Orchestra NG User Manual

VoiSmart

Orchestra NG Release 8.14.0

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Introduction

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1.1 What is Orchestra NG

Orchestra NG is a full featured, multi-tenant IP-PBX that allows to offer standard PBX features over IP networks, using SIP as VoIP protocol for local extensions and hybrid TDM/SIP for interconnecting to telco networks.

1.2 Overview

An Orchestra NG system is normally composed by several items, depending on specific application deployment. In most cases there's an Orchestra NG instance running on dedicated hardware or virtual appliance, several VoIP phones compatible with the SIP protocol and one or more interconnection to PSTN circuits using SIP or TDM technology. Every component is connected to each other using standard IP networks.

1.3 Features

Orchestra NG offers a full fledged Unified Communications server. A brief and not exhaustive list of principal features follows:

- easy to use and understand web-based GUI;
- voice SIP server, supporting either TCP and UDP clients, with optional voice encryption;
- voice conference server, supporting N-party conference rooms, with booking system and live, web-based administration;
- fax server, up to 14400 bps, using T.38 if routed over SIP;
- operator panel, a live web-based GUI to administer and monitor live calls;
- graphical dialplan editor, to create IVRs and the needed call routing logic without having any specific knowledge of telephone systems;
- full featured mobility system to deliver calls on any device, to any location;
- call center grade queues and agents, with powerful analytics;
- all PBX features the company needs, from music on hold to parking and other advanced phone system feature;
- powerful phonebook and calendar features;
- can be integrated to company directory and authentication servers;

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Quickstart

Contents

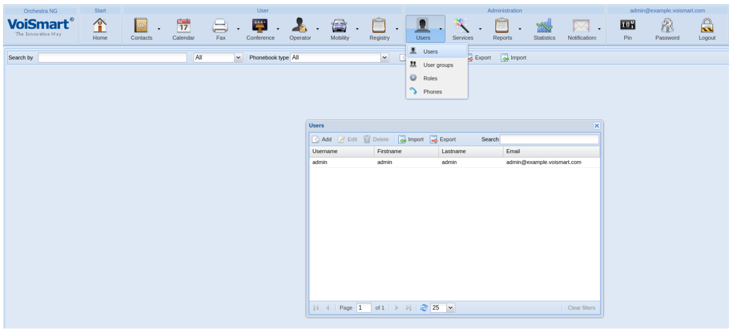


Figure 2.1: Users window

2.1 Introduction

A basic configuration for an Orchestra NG domain, suitable for initial setup of inbound and outbound calling, is comprised of these steps:

- create users, see section 2.2;
- add extensions to that users, see section 2.3;
- configure a phone, see section 2.4 on page 6;
- associate a valid LCR to the user, see section 2.5 on page 7;
- configure inbound calls routing using the *Dialplan editor*, see section 2.6 on page 7.

A simple Call Queue service will also configured in section 2.7 on page 8.

2.2 Create domain users

Open the Users window by clicking on the Users → Users menu and then create a new one by clicking on the Add button in that window, as shown on figure 2.1.

Fill in user details as shown on figure 2.2. Refer to section 5.1 on page 27 for a detailed description of the meaning of the fields.

2.3 Create user's extensions

After closing that window, a new extension must be created. To do so, just click the *Edit* button after having selected the newly created user in the user list window (figure 2.1). A window just like the one in figure 2.3 will appear. After clicking on the *Extension* icon, just enter the extension details and press *Update* (figure 2.4). For further details, see section 5.1 on page 38.

