p>Way of translating incoming numbers. If you translate numbers, you have to use the translated format within your dial plan.</p> <ul style="list-style-type: none"> <li>• no conversion (default)</li> <li>• national format</li> <li>• ITU format</li> </ul>

## Trunks

Field	Description
<u>ROUTE_ON_TO_IN_SIP_HEADER:</u>	On incoming calls, take the destination from the TO header or from the Request-URI header?
<u>WILL_REGISTER_WITH_US:</u>	Will the SIP provider register with us? This is required by some SIP providers.
<u>RELAX_INCOMING_SECURITY:</u>	Select for less tight security. <ul style="list-style-type: none"><li>• No: disabled, tight security. (default)</li><li>• Port: Does not require matching ports for incoming requests.</li><li>• Invite: Does not require authentication for incoming invites.</li><li>• Both: Port and Invite.</li></ul>
<u>SEND_INCOMING_CALLS_OUT:</u>	If a call comes in over this trunk and the dialed number is not recognized by the dial plan, send the call out over an outbound trunk. Can be used to sent out calls over another Voclarion.
<u>ALLOW_CALLS_TO_EXTENSIONS:</u>	If a call comes in over this trunk and the dialed number is not recognized by the dial plan, try to call an internal extension directly?
<u>NOTES:</u>	Any notes that might be helpful to you.

## 3.4. Pickup Groups

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A call pickup group is a group of users, which can answer each others ringing phone. Departments can function as pickup group as well as self defined groups. A person can be part of several groups.

### 3.4.1. Using Departments as Pickup Groups

You can use a department as pickup group. All members within the department are automatically member of this pickup group. To do this, enable this option from the company settings. Once active, each department within the company will also function as a pickup group.

**Web interface menu**      Companies > *Company*

1. Set field USE DEPARTMENTS AS PICKUP GROUPS to “Yes”.
2. Click the SAVE button.

### 3.4.2. Creating Pickup Groups

**Web interface menu**      Companies > *Company* > Dial Plan > Pickup Groups

On the page above you'll see an overview of all available pickup groups.

1. Click the ADD (+) icon to add a new group.
2. Enter a name for the pickup group and click the SAVE button. A new group is created and you'll be taken back to the list of available groups.
3. Click on the new pickup group. Click the MEMBERS tab to add members.

4. Click the EDIT icon.
5. All company members will be shown, sorted by department. Mark the check box behind each person you want to add.
6. Click the SAVE button.

### 3.4.3. Answering a Phone Call

To answer a colleague's ringing phone press \*8. To answer a phone of a specific colleague, press \*8<extension\_of\_colleague>.

## 3.5. Redirect Schedules

---

Some parts of your dial plan are time dependent. Every working day at 17:00 the office closes and the PBX has to play a message asking the caller to call back tomorrow. At the same time the support desk closes and all calls have to be directed to an external call center. Each year on Christmas the PBX has to play a message wishing your customers all the best and tell them to call back after Christmas. All this is done by a redirect schedule. Redirect schedules perform an action on a regular time interval.

After creation, redirect schedules are assigned to an inbound phone number or extension, so calls can be forwarded based on the time schedule. The actions you need to take:

1. Create a redirect schedule.
2. Specify start/end time, date and frequency for this schedule.
3. Select an extension to connect calls with when the schedule is active.
4. Assign the schedule to one or more inbound numbers or extensions.

### 3.5.1. Redirect Schedule examples

Below you'll find some example schedules to show you how schedules work. Later on we'll describe all settings in detail.

#### 3.5.1.1. Office Closing Hours

The office working hours are from Monday-Friday 09.00 – 17.00. When closed, calls have to be connected to voice mail (2000).

#### Configuration

1. Create two redirect schedules.

*Name:* Closed-Weeknights  
*Extension:* 2000 (voice mail)  
*Interval:* Every week (Mon-Tue-We-Thu-Fri)  
*Interval Frequency:* 1

*Start Time:* 17.00  
*End Time:* 09 00  
*Date:* The start date for this schedule  
*End Date:*

*Name:* Closed-Weekend  
*Extension:* 2000 (voice mail)  
*Interval:* Every week (Sa-Su)  
*Interval Frequency:* 1  
*Start Time:* 00.00  
*End Time:* 23 59  
*Date:* The start date for this schedule  
*End Date:*

### 3.5.2. New Year's Day

The office is closed on New Year's day. When a customer calls the office, a message will be played.

#### Configuration

1. Create the following redirect schedule.

*Name* New Year's day  
*Greeting prompt:* "Closed"  
*Interval:* Every year  
*Interval Frequency:* 1  
*Start Time:* 00.00  
*End Time:* 23.59  
*Date:* January 1st  
*End Date:*

### 3.5.3. Various office hours

The main number connects to an IVR menu (extension 2000) with 2 options:

1. Sales (extension 2010)
2. Service desk (extension 2020)

Both departments have different office hours:

- Sales Mo – Fr 09.00 – 17.00. Outside these hours calls have to be connected to voice mail (extension. 2001).
- Servicedesk Mo-Su 09.00 - 20.00. Outside these hours a message is played and the call is ended.

#### Configuration

*Name:* Sales -Closed-Weekdays  
*To Extension:* 2001  
*Interval:* Every week (Mo-Tue-We-Thu-Fri)  
*Interval Frequency:* 1  
*Start Time:* 17.00  
*End Time:* 09 00  
*Date:* The start date for this schedule  
*End Date:*

*Name:* Sales -Closed-Weekend  
*To Extension:* 2001  
*Interval:* Every week (Mo-Tue-We-Thu-Fri)  
*Interval Frequency:* 1  
*Start Time:* 17.00  
*End Time:* 09 00  
*Date:* The start date for this schedule  
*End Date:*

*Name:* Servicedesk -Closed  
*Greeting prompt:* "Closed".  
*To Extension:*  
*Interval:* Every day  
*Interval Frequency:* 1  
*Start Time:* 20.00  
*End Time:* 09 00  
*Date:* The start date for this schedule  
*End Date:*

### 3.5.4. Creating Redirect Schedules

**Web interface menu** Companies > *Company* > Dial Plan > Redirect Schedules

1. To add a new schedule, click ADD (+).
2. Fill out all required fields. Depending on your choice, the options will change.

NAME: Name for this schedule.  
DESCRIPTION: Description for the schedule.  
GREETING: A sound prompt played when the schedule is active.  
PROMPT:  
EXTENSION (1-3): When the schedule is active, calls will be connected to the first extension. If this extension is unavailable, the call will be connected to the second and so on<sup>73</sup>. If no extension is specified, the call will be ended.  
INTERVAL: The schedule interval. This is the time between two schedule activations (see chapter 3.5.5).

---

<sup>73</sup> To connect calls to an external phone number, supply an extension which is forwarded to the external phone number. For more information about call forwarding see chapter 10.8.

## Redirect Schedules

<u>INTERVAL</u>	The frequency of the chosen interval. When no interval is required, set <u>INTERVAL</u> to None.
<u>FREQUENCY</u> :	
<u>START/END TIME</u> :	The start and end time of the schedule. If the schedule has to be active the whole day, use start time: 0:00, end time: 23:59.
<u>TIME_ZONE</u> :	Select the time zone for this schedule.
<u>DATE</u> :	Date of activation.
<u>END DATE</u> :	The date the schedule ends. The end date is default set to No End Date. If an end date is needed, unmark the check box. A field for end date will appear in the page. If 'No End Date' is selected, this schedule is valid forever. The end date is the last day the schedule is active.

3. Click SAVE.

### 3.5.5. Interval and Interval frequency

The interval of a redirect schedule can be described as the time between two activations. If you want to use a schedule only once, choose "none". Choose between the next intervals. Depending on your choice, the screen layout will change.

- **None:** the schedule is only active on the day as set for START DATE. No frequency can be given.
- **Every day:** the schedule is active every day. You can choose this when the schedule has to be daily active for a short period of time.
- **Every week:** if you choose every week, a set of extra options will appear. Select for which days (Mon-Sun) the schedule has to be active.
- **Every month:** If you choose every month, you can choose to repeat the schedule by day (every first Monday) or by date (every 5<sup>th</sup> of December).
- **Every year:** the schedule is active every year.

If an interval is selected, you can also set a frequency for this schedule. See the next section for examples.

### 3.5.5.1. Examples of Redirect Schedule Intervals

#### Every Day

- If the interval is set to 'Every day', and the frequency is 1 the schedule runs every day.
- If the frequency is set to 2, the schedule runs every other days.

#### Every Week

- If the frequency is 3, the schedule runs every three weeks.
- For schedule 'weekend', select Saturday and Sunday.
- For schedule 'weekday', you can select Monday, Tuesday, Wednesday, Thursday and Friday.

#### Every Month

- If the interval is set to 'Every month' and the frequency is set to 3, this schedule repeats every three months.
- Suppose the chosen date is January 25, 2009 and the frequency is set to '1'. The next day this schedule becomes active is February 25, 2009.
- Suppose the chosen date is January 25, 2009, and the frequency is set to "1". This day is the third Wednesday of January. The next time this schedule becomes active is at the third Wednesday of February.

#### Every Year

- If the frequency is set to "1", this schedule repeats every year. This option can be used for holidays such as New Year's day.

## 3.5.6. Assigning Redirect Schedules

After creating a schedule, it must be assigned to an inbound number or an extension. You can assign as many schedules as you like. Sometimes schedules overlap in time. For example schedule 'Weekend' and 'Easter'. What happens on Easter Sunday? Which schedule will be active? To solve this problem we created schedule priority. More about this in chapter 3.5.7.

1. Click on an inbound phone number or extension.
2. Click the [REDIRECT SCHEDULE](#) tab.

## Redirect Schedules

### Web interface menu

Companies > *Company* > Dial Plan > Inbound Numbers

Companies > *Company* > Extensions

3. Choose the redirect schedule(s) and mark the check box. If multiple schedules are overlapping, choose a priority number from 1- 99 (see chapter 3.5.7).
4. Select 'ACTIVATE THE SCHEDULES SELECTED BELOW'.
5. Click SAVE.



To see on which extension or inbound number a redirect schedule is used, go to the schedule and click on the USAGE tab.

---



Expired Redirect Schedules disappear from the Redirect Schedule tab. Schedules with no end date stay visible.

---

### 3.5.7. Redirect Schedule Priority

Sometimes schedules overlap in time. For example schedule 'Weekend' and 'Christmas'. What happens when Christmas is on a Sunday? Which schedule is active? To solve this problem we created a schedule priority. You can assign a priority (1 – 99) to each schedule. The schedule with the *lowest number* has the highest priority and will be used. In this case you probably want to use the schedule 'Christmas' announcing the company is closed until after Boxing day, instead of using schedule 'Weekend' announcing people to call back on Monday.

- Highest priority = priority number 1
- Lowest priority = priority number 99

**Web interface menu**      Companies > *Company* > Dial Plan > Inbound  
 Numbers > *Phone Number* > Redirect Schedule tab

### 3.5.7.1. Example

For your convenience we show you an example with the schedules called 'Closed', 'Weekend' and 'Christmas'.

- “Closed” is activated daily from 18:00 until 9:00. It plays a message like “Sorry, we are closed right now. You can reach us on working days from 9:00 till 18:00”.
- “Weekend” is activated from Friday 18:00 until Monday 9:00, every week. It plays a message like “Sorry we are closed right now. Please call back on Monday after 9:00”.
- “Christmas” is activated from Friday 18:00 (24 December) until Tuesday 9:00 (27 December), because we are also closed on Boxing Day Monday. It plays a message like “Sorry we are closed right now. Our office is opened after Christmas”.

The schedule “Closed” has to run every day. Except for every weekend: the “Weekend”-schedule has to be active in the weekend. Except on the weekend of 24 December. The “Christmas”-schedule has to be active at this time.

In this case, you assign the schedule “Closed” a default priority of “50”. Schedule “Weekend” is assigned a higher priority, let's say “10”. And you give “Christmas” the highest priority: “1”.

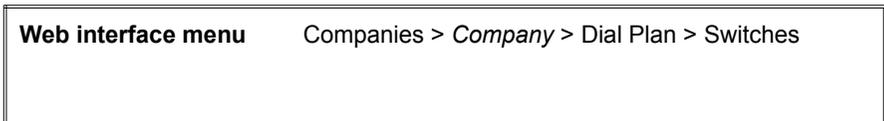
## 3.6. Switch

---

Redirect schedules are used to redirect calls automatically, based on time. With switches you can change the flow of calls manually. Switches also can be used to manually (de)activate a redirect schedule. They can be programmed on a telephone function key and can be manually activated or deactivated. Switches (in contrast to Call Forwarding on an extension) redirect all calls from an extension or inbound number.

### 3.6.1. Add switches

To add (or change) a switch, go to the following page. The page shows a overview of all available switches for this company.



1. Click on the ADD (+) button to add a new switch, or click on the switch' name to change settings. The switch settings are explained below.
2. When finished click SAVE to create the new Switch.

Field	Description
<u>NAME:</u>	Name of the switch.
<u>EXTENSION:</u>	Choose an extension for this switch from the list or add a new one using the <u>ADD (+)</u> -icon.
<u>DESCRIPTION:</u>	Description for this switch.
<u>POSITION:</u>	Set the position of the switch to on/off. This value shows the current setting and will change according to the use of the switch.

After saving you'll be taken back to the overview page. Now select the switch you just created to make some additional changes.

Field	Description
<u>VERRIDE:</u>	Can the switch position as set by a redirect schedule be manually overwritten? See chapter 3.6.4 for information about connecting switches to redirect schedules.
<u>GREETING PROMPT:</u>	This sound prompt is played to a caller when the switch is on (active).
<u>XTENSION (1-3):</u>	If the switch is on (active) calls are connected to the first extension <sup>74</sup> . If the first extension does not accept the call (bounce), the call is connected to the second extension and so on <sup>75</sup> . Special *-codes are accepted.

### 3.6.2. Assign a switch

Now we created a switch, we have to assign it to an extension or an incoming number. When we flip the switch all calls from this incoming number (or extension) will be redirected to the extension as set on the switch. You can assign as many switches as you like.

<b>Web interface menu</b>	Companies > <i>Company</i> > Dial Plan > Inbound Numbers
	Companies > <i>Company</i> > Extensions

1. Click on an inbound phone number or extension.
2. Click the SWITCHES tab.
3. Select the switch(es) by marking the check box.
4. Assign an priority (1-99). If multiple switches are overlapping, the switch with the highest priority (lowest number) will be followed.
1. Select 'ACTIVATE THE SWITCHES SELECTED BELOW'.
2. Click SAVE to activate.

<sup>74</sup> This overrules the redirect schedule settings, see chapter 3.6.4.

<sup>75</sup> To connect the call to an external phone number, connect the call to an extension which is forwarded to the external phone number.

### 3.6.3. Flip the switch

To flip a switch - (de)activate it – simply call the extension of the switch from a phone within the company. The Voclarion will respond verbally with the new position (on / off). The extension can be programmed as a function key (chapter 2.5.2). The switch can now be used by pressing the function key and the BLF will indicate the position of the switch<sup>76</sup>. If the BLF is on, the switch is active and calls will be redirected according to the settings on the switch.



Switches take precedence over redirect schedules (chapter 3.5).

---

The current position of a switch can also be viewed from the web interface (switch overview page). The tab HISTORY will show all changes.

### 3.6.4. Connect a redirect schedule to a switch

We now have a working switch. If you like you can connect a redirect schedule to a switch. This can be used for switching the dial plan to 'night settings' both automatically and manually.

When connected the switch will follow the redirect schedule's time window, but can still be switched manually. When the redirect schedule becomes active, the position of the switch is set to on. When the redirect schedule becomes inactive, the position of the switch is set to off. Keep in mind that the switch always follows its own redirect settings, *not the redirect settings of the schedule*.

1. Go to the switch and click on the tab REDIRECT SCHEDULE.
2. Select ACTIVATE THE SCHEDULES SELECTED BELOW.
3. Select the schedules you wish to connect to the switch and assign priorities (1-99) as described on chapter 3.5.

---

<sup>76</sup> When activating the switch the BLF and the verbal response will instantly change. However it can take up to a minute before the switch is actually activated.

4. Press SAVE to activate the changes.

The switch will now follow the schedule's time window, but also can be (de)activated manually.



- Switch forward settings take precedence over redirect schedules.
  - The switch always follows it's own redirect settings, not the redirect settings of the connected schedule.
- 

### 3.6.4.1. Example

The connection between a redirect schedule and a switch can be a bit tricky. This (simplified) example will clarify things.

<b>Incoming number</b>		
number	1 <sup>st</sup> extension	2 <sup>nd</sup> extension
123456789	100	101

<b>Redirect schedule</b>		
start	end	1 <sup>st</sup> extension
- 18.00	- 09.00	200

<b>Switch</b>		
Extension switch	override	1 <sup>st</sup> extension
400	yes	300

## Switch

<b>History</b>			
<i>time</i>	<i>action</i>	<i>Switch position</i>	<i>Calls forwarded to</i>
- 08.00		ON	300
- 09.00		OFF	100
- 12.00	Dial 400	ON	300
- 13.00	Dial 400	OFF	100
- 18.00		ON	300

## 3.7. Music on Hold

---

When a caller is put on hold or has to wait in a queue, the Voclarion can play music. This music can be a set of MP3<sup>77</sup> or WAV files. The Voclarion offers a default set of music<sup>78</sup>, but you can upload your own files if you like. For each queue you can set a different music category. Voclarion offers three default kinds of music on hold:

- Default<sup>79</sup>: set of pre-installed free music.
- Ringing: the sound you hear when trying to call someone.
- Beepbeep: the sound you hear when you are put on hold.

It is advisable to play some kind of sound when a call is put on hold. If a caller hears no sound at all, he might think something has gone wrong and disconnects the call. If you do not wish to play music, we suggest "Beepbeep" for company music on hold and "Ringing" for queue music on hold.

When the first caller is put on hold, the music will start. When no callers are on hold, the music will be paused to enhance system performance. When a caller is put on hold again, the music will start where it left.

When you have little callers on hold during the day, keep this in mind:

- If you select long songs, people who are calling several times a day may hear the same music. We suggest you to select only short songs.

---

77 MP3's use a lossy compression algorithm is designed to greatly reduce the amount of data required to represent the audio recording and still sound like a faithful reproduction of the original uncompressed audio for most listeners. A MP3 file that is created using the mid-range bit rate setting of 128 kbit/s will result in a file that is typically about 1/10th the size of the CD file created from the original audio source [2].

78 In some countries you have to pay a copyright fee and/or a fee for playing music on a public PBX. The music included is free of copyrights, however you might have to pay for using the music on a public PBX. This fee is not included in your purchase. Visit <http://www.biem.org/> for more information.

79 Cannot be changed or deleted.

### 3.7.1. Activating Music on Hold

Music on hold can be added to a company and a queue. The company music on hold is the music played when a caller is put on hold or is waiting in a conference box. Queue music on hold is played when a caller enters a queue.

We added a default set of music on hold<sup>80</sup>. The default set music on hold contains the following music:

1. "Promise", performed by SlicerXXL
2. "Pure love", performed by SC
3. "Sign from above", performed by Nick Edelstein
4. "Turn to Gold", performed by Andycole
5. "Wonderful Again", performed by Corey Werbb
6. "Home", performed by DenniA
7. "I found you", performed by Molo
8. "Over and over", performed by Jon Roniger
9. "Please", performed by Johnny Rei
10. "Popeline", performed by Diane Tell

### 3.7.2. Uploading Music Files

Before uploading music files it is important you check the file size, bit rate and the frequency of your music file. Most common phone lines are mono and in most cases Voclarion uses codecs designed for speech, so your music doesn't have to be recorded in the highest quality. Using less quality files uses less bandwidth, enhances system performance and will save disk space. We suggest you use MP3 files with a file size of 5 MB or less and a bit rate of 128 Kb/s or less.

**Web interface menu**      Companies > *Company* > Dial Plan > Music on Hold

1. Go to the page above. You see a list of all available music.
2. Click [Add](#).

<sup>80</sup> This music (and more) is available from [tribeofnoise.com](http://tribeofnoise.com)

## Music on Hold

3. Click on the 'browse' button and select a MP3-file.
4. Enter a category name for this file. Music files are grouped by category.
5. Click SAVE.



- Music files are played in a random order.
  - If available the PBX will show the MP3 meta tags in the music on hold overview.
- 

### 3.7.3. Deleting Music Files

**Web interface menu**      Companies > *Company* > Dial Plan > Music on Hold

1. Go to the page above. You see a list of all available music.
2. Click on a music name to see details.
3. Click on the speaker icon to listen to the music file<sup>81</sup>.
4. Click on the DELETE (X) button on top of the page to delete the file.

---

<sup>81</sup> If you do not hear music or if you see an error message, your browser does not have the proper software installed to play music files. Consult the documentation of your browser for more information.

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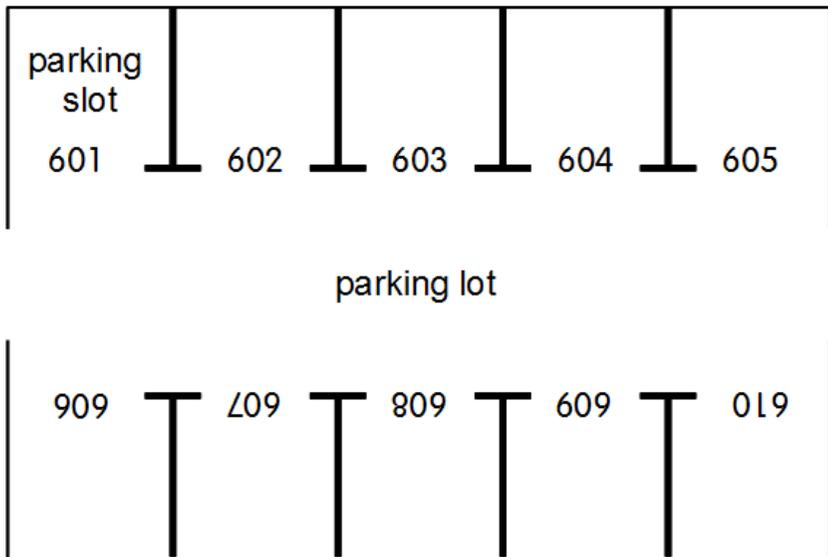
## 3.8. parking lot

---

A parking lot is a space where calls can be parked – similar to a car park. The parking lot is divided into parkingslots. On each slot a call can be parked. Once parked a colleague can pick up the call by dialing the assigned number or by pressing a function key on the phone. The BLF on the phone indicates if there are calls waiting.

The parking lot function is more advanced than the park function (\*38). Both functions can be used simultaneous. A parking lot has the following benefits:

- Similar calls can be parked on the same parking lot. You can create several parking lots, so parked calls can be grouped.
- parkingslots have a fixed extension and can be programmed on a function key. The BLF will indicate if a call is waiting.
- Fall back time can be adjusted and calls can fallback to a specified extension.



*Illustration 8: A parking lot looks a bit like a car park.*

### 3.8.1. Creating a parking lot

Before you can use parkinglots, you have to create at least one. Each parking lot has several parkingslots which can hold a call. You might want to create different parkinglots for each department. This makes it easier for employees to monitor calls for their own department.

<b>Web interface menu</b> <i>Companies &gt; Company &gt; Dial Plan &gt; parkinglots</i>
---

The page above shows a list of parkinglots created for this company. Click on the ADD (+) button to add a new parking lot. Supply the requested information as described next and click on the SAVE button to create one.

<u>NAME:</u>	The name of the parking lot.
<u>DESCRIPTION:</u>	The description for this parking lot.
<u>EXTENSION:</u>	Select a free extension for this parking lot. The Voclarion will suggest one.
<u>FIRST EXTENSION:</u>	The extension of the first slot.
<u>LAST EXTENSION:</u>	The extension of the last slot. The first and last extension define the range of slots in which calls for the lot can be parked.
<u>MUSIC ON HOLD:</u>	The music played when a call is parked.
<u>MAXIMUM TIME IN PARKING LOT (SECONDS):</u>	The maximum time (in seconds) a call can be parked. When this time elapse, the call is connected to the Fallback extension.
<u>FALLBACK EXTENSION:</u>	After the time out, the caller is connected to this extension. When left blank, the caller will be connected to the extension of the user who parked the call.
<u>ANNOUNCE PARKING EXTENSION:</u>	If set to "Yes" (default), the Voclarion will announce the extension of the parking slot the call will be parked on. If you use Monitor/Transfer BLF keys, or blind transfers, it is recommended to set this to "No".

### 3.8.2. Parking a call

If you like to park a call follow the steps below. A parked call can be picked up from every phone within the company. A call will be parked on the first available parking slot.

1. Connect the current call with consultation to the extension of the parking lot.
2. The Voclarion will announce the extension of the parking slot the call will be parked on, followed by on hold music.
3. Connect the call on hold.
4. The call is now parked.
5. Announce the parking slot extension to your colleague.

### 3.8.3. Pick up a parked call

A parked call can be picked up from every phone within the company.

1. Dial the parking slot extension announced by the Voclarion as described on paragraph 3.8.2.
2. The parked call is now connected to your phone and you can have a conversation.

### 3.8.4. Monitor parking slot

parkingslots can be programmed on a phone's function key. A lit BLF indicates a waiting call. Simply press the function key to answer the call. When using BLF keys you don't have to announce a waiting call.

On chapter 2.5.2 you'll find instructions on how to program a function key. Select the function MONITOR and set the extension of the parking slot as PHONE NUMBER. It's advisable to program all parkingslots of the parking lot to your phone.

**parking lot**

## 3.9. Ring groups

---

A ring group (hunt group) rings the phones of all agent which are in the group. One ring group can hold multiple agents and an employee can be an agent for one or more groups within the same company.

### 3.9.1. Ring group vs. Queue

Compared with a queue (chapter 3.10) a ring group offers only one strategy of distributing phone calls to agents: ring all agent phones at the same time. A ring group can not play messages indicating the waiting time or the number of callers waiting. Also there are no reports. If you like a more advanced waiting line, read chapter 3.10 about queues.

### 3.9.2. Creating a ring group



On the page you'll see a list of all available ring groups.

1. To create a new ring group, click [ADD \(+\)](#).
2. Fill out all required fields, as explained next.
3. Click [SAVE](#).
4. A new group is created and available. To fine tune it, click on the group's name to edit it.

Explanation of the ring group settings. Some settings are only available when editing the ring group.

<b>Field</b>	<b>Description</b>
<a href="#">NAME:</a>	The name of this ring group.

Field	Description
<u>EXTENSION:</u>	Choose an empty extension for this ring group from the list or add a new one using the <u>ADD</u> (+) icon.
<u>DEPARTMENT:</u>	Each ring group can be assigned to a specific department. If so, the department manager can edit it.
<u>DEPARTMENT MEMBERS ONLY:</u>	If set to “Yes”, only department members can login to this ring group as agent. Otherwise everybody within the company can login to this ring group.
<u>SHOW ON PHONE DISPLAY:</u>	The group's name can be shown on the phone display. The result may vary depending on the type of phone and other settings.
<u>RINGING TIME:</u>	The length (in seconds) of a ringing sequence. After this period Voclarion will bounce the call. Depending on the extension's call forward setting (chapter 2.3.4.2) the next step will be taken.

### 3.9.3. Adding agents to a ring group

<p><b>Web interface menu</b>      Companies &gt; <i>Company</i> &gt; Dial Plan &gt; Ring Groups &gt; <i>Ring Group</i></p>
--

After creating a ring group, you have to assign agents to it. An agent can be member of more groups. When an agent logs in, he automatically logs in to all ring groups.

Go to the tab AGENTS. The tab shows the agents currently assigned to this group and some basic information about the agents. To change the assigned agents, click the EDIT BUTTON (.)

1. All users for the company are shown, grouped by department. Select the users you wish to make agent for this group by marking the IN RING GROUP check box. You can also set the following settings:

## Ring groups

IN RING GROUP:

If checked, the user is added as agent to the ring group. If you uncheck the user, the user will be removed.

HONOUR FORWARDING:

By default the call forwarding settings are *not* followed for ring group calls. If you select this option, ring group calls will follow call forwarding settings. **Be very careful. Read the warning below.**

2. Click SAVE.



It is highly recommended to disable HONOUR FORWARDING for all agents. If used, try this option in a test situation first.

**Example 1:** if an agent has a call forwarding 'busy' to voice mail, ring group calls are accepted even if the agent is on the phone. All ring group calls are direct accepted by the mobile phone's voice mail. In most cases this is not desirable.

**Example 2:** if an agent sets call forwarding to his mobile phone number, and it's turned off, all queue calls will end up in the phone's voice mail box. Other agents will not receive any incoming calls anymore. Even if you disable voice mail on the mobile phone, you still cannot avoid the unavailable system message.

**Example 3:** if an agent has set a forwarding 'when logged out' on his extension and is logged out, forwarding stays active. Meaning that if the agent has set a forward 'when logged out' to his mobile number and he logs out he still receives queue calls on his mobile phone.

---

### 3.9.4. Forwarding Behaviour

If nobody is logged on in a ringgroup (or everybody is on DND), jumps to Forwarding When Logged Off setting of the ring group extension.



## 3.10. Queues

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A queue handles large quantities of incoming phone traffic. All callers are placed in a queue until an *agent* is available to answer the call. Employees who answer calls from a queue are called agents. Queues are essential for call centers and companies with lots of incoming calls. One queue can hold multiple agents and an employee can be an agent for one or more queues within the same company. The queue weight determines which queue gets to such agents first.

A queue offers multiple strategies of distributing phone calls to agents and can play messages and music while callers are waiting. There are numerous reports available about processed calls, waiting time and productivity of agents.

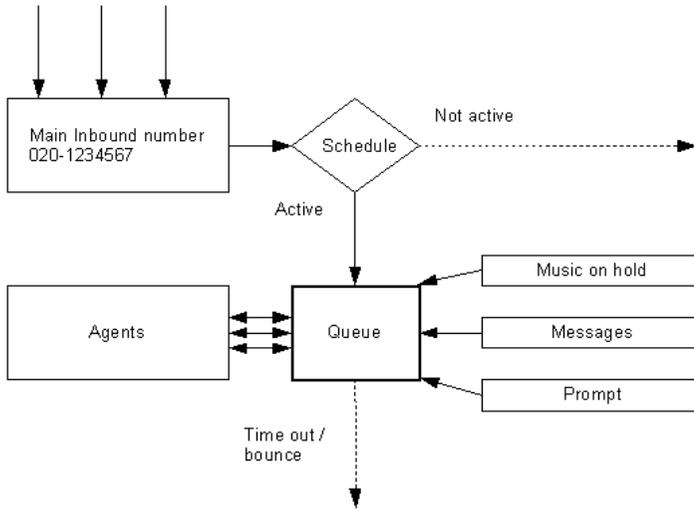
You can use queues as hunt groups. One Voclarion can support up to 250 queues<sup>82</sup>.

A caller can leave a queue in a few ways:

- The caller can end the call while waiting. We call this an abandoned call and it will show up in the reports as such.
- The caller can leave the queue by pressing `*` while waiting. The dial plan will interpret this as a non-answer. It is up to the administrator to set up call forwarding from a queue as a no-answer situation.
- The caller can be removed from the queue when the last agent is logged of (if set). The dial plan will interpret this as a non-answer. It is up to the administrator to set up call forwarding from a queue as a no-answer situation.
- The caller can be refused when the queue is full (if set). The dial plan will interpret this as busy. It is up to the administrator to set up call forwarding from a queue for a busy situation.
- The caller can be removed from the queue when the waiting time expires. The dial plan will interpret this as busy. It is up to the administrator to set up call forwarding from a queue for a busy situation.

---

<sup>82</sup> This is a limitation of Asterisk and will be improved in the near future.



*Illustration 9: Typical Queue Configuration*



<b>Web interface menu</b>	Companies > <i>Company</i> > Dial Plan > Queues > <i>Queue</i>
---------------------------	--

### 3.10.2. Creating Queues

1. Click ADD.
2. Fill out all required fields, as explained next.
3. Click SAVE.
4. A new queue is now created and available. To fine tune, click on the queue name to edit it.

Explanation of settings. Some settings are only available when editing the queue.

<b>Field</b>	<b>Description</b>
<u>NAME:</u>	The name of this queue.
<u>EXTENSION:</u>	Choose an extension for this queue from the list or add a new one using the <u>ADD</u> (+)-icon.
<u>DEPARTMENT:</u>	Each queue can be assigned to a specific department. If so, the department manager can edit it.
<u>DEPARTMENT MEMBERS ONLY:</u>	If set to "Yes", only department members can login to this queue as agent. Otherwise everybody within the company can login to this queue.
<u>LANGUAGE</u>	Language for spoken messages like waiting time. Select a language to change the current language of the call, or select "Don't Change" (default) to keep the current language.
<u>GREETING PROMPT</u>	Select a sound prompt from the list, which will be played on entering the queue. To record a sound prompt, see chapter 3.11.

## Queues

### Field

### Description

MUSIC ON HOLD:

You can select a category of music which will be played while the caller is waiting. Upload your own music (chapter 3.7) or use the Voclarion default music. It's a good idea to play some kind of sound, otherwise the caller may think something went wrong and disconnects. If you don't want to play music, choose "Ringing" (chapter 3.7).

SHOW ON PHONE DISPLAY:

The queue's name can be shown on the phone display. The result may vary depending on the type of phone and other settings.

AGENT ANNOUNCEMENT:

When an agent answers a call, this sound prompt will be played first. This can be useful when employees are agent for more queues. For example: before starting the conversation the employee hears the department name the customer is calling. To record a sound prompt, see chapter 3.11.

QUEUE WEIGHT:

Set a priority for calls from this queue. Calls on queues with a higher weight are first offered to available agents. Priority (low - high): 1 – 99.

<b>Field</b>	<b>Description</b>
<u>STRATEGY:</u>	<p>A ringing strategy determines on how calls are divided between agents. You can choose one of the following ringing strategies :</p> <ol style="list-style-type: none"> <li>1. Ring all available phones until one answers (default)</li> <li>2. Take turns ringing each available phone. Phones will ring one at a time, starting with a random phone.</li> <li>3. Ring the phone which was least recently called by this queue.</li> <li>4. Ring the one with the fewest completed calls from this queue. Voclarion counts the completed calls. If an agent logs out or uses Do Not Disturb and comes available again, the agent needs to catch up with the rest.</li> <li>5. Ring random phone</li> </ol>
<u>RINGING TIME:</u>	The length (in seconds) of a ringing sequence for each phone. After this period Voclarion will try to connect to the next phone, based on the strategy <sup>84</sup> .
<u>WAITING TIME BEFORE RETRY:</u>	The time (in seconds) after which the Voclarion repeats the ringing strategy. During this pause, it is not possible to answer a phone. The lowest value for this field is 3 seconds.
<u>MAXIMUM TIME IN QUEUE:</u>	The time (in seconds) a caller is allowed to stay in this queue. After this, the queue (or better: the queue's extension) will bounce the call as 'not answered'. The call forwarding settings of the queue's extension determine what happens next.

---

<sup>84</sup> If your strategy is set to "ring all available phones until one answers", and your ringing time is set to 15, all phones will ring for 15 seconds. Then all phones will pause for a moment and all phones will ring again for 15 seconds, and so on until the queue is empty.

## Queues

### Field

WRAP-UP TIME:

VARIABLE WRAP-UP TIME:

MAXIMUM CALLERS IN QUEUE:

ANNOUNCE NUMBER OF

WAITING CALLERS:

### Description

The time (in seconds) the agent's phone doesn't ring after ending the last call. This period can be used to write a report. Wrap up-time is valid for all queues.

Allows each agent to decide when to receive queue calls again after. At the end of a call the agent is set to queue pause automatically and has to disable this manually to end the pause. See page 356 for more information about Queue Pause.

The maximum number of callers for a queue (set "0" for unlimited). If the queue is full, the queue (or better: the queue's extension) will bounce the call as 'busy'. The call forwarding settings of the queue's extension determines what happens next.

Should the queue announce the number of waiting callers? Set the time (in seconds) between announcements<sup>85</sup>. To disable set "0" (default). If callers can have different priorities, it is advisable to not use this because new calls can be placed at the top of the queue.

---

85 Note when testing: the first caller in line will not hear this message, the following callers will hear it upon entering. Then the message repeats every X seconds as set for this field.

<b>Field</b>	<b>Description</b>
<u>ANNOUNCE AVERAGE WAITING TIME:</u>	<p>Should the queue announce the average waiting time after announcing the number of waiting callers<sup>86</sup>? Choose between:</p> <ul style="list-style-type: none"> <li>• Yes – The average waiting time is announced after each announcement about the number of people waiting.</li> <li>• No – No average waiting time is announced.</li> <li>• Once – The average waiting time is announced only when entering the queue.</li> </ul> <p>If <u>ANNOUNCE NUMBER OF WAITING CALLERS</u> is disabled, <u>ANNOUNCE AVERAGE WAITING TIME</u> is disabled too.</p> <p>The average waiting time is based on this day's calls and will become more accurate during the day. The first caller of the day will not hear this message, because there is no information available yet.</p>
<u>ANNOUNCE ROUND SECONDS:</u>	<p>Set how precise the average waiting time is announced. For example if you set this field to "30", the queue will announce: "The average waiting time is one minute and thirty seconds" instead of "The average waiting time is one minute and 33 seconds".</p>
<u>PERIODIC ANNOUNCEMENT FOR CALLERS:</u>	<p>Announcement played to callers while waiting in a queue, like product information. Choose a sound prompt from the list. See chapter 3.11 for more information about uploading sound prompts. The first announcement is played after the first "Ringing time".</p>

---

<sup>86</sup> Note when testing: everybody but the first caller in line of the queue will hear the average waiting time.

## Queues

### Field

TIME BETWEEN PERIODIC  
ANNOUNCEMENTS:

### Description

Time in seconds between the start of two announcements. The first announcement starts after RINGING TIME and will repeat itself after the time in seconds set for this field, while all agents are busy. If an agent is available, no announcements are played, a new ringing time starts. During ringing time, it is not possible to play periodic announcements, only in between. See illustration 10 on page 193. To disable periodic announcements, enter "0" (default).

Example: If you want to play an announcement every 30 seconds, and the announcement is 15 seconds, then you should set this field to 45 seconds. The system counts from the beginning of the announcement, not from the end.

CALLER ALLOWED TO BE IN  
EMPTY QUEUE:

When set to "yes", the callers are allowed in a queue with no agents logged in. Otherwise the queue (or better: the queue's extension) will bounce the call as 'busy'. The call forwarding settings of the queue's extension determines what happens next.

HASH TRANSFER ON INCOMING  
CALLS:

Allows you to transfer incoming queue calls by pressing the hash (#) key on your phone. To disable this, set it to 'No'. To use callee transfer within queues, you should also activate it on your phone (chapter 2.5.1).

CALL AGENTS WHEN BUSY:

Send queue calls to the agent's phone, even if the agent is on the phone. Only useful if Call Waiting is enabled (chapter 2.5.1).

### 3.10.3. Adding Agents to a Queue

**Web interface menu**      Companies > *Company* > Dial Plan > Queues >  
*Queue*

After creating a queue, you have to assign agents to the queue. An agent can be member of more queues. When an agent logs in, he automatically logs in to all queues.

Go to the tab AGENTS. The tab shows the agents currently assigned to this queue and some basic information about the agents, like if the agent is on a queue pause. To change the assigned agents, click the EDIT BUTTON.

1. All users are shown, grouped by department. Select the users you wish to make agent for this queue by marking the IN QUEUE check box. You can also set the following settings:

<u>IN QUEUE:</u>	If checked, the user is added as agent to the queue. If you uncheck the user, the user will be removed.
<u>PRIORITY GROUP:</u>	Agents can have a priority. Agents with a lower priority <u>only</u> receive calls when agents with a higher priority are busy or logged out. This way you can create backup agents. 1 = highest priority, 99 = lowest priority. This does not work with the ringing strategy "Ring all phones".
<u>QUEUE PAUSE:</u>	If checked, the agent is able to take a break from the queues, without logging out of his phone entirely. The user is still reachable for other calls. A pause is set for <i>all</i> queues.
<u>HONOUR FORWARDING:</u>	By default the call forwarding settings are <i>not</i> followed for queue calls. If you select this option, queue calls will follow call forwarding settings. <b>Be very careful. Read the warning below.</b>

## Queues

2. Click SAVE.



It is highly recommended to disable HONOUR FORWARDING for all agents. If used, try this option in a test situation first.