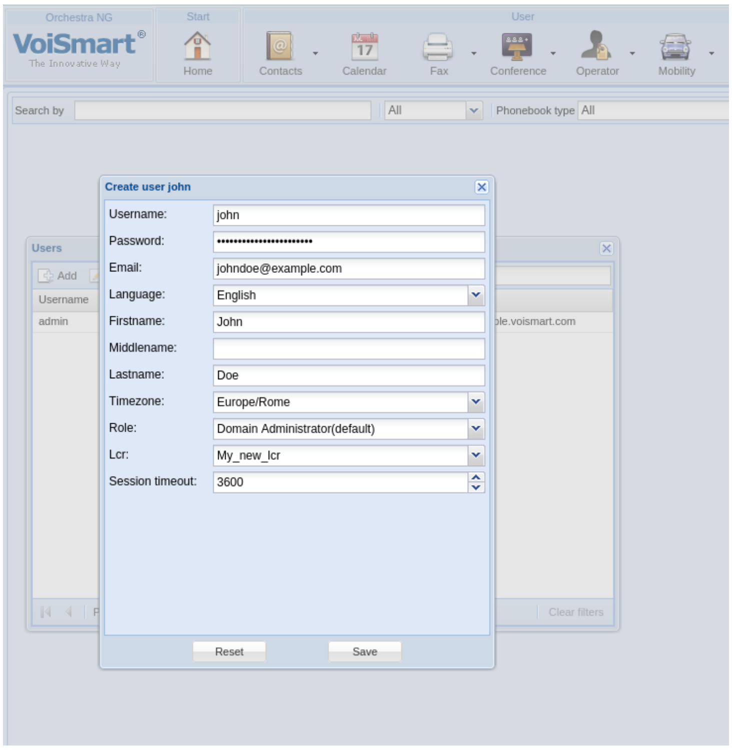


Figure 2.2: Users details

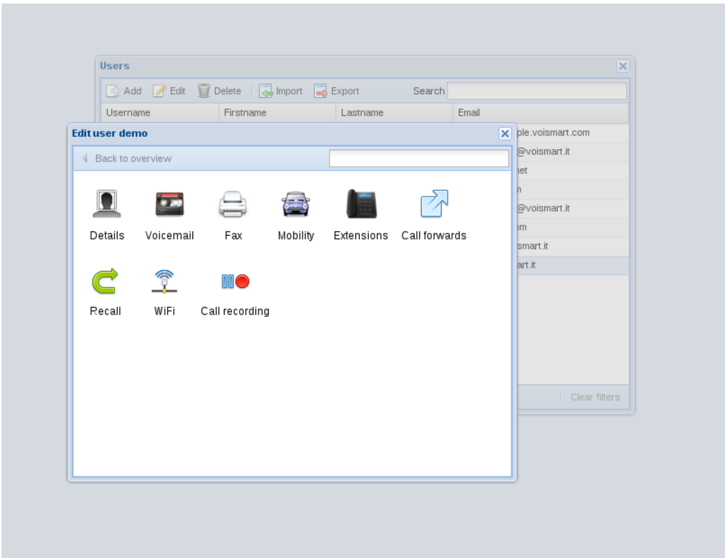


Figure 2.3: Users options window

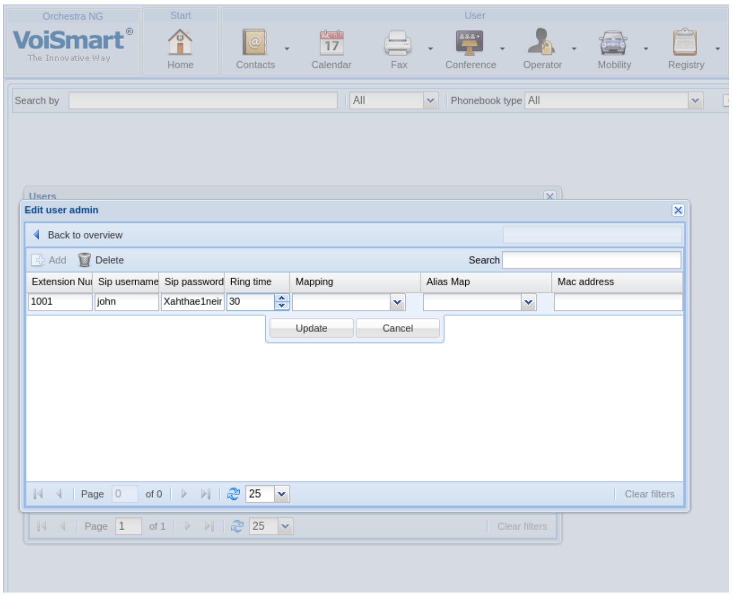


Figure 2.4: User extensions window

2.4 Configure a phone

After having created a new extension, a phone must be configured. All VoiSmart phones can be configured either from the phone menu or from the web interface reachable by pointing a web browser to the phone IP address.

The default credentials for the web configuration interface are:

- username: *admin*;
- password: *admin*.

Here will be shown how to correctly configure a VoiSmart VEP-2100.1 phone, other SIP-compatible phones will a have different interface. Please refer to the their user manual.

The account configuration interface (figure 2.5) is reachable by clicking on the VoIP menu.

The main configuration parameters are:

Server name The Orchestra NG domain;

Server address The Orchestra NG network address, usually an IP address or an IP name (the phone must be able to resolve it);

Server port The port of the *SIP profile*for phone registration, usually 5060;

Account name and password The credentials for this account, as configured in section 2.3 on page 4;

Phone number and Display name The display name and number for this phone.