**Example 1:** if an agent has a call forwarding 'busy' to voice mail, queue calls are accepted even if the agent is on the phone. All queue calls are direct accepted by the mobile phone's voice mail. In most cases this is not desirable.

**Example 2:** if an agent sets call forwarding to his mobile phone number, and it's turned off, all queue calls will end up in the phone's voice mail box. Other agents will not receive any incoming calls anymore<sup>87</sup>. Even if you disable voice mail on the mobile phone, you still cannot avoid the unavailable system message.

**Example 3:** if an agent has set a forwarding 'when logged out' on his extension and is logged out from the queue, forwarding stays active. Meaning that if the agent has set a forward 'when logged out' to his mobile number and logs out from the queue, he still receives queue calls on his mobile phone.

---

### 3.10.4. Queue Status

The tab CURRENT STATUS shows real time information about the selected queue. The page contains the following information:

#### State

<u>CALLERS:</u>	The number of callers in this queue.
<u>HOLD TIME:</u>	The longest waiting time.

---

<sup>87</sup> With strategy "Ring all phones".

**Callers**

<u>POSITION:</u>	Position in the queue. Call number 1 will be answered next <sup>88</sup> .
<u>CALLER ID:</u>	caller ID and name of the caller (if available).
<u>TIME WAITING:</u>	Time this caller is waiting in the queue (min:sec).
<u>PRIORITY:</u>	Priority of this caller. Callers with higher priority are placed on top of the list.

**Agents**

<u>AGENT:</u>	Name and availability of the agent. <ul style="list-style-type: none"> <li>• Unavailable</li> <li>• Not in use</li> <li>• In use</li> </ul>
<u>CALLS TAKEN:</u>	Number of calls answered by this agent.
<u>MINUTES SINCE LAST CALL:</u>	Time elapsed since last answered call (min:sec).

**3.10.5. Queue Report**

The tab REPORT shows the real time daily queue report. It summarizes the use of a selected queue over the selected time span. More reports are available from the reports menu at the left of the screen.

First select a start and end date. You also can select which queue you would like to see and the incoming number from which the calls have to come. When done, press the GENERATE button and the report will be generated. This can take a few seconds if you selected a wide time span.

The report contains the following information per day and summations over the selected time span.

<b>Field</b>	<b>Description</b>
<u>TOTAL CALLS:</u>	The number of connected calls to this queue.
<u>ANSWERED:</u>	The number of answered calls.

---

<sup>88</sup> If all new incoming calls have the same or lower priority.

## Queues

<b>Field</b>	<b>Description</b>
<u>ABANDONED:</u>	The number of calls ended by the caller before the call was answered by an agent or the maximum waiting time had expired.
<u>TIME OUT:</u>	The number of calls removed from the queue because the maximum waiting time had expired.
<u>TOTAL DURATION:</u>	Total speaking time of all calls (m:s).
<u>AVERAGE DURATION:</u>	Average call duration (m:s)
<u>TOTAL WAITING TIME:</u>	Total waiting time of all callers (m:s)
<u>AVERAGE WAITING TIME:</u>	Average waiting time of all callers (m:s)

### 3.10.6. Queue Fall Back

<b>Web interface menu</b>	<i>Companies &gt; Company &gt; Dial Plan &gt; Inbound Numbers</i>
	<i>Companies &gt; Company &gt; Dial Plan &gt; Queues</i>

Sometimes you want to have a fall back for a queue: agents who only receive calls when all other agents are busy or when the waiting time for your customers exceeds a certain limit.

Note the difference between both approaches. In the first case backup agents are contacted as soon as all other agents are on the phone. In the second case backup agents are contacted only after the maximum waiting time has expired.

In the next chapters we explain how to set up both scenarios.

#### 3.10.6.1. Agent priority

The first case can be achieved by setting priorities on agents. If all 'high priority agents' are on the phone, the queue will call lower priority agents. Note: this does not work with the ringing strategy "Ring all phones" (page 196).

The agents with the highest priority receive incoming calls first. When all agents with the highest priority are busy or logged out, new incoming calls are sent to

agents with a lower priority. You can set the priority of agents from the queue settings. Agents can have a priority from 1 to 99, where 1 is the highest (most important) priority. Priority 1 agents will receive calls first. Agents can have a different queue priority for each queue. See chapter 3.10.3 for more information about how to add agents.

### 3.10.6.2. Fall Back queue

We achieve the second scenario by creating a fall back queue. When a caller has waited for X seconds in the main queue (A), he will be connected to a fall back queue (B). Queue A and B have both different agents.

The overflow is created by setting a limit on queue A. In the example above we have set the maximum waiting time for queue A. However you can also set a limit on the number of accepted callers. The call forwarding settings on the extension of queue A will redirect callers to queue B.

1. Create a new queue with fall back agents (see chapter 3.10). We call it the fall back queue.
2. Now go to extension of the main queue and set the call forwarding settings for “When Not Answered”<sup>89</sup> or “When Busy”<sup>90</sup> to forward to the fall back queue (see chapter 2.3.4.2 on call forwarding settings).
3. Set the maximum waiting time (or maximum callers) for the main queue.

---

<sup>89</sup> Will be triggered when maximum waiting time expires.

<sup>90</sup> Will be triggered when queue is full.

## Queues

---



- Periodic announcing the number of waiting callers or waiting time can have undesirable effects when connecting callers to a different queue. The waiting time and number of waiting callers can change rapidly.
  - Do not use the same agents for both queues.
  - Setting an overflow to the fall back queue based on number of people waiting in the first queue can have an undesirable effect: last callers are helped first.
-

### 3.10.7. Queue Priority on Inbound Numbers

Several inbound numbers can be connected to one and the same queue. It is possible to put calls from a specified inbound number (a hotline for example) in front of other queue calls.

We do this by using priorities. Not only agents have a priority, calls also have a priority. Priority on inbound numbers can be used to give incoming calls from a specific inbound number priority over other calls within a queue. For more information see the chapter about incoming numbers (3.2.1).



## 3.11. Sound Prompts

---

With Voclarion you can easily record sound files, named sound prompts. You can also upload sound prompts. Both procedures are described below. Sound prompts can be used to:

- Welcome callers
- Guide callers through an IVR menu
- Inform callers waiting in a queue
- Other messages



A personal unavailable messages for your voice mail is not a voice prompt. Dial \*25 from your phone to access the voice mail menu and to set a message.

---

### 3.11.1. Uploading Sound Prompts

**Web interface menu**      companies > *Company* > Dial Plan > Sound Prompts

1. Click ADD (+)
2. Select a company.
3. Click on the BROWSE button and select the desired sound prompt in WAV<sup>91</sup> or GSM file format. Do not upload files with names containing spaces and other special characters like & and %, as the Voclarion will not be able to play them.
4. Enter a description.
5. Select the language of the sound prompt.
6. Click SAVE. The file will be uploaded, this can take a few seconds, depending on the size of the file.

---

91 RIFF (little-endian) data, WAVE audio, Microsoft PCM, 16 bit, mono 8000 Hz

### 3.11.2. Changing sound prompts

To change or delete a sound prompt, go to the page below, and follow the steps.

**Web interface menu**      Companies > *Company* > Dial Plan> Sound Prompts

1. Click on the ID of the sound prompt you wish to edit or delete.
2. Click on the DELETE (X) button to delete the sound prompt or click the EDIT (  ) button to edit.

### 3.11.3. Recording sound prompts

Recording sound prompts is easy. Just use your phone (and headset if available) as a microphone. Recorded sound prompts will be saved on the Voclarion. The Voclarion will use date and time as a description. Do not forget to edit the description to something more meaningful.

**Web interface menu**      Companies > *Company* > Dial Plan> Sound Prompts

1. Dial **\*55** on your phone. When connected, you will hear a tone.
2. Record your message. When done, press **#**.
3. Choose one of the options:
  - 1: save the recorded message
  - 2: listen to the recorded message
  - 3: re-record a message
4. After saving the sound prompt, you will hear the ID number assigned to this new file. Go to the list of Sound Prompts and click on the newly added sound prompt.

## Sound Prompts

5. Click EEDIT and give the sound prompt a proper description so you can easily select the right prompt when needed. It is recommended to write out the whole text of the recorded message.
6. Click SAVE.

To listen to a recorded message or rerecord a message, dial:

**\*55 <ID of the sound prompt>**. This ID is automatically assigned to a recorded message and can be found in the sound prompt overview.



- To record a welcome message for your voice mail, dial \*25 for voice mail settings.
  - For best sound quality, it is recommended to use the headset or handset for recording sound prompts. Do not use the hand free mode.
-



## 3.12. Interactive Voice Response Menus

---

Interactive Voice Response (IVR) menus lead callers to a department or person, without the help of an operator. An IVR menu usually contains:

- A welcome message
- Menu options (optional)
- Verbal feedback on an action (optional)
- Time out setting (optional)

Example of a voice mail menu:

*Welcome at My Company,*

*Press 1 to speak with the Sales department*

*Press 2 to speak with the Support desk*

*Press 3 to speak with the Finance department*

*Press 4 to leave a message*

*Press 5 to repeat this menu*

*Press 9 to speak with the operator*

*Or dial the extension of your contact.*

*Illustration 11: Typical IVR menu structure*

### 3.12.1. Creating IVR Menus

**Web interface menu**      Companies > *Company* > Dial Plan > IVR Menus

We'll start with creating an IVR menu without any menu options. In the next paragraph we'll show you how to add menu options and how to add the menu to your dial plan.

1. Record or upload the desired sound prompt for the IVR menu. See chapter 3.11 for more information about sound prompts. The prompt contains a welcome message and the menu options the caller can chose from. By not naming all options, you can create hidden menu option. For example you can make a menu option "99" to participate in a conference, but not name the option in the voice prompt.

## Interactive Voice Response Menus

2. Add a new IVR menu. Click the ADD (+) button.
3. Fill out all required fields.

<b>Field</b>	<b>Description</b>
<u>NAME:</u>	The name of the IVR menu, needs to be unique.
<u>DESCRIPTION:</u>	Description of the IVR menu, needs to be unique.
<u>EXTENSION:</u>	Lookup a free extension or create one.
<u>LANGUAGE:</u>	Select the language for this IVR menu.
<u>DIALING EXTENSIONS:</u>	If enabled, callers can dial extensions while in the IVR menu. This is useful for employees who want to call colleagues directly and ignore the menu options.
<u>GREETING PROMPT:</u>	The sound prompt played when the caller enters this menu.
<u>TIME OUT:</u>	The time (seconds) within which a caller can make a choice. After this time expires the caller will be connected to the first extension (see below). Enter a digit > 0 to activate this option. If time out is set to "0" (default) the menu repeats after 5 seconds, with a maximum of 5 repeats. After this the call will be bounced.
<u>1<sup>ST</sup> / 2<sup>ND</sup> / 3<sup>RD</sup> EXTENSION:</u>	Order of extensions the PBX contacts when no choice is made after the given time out. If the first extension bounces the call, the Voclarion will try the second extension and so on.

4. Click SAVE.

It is a good idea to set a Time Out extension, so that even callers with a non standard tone set (old phones) can reach your company if making a choice fails. If you set a time out to anything but "0", and you do set any menu options or extensions, the menu will repeat the greeting prompt 5 times. After that, the connection will be broken.

### 3.12.2. Adding Options to an IVR Menu

In the previous paragraph we have created an IVR menu. Now we can add some menu options. For each menu option you can specify a "key press". Pressing this key triggers an action: a sound prompt will be played and/or the call will be

**Web interface menu**      Companies > *Company* > Dial Plan > IVR Menus

transferred to an extension. As a safety a caller is allowed to make up to 100 key presses in one call.

1. Click on the IVR menu for which you want to add menu options.
2. Click the ADD button on the bottom of the screen to add a new menu option to the selected menu.
3. Fill in all required fields:

**Field**

**Description**

KEY PRESS:

The number a caller has to dial on his phone to select the menu option. Choose from 0-999, \* and #. Other entries are not allowed. The number of key presses per call is limited to 1000.

**Note:** If you created a menu option that is equal to an existing extension and you have enabled ALLOW EXTENSIONS, Voclarion will connect to the menu option.

DESCRIPTION:

Description for this menu option.

LANGUAGE:

Change the language of this call.

GREETING PROMPT:

This is a sound prompt announcing your choice and will be played right before the call is transferred, for example: "You will now be connected to the sales department..."

1<sup>ST</sup> / 2<sup>ND</sup> / 3<sup>RD</sup>:

EXTENSION:

Select the extension to transfer the call to after the caller chose this option. If the 1<sup>st</sup> extension is unreachable Voclarion will connect to the 2<sup>nd</sup> and so on. It is advisable to set at least one extension for each menu option.

4. Click SAVE.

### 3.12.3. Connecting to a Voice Mail Box

It is also possible to connect a caller to a voice mail. Announcing that the caller can leave a message, can be done in two ways:

- Connect to the mail box of a user. Connect the menu option to **\*26<extension\_voicemailbox>**. This user already has recorded an unavailable message by dialing his mail box options (using **\*25**). For more information about voice mail see page 363.
- Otherwise you can record a sound prompt by dialing **\*55** and insert this into the menu option field GREETING PROMPT. Provide the mail box extension as described before. By default the mail box gives a system unavailable message, which has to be replaced by an empty unavailable message (silence). For more information about voice mail see page 363.

### 3.12.4. Connecting to an IVR Menu

Now we have a fully functional IVR menu. You can test it by dialing the extension of the menu from your office phone. To connect a dial plan item (for example an incoming number) to an IVR menu, simply set the menu's extension as first of the three extensions<sup>92</sup> within the incoming number.

If you would like to connect a call to an extension if the caller doesn't press a key (or there aren't any options) set this extension in the *IVR menu* settings, not in the dial plan item referring to the menu. To clarify this we will show you two examples. The first will not work. The second example is the right way to set an alternative extension.

#### Example of an Incorrect Configuration

In the example below you see that extension 1001 and 1002 are entered as second and third extension, *after* the IVR menu in the dial plan. This will **not** work because an IVR menu always accepts the call. It will never bounce calls, so extension 2 and 3 are never used.

---

<sup>92</sup> An dial plan item has always three extensions to connect to. If the first extension cannot be reached, the PBX tries to connect the call to the next and so on. However once connected to an IVR menu, the call cannot return. An IVR menu simply always accepts a call, it never bounces a call. Therefore adding extensions after an IVR menu extension has no use.

Inbound Number	1st extension	2nd extension	3rd extension
020-1234567	1000 (IVR menu "Welcome")	1001 (queue)	1002 (operator)

### Example of a Correct Configuration

The next example shows a correct configuration. This menu connects to extension 1001 automatically after 5 seconds, when no key is pressed by the caller. If the call is bounced by 1001, it will be connected to extension 1002.

Inbound number	1st extension	2nd extension	3rd extension
020-1234567	1000 (IVR menu Welcome)	-	-

IVR menu "Welcome"	
Extension	1000
Time Out	5
1st Extension	1001 (queue)
2nd Extension	1002 (operator)
3rd Extension	

### 3.12.5. Fail-safe IVR

Some old phones cannot send DTMF-tones or the Voclarion will not recognize them. In this case the caller can not select a menu option. So it is wise to create an escape. Note: the default timeout setting will disconnect the call after five repeats.

You can create the following menu:

1. Set the timeout to 10 seconds and the first extension to the extension of the operator. The menu will not repeat anymore and if no choice is made within 10 second, the caller will be connected to the operator (escape).

## **Interactive Voice Response Menu**

2. Because the menu will not repeat (which can be useful on larger menus), set a key press, for example 9 to repeat the menu. The extension for this key press is the menu's extension. When the caller chooses to hear the menu again, the menu will start again and the caller has again 10 seconds to make a choice.

## **3.13. Direct Inwards System Access (DISA)**

DISA (Direct Inward System Access) allows an employee from outside the Voclarion to obtain an "internal" dial tone and to place calls as if the caller is calling from inside the office.

Advantages of DISA:

- When making a phone call using DISA, the company's caller ID is used.
- The caller only has to pay the phone costs to the PBX. The phone costs from PBX to the dialed number are billed to the company.
- DISA makes voice mail accessible from abroad.
- Calls that are recorded will be saved in the your personal detail records.

### **3.13.1. Using DISA**

There are two kinds of DISA:

**DISA-Authorized:** The PBX tries to identify the employee with his caller ID. If it matches with the number set as mobile number, the employee will be given a system dial tone. If no match is found, the connection is terminated by the PBX.

**DISA-PIN:** Always require a PIN. You can use this if you want the caller always to enter his PIN. Once you entered your PIN, you can make a phone call with your own caller ID and permissions.

Call an inbound number connected to the DISA application on the PBX. If the PBX identifies the caller by his ID, the user is given a system dial tone on which a call can be placed. You have up to 20 seconds to dial a number. If no match is found, the call will be bounced by the PBX.



If an employee frequently uses DISA, it can be more efficient to provide the employee with a hard- or soft phone. The employee can now more easily make phone calls and receive calls, in many cases at no costs at all.

### 3.13.2. Setting up DISA

There are two ways to use DISA: you can add the service to an inbound number, or you can add the service to an IVR menu. In both cases you connect the call to (one of these) DISA codes:

- \*9001 – DISA identifies caller by caller ID.
- \*9002 – DISA asks for a PIN.

If identification fails, the service will bounce the call. For more information see the following examples.

#### 3.13.2.1. Adding DISA to an Inbound Number

You can assign the DISA service to any inbound number.

**Web interface menu**      Companies > *Company* > Dial Plan > Inbound Numbers

1. Click on an inbound number to edit the settings.
2. Set field IDENTIFY CALLER BY ID to “Yes”.
3. For field 1ST EXTENSION select extension **\*9001** (DISA-Authorised). The PBX will compare the caller's ID with the mobile number as set in his personal information. If they match, the caller is connected to the DISA service and will be given a dial tone. Otherwise the call will bounce.
4. For field 2ND EXTENSION select extension **\*9002** (DISA-PIN). If the call has bounced, the second extension is called and the caller is asked to enter a

## **Direct Inwards System Access (DISA)**

PIN. If there is a match, the caller is connected to the DISA service. Otherwise the call will bounce.

### **3.13.2.2. Adding DISA to an IVR menu**

You can also assign DISA to an IVR menu option. The configuration is similar to setting up an incoming number (chapter 3.13.2.1). Chapter 3.12 describes how to set up an IVR menu.

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## 3.14. Conference Boxes

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Most phones support 3-way calls. You can make a phone call with three people. One person has to call two others to start a conversation. Sometimes you want to talk with more than two people. In such cases conference boxes are quite useful. This feature is supported by the PBX and can hold an unlimited number of participants.

Conference boxes are created within the Voclarion web interface or by phone. The user's department is owner of the conference box. The list of conference boxes is visible to anyone, but only members of the same department can view and edit PIN (Personal Identification Number) information. You can also create conference boxes from your office phone. A box is assigned an extension from the conference box range. Phone-made conference boxes are by default valid for only one week. However from the PBX web interface the expire date can be changed.

### 3.14.1. Conference Box Ranges

If multiple companies are registered to the PBX, an extension range must be configured for each company. Conference boxes need PBX unique extensions, so two boxes cannot have the same extension, even if they are created in different companies. In case of a single company, defining ranges is useful to help you organize your extensions.

### 3.14.2. Creating Conference Boxes by Phone (\*30)

To create a conference box by phone, dial **\*30** and follow instructions. The PBX will ask you for a 4-digit PIN and will return a conference box number and the PIN. The conference box is now created.

### 3.14.3. Creating Conference Box from Web Interface

To create a new conference box from the Voclarion web interface. Go to:

1. Click the ADD (+) button to add a new box.
2. Fill in all required fields:

<b>Web interface menu</b>	Companies > <i>Company</i> > Dial plan > Conference Boxes
	My Voclarion > Conference Boxes

**PIN:** A 4-digit PIN code required to enter the conference. To accept callers without a PIN, leave the field empty.

**START DATE:** Select the first date on which the conference box is available.

**END DATE:** Select the last date on which the conference box is available.

**DESCRIPTION:** Description for this conference box.

3. Click SAVE.

### 3.14.4. Attending to a Conference

A conference can be attended to from within the company but can also be made available from outside. Both will be discussed in the next chapter.