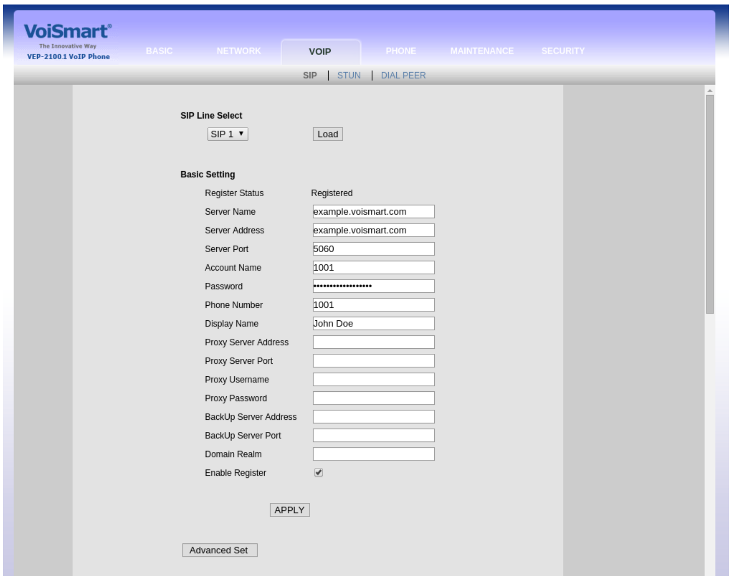


Figure 2.5: VoiSmart Vep 2100.1 configuration

Make sure the *Enable register* is flagged, and press *Apply*. If configured correctly, the *Register status* will be *Registered*.

2.5 Associate an LCR to the user

To allow outbound calls for the user, a valid LCR must be selected from the user creation dialog (figure 2.2). If no valid choices are available, please refer to the Orchestra NG Administrator Manual on how to create a new LCR.

2.6 Configure inbound calls routing

Inbound calls routing is configurable via a graphical Dialplan Editor, reachable by clicking on the Services → Dialplan Editor menu.

The configuration interface is divided in three main panes, shown in figure 2.6. The *Dashboard* section is the main component and consists in an entry point object on the left, which represents the source of the inbound calls for this domain. To change the call flow and direct them to specific services, one of the tools from the *Tools* pane on the left must be dragged and dropped on the *Dashboard* and then connected to the entry point, until the desired call routing is reached. A set of common actions, such as save and delete object operations, are found on the top *Toolbar*.

To simply direct a call to the newly created user, just drop the *Inbound callrouter* element from the *Tools* pane into the *Dashboard*. This element, allows to direct calls from the domain's inbound numbers to different services.

To route calls to the user, connect this element to the entry point, drag and drop the *User* tool and connect it to the desired number. The resulting configuration will be similar to the one shown on figure 2.7.

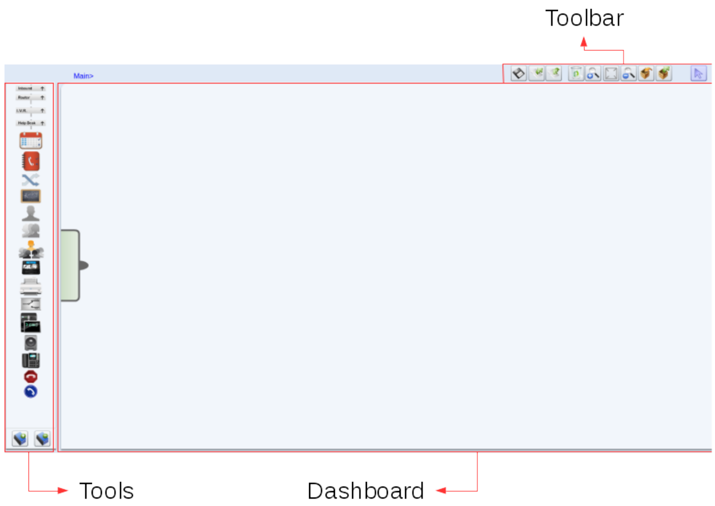


Figure 2.6: Dialplan editor main sections

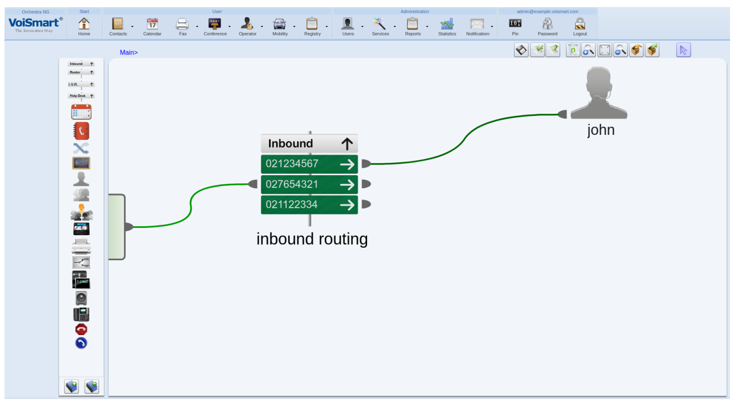


Figure 2.7: Simple dialplan configuration

After clicking on the *Save* button on the top *Toolbar*, the configuration will be saved, and the user will start receiving calls.

For further details on configuration and some more detailed examples, refer to section 5.2 on page 41.

2.7 Call queues

To configure a simple service of call distribution among several agents, a new queue must be created.

First of all, the queue's agents must be created, just like done on section 2.2. Then, a new sound file must be uploaded using the dialog window available after clicking on the Services → Sound Files menu. In that window, load the desired file with a relevant description, as shown in the example on figure 2.8.

To create a new queue, click on the Services → Queue menu and press the

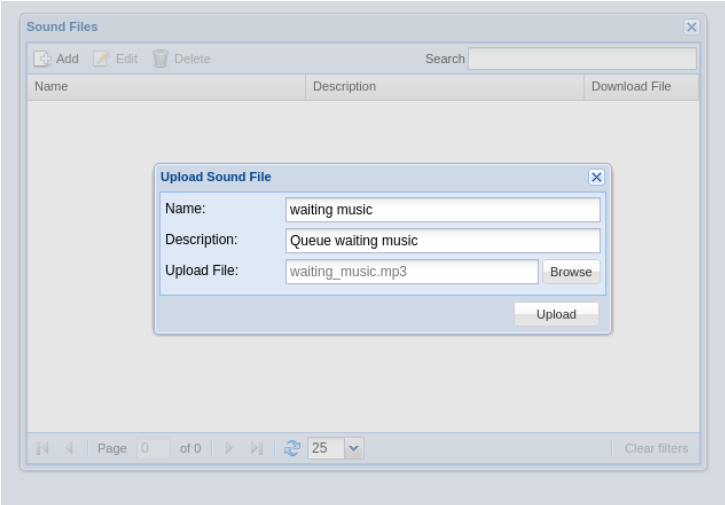


Figure 2.8: Sound file upload dialog

Add button on the window that will open. A dialog like the one in figure 2.9 will be shown. See section 5.2 on page 41 for the detailed meaning of all the configuration fields.

After having created the call group, your new service will be visible in the queue list windows (figure 2.10). Click on the icon at the right of the queue record to open add new agents. This new window (figure 2.11) contains the list of all users and groups associated with this call group and their specific configuration parameters. To add new users, click on the *Add users* button, and then move the desired users from the right grid to the left one by selecting them and clicking on *Add*, as shown in figure 2.12.

The only remaining configuration to do now is to open the Dialplan editor and add a queue service from the tools on the left, choose a valid call group from the list and connect it to the desired entry in the inbound callrouter (figure 2.13). As soon as the configuration is saved, any call to the selected number will be directed to the new queue.