The first person entering a conference will hear a message he is the first one and will hear music until another person enters. Each time a person enters the conference you'll hear a soft beep. When two people are in the conference, the music will now stop and both parties can hear each other. The music used is the company default music on hold and cannot be disabled.

#### 3.14.4.1. From within the company

There are two ways to let callers attend to a conference box:

1. Within the company<sup>93</sup> you dial **\*31** to attend to a conference. Follow the instructions and enter the conference box number followed by **#** and your PIN (if set) followed by **#**.
2. Within the company<sup>93</sup> you dial the extension of the conference box to attend to a conference. Follow the instructions and enter your PIN (if set) followed by **#**.

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<sup>93</sup> Or from a DISA connection.

## Conference Boxes

### 3.14.4.2. External callers

Usually, you also want people from outside the company attending a conference. They cannot dial **\*31**, except if they have DISA access. There are two ways to make your conference box available for the outside world:

1. Add a menu option to an IVR menu, which connects to **\*31** (for different conference boxes) or to the extension of a conference box for a fixed conference box<sup>94</sup>. IVR's are discussed in chapter 3.12.
2. Connect an inbound number to **\*31** (for different conference boxes) or to the extension of a conference box for a fixed conference box<sup>94</sup>. Inbound numbers are discussed in chapter 3.2.1.

### 3.14.5. Managing a Conference (\*32)

Sometimes an administrator is needed to manage a conference. An administrator has some additional options to moderate the conference. Only one person can join the conference as administrator. The administrator dials **\*32 <extension conference box>** to attend to a conference. See paragraph 3.14.6 for administrator options.

### 3.14.6. Menu (\*)

While in a conference box, press **\*** to get access to the menu. A regular users has access to all options, except for menu options 2 and 3: only the administrator is able to reject a caller and (un)lock the conference. If a conference box is locked, no new participants are accepted. New calls are bounced.

When accessing the menu, the PBX will play all available options. Make a choice by pressing a key on your phone. After you pressed a key, you will automatically leave the menu, so you have to press **\*** again to make another choice.

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<sup>94</sup> Make sure it does not expire!

## Conference Boxes

1 (un)Mute the conference	2 <i>(un)Lock conference</i>	3 <i>Eject last participant</i>
4 Decrease volume	5	6 Increase volume
7 Decrease mic. volume	8 Leave menu	9 Increase mic. volume
* Enter menu	0	#

Table 11: Conference box menu

## 3.15. Page Groups / Intercom

---

Paging allows you to speak through the speaker phone of several phones at the same time. When enabled and supported intercom (talking back<sup>95</sup>) is also possible. Contact your phone supplier for more information about supported phones.

### 3.15.1. Creating a page group

**Web interface menu**      *Companies > Company > Dial Plan > Page Groups*

1. Click ADD.
2. Enter all fields:

<b>Field</b>	<b>Description</b>
<u>NAME</u> :	The name for this page group.
<u>EXTENSION</u> :	The extension for this page group. Add a new one (+), or select a free extension from the list.
<u>TALKING BACK</u> :	This enables the intercom feature, the receiving person is able to talk back to the person who is speaking.

3. Click SAVE to create a new page group and return to the list with page groups.

### 3.15.2. Add phones to a page group

After creating a page group, you can add phones to the group. Keep in mind that you add *phones* instead of *users*, because this feature is phone dependent.

1. Click on the page group.
2. Click the 'PHONES' tab to see an overview of all phones in this page group.
3. Click on the EDIT button, at the top of the page to see an overview of all available phones for this company.

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<sup>95</sup> The phone switches to hands free.

4. Select the check boxes of the phones you want to add to this page group. The phones are sorted by location.
5. Click SAVE at the bottom of the page to add the selected phones.

### 3.15.3. Calling a page group

To page a group dial the page group extension.

### 3.15.4. Paging one person

To page just one person, dial \*36<extension>. The person you are paging is able to talk back to you over the speaker phone (intercom). If you wish to page your colleague with extension 6234, dial: \*366234.

In case of paired phones, only the phone that the user actually logged into, will be paged.



## 3.16. Outbound Call Routing

---

Outbound Call Routing defines the way outbound calls are handled and by which trunks. If the Voclarion has trunks from multiple operators, setting the appropriate routes achieves Least Cost Routing.

Least Cost Routing (LCR), also known as Automatic Route Selection (ARS), is a feature that enables the system to route a call over the most appropriate carrier and service offering based on factors such as the type of call (local, local long distance), the time of day (prime time, non prime time), and the day of the year (weekday, weekend day). In countries with lower rates for cellular-to-cellular calls than for calls between cellular phones and landlines, LCR sometimes is used to route the landline leg through a cellular interface (GSM Gateway) to take advantage of the lower rates. LCR is of greatest value if the telecom environment is liberalized or deregulated and there are multiple competing carriers and rate plans from which to choose.

### Examples of Least Cost Routing

1. Sometimes there are lower rates for cellular-to-cellular calls than for calls between cellular phones and landlines. Dutch mobile numbers start with "06". We can create a call route that routes phone numbers starting with "06" to the "GSM Gateway"-trunk. Now we automatically save on mobile phone calls.
2. You route all USA phone calls through a SIP provider with lower USA rates. You route UK calls through another SIP provider.
3. Calls from our company in the UK to phone number in the USA will be first routed over the internet to the Voclarion at our office in the USA. This Voclarion puts the calls through to the desired phone number at national phone rates. There are no phone costs for calls between the two Voclarion.
4. You can configure a trunk so international phone numbers will be prefixed for using a carrier select.

### 3.16.1. First Outbound Call Route

The first route you create will be your default route. This route will be followed when no other routes apply<sup>96</sup>. All other routes are exceptions to the first route. Trunks must be created, before you can define call routes.

### 3.16.2. Creating Outbound Call Routes

<b>Web interface menu</b>	Companies > <i>Company</i> > Dial Plan > Outbound Call Routing
	Quick Setup > Outbound Call Routing

1. Click the ADD button (+) at the top right side of the screen.
2. Fill out the required fields: the target phone numbers and the primary trunk for that target. Other fields are optional.

<b>Field</b>	<b>Description</b>
<u>TARGET:</u>	The selection of phone numbers for this route. We suggest you start with a default route, indicated by the target wildcard “.“ (dot). Wildcards are explained on page . You can find more information about targeting in the next paragraph.
<u>FIRST (SECOND, THIRD) TRUNK:</u>	Select the trunk these phone numbers should be routed over. The pull down shows all trunks available. If the Primary Trunk is unavailable <sup>97</sup> , the Voclarion will try the Secondary Trunk, and so on.
<u>SET EXTERNAL CALLER ID:</u>	Set to "Yes" (default setting) if the caller ID has to be the external caller ID (see also caller ID priority on page 38). Otherwise the caller ID is set to the local extension.

96 The default route has target “.“ (dot), the wildcard representing all digits.

97 This is no protection against a bad line or a disconnection by your phone company. If your phone company accepts the call the PBX marks the call as successful routed.

## Outbound Call Routing

Field	Description
<u>STRIP DIGITS:</u>	The number of digits that should be removed from the front of the dialed phone number, before it is sent over this trunk. You achieve a different prefix by first stripping a number of digits from the front of the target and then prefixing the result using the <u>PREFIX WITH</u> field. Note: the trunk can strip and prefix digits too.
<u>PREFIX WITH:</u>	Enter the sequence of digits that needs to be added in front of a phone number, before sending it over the trunk. This is useful if you use a carrier select service.

3. Optionally, add a second and third trunk if more than one trunk is available. The secondary trunk will be used if the primary trunk is not available, and so on.
4. Click SAVE.

### 3.16.3. Targeting

Every route starts with a target which 'describes' the phone numbers used for this route. A target can contain one or more wild 1cards (see page 39)

#### 3.16.3.1. Examples of Outbound Call Routes

- If you like to capture a mobile number (Dutch mobile numbers start with "06") you can use: "06." (there's a dot at the end!).
- International phone numbers start with "00", like "0031205628292". If you like to capture all phone numbers *except* international phone numbers, you can use a target like "ZZ." (there's a dot at the end!). It selects any phone number that does not start with "00".
- The two examples combined: Your phone company has expensive rates for international and mobile calls. Your SIP provider offers lower rates for international calls. A carrier select service has low mobile phone rates.

1. Make a default route (.) to a SIP provider's trunk  
(all calls go through this trunk...)
  2. Make a route (06.) to the carrier select trunk  
(...except mobile calls starting with 06...)
  3. Make a route (ZZ.) to your phone company's trunk.  
(...and except non-international calls)
- If you want to allow users to dial '1649' (carrier select) in front of a phone number. Create a target (1649.). Now the PBX will allow calls starting with this prefix.



## **Chapter 4. Message Plan**

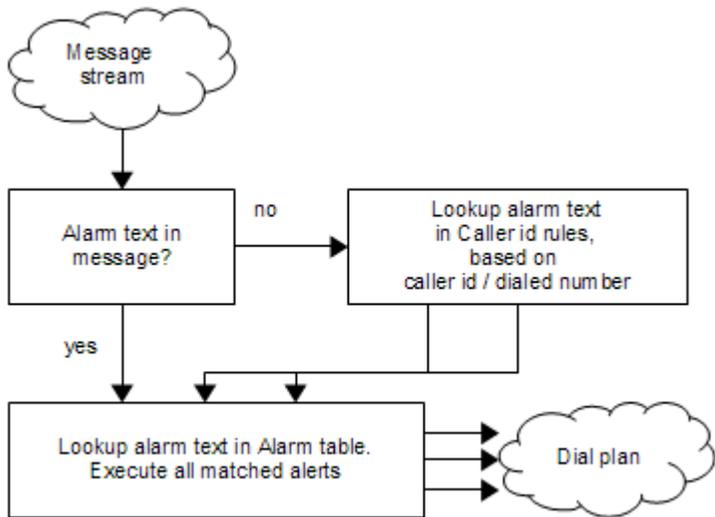
*Starting from Voclarion 2.2 the PBX can take action on  
alarm messages.*

## 4.1. Introduction

---

Starting from Voclarion 2.2. the PBX can act on alarm messages send by alarming systems like fire alarms, intruder alarms, nurse call systems or any other remote I/O controller. The Voclarion can for example call a group of people when a fire alarm comes in. Or a nurse can be called on an emergency call.

With a beeper system the emergency message comes in on a beeper. You have to find a phone and call in to know exactly what's going on. The Voclarion is far more efficient. It can use all sorts of end points like a mobile phone, a pc or a tablet pc. Most end points can show more information on the display and can establish two-way communications with a single key press. Some devices even support video. Also Voclarion can efficiently distribute emergency calls among available staff.



*Illustration 12: Message Plan*

### 4.1.1. How it works

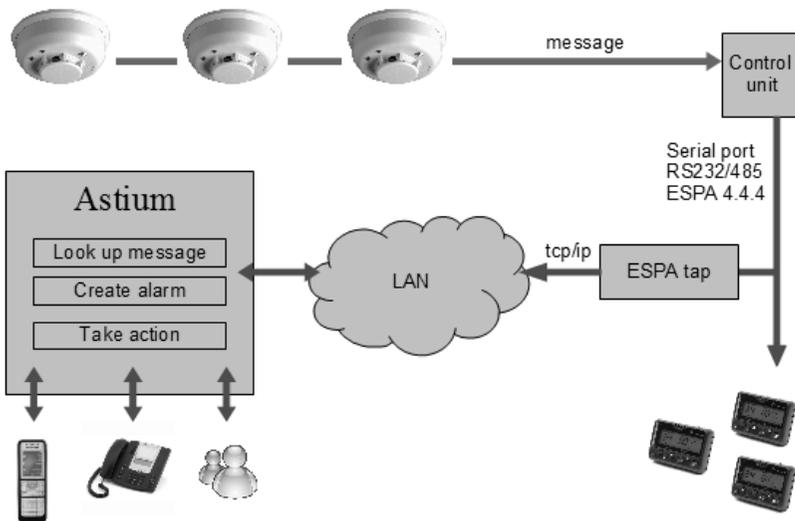
- Voclarion receives messages from different sources. A message contains an alarm text, a caller ID and a dialed number. Some messaging systems

do not supply all values. For such systems Voclarion provides translation tables, depending on the messaging system.

- If no alarm text is available, Voclarion will lookup the caller ID and dialed number from the caller ID table to find one or more alarm texts.
- All alarm text are looked up in the alerts table. The alerts table contains a list of phone numbers to be called for a given alarm text. All matches are now executed.

### 4.1.2. Sources

There are several sources which can be used as message stream for Voclarion.



*Illustration13: Voclarion taps ESPA messages*

#### 4.1.2.1. ESPA tap

Voclarion supports the ESPA 4.4.4 protocol. In 1984 the European Selective Paging manufacturer's Association (ESPA for short) defined the serial data interface ESPA 4.4.4 for radio paging systems. Since then, many producers of call systems, paging systems and alarm servers have implemented this interface and have made it a European standard interface that is widely applied, especially to

## Introduction

realize nurse and emergency calls to mobile caregivers. The protocol is supported by a large number of systems, such as fire alarms, intruder alarms, nurse call systems and cordless telephony.

In the recent past the requirements placed on the mobile alerting processes have, however, changed dramatically. Today, TCP/IP-LAN complements if not replaces the proprietary cabling, WiFi is taking the place of the DECT radio technology, VoIP technology is substituting the ISDN, and simple telephone sets or pagers are being replaced by multifunctional phones. In addition, sites are being increasingly linked and services are centralized.

ESPA 4.4.4 defines a serial protocol intended for the interconnection of paging equipment with telephone exchanges or computer systems. The protocol used is message-based and is built upon the ISO1745 transport specification. The protocol is designed to support multi-drop communications on an RS485 data bus, however it is also often used point-to-point on V24/RS-232 connections. The general principle is that two equipment designed the standard should inter work satisfactorily and sanely without extensive configuration, despite the significant variations in operation of both PBX and paging systems.

ESPA normally just provides a simple message text. The Voclarion ESPA Tap drivers therefore contain a table which can add a callerid, called number to any message text.

### 4.1.2.2. CIE-H10 remote I/O controller

CIE-H10 is a remote I/O controller. It helps to monitor and control digital inputs and outputs remotely. By applying it, not only you can reduce cost and risk, but also you can shorten development time to add the network capability in your system. Because CIE-H10 allows to extend the distance of your I/O control system, you are able to remotely control and monitor the I/O devices over the Internet



*Illustration 14: CIE-H10 remote I/O controller*

anywhere you are. Since CIE-H10 has various methods for I/O control such as HTTP, Modbus/TCP and Serialized Modbus/TCP, it is available on various environments.

A caller ID, called number and optionally an alert text can be assigned to each input port. Note that with an I/O controller the port can be connected to a switch (chapter 3.6), so a BLF can light up, and the dial plan can be influenced while the input port is activated.

### Highlights

- Remote I/O controller
- RS232 to Ethernet Converter
- Ethernet 10Base-T or 100Base-TX (Auto-Sensing)
- 8 Digital Input Ports (photo-coupler interface)
- 8 Digital Output Ports (relay interface)
- 1 x RS232 (up to 230,400bps, RS232 <--> TCP/IP processing)
- Access Restriction : IP and MAC address filtering, Password
- Stored Web server for simple management (custom web page)
- Flexible configuration and network communication by 'AT Commands' (patent)
- Telnet COM Port Control Option (RFC 2217)

#### 4.1.2.3. URL

An alarm can also be triggered by accessing a url containing some parameters. This can be useful for testing your message plan. To trigger an alert, use:

`http://<ip address pbx>/ps/alert.php?shortname=oneip&alert=Burglar`

Force the alert to use the caller ID rule table by not specifying an alert text, but

## Introduction

specifying caller ID and called number instead:

`http://<ip address pbx>/ps/alert.php?shortname=oneip&callerid=151&exten=600`

## 4.2. Setting up messaging

---

1. Voclarion receives messages from a message stream. You can add a new stream by creating a messaging channels (chapter 4.2.1). Each message can contains a caller ID and/or a called number and/or an alarm text.
2. Make a translation of message from the stream to add additional information
3. Create alerts which will be triggered by alert texts. Alerts define which phone numbers are called upon a certain alarm.
4. If the message does not contain an alarm text, an alarm text is looked up by matching caller ID and dialed number from the caller ID rules. If multiple rules match, multiple alarms will be generated!
5. Now all messages have an alarm text which will trigger an alert.

### 4.2.1. Adding a messaging channel

A messaging channel is the source of a messages stream. In most cases this is an ESPA tab or a CIE-H10 controller. To add a new channel, go to:

<b>Web interface menu</b>	Companies > <i>Company</i> > Message plan > Messaging Channel
---------------------------	---

1. Click the ADD button (+) at the top right side of the screen.
2. Supply the requested information as described next.

<b>Field</b>	<b>Description</b>
<u>NAME:</u>	The name for this channel.
<u>TYPE:</u>	Select a channel type.
<u>MAC ADDRESS:</u>	The MAC address of the device. This is used to identify the hardware.
<u>IP ADDRESS:</u>	The device's ip address.

Field	Description
<u>PORT:</u>	The port number to connect to this device, default port number is 5655.

Click SAVE to create a new channel. You'll be taken back to the list of message channel. To fine tune the trunk, click on the name of the created message channel. The following new fields will appear:

Field	Description
<u>DEFAULT GROUP:</u>	Incoming messages are send to this group if no target is found for the message. When in learning mode, new messages are sent to this group in the translation table.
<u>LEARNING MODE:</u>	Automatically build a translation table. New messages are automatically added.
<u>NOTES:</u>	Notes for this trunk.

#### 4.2.1.1. Translation table

A translation table is connected to a messaging trunk. It translates the messages from the stream. You can add translations one by one (chapter 4.2.1.2) or you can import translations from a spreadsheet (information on importing files on chapter 1.8).

#### 4.2.1.2. Add translations manually

To manually add translations to the translation table, go to:

**Web interface menu**      Companies > *Company* > Message plan > Messaging Channel

1. Select a message channel, and click on the tab TRANSLATION.
2. Click the ADD (+) button at the top right side of the screen.

## Setting up messaging

3. Supply the requested information as described next.

<b>Field</b>	<b>Description</b>
<u>MESSAGE:</u>	The text of the message to be translated.
<u>IGNORE:</u>	If set to <u>YES</u> , this translation is ignored.
<u>CALLER ID:</u>	caller ID used for this message.
<u>CALLER ID NAME:</u>	caller ID name used for this message.
<u>DIALED NUMBER:</u>	This sets the dialed number. Used in the alert table to match alerting rules.
<u>ALERT:</u>	The alert this message will trigger. Used in the alert table to match alerting rules.

Click SAVE to save the new translation.

### 4.2.2. Alerts

An alert is activated by an alarm text and calls up to 3 extensions. First, Second and Third extension will be called and whoever answers will be connected to an given extension. People who are called, will see the alarm's caller ID (name) from the alarm in their phone display.

**Web interface menu**      Companies > *Company* > Message plan > Alerts

1. Click on the tab ALERTS.
2. Click the ADD (+) button at the top right of the screen to add an alert.
3. Supply the requested information as described next.

Field	Description
<u>ALERT TEXT:</u>	The alert text which triggers this alert.
<u>1<sup>ST</sup>, 2<sup>ND</sup>, 3<sup>RD</sup> EXTENSION:</u>	The PBX will try to set up a call to the first extension. If it fails, the second extension will be called. And so on. Use phones, ring groups or extensions forwarded to mobile phones. Do not use queues, conference boxes , IVR's or sound prompts.
<u>CONNECT TO:</u>	If one of the extensions above accepted the call, it is immediately forwarded to this extension. You can use all sorts of extensions.

Click SAVE to save the new translation.

### 4.2.3. caller ID rules

caller ID rules activate an alert based on the caller ID and dialed number. You can add rules one by one (see below) or import a set of rules from a spreadsheet (see chapter 1.8 for more information on importing a spreadsheet).

**Web interface menu**      Companies > *Company* > Message plan > Alerts

1. Click on the tab CALLER ID RULES
2. Click the ADD (+) button at the top right of the screen to add a rule.
3. Supply the requested information as described next.

## Setting up messaging

<b>Field</b>	<b>Description</b>
<u>CALLER ID:</u>	Select the caller ID(s) which trigger this alert. Wildcards (chapter 1.4.2.7) can be used.
<u>DIALED NUMBER:</u>	Select the dialed number which trigger this alert. Wildcards (chapter 1.4.2.7) can be used.
<u>ALERT:</u>	Select the Alert which is activated when both caller ID and dialed number match.
<u>DESCRIPTION:</u>	A description for this rule.

Click SAVE to save the new translation.

## 4.2.4. Examples

### 4.2.4.1. Burglar Alarm

1. Connect the Burglar Alarm to the I/O Controller Input Port 1.
2. Generate the alarm text 'Burglar' in the Input Ports port 1.
3. In the Alerts Screen create multiple Alerts with the alert text 'Burglar', each calling some employees. Use a conference box as the Connect To extension. When the alarm goes off, the system calls staff members and connects them with each other in a conference box.

### 4.2.4.2. An analog alarm calls for a nurse

This explains how to connect an analog alarm phone to the Voclarion, parking the call and dialing for a nurse. The nurse will hear an IVR and when she accepts the call by pressing '2', she will be connected the alarm phone in speaker mode.

In short, the behavior you desire is as follows:

1. Alarm button is pressed on alarm phone.
2. This device dials a pre-configured number. It's connected to an ATA.
3. The pre-configured number is a plugin, which switches the device to audio-mode, generates the internal alarm, and parks the call.
4. The alarm looks into the alarm table, and takes appropriate action, like dialing nurse groups.

5. The nurse gets connected to an IVR, and pressing '2' takes the picks, and picks up the waiting console.
6. At the end she needs to dial \*0 to indicate to the console the alarm is handled.

### Defining consoles

The alarm consoles should be programmed with a number to dial. Also they should bear callerid. The system does not work, when we do not have caller ID functioning. Consoles can be connected through an ATA, or can come in over for example analog lines. For this example we assume we have a console connected to an ATA and logged in as user 'Room Alarm' with caller ID 601.

### Entering the necessary configuration

Create one parking lot, 3 plugins, caller ID rules, phone book entries and sound prompts for each alarm console. And Alarm rules for each group of consoles.

#### *parking lot*

Create the parking lot. Assign an extension (say 910), and define 10 parkingslots. Set the waiting time to 250 seconds (chapter 3.8).

#### *Nurse ring groups*

All nurses should be logged in to ring groups. The system will fail over to another group when no nurse answers the call. Create ring groups with number 920, 921 and 922 and place the nurses in those groups.

#### *Plugins*

Create the following plugins in *Company > Advanced > Plugins*

1. CatchAlarm. For the context enter 'catch-console-alarm-entry'. Assign an extension to this alarm. This extension should also be programmed into the alarm console boxes. Parameter 1 is the extension number for the parking lot (910 in our example).
2. ConnectAlarm. Context should be 'connect\_alarm\_console-entry'. Choose any extension number.
3. PlayUserPrompt. Set context to: 'playback\_userprompt-entry'. Choose any extension number.

## Setting up messaging

### *caller ID rules*

For every possible caller (read: for every console installed) you need to define a rule. These rules define which alarm should be generated, based on the caller ID, and the number dialed. In our example we have a console with caller ID 601, preconfigured to dial 902. Define your own text for alarm, say 'HallWay 1'.

### *Sound prompts*

For each alarm console, create a sound prompt stating for example the room number or street address. Also create the following prompts:

- to repeat press '1' to connect press '2'
- alarm

### *Phone book entries*

Based on the caller ID the Voclarion determines two things:

1. the phone brand and model (to know how to reset the alarm, or put it into speaker mode)
2. which sound prompt to play (to tell the nurse who this is)

For each console, create a phone book entry and set the phone model.

### *The Nurse IVR menus*

You need to create two menus and link them together:

1. the first IVR which only says 'Alarm' and links to the second
2. the second says: press 1 to repeat, press 2 to accept

For the first IVR, set the timeout to 1 second, and as timeout choices the plug in playuserprompt, and as the second choices the other IVR.

For the second IVR, set the 'to repeat press 1' as the welcome message, timeout to '5', and define the following IVR key presses:

- '1' - point to the other IVR
- '2' - point to plugin ConnectAlarm

### *Alarm Rules*

As the final step, you should now create Alarm Rules. These define what will happen when a particular alarm is received. In our example create an alarm for

## Setting up messaging

Hallway 1, tell the system to dial the nurse ring groups, and connect them to the Nurse IVR.



## **Chapter 5. My Voclarion**

*This chapter describes the My Voclarion section. My Voclarion contains setting of the user currently logged in: you.*

## 5.1. Introduction

---

The PBX Manager is the Voclarion's Graphical User Interface (GUI). You can access the PBX Manager by surfing to your Voclarion's ip address or hostname. Contact the system administrator if you need help with this.

With the PBX Manager you can easily change all settings or view call information. A section of the PBX Manager is called My Voclarion and it gives you direct access to your personal settings. When logging in to the PBX manager, you will automatically be taken to My Voclarion.

Depending on your role you might see slightly different options.

## 5.2. My Voclarion Overview

---

### 5.2.1. My Voclarion® Dashboard

Once successfully logged in, you will be taken to the overview page, which contains all basic account information. We call this the dashboard. Here you can<sup>99</sup>:

- View your last calls
- Set your presence information
- Manage your voice mails
- Manage your extensions
- Manage your queues
- View your personal settings

All information is grouped in so called *information blocks*. We did great effort to keep the dashboard page nice and clean, so only information useful to you will be shown. So, if you are not logged in to any queues, the queue block will not be shown at all. However, if you would like to change queue settings, simply use the menu at the left of the page.

All blocks on this page can be contracted and expanded by clicking on the icon  on top of the block. In some cases the title bar contains information about the block, like the number of (new) voice mails.

#### 5.2.1.1. My Voice Mail

When you have (new) voice mails, the voice mail block will be shown. Here you can view your voice mails. You also can listen to () , download () and delete () voice mails. Click the corresponding icon at the end of the row and follow the instructions. Only the last 15 messages will be shown.

---

<sup>99</sup> Depending on user role.



Once a voice mail is deleted, you cannot undo it.

---

### 5.2.1.2. My Last Calls

This is an overview of your last 15 calls<sup>100</sup>, incoming and outgoing. To listen to a recording click on . To see the call details, click on .

### 5.2.1.3. My Extensions

This block will be shown when you have extensions assigned. If not, you can assign extensions using the menu at the left. For each extension you see the call forwarding settings, voice mail setting and recording settings. Click the EDIT button to change the call forwarding settings.

### 5.2.1.4. My Queues

This is an overview of all queues you are currently logged in to. You can also see the amount of calls and the time of your last queue call.

### 5.2.1.5. Personal settings

This block shows your personal settings, like language settings and some phone settings and a link to all manuals. In this block you can also set your presence status: select a status and type a message.

## 5.2.2. Menu

All other settings can be accessed from the menu at the left:

- **Personal Settings:**  
An overview of your user setting like name, user name, password and language settings (see chapter 2.8).
- **Departments:**  
An overview of your department(s). See also chapter 2.2.

---

<sup>100</sup> Based on your current extensions.

- **Extensions:**  
An overview of all extensions assigned to you. The extension settings and call forwarding settings are editable by clicking on the extension (chapter 2.3).
- **Phone:**  
All information about your current phone. This is also the place to set your function keys (see chapter 2.4).
- **Queues:**  
An overview of all queues you are agent for (see chapter 3.10).
- **Conference Boxes:**  
An overview of all conference boxes of your department(s). Conference boxes created by you are editable (chapter 3.14).
- **Phone Book:**  
Your personal phone book. You can manage your personal contacts here.
- **Status options:**  
The place where you can setup your status and messages .
- **Reports:**  
Reports about calls from and to you. Recorded calls can be downloaded here.
- **History:**  
History of changes made by you.



## **Chapter 6. Manager Features**

*This chapter describes the manager features call reports. The call monitoring and call recording features are discussed here as well.*

# 6.1. Reports

---

All call information is saved by the PBX as Call Detail Records (CDR). This information is used to generate reports. Every employee can see his own personal call reports. Based on a user's role, more reports are available.

There are different kinds of reports. When you visit the Reports overview page, you can choose which report you would like to see. We describe the reports in the following paragraphs.

Reports can be viewed within the web interface, downloaded as a CSV file and can be accessed using ODBC.

## 6.1.1. Live information

All information is live generated, however it can take up to 30 seconds after finishing a call before it is added to the reports. To view a report, simply click on a report icon from the overview page and fill in the header.

Depending on the report you can select:

- Start date and End date (End date is included)
- Category (type of call)
- Direction (incoming, outgoing, both)
- Search (show only results with these digits as caller, dialed or answered phone number)

All selected settings will narrow the number of results. When done, click GENERATE and a live report will be generated.



Note that generating reports will have great impact on the performance of the Voclarion. Do NOT generate reports when much phone traffic takes place. This shall deteriorate the call quality.

---

## 6.1.2. Export

At the end of the report you can download it as a CSV formatted file. By using ODBC you can make CDR records available to other applications.

## 6.1.3. Phone numbers

Reports contain up to three different phone numbers. Every number is looked up in the phone books. If a name is found, it will be shown behind the number.

1. Caller: the caller id of person who made the phone call.
2. Dialed: the number this person dialed.
3. Answered: the caller id of the person who answered the call.

When no phone number is available, the report shows why:

- Anonymous: the caller id was blocked by the caller
- No answer: the call was not answered.
- Busy: the call was not answered because the other side was busy.
- Normal Clearing: the call was not answered but was ended as expected (for example dialing a \*-code will not result in a call).

## 6.1.4. Call Details

Some reports contain call details. The call details are a user friendly log, showing you how the call was handled from start to finish. It's a great tool for debugging your dial plan. The call details also show the *cause codes*, explaining why a call was disconnected. An overview can be found on page Error: Reference source not found. To see the details of a specific call, click on the icon () , right behind the date. You will now see the details, looking like this:

## Reports

- 07/27/09 15:32:57  
Call from 0123456789 to 0201234567
- 07/27/09 15:32:57  
Go to ivr Helpdesk-welcome
- 07/27/09 15:33:05  
Presses '2' in ivr Helpdesk-welcome
- 07/27/09 15:33:05  
Go to queue Helpdesk1
- 07/27/09 15:33:05  
0704020609 joins queue Helpdesk1
- 07/27/09 15:33:05
- 0123456789 gets music on hold (company-80-90)
- 07/27/09 15:33:10  
0123456789: music on hold ends
- 07/27/09 15:33:10  
Answered by extension 1300 (John Smith)
- 07/27/09 15:37:55  
Agent hangs up on 0123456789. Call lasts 285 secs  
(music on hold 5 secs)
- 07/27/09 15:37:55  
Call ends.

*Table 12: Call details example*

### 6.1.5. Billing

Some reports contain information about phone costs. Price information is only available when billing is activated and rate plans are applied to the trunk, company, department and/or user.

The report shows the fields Costs (cost price) and Amount (cost price + allowance). Both information I generated from the applied rate plans.

## 6.1.6. Recorded Conversations

Some reports contain recorded conversations. A recording is marked by an icon (  ), right behind the date. Click on the icon to play the conversation. Or right-click the icon to save the conversation to your computer. The Call Details contain the start and the end time of (manual) recordings.



- Before conversations are recorded, the feature must be activated. See chapter [Error: Reference source not found about Call Recording](#) for more information.
-

## 6.2. Reports

---

<b>Web interface menu</b>	<i>company reports:</i> Companies > <i>Company</i> > Reports
---------------------------	---

All reports are grouped in tabs. Click on a tab to see the reports. Reports can be found on the location below.

## Reports

### 6.2.1. Detailed report

This is a detailed overview of all calls in the specified period. The general report shows:

- Call Date: the date and time the call started. The date can be followed by one or more icons.
  - Click on the icon  to see the details of this call.
  - Click on the icon  to play a recorded call.
  - The icon  indicates an alarm call.
- Caller
- Destination
- Answered
- Call Duration
- Costs
- Total Amount of Calls

You can change the information showing from this report by changing the Show field. Select one of the following menu options:

- General report: see above.
- Most called from: the caller ids most called the company.
- Most called to: phone numbers employees called the most.
- Calls per day of week (graph): the number of calls per week day (Mon - Sun). Only week days with calls are shown.
- Calls per day of month (graph): the number of calls per month day (01 - 31). Only days with calls are shown.

- Calls per hour of day (graph): the number of calls per hour (00 - 23). Only hours with calls are shown. 9 means 9:00-9:59.

## Reports

### 6.2.2. Call Recording Samples

Select (a sample of) recorded calls. You can use additional filters to narrow the results and take a sample of all available recordings.

This report shows:

- Start time of the recording.
- Call details.
- Call recording
- The caller's phone number, the dialed number and the person who answered the call.
- Duration of the call (m:ss)
- Cost / Amount of the call.

## Reports

### 6.2.3. Outgoing by Cost Class

This report shows outgoing calls sorted by cost class (like local, mobile, toll free).

The report shows the following information per cost class:

- Totals Calls
- Total Duration
- Average Duration

## Reports

### **6.2.4. Outgoing with CMC / WrapCode**

This report shows outgoing calls by a wrap code.

The report shows the following information per cost class:

- Representative
- Wrap code
- Dialed number
- Duration of the call

## Reports

### 6.2.5. Outgoing by Number

This report shows the most dialed phone numbers. The report contains the following information:

- Dialed phone number / extension.
- Totals calls per number
- Total Duration per number
- Average call duration per number
- Totals of the information above.

### 6.2.6. Summary by day

This report shows call information, grouped by week. The report contains the following information:

- Incoming calls
  - Answered calls: the number of incoming calls answered by employees.
  - Abandoned calls: the number of incoming calls disconnected by the caller while waiting in a queue or on hold.
  - Total calls: the total number of incoming calls.
  - Total duration: the total time of all incoming calls (mm:ss).
  - Average duration: the average time of an answered incoming call (mm:ss).
- Outgoing calls
  - Total calls: the total number of outgoing calls.
  - Total duration: the total time of all outgoing calls (mm:ss).
  - Average duration: the average time of an answered outgoing call (mm:ss).
- Internal calls
  - Total calls: the total number of internal calls.
  - Total duration: the total time of all internal calls (mm:ss).
  - Average duration: the average time of an answered internal call (mm:ss).
- Totals

- Total calls: the total number of all calls.
- Total duration: the total time of all calls (mm:ss).
- Average duration: the average time of an answered call (mm:ss).

### 6.2.7. Summary by hour

This report shows call information, grouped by hour. Also averages over the time span are shown. The report contains the following information:

- Incoming calls
  - Answered calls: the number of incoming calls answered by employees.
  - Abandoned calls: the number of incoming calls disconnected by the caller while waiting in a queue or on hold.
  - Total calls: the total number of incoming calls.
  - Total duration: the total time of all incoming calls (mm:ss).
  - Average duration: the average time of an answered incoming call (mm:ss).
- Outgoing calls
  - Total calls: the total number of outgoing calls.
  - Total duration: the total time of all outgoing calls (mm:ss).
  - Average duration: the average time of an answered outgoing call (mm:ss).
- Internal calls
  - Total calls: the total number of internal calls.
  - Total duration: the total time of all internal calls (mm:ss).
  - Average duration: the average time of an answered internal call (mm:ss).
- Totals

- Total calls: the total number of all calls.
- Total duration: the total time of all calls (mm:ss).
- Average duration: the average time of an answered call (mm:ss).

### 6.2.8. Agent Performance

The report shows the activities of a queue agent during the selected time period. Use the pull down menu to select one of the agents.

- Time logged on: the amount of time the agent is logged in to a queue, including pause, talk and wrapup time (hh:mm:ss).
- Pause time: the amount of time queue pause was active (m:ss).
- Talk time: the amount of time an agent was connected to a queue call (m:ss)
- Wrapup time: the amount of time wrap up was active, manually or automatic.
- Idle time: the amount of time the agent was available for incoming queue calls. Direct calls are not taken in account.

## Reports

### 6.2.9. Abandoned Calls

This report shows information about abandoned calls for one or more selected queues within a selected date or time period. Abandoned time is defined as the time a caller has waited in a queue before hanging up without the call being answered. The report includes:

- Queue name
- Number of abandoned calls
- Average and total waiting time in (per queue) seconds
- Average waiting time grouped. The report shows the number of abandoned calls within 0-10,11-20,21-30,31-60,61-120,121-180,181-300 and after 300 seconds.

## Reports

### 6.2.10. Agent performance

The report shows the productivity of queue agents. The following information is shown:

- Agent name and extension(s).
- Number of received calls from a queue (including unanswered calls) per agent.
- Number of answered calls per agent.
- Number and percentage of unanswered calls per agent.
- Average waiting time on answered calls
- The service level: percentage of answered calls per agent.
- Totals for all queues

## Reports

### 6.2.11. Answered Calls

The Answered Calls Report shows answered call information for one or more selected queues for a selected period of time.

- Queue Name
- Answered Calls: the number of answered calls.
- Average Time: the average time before the call was answered.
- Number of Calls Answered within: the number of answered calls within 0-10, 11-20, 21-30, 31-60, 61-120, 121-180, 181-300 and after 300 seconds.

## Reports

### **6.2.12. Queue average speed of answering**

The average answering time of queue calls. The report shows:

- the number of calls
- Average speed of answering
- The telephone service factor
- The longest waiting time
- Number of abandoned calls

### 6.2.13. Queue Call Flow

The Queue Call Flow report shows how calls flow after being offered to a queue, within the selected time period.

- Offered: the total of phone calls offered to the queue (Overflowed In + Offered Direct In)
- Overflowed In: the number of calls originally offered to another queue. The other queue did not accept the calls.
- Offered Direct In: the number of calls offered to this queue directly.
- Rejected: the number of calls rejected from this queue due to congestion.
- Overflowed Out: the number of calls overflowed to another destination.
- Abandoned: the number of calls ended by the caller while waiting in the queue.
- Answered: the number of calls answered by a queue agent.

## **6.2.14. Queue Inbound Calls**

The Queue Inbound Calls Report shows incoming call information for one or more selected queues for a selected date or time period. The report contains the following information:

- **The Caller Type:** identifies whether the incoming call is an internal call (including calls initiated intern or transferred an operator) or an outside call.
- **Number of Offered Calls:** the number of incoming queue calls.
- **Number of Answered Calls:** the number of incoming queue calls that have been answered by an agent.
- **Number of Lost Calls:** the number of incoming queue calls that not have been answered. Lost calls include abandoned calls, rejected calls and overflow (timeout).

### 6.2.15. Queue Information

The Queue Information Report shows queue statistics for one or more selected queues for a selected time period. The report shows the following information:

- Queue Name: the chosen queue names
- Maximum Queue Size: the maximum number of calls that could be queued (0 = unlimited). This is not the maximum number of callers waiting.
- Longest Queue Size: the maximum number of calls queued at the same time within the selected time frame.
- Average Queue Time Answered: the average time a call was queued before being answered by an agent.
- Average Queue Time Abandoned: the average time a call was queued before it was abandoned by the caller (hung up).
- Longest Queue Time Answered: the longest time a call was queued before being answered by an agent.
- Average Queue Time Abandoned: the longest time a call was queued before it was abandoned by the caller (hung up).
- Queued Calls: the total number of calls queued. This number includes calls abandoned and answered.
- Totals and averages of the information before.

### 6.2.16. Queue performance

The Queue Performance Report shows queue performance and service level of the selected queues for a selected period of time.

At the header you can make some additional settings:

- **Threshold:** set the threshold (minutes and seconds) for the service level, see below.
- **Show the information grouped per day, week or month,** within the given time frame. The selected time period will change according to the week or month setting. NB: a too wide time frame results in a “too many results” - error.

The report shows the following information:

- **Queue Name:** the chosen queue names
- **Extension:** the queue's extension.
- **Number of calls:** total number of calls offered to the queue.
- **Answered:** the number of calls answered by a queue agent.
- **Timeout:** the number of calls that left the queue because the maximum time has expired or the maximum number of calls had been exceeded. This number does NOT includes abandoned calls.
- **Abandoned:** the number of calls disconnected by the caller while waiting.
- **Average waiting time:** the average time (m:ss) a call was in the queue.
- **Service level:** percentage of answered calls within the given time (threshold).