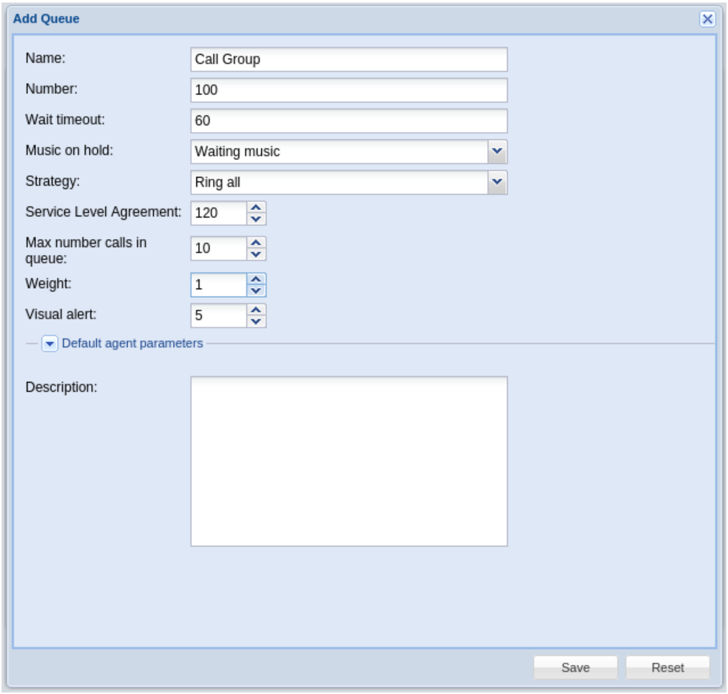


Figure 2.9: Queue creation dialog

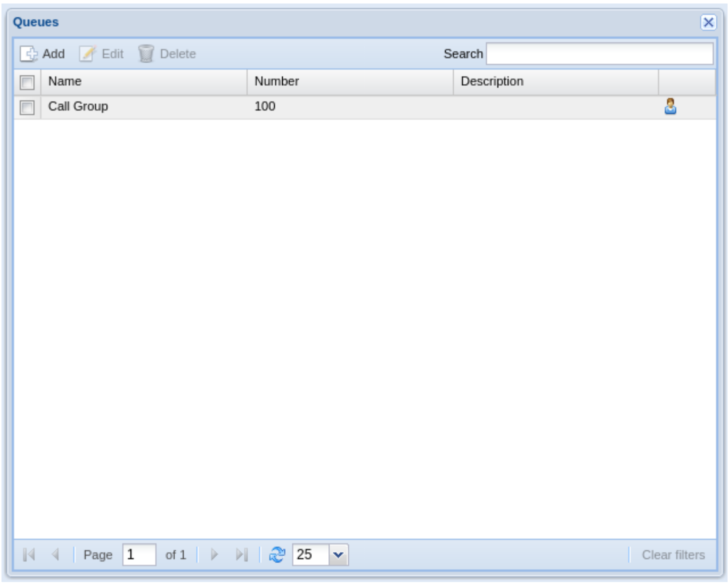


Figure 2.10: Queue list window

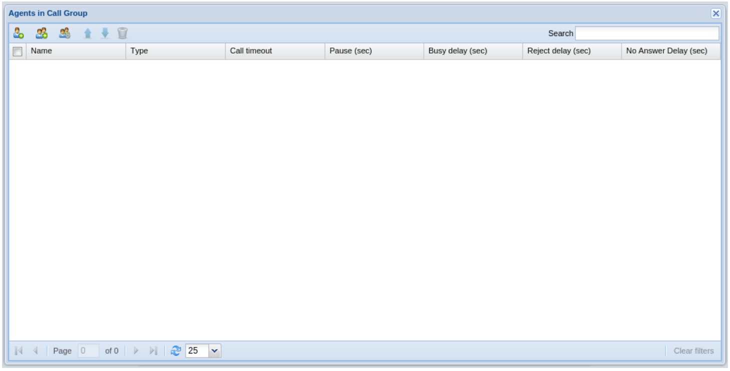


Figure 2.11: Agent list window

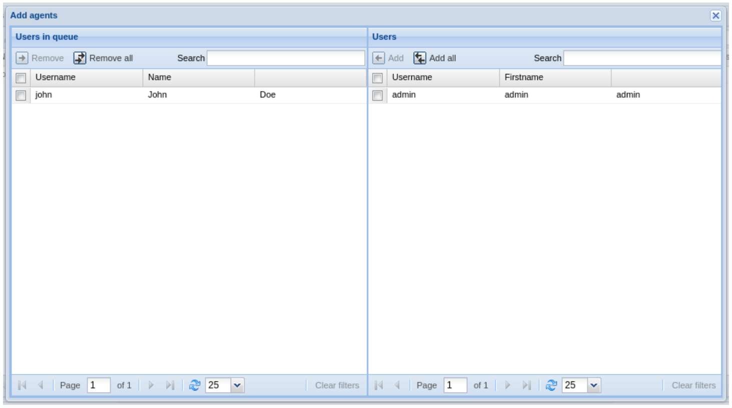


Figure 2.12: User association to a queue

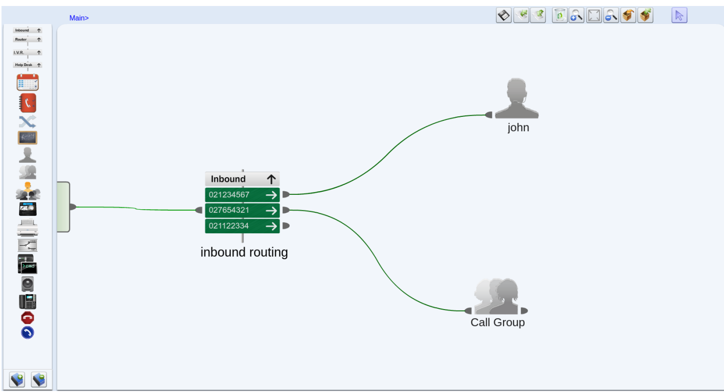


Figure 2.13: Dialplan with a queue service

3

Orchestra NG web interface

Contents

3.1 Introduction

Orchestra NG is web based system, where all services are used and managed using an HTML5 compliant web interface. There are two web GUIs, one for the domain or tenant, the other one for platform configurations that are on top of domains.