## Reports

### 6.2.17. Queue pause reasoning listing

This report shows the queue pause with reason.

- The name of the agent
- The reason number
- The time the agent has been on pause

### 6.2.18. Queue Service level

This report shows the service level per day, for the given time period. You can narrow down the results by selecting a queue and an inbound number. The report shows:

- Total calls: the number of calls for this queue.
- Answered: the number of calls answered by a queue agent.
- Unanswered: the number of calls not answered by an agent (overflow + abandoned calls).
- Average waiting time: the average time (m:ss) a call was in the queue.
- Service level: percentage of answered calls per day.

## 6.2.19. Queue Unoccupied Time

This report shows the time periods (within the selection) when there are no agents logged in to the selected queue. The report shows the following information:

- Date, start and end time of the period with no agents logged on.
- The duration no agents were logged in (hh:mm:ss)
- Calls missed. Only available if the queue accepts calls when no agents are logged in.

## Reports

### 6.2.20. Queues by Day

This report shows queue information, grouped by day. Use the pull down menus to select a time period, a queue and an inbound number. The report contains the following information:

- Total calls: the total number of calls for this queue.
- Answered: the number of calls answered by a queue agent.
- Abandoned: the number of calls disconnected by the caller while waiting.
- Timed out: the number of calls that left the queue because the maximum time has expired or the maximum number of calls had been exceeded. This number does NOT include abandoned calls.
- Total duration: the total time of all answered calls (m:ss).
- Average duration: the average time of an answered call (m:ss).
- Total waiting time: the total waiting time (m:ss) for all calls in the queue.
- Average waiting time: the average time (m:ss) a call was in the queue.
- Totals of the information above.

## Reports

### 6.2.21. Queues by Hour

This report shows queue information, grouped by hour. Use the pull down menus to select a time period, a queue and an inbound number. The report contains the following information:

- Total calls: the total number of calls for this queue.
- Answered: the number of calls answered by a queue agent.
- Abandoned: the number of calls disconnected by the caller while waiting.
- Timed out: the number of calls that left the queue because the maximum time has expired or the maximum number of calls had been exceeded. This number does NOT includes abandoned calls.
- Total duration: the total time of all answered calls (m:ss).
- Average duration: the average time of an answered call (m:ss).
- Total waiting time: the total waiting time (m:ss) for all calls in the queue.
- Average waiting time: the average time (m:ss) a call was in the queue.

Totals of the information above.

### 6.2.22. Queues by Week

This report shows queue information, grouped by week. Use the pull down menus to select a time period, a queue and an inbound number. The report contains the following information:

- Total calls: the total number of calls for this queue.
- Answered: the number of calls answered by a queue agent.
- Abandoned: the number of calls disconnected by the caller while waiting.
- Timed out: the number of calls that left the queue because the maximum time has expired or the maximum number of calls had been exceeded. This number does NOT include abandoned calls.
- Total duration: the total time of all answered calls (m:ss).
- Average duration: the average time of an answered call (m:ss).
- Total waiting time: the total waiting time (m:ss) for all calls in the queue.
- Average waiting time: the average time (m:ss) a call was in the queue.
- Totals of the information above.

### 6.2.23. Notifications

This report shows all notifications for the given time frame. The report includes:

- Start Time
- Sender of the message
- Location
- Target
- Shortname
- Message
- Status
- The name of the person who resolved the matter.
- Duration of the notification.

## Chapter 7. Company Advanced Settings

*The company advanced settings are a group of settings you probably won't need every day. They include importing data for the dial plan, setting contact types, setting user permissions and more.*

## 7.1. File Import

---

**Web interface menu**      Companies > *Company* > Advanced > File Import

After registration of the new company, you can add departments, users, phones and extensions. At first time setup it is convenient to use a spreadsheet to import this information. In this paragraph we will show you how to import large quantities of data by spreadsheet (in CSV file format). Later on we will describe how to add a department, extension, user or phone by hand.

There are example spreadsheets available in the PBX web interface which you can download and change. The PBX offers different kinds of spreadsheets for importing different kinds of information.

### 7.1.1. Creating a Spreadsheet

At the location as described before, you can download an example spreadsheet. After downloading, it you can edit the content with a spreadsheet program<sup>101</sup> and upload it to the PBX.

**Web interface menu**      Companies > *Company* > Advanced > File Import

Go to the location described above. Here you can find a link to example spreadsheets. You can download the files by right-click on the link and choosing “Save link as”<sup>102</sup>. Save the file on an location that is easy to find, your desktop for example. The link will save it as a CSV file.

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<sup>101</sup> You can choose your favorite program, like Microsoft Excel or OpenOffice.org Calc. Make sure the software is able to import and export CSV (text-) files.

<sup>102</sup> The exact description may vary in some browsers.

## File Import

### 7.1.1.1. Opening the Spreadsheet

The downloaded file is in CSV-format (comma separated value). If you open the file in a text editor you will see it is *human readable*. This makes it very easy to edit and therefore CSV is supported by almost all spreadsheet software. In a CSV all fields are saved as text, separated by a comma. The first line is the header, the next lines are content.

In most cases your spreadsheet software will start automatically when you double-click the file. If not, you have to start your spreadsheet software manually and open the CSV file. If all goes well, you'll see a few comments on top (starting with #<sup>103</sup>), and below the comments some headers nicely formatted in columns<sup>104</sup>. On the rows below the header you can add the requested information. For an example see table 13.

If this is not the case, for example all headers are put into one cell, your software doesn't read the file correct. Close the file *without saving* it, and try the following steps:

1. Rename the file to \*.txt
2. Open your spreadsheet software
3. Choose "Import" or "Open" form the menu. If you can choose a file type, choose "CSV – comma separated value" or "Text". Open the file.
4. In most cases an import wizard starts which will ask you some information about the file. Often the software will show you a preview when editing the separator settings. Use the settings below:
  - separated by: “,”
  - text delimiter: (none)

---

103 All texts starting with # will be ignored. You can use it to add comments. Start every comment cell with #.

104 Don't worry if you cannot read the whole text because the cell is not wide enough. This means the import was successful. You can change the column width if you like. When you select the cell you can read all text.

- from row: 1

5. If this doesn't provide the desired effect, please check your software manual for more information.

### 7.1.1.2. Editing the Spreadsheet

The spreadsheet contains some comments (start with #) which will be ignored by the PBX. The comments explain how the user the spreadsheet. Below the comments you will find the headers of the columns, as described below.

Each row is a set of information. For example if you add a user to the company spreadsheet (see table 13), all information about a particular user is on one and the same row.

# this file was exp...		
#		
# When importing...		
#department_name	department_short..	department mana..
sales	sales	yes
sales	sales	yes
sales	sales	no
public relations	pr	yes
public relations	pr	no
public relations	pr	no

Table 13: Detail example company spreadsheet



All extensions entered in the Inbound Numbers sheet must also be present in the Company sheet/PBX. If not, extensions of the Inbound Numbers sheet will not be recognized and imported by the PBX.

## File Import

### 7.1.1.3. Saving the Spreadsheet

When finished, save the spreadsheet as CSV or text file, not as the software's own format. Choose SAVE AS... or EXPORT, depending on the software you use. For more information check the manual of your spreadsheet software.

### 7.1.2. Importing Data (File Import)

Select the company for which you want to import a spreadsheet. Always start with the Company spreadsheet , so all extensions are already present in the PBX.

It is not possible to undo an import. Once you have chosen to overwrite the existing data, you cannot retrieve it anymore or undo the import. So be careful when selecting “Keep Old Data” or “ Delete Old Data”.

**Web interface menu**      Companies > *Company* > Advanced > File Import

Click BROWSE and select the saved spreadsheet.

1. To keep the data already in the system, check the radio button KEEP OLD DATA. To overwrite all data already in the system, check the radio button DELETE OLD DATA.
2. Click SAVE.
3. The spreadsheet will now be imported and processed. When an error occurs, a warning message will be displayed. In that case no data will be imported. You can try again.



Make sure that you always start with importing the Company spreadsheet with the extensions, followed by the Inbound Numbers sheet. The company sheet contains the extensions and must be present before you can import inbound numbers.

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## 7.2. File Export

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**Web interface menu**      Companies > *Company* > Advanced > File Export

Information can also be exported. Select the filename which you would like to receive. The CSV file is now downloaded.

**Web interface menu**      *Company* > Advanced > External sites

## 7.3. Contact Types

---

**Web interface menu**      *Company* > Advanced > Contact Types

With contact types you are able to group contacts, like “customers” and “VIP”. This can be used for:

- Assigning a ringtone. By using different ringtones, employees will be able to hear what kind of contact is calling. For more information about ringtone support, see chapter 7.6.3.
- Contact types can be used to set different call forwarding for different types of callers. For example: when a customer calls an employee, but he isn't able to answer his phone, the customer is connected to the operator. When a colleague calls he is forwarded to the voice mail.

There are two default contact types available: *default* and *internal call*. As administrator you can add unlimited new contact types.

### 7.3.1. Internal call

An internal call is a call between two employees. This can be a call between two extensions but also a call from/to an employee's mobile phone<sup>105</sup>. Internal calls are not recorded by default. Keep this in mind when testing call recording with your mobile phone. Internal call recording must be activated first.

### 7.3.2. Assign Contact Type

You can create an unlimited amount of contact types. From the phone book you can assign contact to your contact types. We strongly suggest to keep the number of contact types as low as possible.

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<sup>105</sup> The mobile number has to be added to the employee's settings.

### 7.3.3. Creating a Contact Type

**Web interface menu**      *Company* > Advanced > Contact Types

1. Press the ADD (+) icon.
2. Enter a name for the contact type
3. Enter a description.
4. Click the SAVE button.

## 7.4. Company Phone Book

---

The PBX has two phone books: a company wide and a personal phone book. All contacts added to the company phone book are available (when logged into the web interface) for all employee. The company phone book contains all phone numbers of colleagues by default. A user cannot edit this phone book by default<sup>106</sup>, only a system administrator and the company manager can do this.

<b>Web interface menu</b>	Companies > <i>Company</i> > Advanced > Company Phone Book
---------------------------	--

The phone books are sorted alphabetically. If the phone book contains a lot of contacts, the top contains a bar to select addresses. You can click on a specific character to see all names of contacts starting with that character.

The phone book contains an ALL function. When clicked, the web interface shows the complete list of phone numbers, sorted by name. This function is only visible when the phone book does not contain more than 1000 phone numbers.

For each incoming call the PBX checks both phone books. If a corresponding number is found<sup>107</sup>, the name is shown on the display of the ringing phones<sup>108</sup>.

### 7.4.1. Fast Dial Numbers

When adding a new contact, you have to supply a fast dial number, which is a short number to dial the phone number. The minimum required length of a fast dial numbers is two digits, however we recommend using at least three digits. Some

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<sup>106</sup> You can use roles to change this.

<sup>107</sup> You can, for example, copy your contacts database overnight to the PBX. If a customer calls, this name will appear on the display of all phones. How this is done depends on several settings. Contact your dealer for more information.

<sup>108</sup> Depending on the phone type.

fast dial numbers, like emergency numbers cannot be used. Fast dial numbers can starting with "0". In this case the 0 has to be dialed too.

In some companies the 9 is used to call the attendant. Because fast dial numbers have at least two digits, 9 can't be used as a fast dial number. Voclarion now translates dialing a 9 to 99, which can be used as a fast dial number.

## 7.4.2. Adding Contacts to the Company Phone Book

This paragraph describes how to add contacts one by one. You can also import a data sheet with contacts. See chapter 7.1.

**Web interface menu**      *Company > Advanced > Company Phone Book*

1. Click ADD.
2. Enter the following information:

FAST DIAL NUMBER:      The system does a suggestion. This can be changed. Choose a fast dial number 2-16 digits, and can starting with a "0". Also the wildcards '%' and '\_' are allowed.

## Company Phone Book

### PHONE NUMBER:

- Is dialed by the PBX when a user dials the fast dial number
- Is used as caller ID lookup for incoming calls.
- It is used to connect callers to a Direct Contact.

It can be:

- external phone number
- internal extension
- PBX \* code
- Skype address

There are two wildcard characters:

You can use the '%'. This wild card stands for any sequence of numbers. For example: "0032%", selects all numbers starting with "0032" (Belgian phone numbers) like "0032555678".

You can also use the '\_' (underscore). This wild card stands for any single digit. For example: "10\_\_" allows all 4-digit number which start with '10'.

### NAME:

Enter a name for this contact.

### CONTACT TYPE:

Choose one of the available contact types. Contact types can have specific ringtones and call forwarding options.

### ADDRESS, ZIP, CITY,

Address information.

### COUNTRY

- CUSTOMER ID: This field can be used to specify a customer's ID. This ID corresponds with your database and can be sent to the Voclarion Switchboard, which will include it while opening an external program for incoming calls (See Switchboard Manual).
- DIRECT CONTACT 1,2,3: Enter any phone number or extension. When the Direct Contact feature is used, the caller will directed to this number.
- USER FIELDS 1-5: Up to 5 personal user fields (see chapter 2.9 for more information about user fields).

3. Click SAVE.

### 7.4.2.1. Fast Dial Numbers Example

To create a one-button intercom<sup>109</sup> to your secretary's phone, do the following:

1. Create a fast dial number like 1002 with phone number \*361234. This will activate the intercom feature(\*36) and dial the extension of your secretary (1234).
2. Assign the created fast dial number (1002) to a key on your phone for easy use.



Note that certain fast dial numbers and extensions are not allowed, in order to reach emergency services and use of internal operations.

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<sup>109</sup> Not all phones support intercom/paging, please contact your dealer for more information. For more information about page groups / intercom see the Operation Manual.

### 7.4.3. Editing Contacts

**Web interface menu**      *Company > Advanced > Company Phone Book*

1. Click on the fast dial number of the contact you would like to edit.
2. Click EDIT.
3. Edit the contact and click on SAVE to store the changes, or click REMOVE to delete the contact.

### 7.4.4. Personal Phone Book

A user can also add contacts to a personal phone book. These contacts are only visible to the user.

Keep in mind that fast dial numbers of personal contacts are overridden by company fast dial numbers.

To add contacts to your personal phone book, open your personal phone book. Personal Phone Book contacts are visible on a few phone types. Ask your phone supplier which phone types support this.

### 7.4.5. Direct Contact

With Direct Contact you can connect a caller to a pre-defined contact within your company. For example when a customer calls, he can be connected with his account manager automatically without speaking to some one else first (e.g. the operator). Setting up Direct Contacts is a two step process:

1. In the phone book you can provide three Direct Contacts for every entry. The PBX looks up the caller's ID in the phone book. If it finds one and also finds an extension for Direct Contact, the caller will be connected to

this extension<sup>110</sup>. If the PBX cannot find a matching caller ID, the caller will be connected to the next extension set in the dial plan. So, do not forget to supply a fall back extension in case no caller ID is found. This feature works only on incoming calls.

2. In the dial plan (for example within an IVR-menu) you can specify which of the three Direct Contacts you would like to use on the current call. You do this by specifying a module. The next examples will clarify this.

Module	Looks up from the phone book
*9000	Direct contact 1 till 3 <sup>111</sup>
*9003	Direct contact 1
*9004	Direct contact 2
*9005	Direct contact 3

Table 14: module vs direct contact

### 7.4.5.1. Setting Up Direct Contact

<b>Web interface</b>	Company > Advanced > Company Phone Book
<b>menu</b>	My Voclarion > Phone Book > tab Company Contacts

1. Select a contact from the phone book.
2. Enter an extension (or phone number) for field "Direct Contact 1, 2 and/or 3".
3. Click SAVE.  
Now activate Direct Contact.
4. Go to the dial plan and select an inbound number or IVR menu.
5. Enter a module as first extension for an inbound number, as first extension in an IVR menu, or as first extension after a key press.

<sup>110</sup> Keep in mind that if this extension has a call forwarding setting, the PBX will honor them, which can lead to unexpected results.

<sup>111</sup> The first set will be used

## **Company Phone Book**

6. Enter a second (fall back) extension in case a direct contact cannot be found.

**Example: UK customers connected with English speaking agents**

We would like to direct all calls from the UK to a group of agents who speak English.

- UK numbers start with 0044.
- The extension of queue “English” is 500.
- If not calling from the UK, the customer will be connected with the operator: extension 203.

**How to configure:**

<b>Company Phone Book</b>	
Fast Dial Number:	Enter a fast dial number
Phone Number:	0044% (% is a wildcard)
Name:	UK numbers
Direct Contact 1:	500 (queue “English”)

<b>Inbound Number</b>	
Inbound Number:	The main number of the company
1 <sup>st</sup> extension:	*9003 (direct contact)
2 <sup>nd</sup> extension:	203 (Operator)

## Company Phone Book

### Example: A customer connected to his account manager

Sometimes it is convenient to connect a customer directly to his account manager when calling for support. Customers without a specific contact will be immediately connected to the second extension: the support desk queue.

### How to configure:

<b>Company Phone Book</b>	
Fast Dial Number:	Enter a fast dial number for this customer
Phone Number:	Caller ID of the customer
Name:	Name of the customer
Direct Contact 1:	The extension of the account manager

<b>Inbound Number</b>	
Inbound number:	Number of the support desk
1 <sup>st</sup> extension:	*9003
2 <sup>nd</sup> extension:	extension of the support desk queue.

**Example: Customer connected with the department contact manager**

Usually a customer has different contacts on different departments. For example:

- If the customer calls sales, he needs to speak to his sales contact (extension 250),
- If the customer calls finance, he needs to speak to his finance contact (extension 260)
- If the customer has a technical question he needs to speak with the support desk (extension 270).

**How to configure:**

<b>Company Phone Book</b>	
Fast Dial Number:	Enter a fast dial number for this customer
Phone Number:	Caller ID of the customer
Name:	Name of the customer
Direct Contact 1:	Extension 250: Sales contact
Direct Contact 2:	Extension 260: Finance contact
Direct Contact 3:	Extension 270: Support contact

<b>Inbound Number</b>	
Inbound number:	Company main number
1 <sup>st</sup> extension:	300 (IVR menu "Welcome")

<b>IVR Menu "Welcome"</b>	
Key press 1 (Sales)	
1 <sup>st</sup> extension:	*9003 (refers to Direct Contact 1)
2 <sup>nd</sup> extension:	Extension of Sales queue
Key press 2 (Finance)	
1 <sup>st</sup> extension:	*9004 (refers to Direct Contact 2)
2 <sup>nd</sup> extension:	Extension of Finance queue
Key press 3 (Support Desk)	
1 <sup>st</sup> extension:	*9005 (refers to Direct Contact 3)
2 <sup>nd</sup> extension:	Extension of Support queue

### 7.4.5.2. Blocking Specific Incoming Calls

It is possible to block incoming calls with a specific caller ID. The PBX can identify the caller and send the call directly to a specific extension that is set to busy tone. Or you can send the call to a voice mail. Per caller ID you need to specify a destination extension.

**Web interface menu**

*Company > Advanced > Company Phone Book*

*Company > Inbound Numbers*

1. Go to the Company Phone Book and add a new contact.
2. Enter the phone number of the caller you'd wish to block.

Enter for field DIRECT CONTACT 1 an extension set to busy tone.

3. Go to the inbound number and enter as first extension **\*9000**, instead of the default extension. This will lookup the Caller ID and if found redirect. The default extension can be set at the second and/or third extension.

The PBX looks for the caller ID in the phone book. If a match is found, the PBX connects the call directly to the extension set for Direct Contact 1. This can be a free extension, that is set to busy tone or any extension such as one with voice mail activated.

## 7.5. Presence

---

For each company, department and user it is possible to create and assign statuses. The default status is Available. Statuses made in the Company settings are also available for departments and users. Presence information is shown in the switchboard and PBX Manager.

### 7.5.1. Creating a Status

**Web interface menu**      *Company > Advanced > Presence*

1. Click the ADD icon.
2. Enter a status name such as “Lunch” or “Coffee break”.
3. Click the SAVE button.

To delete an existing status, press the DELETE icon in front of the status.