3.2 Supported Browsers

Because of leveraging on advanced JavaScript/CSS techniques, some restrictions on supported browser are applied, in order to guarantee the best user experience. Currently supported browsers are:

- Google Chrome, from version 24;
- Mozilla Firefox, from version 30;
- Microsoft Internet Explorer, from version 11.

3.3 Access to the web interface

Orchestra NG has some basic params, like IP address or login details.

- default username: *admin@example.voismart.com*;
- default password: *Voismart*;
- default ip address if installed as software: *DHCP*, unless differently specified during operating system installation;
- default ip address if bought as appliance: *192.168.0.250*;
- domain web interface url: `http://<ip address>`;
- system administrator web interface url: `http://<ip address>/admin`.

> ✍ <default ip address> means the currently configured IP address of the Orchestra NG installation.

On figure 3.1 on the next page a sample login interface is shown.

3.4 Main toolbar

The main menu, as shown on figure 3.2 on the facing page, is a set of submenus used to access all domain-specific configurations.

This toolbar is divided in three main sections: the user section, which provides user-related functionalities, the admin section, which contains services for the domain administrator and a last group of buttons, which give access to credentials of the logged-in user and the button to terminate the current session.

A summary of all menus follows:

- Orchestra NG: when clicked displays the build version;

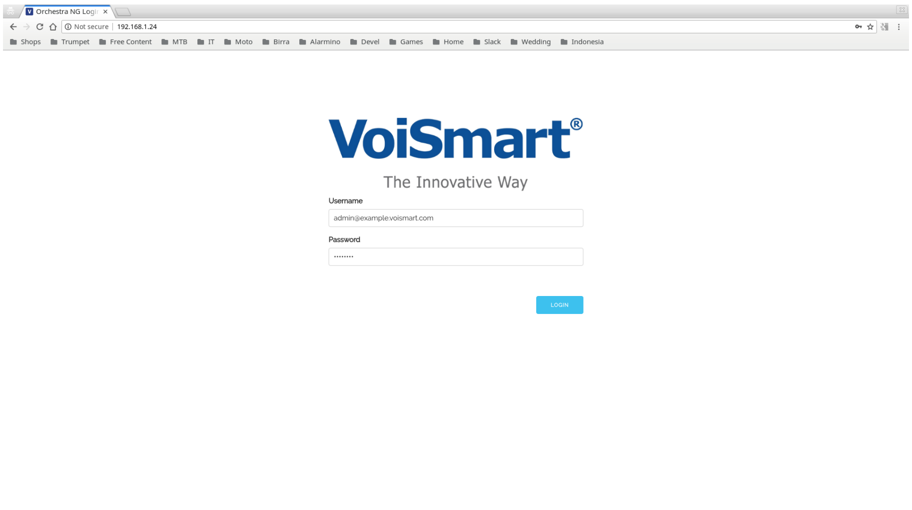


Figure 3.1: Orchestra NG login page.

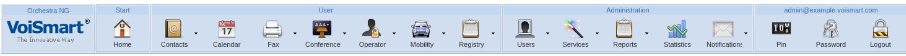


Figure 3.2: Orchestra NG toolbar.

- Home: clears the dashboard;
- Contacts: access to Contacts and distribution lists;
- Calendar: opens the user calendar;
- Fax: access to fax functionalities;
- Conference: access to conference panel and booking;
- Operator: access to the Operator panel;
- Mobility: access to the Mobility features;
- Registry: access to the user call and fax logs;
- User: access to users, groups, roles and extension management interfaces;
- Services: access to domain administrator functionalities;
- Reports: access to domain call and fax logs;
- Statistics: access to the statistics panel;
- Notifications: opens to the configuration menu for user notifications;
- Pin: shows or generate the Pin for the currently logged-in user;
- Password: access to dialog to change to password of the logged-in user;
- Logout: logs out the current user.

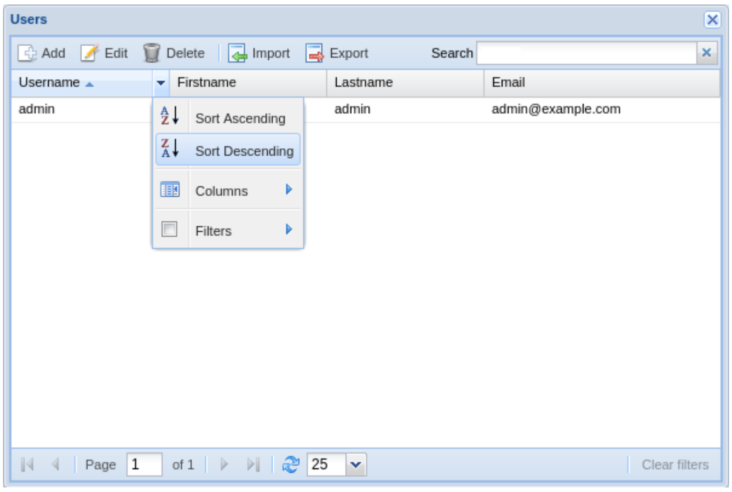


Figure 3.3: Context menu for grid column widgets.

3.5 Common GUI elements

This section describes some common elements and functions that are used extensively throughout the Orchestra NG web interface.

Grid column menu

The grid interface is commonly used to show a list of records with a varying level of details and functions. One common function is the possibility to filter the records according to some search criteria and to hide some of the fields. These operations are possible by opening the context menu shown in figure 3.3 by clicking the small triangle on any of the column headers.

The first two items in that menu sort the records in ascending or descending order with respect to that column. It is also possible to toggle the sort order by simply clicking on the column label repeatedly.

The columns submenu (figure 3.4), can be used to show and hide the visible columns.

The last submenu is used to filter records according to the chosen field. When a filter is active on one or more column, the filtered columns are shown in a different font, and a message is shown in the bottom-right corner of the window to warn the user that only a subset of all the records is currently being displayed. For example, in figure 3.5, there are two filters active on the fields "Sender" and "Recipient".

The application of a column filter is slightly different for different kind of fields:

Textual fields: simply enter the desired text to filter, figure 3.6;

Numerical fields: there are three input fields available (figure 3.7), to filter for records with values greater than, less than or strictly equal to the desired value;

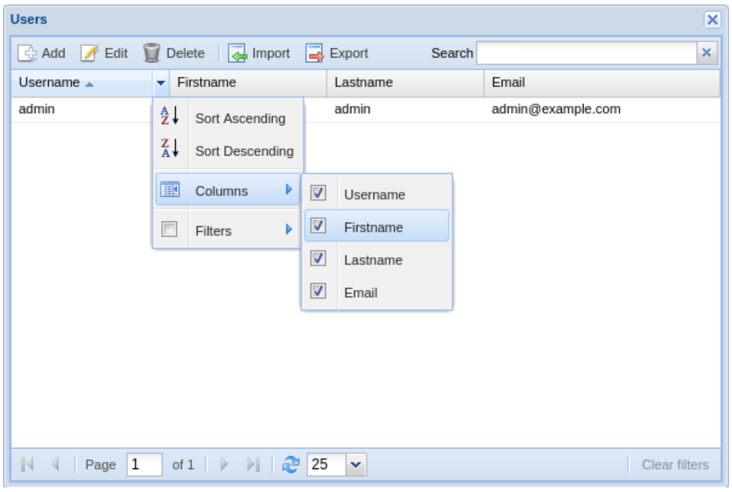


Figure 3.4: Context menu for grid column widgets, visible columns selection.

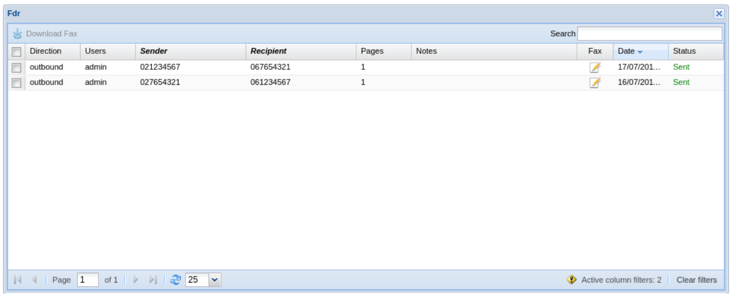


Figure 3.5: Window with two active column filters.

Date fields: there are three date pickers to filter for records created before, after or on the selected date, figure 3.8;

Fixed choice: here, the filtering is done by selecting among the available choices. The selection can be exclusive (only one possible value can be selected, as in figure 3.9), or multiple (more than one values can be selected, as in figure 3.10).

To remove a column filter, simply deselect them from the same context menu used to create it, or just press the "Clear filters" button on the lower-right corner of the window (see figure 3.5).

Grid's paging toolbar

Every grid dialog, has a paging toolbar to browse through grid's items (figure 3.11).