### 7.5.2. Choosing a Status

**Web interface menu**      *My Voclarion > Dashboard > Personal Settings*

1. Click on the current selected status.
2. If there are more statuses available, a pull down menu will appear. Choose one and press the OK button.

### 7.5.3. Presence Messages

Besides setting your status you can also assign a personal message to that specific status. This can be done at the dashboard of a user.

**Web interface menu**      My Voclarion > Dashboard > Personal Settings

1. Select the right status.
2. Type a self chosen text in the field for "What are you doing?".
3. Press the OK button.
4. If you want to change the text, replace the current message by the new text and press the OK button.

## 7.6. Phone Types

---

**Web interface menu**     *Company > Advanced > Phone Types*

Phone Types shows an overview of all supported telephones. If you click on a phone, you can see basic information about how to use this phone.

For each phone type there are one or more tabs available:

- Overview
- Function Keys
- Ringtones
- Call
- Advanced

### 7.6.1. Overview

How to use your phone, the very basics:

- Login to a phone
- Answer a ringing phone of a colleague
- Dial a number
- Set up voice mail and listen to voice mail messages
- Put a call on hold
- Join a conference
- Use the Paging feature
- Use Dial by Name
- Use the Call Parking feature
- Use a fast dial number

### 7.6.2. Function Keys

The function keys are programmable for all phones of a the same type (if available). Once programmed per phone type, it cannot be changed by an

## Phone Types

individual phone. See chapter Function Keys of the Operation Manual for more information.

### 7.6.3. Ringtones

For each phone type you can assign ringtones. After saving, it is applied to each phone of this type. Some of the phone types must be rebooted first. For each contact type you can assign a preferred ringtone. As company manager/administrator you can create more contact types.

Ringtones are not supported by all phones. For more information, ask your phone supplier.

**Web interface menu**      *Company* > Advanced > Phone Types

1. Click on the icon of the phone type for which you want to set ringtones.
2. Click the RINGTONES tab, when available.
3. Click EDIT.
4. Choose a ringtone for each contact type.
5. Click SAVE.



All calls coming from a queue are marked as external call. So calls from a queue can only have one ringtone.

---

### 7.6.4. Call Settings

The call settings of a phone type are the same as for each individual phone type. Each phone type has its own different settings. The configuration of a phone type is applied to all new phones. To copy these settings to all existing phones of this

type in the PBX, you can use the COPY TO ALL EXISTING PHONES button. See chapter Phones of the Operation Manual for more information about phone settings.

### **7.6.5. Advanced Settings**

The call settings of a phone type are the same as for each individual phone type. To copy these settings to all existing phones of this type in the PBX, you can use the COPY TO ALL EXISTING PHONES button.

See chapter Phones in the Operation Manual for more information about phone settings.

## 7.7. Plugin

---

Plugins can add third party functions to the PBX. In the dialplan you can connect a call to the extension of the plugin to activate the function. Some plugins have an \*-code to activate the function. For more information about a plugin, consult the developer.

To add a new plugin, press the button ADD (+) . Supply the next information:

<b>Web interface menu</b> <i>Bedrijven &gt; Bedrijf &gt; Geavanceerd &gt; Plugins</i>
---

**Veld**

NAME:

DESCRIPTION:

EXTENSION:

CONTEXT:

**Omschrijving**

The name of the plugin

Description for this plugin

Extension for this plugin

This is the context name for the plugin

## 7.8. Client matter code

---

A Client Matter Code (CMC) adds an certain reference code to a call, for example a client ID. The CMC can be added from the Switchboard software.

toegevoegd aan een gesprek middels de Switchboard bedienpost. Click on the button ADD (+) to create a new Client Matter Code.

<b>Field</b>	<b>Description</b>
<u>NAME</u>	Name of the CMC
<u>DESCRIPTION</u>	Description of the CMC
<u>CLIENT MATTER CODE</u>	This is the CMC, in most cases a numbver like a contact ID.

<b>Web interface menu</b>	Bedrijven > <i>Bedrijf</i> > Geavanceerd > Client matter code
---------------------------	---

## 7.9. Queue pause Reason Code

---

A queue pause reason code is a code which indicates the reason why an agent took a pause from the queue. Click ADD (+) to add a new one. You have to supply the following information:

<b>Field</b>	<b>Description</b>
<u>N</u> AME:	Nome for this code
<u>S</u> HORT <u>N</u> AME:	Short name for this code
<u>C</u> ODE:	The code, normally a number
<u>D</u> ESCRIPTION:	The description of this code

<b>Web interface menu</b>	Companies > <i>Company</i> > Advanced > Queue reason codes
---------------------------	--

## 7.10. Function Key templates

---

With Function Key Templates you create a template containing a set of Function Keys. Once done, you can assign this template to phones, so you don't have to configure all keys manually. Because the (number of) available Function Keys are different on every phone, you have to do a key mapping for every type of phone. So if you have a template called Sales, you have to create a key mapping for Aastra 6751i-phones, Aastra 6753i-phones, etc.

If there are still Function Keys available on the phone after applying the template, users can apply a function themselves.

<b>Web interface menu</b> Companies > <i>Company</i> > Advanced > Function Key templates
--

The page shows an overview of available templates. The first template is called "Default" and cannot be deleted. To create a new template:

1. Click on ADD (+)
2. Supply a template name, for example "Sales"
3. Click SAVE

A new empty template is created and you will be taken back to the overview screen. In the next paragraph we add keys to the template. When done you have to assign the temple to users.

## Function Key templates

### 7.10.0.1. Defining Function keys

**Web interface menu**      Companies > *Company* > Advanced > Phone types

To add Function Keys to a template for a specific phone, first select the phone type. You'll see information about this phone on your screen. At the top of the page look for the tab "Function Keys".

1. Open the tab. An overview of all available keys is shown. By default all keys have "No function".
2. Select the template you would like to edit from the pull down menu at the top right. The changes you are about to make will apply to this template in combination with this type of phone.
3. Click EDIT () to change the keys.
4. Choose for each button a label, a function and provide a phone number / extension if needed. The label will be displayed on the phone screen (when available).
5. Click SAVE.
6. Reboot the phone before use.



When you have created a new Function Key Template, you have to assign this template to a user.

---

## 7.11. Company Public Key

---

The public key is used to authenticate IAX and SIP sessions. The key cannot be changed. RSA is an algorithm for public key encryption. It was the first algorithm known to be suitable for signing as well as encryption, and one of the first great advances in public key cryptography. RSA is widely used in electronic commerce protocols, and is believed to be secure given sufficiently long keys and the use of up-to-date implementations.

**Web interface menu**

Companies > *Company* > Advanced > Public Key

## 7.12. Server Management

---

The Server Management page offers options to restart Voclarion services for the company. There are also services available for the whole system, for more information see chapter System Settings of the Installation Manual.

### 7.12.1. Restart Configuration Daemon

If you have made changes to the configuration, but changes are not automatically activated you can force a restart of the configuration daemon. It will stop, start again and will reload all settings.

### 7.12.2. Restart All Phones

Sometimes it can be necessary to restart phones. Phones will automatically restart after the current call is finished. After restart the phones will check for new firmware and install it if available.



## **Chapter 8. Download Center**

*The download center contains Voclarion manuals and software you can use with your PBX.*

## 8.1. Introduction

---

The Download Center contains a wide variety of files for you to download. Here you can find:

- All sorts of (end user) manuals<sup>112</sup>
- Software included with Voclarion
- Third party software you can use with Voclarion
- Other information

All files are grouped by language and Voclarion version. We update this information on a regular basis so you always download the latest software and manuals.

<b>Web interface menu</b> Download Center
---

Most manuals are in the popular PDF file format. The Portable Document Format (PDF) is a file format created by Adobe Systems for document exchange. PDF is used for representing two-dimensional documents in a device-independent and display resolution-independent fixed-layout document format. To view PDF documents you need the free Acrobat PDF reader, which you can download at <http://www.adobe.com>.



- To view and download information on this page you need an Internet connection.
  - The use of third party software is at your own risk and is not supported by Voclarion. Third party software may require additional license fees. These are not included with the Voclarion by default. Contact the software distributor for more information.
- 

<sup>112</sup> Manuals are also available in print. Contact us for more information.

## Introduction



## **Chapter 9. Message Center**

*Read news about Voclarion*

## 9.1. Introduction

---

As part of our support we inform you about new features and updates. We do this by sending small text messages to the Voclarion PBX manager. A new message will be indicated by a blinking icon on top of the page. You'll need a working Internet connection.

### 9.1.1. Message Center

When you click on the (blinking) icon, you will be taken to the Message Center. Here you can see the messages of the last 30 days. The number of messages is set on the Manager Features.

### 9.1.2. Disabling Messages

To block all (upcoming) messages, set `RSSFEED_MAXITEMS` to "0". No messages will be shown anymore. See the Manager Features for more information. Changes will take place the next time you login.



## **Chapter 10. PBX Features**

*This chapter describes how to use common PBX features.*

## 10.1. Introduction

---

This chapter will discuss some common features, which can be used by dialing special \*-codes.

### 10.1.1. Using Your Phone or the PBX Manager?

Some features like call forwarding can be set from the graphical PBX Manager, but also by dialing \*21 from your phone. In most cases it makes no difference in which way you change settings and you can use them both if you like.

However, in some cases the PBX Manager is more advanced and more efficient. If we look at call forwarding, you can set different types of forwarding and there are more options available. If there is a difference between the PBX Manager settings and the settings by phone we will mention this.

### 10.1.2. Phone Features vs. Voclarion Features

There are two kinds of features: *phone features* and *Voclarion features*. Before we start it is good to understand the difference between phone features and Voclarion features. Some features such as Call Forwarding can be controlled by the Voclarion (we call these Voclarion features) and by your phone (we call these phone features).

Action	Controlled by
Settings in PBX Manager	Voclarion
* and # codes entered on your phone	Voclarion
Setting from your phone menu	Phone

Table 15: Phone vs Voclarion features

- I. Phone features are stored on your phone and will be lost after the phone resets. Phone features do not work when the phone is unreachable and do not transfer if you login on to another phone. The Voclarion is not aware of these settings.

## Introduction

For example: using the menu on your phone you can forward calls to another extension.

- II. Voclarion features are stored on the Voclarion and will still work when a phone is unavailable or is reset. When you login on another phone, settings will stay active. For example: you can use the Voclarion PBX Manager or \*21 to forward calls to another extension.



This manual only describes Voclarion features. The manufacturer's manual included with your phone describes Phone Features.

We strongly suggest you to use Voclarion Features only. The Voclarion is not aware of phone settings which can result in hard to track errors.

---

## 10.2. Phone Login and Logout (\*1)

---

To use your phone, you have to log in first. If you try to dial a number and you are not logged in, the system will ask you to enter your login and PIN.

It is not possible to login a user which has no extension assigned . It is also not possible to login with an extension of a ringgroup or IVR menu.

When you have two phones linked to each other (twinning), you only have to log in on one phone. The other will be logged in (or out) automatically.

### 10.2.1. Login

Code	Description
*1	Log in. Enter the <b>phone login code, PIN, #</b> . To log out, dial *1.

Enter your login code (usually this is your extension number, unless set differently), directly followed by your PIN and a '#'. The phone will confirm your login or repeat the question to login if the entry is not correct. After three failed attempts, the account is locked.

If you log in to a phone while another user is logged in, the current user will automatically log out. Also, if you are already logged in on another phone, you will be automatically logged out from the other phone.

Each time after logging in, your phone restarts and check for updates. Depending on the phone type this process takes from a few seconds till up to a few minutes.

### 10.2.2. Logout

Code	Description
*1	To log out, dial *1 again

### Phone Login and Logout (\*1)

After you dial \*1 the phone will respond with “Goodbye”. All calls to the phone will now be handled according to the call forward setting 'logged out'. Logging out can be blocked from the user settings.

### 10.2.3. Switch user

Code	Description
*1*	Log out current user, prompt for new user.

## 10.3. ringgroup Login and Logout (\*42)

---

As ringgroup agent you do not need to log in to your ringgroup. After phone login, you are automatically logged on to all your ringgroups. If you want to log off and on to one specific ringgroup, you can use the ringgroup logout and login. ringgroup pause can be used if you want to take a break.

Key	Function
*40xxxx	Log in to the ringgroup. The Voclarion will respond accordingly.
*41xxxx	Log out of the ringgroup. The Voclarion will respond accordingly.
	(xxxx = ringgroup extension)

### 10.3.1. ringgroup Pause

While logged in to one or more ringgroups, you may want to take a break without logging out completely. The pause feature makes it possible to log out from *all* ringgroups temporarily. If pause is enabled, you will not receive any calls from ringgroups. You can still be called on your extension directly.

There is a ringgroup pause function key available to program on the phone. Pressing it toggles the ringgroup pause. When you activate DND on your phone, ringgroup pause will automatically become active.

ringgroup pause is automatically enabled after each finished call, when variable wrap-up time is enabled.

Key	Function
*42	Enable/Disable ringgroup Pause
*42*	ringgroup status
*421	Take a pause – log out of all ringgroups
*420	Resume – login to all ringgroups

## 10.4. Ringgroup Login and Logout (\*43)

---

As ringgroup agent you do not need to log in to your ringgroup. After phone login, you are automatically logged on to all your ringgroups. If you want to log off and on to one specific ringgroup, you can use the ringgroup logout and login. Note there is no such thing as ringgroup pause.

Key	Function
*43xxxx	Log in to the ringgroup. The Voclarion will respond accordingly.
*44xxxx	Log out of the ringgroup. The Voclarion will respond accordingly.
	(xxxx = ringgroup extension)
*44*	Ringgroup status

## 10.5. Placing a Phone Call

---

To place a call, simply enter the number on your phone. Some phones will automatically dial the number after a few seconds. Other phones request that you press 'enter' or '#' after you are finished, just like on your mobile phone. Check your phone manual for more information.

### 10.5.1. Dialing an Extension

To call any extension simply dial the extension as described before. You can call virtually any extension, like a phone, a ringgroup or an IVR menu.

### 10.5.2. Dialing an External Number

To call an outside phone number, you *don't* have to call a prefix like "0". The Voclarion will figure oVoclarionut itself if you have dialed an extension or an external number. If the number of digits you dialed is the amount as the number of digits of your extensions, the Voclarion will try to call an extension. Otherwise the Voclarion will open an outside line.

#### 10.5.2.1. Forcing Outside Phone Numbers (\*\*\*)

If you want to dial an external phone number that has the same amount of digits as your extensions, you can force the Voclarion to call an external number by using the \*\*\* prefix. You can also use this prefix in your phone book.

For example:

- 1888: extension of your colleague
- 1888: special telephone information number

If you call 1888 the Voclarion will call your colleague because the PBX thinks you are calling an extension. If you call \*\*\*1888 the Voclarion will open an external line and dial the number.

## 10.6. Manual Recording (\*3)

---

With the call recording feature you can activate and deactivate call recording within a call. Press \*3 or use a function key on your phone. The recording can only be activated during an active call, it is not possible to enable recording before starting the call. To use this feature, the correct user permissions must be set and call recording must be disabled on the extension.

To only monitor a recording program \*3\* <ext> to a function key.

## 10.7. Call Pickup (\*8)

---

If a colleague's phone within the same pickup group rings, you can answer it from your own phone.

1. To pickup a random ringing phone from the pickup group, dial **\*8**.
2. To answer a ringing phone of a specific colleague within the same pickup group, dial **\*8<extension\_colleague>**.

## 10.8. Call Forwarding

---

When you are away or on the phone you can redirect calls. This can be done by the Voclarion PBX web interface or by using \*-codes on your phone (see below). In the PBX Manager you can select which contact types are forwarded when using \*-codes. Below you can see an overview of forwarding options.



- Some call forwarding features such as Ask Caller, Find Colleague, Mobile are only available from the PBX Manager. Go to the PBX web interface to activate these features.
  - All codes can be programmed on a phone's function key.
- 

### 10.8.1. Call Forward Unconditional (\*21)

Call forward unconditional will forward all calls in any case<sup>113</sup> to another extension or phone number. You can change this setting from your own phone.

<b>Unconditional Call Forward</b>	<ul style="list-style-type: none"><li>• Status: <b>*21*</b></li><li>• Set to Ringing: <b>*210</b></li><li>• Enable: <b>*211</b></li><li>• Always Forward to Operator: <b>*212</b></li><li>• Always Forward to Voice Mail: <b>*213</b></li><li>• Always Give a Busy Signal: <b>*214</b></li><li>• Always Bounce to the PBX: <b>*215</b></li><li>• Set Phone Number: <b>*21&lt;phone number&gt;</b></li></ul>
-----------------------------------	---

### 10.8.2. Call Forward when Busy (\*22)

Forward calls when your phone is busy. You can change this setting on your own phone.

---

<sup>113</sup> Call forwarding unconditional overrules all other settings, including call forwarding when logged off.

<b>Call Forward When Busy</b>	<ul style="list-style-type: none"><li>• Status: <b>*22*</b></li><li>• Enable: <b>*221</b></li><li>• Always Forward to Operator: <b>*222</b></li><li>• Always Forward to Voice Mail: <b>*223</b></li><li>• Always Give a Busy Signal: <b>*224</b></li><li>• Always Bounce to the PBX: <b>*225</b></li><li>• Set Phone Number: <b>*22&lt;phone number&gt;</b></li></ul>
-------------------------------	---

### 10.8.3. Call Forward when Not Answered (\*23)

Forward calls when you do not answer your phone within the given number of seconds. You can change this setting on your own phone.

<b>Call Forward No Answer</b>	<ul style="list-style-type: none"><li>• Status: <b>*23*</b></li><li>• Enable: <b>*231</b></li><li>• Always Forward to Operator: <b>*232</b></li><li>• Always Forward to Voice Mail: <b>*233</b></li><li>• Always Give a Busy Signal: <b>*234</b></li><li>• Always Bounce to the PBX: <b>*235</b></li><li>• Set Phone Number: <b>*23&lt;phone number&gt;</b></li></ul>
-------------------------------	---

### 10.8.4. Call Forward when Logged Off (\*24)

Forward calls when you are not logged in on a phone. You can change this setting on your own phone.

<b>Call Forward When Logged Out</b>	<ul style="list-style-type: none"><li>• Status: <b>*24*</b></li><li>• Enable: <b>*241</b></li><li>• Always Forward to Operator: <b>*242</b></li><li>• Always Forward to Voice Mail: <b>*243</b></li><li>• Always Give a Busy Signal: <b>*244</b></li><li>• Always Bounce to the PBX: <b>*245</b></li><li>• Set Phone Number: <b>*24&lt;phone number&gt;</b></li></ul>
-------------------------------------	---

## 10.9. Voice Mail (\*25)

---

If you cannot be reached by phone, the caller can be offered to leave a voice mail. Voice mail has to be switched on (see below) and can be referred to by the call forwarding settings (chapter 10.8). Accessing voice mail is not only possible from your own phone, but also from another extension or outside line (with DISA). The number of messages for each voice mail box is limited to 99. Only from the PBX Manager you can configure the time period a message will be kept.

When someone leaves a message, you'll receive a notification e-mail. This e-mail is sent to the address provided in the PBX Manager (SEND E-MAIL TO). Many phones support a visual and/or audible message indicator. This can be configured from the PBX web interface.

Code	Description
<b>*25</b>	Voice mail menu to listen messages and change settings. Asks for PIN first. Enter your phone PIN, followed by the # key.
<b>*25*</b>	Reports current voice mail status.
<b>*25 &lt;ext&gt;</b>	Voice mail menu to listen messages and change settings. Asks for PIN first. Enter your phone PIN, followed by the # key.
<b>*213</b>	Switch voice mail ON
<b>*210</b>	Switch voice mail OFF
<b>*26 &lt;ext&gt;</b>	Dial the voice mail box of a specific extension to leave a message.

## 10.10. Easy Transfer / Callee Transfer (#)

---

There is an easy way to transfer calls that works on any phone for calls received from the Voclarion<sup>114</sup>. Callee Transfer can also be used on mobile phones. Simply press '#', to transfer a call. You will hear 'Transfer'. Now you can dial any number, to transfer the call to. When the call is answered, announce the call, and hangup. The call will be transferred. If the call is not answered, press '\*\*' to take the call back.

If the transfer is not answered within 45 seconds, the call will be transferred back to your phone. A beep indicates the call is returned to you.



To use this feature it must be activated in the advanced features of the phone first. Go to Companies > *Company* > Phones > *Phone* > tab Advanced. Look for "HASH TRANSFERS INCOMING (OR OUTGOING) CALLS".

Be aware there can be undesirable effects when calling services using the \* and #, like IVR menus.

---

---

<sup>114</sup> Does not work with Direct Media.

## 10.11. Paging / Intercom (\*36)

---

Paging is used to make announcements to one person or to a group by paging over the speaker phone. If you would like to page more phones at the same time, create a page group first. It is also possible to allow people to talk back to you; this is called intercom. To use paging you'll need the correct permissions.

Dial the paging code. If connected you will hear a beep. Supported phones will automatically enable speaker phone mode and you can make an announcement.

The paging/intercom feature is not supported by all phones. Ask your supplier if your phone supports paging.

Code	Description
*36<ext>	Page a person's phone
<Page group number>	Page a group of phones

## 10.12. Call Parking (\*38)

---

Call Parking is used if you have a phone call for a person who is in the office, but not at his desk. The call can be parked and picked up from any other phone within the company by dialing the parking number.

### 10.12.1. Park a call

1. Put the call through with consultation to \*38.
2. You will receive a call number. Now connect the call.

### 10.12.2. Pick up a parked call

1. The person the call is intended for, has 90 seconds to pick up the call by dialing \**<parking\_number>*.
2. After 90 seconds, the call will be returned to you.

Starting from Voicemail 2.2. there is a more advanced parking feature called Parking lot (chapter 3.8).

## **10.13. Parking lot**

---

A more advanced call parking function is the Parking lot. A parked lot is divided in parking slots, which can be monitored. The parked call can be picked up from every phone within the company.

### **10.13.1. Park a call**

1. Connect the current call with consultation to the extension of the parking lot.
2. The Voclarion will announce the extension of the parking slot the call will be parked on, followed by on hold music.
3. Connect the call on hold.
4. The call is now parked.
5. Announce the parking slot extension to your colleague.

### **10.13.2. Pick up a parked call**

A parked call can be picked up from every phone within the company.

1. Dial the parking slot extension announced by the Voclarion.
2. The parked call is now connected to your phone and you can have a conversation.

## 10.14. Call Forward Override (\*60)

---

If an extension has a call forward enabled, a caller can bypass this forward and reach the person on his phone. This is only when the user is actually logged in to a phone.

For example, the user is logged in, and has set a default forward to his mobile number. The caller dials **\*60<extension\_of\_the\_user>**. Instead of being forwarded to the mobile number, the phone of the user will ring.

This feature can only be used by someone if Call Forward Override on his colleague's extension is activated.

## 10.15. Call Back on Busy (5)

---

If you call someone that turns out to be busy, you will have to try again to see if he is available (unless you monitor his phone status of course). With Call Back on Busy you don't have to try over and over again. Instead your phone will ring and setups a call when the person is available again. This only works on internal calls.

When you dial an internal extension and get the busy tone, press “5”. A beep acknowledges your key press. Hang up the phone. When the person is available again, your phone will ring. If you pick up the phone, the PBX will automatically setup a call to this person.

If you are on the phone when the PBX calls back, the PBX will try again in reverse order.

Please note:

1. This feature can only be used on internal calls.
2. This feature can only be used if a person is actually on the phone and has configured a “busy tone” on internal call forward settings. This feature also works with Do Not Disturb.
3. Call Back on Busy expires when a call has been setup successfully or after 30 minutes.

## 10.16. Dial With Return (\*61)

---

With this option you can transfer a caller to an external phone number. When the call has finished, the caller will be connected to the extension automatically.

1. Connect your active call to **\*61<extension><phone number>**  
(you can use blind or consultative transfer)
2. The call will be transferred to *<phone number>*
3. After the call is finished, the person will be connected to *<extension>* automatically.

## 10.17. Wake Up Call (\*62)

---

The PBX provides a wake up service. To activate the service, dial \*62. You will be asked to enter the time you want to receive the wake up call.

The alarm has to set again daily.

Code	Description
*62	Set the alarm. The Voclarion will ask for a time. Always supply four digits, like 0600 for 6 am.
*63	Check alarm setting.
*64	Delete current alarm.

## **10.18. Direct Inward System Access (DISA)**

DISA (Direct Inward System Access) allows callers from outside the Voclarion to obtain an "internal" dial tone and to place calls as if they are placing it from the office.

Your company needs to have allocated a special incoming number for DISA. Users dial the DISA phone number and enter a PIN followed by #. Than they'll hear a dial tone and have 20 seconds to dial a number.

## 10.19. Dial by Name (\*35)

---

If you do not know the extension of the person you are looking for, you can dial **\*35**. You will be asked to enter the first three characters of the name of the person you are looking for. You have 5 seconds to supply an entry. The Voclarion will now look for a match on first name, last name or both based on the settings.

1. Dial **\*35**
2. Use the keys of your phone to specify the first three characters of the name you are looking for. For example: if you are looking for Mr. Anderson, press 2-6-3 (see table 16).
3. The Voclarion will now lookup the first person who's name starts with the characters you entered. If this person has his name recorded (voice mail menu, see page 363) the PBX plays the recording. If not Voclarion will spell the person's full name.
4. In case of more than one match, you can cycle through all matches using **\***.
5. If the PBX mentions the right name, press "1" to dial this person's number.

<b>1</b>	<b>2</b> <b>ABC</b>	<b>3</b> <b>DEF</b>
<b>4</b> <b>GHI</b>	<b>5</b> <b>JKL</b>	<b>6</b> <b>MNO</b>
<b>7</b> <b>PQRS</b>	<b>8</b> <b>TUV</b>	<b>9</b> <b>WXYZ</b>
<b>*</b>	<b>0</b>	<b>#</b>

Table 16: Key pad characters.

## 10.20. Main PBX Menu (\*99)

---

There are four previously described PBX features which can be accessed by dialing \*99. Press:

- 1 for Voice Mail.
- 2 for Call Forwarding.
- 3 for Conference Boxes.
- 4 for Phone Book (Directory).



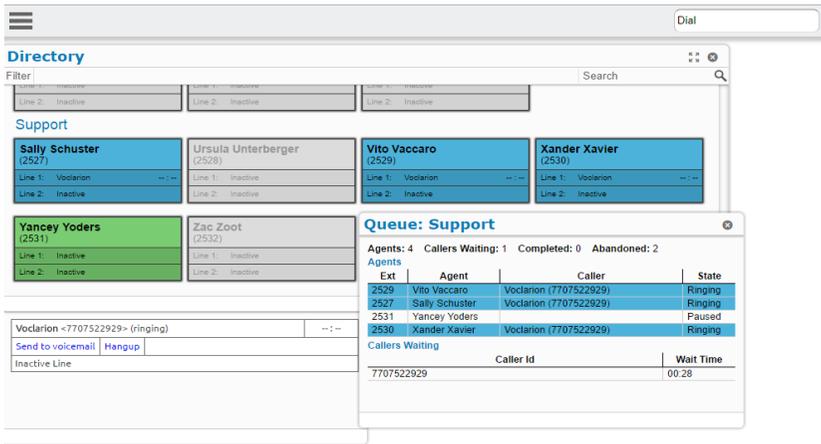
## **Chapter 11. Web-based Switchboard**

*Describes the switchboard, usable through your  
browser.*

# 11.1. Switchboard

This web-based application is a powerful tool that lets you see activity in your Voclarion system. You are able to see users logged in, who is talking and to whom, and queue information. You can also preform phone actions such as Originate a call, Transfer a call, Hangup or Send to Voicemail.

The application includes several panels to display different activity in your system. The panels can be closed and opened from the Menu button.



*Illustration 15: Switchboard with the Directory, Current Calls, and Queue panels*

- **Directory:** View your entire staff's presence split up by department. You can preform Click to Dial, Drag and Drop your calls to transfer.
- **Current Calls:** For each of your extensions you can view your active calls, drag and drop the call to the directory to Transfer, or Hang the call up.
- **Queues:** View Completed/Abandoned Call for the queues, Agents in the Queue and their status. Also see any callers waiting in line and how long they've been waiting.

### 11.1.1. Directory

The Directory panel shows all contacts in your company with their current state. There are different colors for what the users are doing.

GREEN The user is logged in and able to receive calls.

BLUE The user is on a call, on hold, ringing or dialing.

GREY The user is not logged into a phone.

You can filter what displays in the Directory by clicking the Filter button. Here you can show/hide departments and show/hide user's logged out.

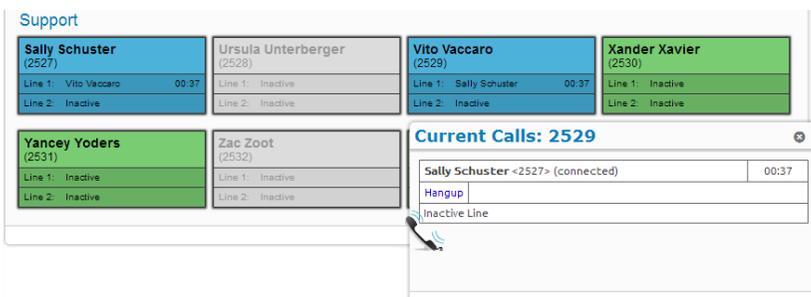
Each extension that is logged in may be clicked to initiate a call with the user.

### 11.1.2. Current Calls

For each extension you have, a Current Call panel is available. This will show the current information of your extension.

Depending on the current state of the extension you will have certain actions available to you. i.e. Hangup, Send to Voicemail, Transfer.

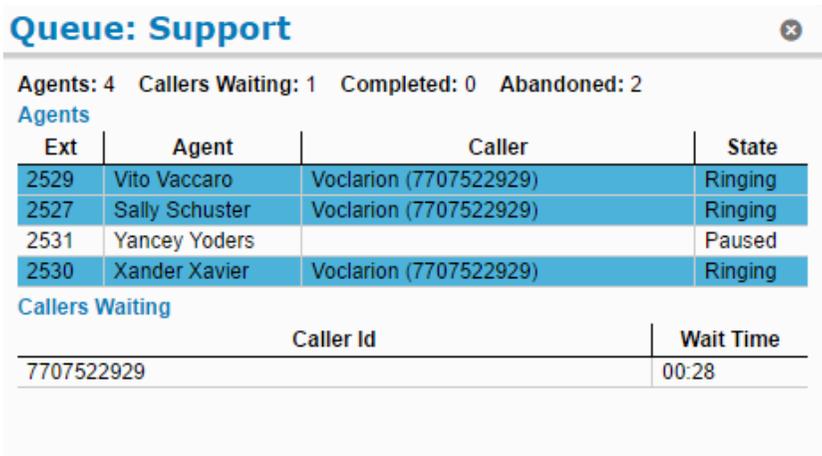
While on a call you can click and hold the call info and drag and drop in the directory to transfer to the extension you drop on.



*Illustration 16: Transferring a call to someone in the directory.*

### 11.1.3. Queues

You can view real-time information for a queue and it's agents. You'll be able to see the Agents in the queue and their status. As well as caller id information for the caller the agent is on the phone with. Callers who are waiting in line can be seen and the time they have been waiting for.



The screenshot shows a panel titled "Queue: Support" with a close button in the top right corner. Below the title, it displays summary statistics: "Agents: 4 Callers Waiting: 1 Completed: 0 Abandoned: 2".

**Agents**

Ext	Agent	Caller	State
2529	Vito Vaccaro	Voclarion (7707522929)	Ringling
2527	Sally Schuster	Voclarion (7707522929)	Ringling
2531	Yancey Yoders		Paused
2530	Xander Xavier	Voclarion (7707522929)	Ringling

**Callers Waiting**

Caller Id	Wait Time
7707522929	00:28

Illustration 17: The Queue Panel.



## **Chapter 12. Quick Card PBX Features**

*An overview of the most used features for your  
reference.*

## Quick Card PBX Features

This Voclarion Quick Card shows frequently used PBX functions. All actions can be performed from your phone. All codes can also be programmed to function keys on your phone.

Feature	Code
<b>Main PBX Menu</b>	<ul style="list-style-type: none"> <li>• <b>*99</b></li> </ul>
<b>Phone Login</b>	<ul style="list-style-type: none"> <li>• <b>*1</b> Log in to your phone. Enter <b>&lt;phone_login&gt;&lt;PIN&gt;#</b></li> <li>• <b>*1*</b> Switch user on a phone. Enter <b>&lt;phone_login&gt;&lt;PIN&gt;#</b></li> </ul>
<b>Phone Logout</b>	<ul style="list-style-type: none"> <li>• <b>*1</b> Log out. Can be blocked.</li> </ul>
<b>Call Recording</b>	<ul style="list-style-type: none"> <li>• <b>*3</b></li> <li>• To enable/disable recording on active conversation on your phone. Only with proper permission and automatic call recording switched off.</li> </ul>
<b>Call Pickup</b>	<ul style="list-style-type: none"> <li>• <b>*8</b> Answer a random call from a pickup group.</li> <li>• <b>*8&lt;extension_of_colleague&gt;</b> Answer a call from a specific colleague within the same pickup group.</li> </ul>
<b>Easy Transfer / Callee Transfer</b>	<ul style="list-style-type: none"> <li>• <b># &lt;ext&gt;#</b> To transfer, dial <b># &lt;ext&gt;#</b> and wait. When connected, hang up. To take the call back, press <b>*</b>.</li> </ul>
<b>Dial by Name / Directory</b>	<ul style="list-style-type: none"> <li>• <b>*35</b> Enter the first three letters of the name. Follow the instructions.</li> </ul>

Feature	Code
<b>Call Parking</b>	<ul style="list-style-type: none"> <li>• <b>*38</b> Put call on hold, dial <b>*38</b>. You will hear a parking number, followed by music on hold. Transfer call and hang up.</li> <li>• To pick up call, dial * <b>&lt;parking_number&gt;</b> from a phone within the company. After 90 seconds, the call on hold will be returned.</li> </ul>
<b>Call Forward Override</b>	<ul style="list-style-type: none"> <li>• <b>*60&lt;extension&gt;</b></li> <li>• Ring the phone of a user while ignoring all active call forward settings (only when user is logged in on a phone) .</li> </ul>
<b>Call Back when Busy</b>	<ul style="list-style-type: none"> <li>• When employee is busy, press <b>5</b> and hang up the phone. When employee comes available, your phone will ring. When you pick up, a call is set up.</li> </ul>
<b>Dial with Return</b>	<ul style="list-style-type: none"> <li>• Transfer a call to <b>*61&lt;extension&gt;&lt;phone number&gt;</b></li> <li>• Call will be connected to <b>&lt;phone number&gt;</b>. When call ends, the caller will be connected to <b>&lt;extension&gt;</b>.</li> </ul>
<b>Wake Up Call</b>	<ul style="list-style-type: none"> <li>• <b>*62</b> Enter the time using 4 digits.</li> <li>• <b>*63</b> to check the status of the call</li> <li>• <b>*64</b> to cancel the wake up call</li> </ul>
<b>Paging/Intercom</b> For paging single phone.	<ul style="list-style-type: none"> <li>• <b>*36&lt;extension&gt;</b></li> <li>• For a paging group, dial the group number (without *36).</li> </ul>
<b>Call back on busy</b>	<ul style="list-style-type: none"> <li>• <b>5</b></li> <li>• Press 5 during busy tone. Your phone will ring when employee is available. Internal calls only.</li> </ul>

## Quick Card PBX Features

Feature	Code
<p><b>Voice Mail</b></p>	<ul style="list-style-type: none"> <li>• Listen to voice mail: <b>*25</b></li> <li>• Status voice mail setting: <b>*25*</b></li> <li>• Listen to voice mail from another extension: <b>*25&lt;ext&gt;</b></li> <li>• Voice mail on: <b>*213</b> (see also Call Forward options)</li> <li>• Voice mail off: <b>*210</b></li> <li>• Dial voice mail box of colleague: <b>*26&lt;ext&gt;</b></li> </ul>
<p><b>Unconditional Call Forward</b></p> <p><i>Use the PBX Manager to select contact types.</i></p>	<ul style="list-style-type: none"> <li>• Status: <b>*21*</b></li> <li>• Enable: <b>*211</b></li> <li>• Set back to Ring Phone: <b>*210</b></li> <li>• Always Forward to Operator: <b>*212</b></li> <li>• Always Forward to Voice Mail: <b>*213</b></li> <li>• Always Give a Busy Signal: <b>*214</b></li> <li>• Always Bounce to the PBX: <b>*215</b></li> <li>• Set Phone Number: <b>*21&lt;ext&gt;</b></li> </ul>
<p><b>Call Forward When Busy</b></p> <p><i>Use the PBX Manager to select contact types.</i></p>	<ul style="list-style-type: none"> <li>• Status: <b>*22*</b></li> <li>• Enable: <b>*221</b></li> <li>• Always Forward to Operator: <b>*222</b></li> <li>• Always Forward to Voice Mail: <b>*223</b></li> <li>• Always Give a Busy Signal: <b>*224</b></li> <li>• Always Bounce to the PBX: <b>*225</b></li> <li>• Set Phone Number: <b>*22&lt;ext&gt;</b></li> </ul>
<p><b>Call Forward No Answer</b></p> <p><i>Use the PBX Manager to select contact types.</i></p>	<ul style="list-style-type: none"> <li>• Status: <b>*23*</b></li> <li>• Enable: <b>*231</b></li> <li>• Always Forward to Operator: <b>*232</b></li> <li>• Always Forward to Voice Mail: <b>*233</b></li> <li>• Always Give a Busy Signal: <b>*234</b></li> <li>• Always Bounce to the PBX: <b>*235</b></li> <li>• Set Phone Number: <b>*23&lt;ext&gt;</b></li> </ul>

Feature	Code
<b>Call Forward When Logged Out</b> <i>Use the PBX Manager to select contact types.</i>	<ul style="list-style-type: none"> <li>• Status: <b>*24</b></li> <li>• Enable: <b>*241</b></li> <li>• Always Forward to Operator: <b>*242</b></li> <li>• Always Forward to Voice Mail: <b>*243</b></li> <li>• Always Give a Busy Signal: <b>*244</b></li> <li>• Always Bounce to the PBX: <b>*245</b></li> <li>• Set Phone Number: <b>*24&lt;ext&gt;</b></li> </ul>
<b>Conference Box</b>	<ul style="list-style-type: none"> <li>• Create a conference box: <b>*30</b></li> <li>• Join a conference: <b>*31</b> Enter box number and password.</li> <li>• Join a specific conference: <b>*31&lt;box_number&gt;</b></li> <li>• Join a conference as administrator: <b>*32&lt;box_number&gt;</b></li> </ul>
<b>Queue</b>	<ul style="list-style-type: none"> <li>• Status: <b>*42*</b></li> <li>• Login: <b>*40&lt;queue ext&gt;</b></li> <li>• Logout: <b>*41&lt;queue ext&gt;</b></li> <li>• Pause off: <b>*420 (all queues)</b></li> <li>• Pause on: <b>*421 (all queues)</b></li> </ul>
<b>Ring group</b>	<ul style="list-style-type: none"> <li>• Status: <b>*43*</b></li> <li>• Login: <b>*43&lt;queue ext&gt;</b></li> <li>• Logout: <b>*43&lt;queue ext&gt;</b></li> </ul>
<b>Do not disturb</b>	<ul style="list-style-type: none"> <li>• <b>*28</b> switch on / off</li> </ul>
<b>Recording Sound Prompts</b>	<ul style="list-style-type: none"> <li>• Recording a sound prompt: <ol style="list-style-type: none"> <li>1. press <b>*55</b></li> <li>2. wait for a beep and record a message</li> <li>3. press <b>#</b> to end recording</li> <li>4. save or rerecord the message.</li> </ol> </li> <li>• Listening to and rerecording a sound prompt: <b>*55&lt;sound-prompt-ID&gt;</b></li> </ul>
<b>Monitoring a Colleague</b>	<ul style="list-style-type: none"> <li>• <b>*50&lt;ext&gt;</b> only if permitted by administrator.</li> </ul>

## Quick Card PBX Features

Feature	Code
Phone Information	• *9991
User Information	• *9992
Echo Test	• *9993
Milliwatt Test	• *9994
External Waiting Music	• *9995
Reprovision Phone	• *9996



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