All of the items currently visible in a grid are collectively called a page, and the paging toolbar's main purpose is to browse through pages, using the little arrows which allow to move to the first, previous, next and last page. The

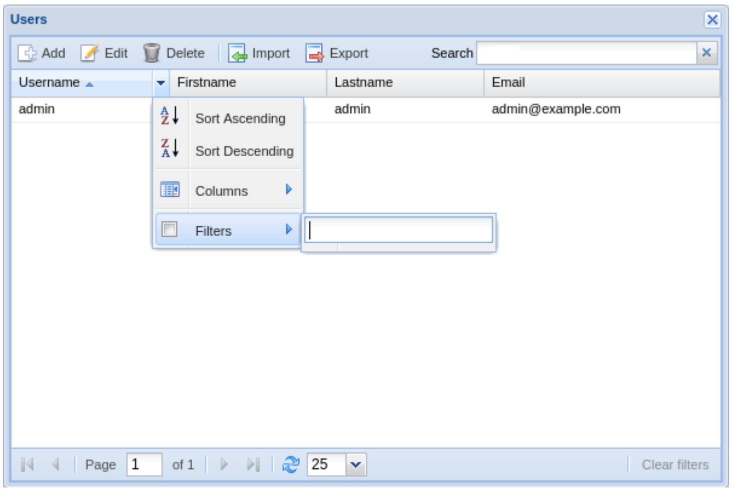


Figure 3.6: Column filter for textual fields.

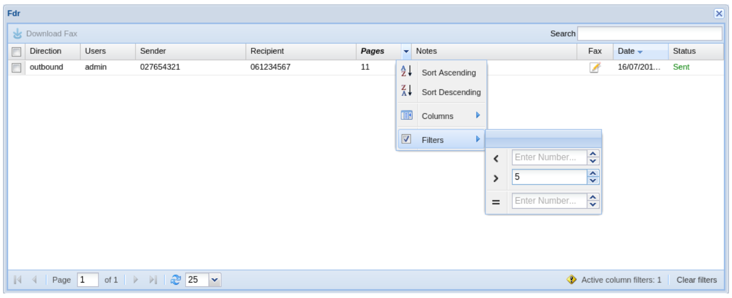


Figure 3.7: Column filter for numeric fields.

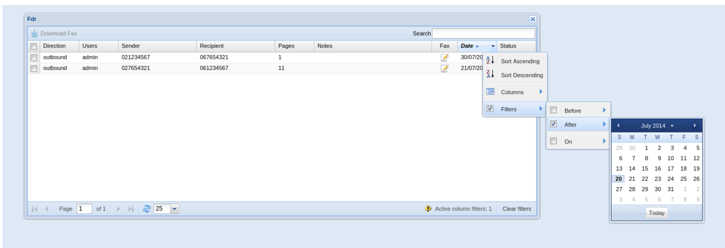


Figure 3.8: Column filter for date fields.

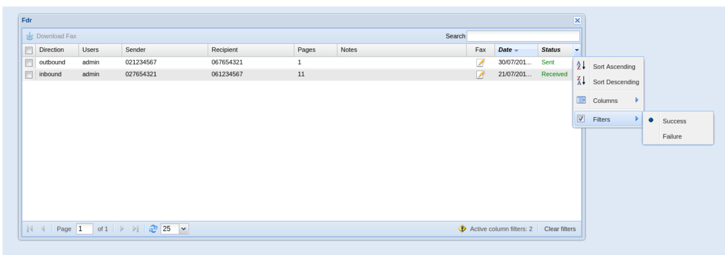


Figure 3.9: Column filter for exclusive choice fields.

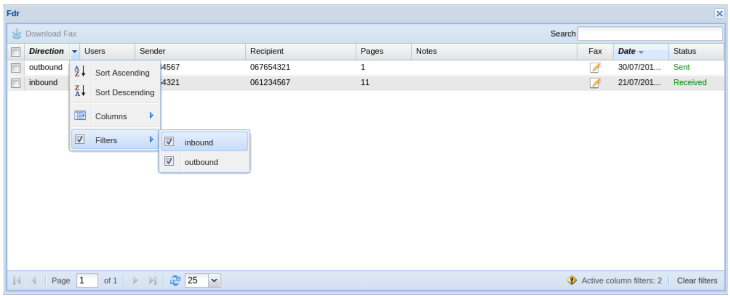


Figure 3.10: Column filter for multiple selection fields.

Figure 3.11: Grid's paging toolbar.

currently displayed page is shown in the center along with the total number of pages.

The small curved double arrow, reloads grid's items, and the small selector near it, modifies the page size, i.e. the number of items which form a page.

The last button, clears the currently active column filters, see section 3.5 on page 16 for details.

Live search feature

This feature is available in many of the dialog windows in the Orchestra NG interface. Searching content is a pervasive feature, and by entering text in a live search field, the content will be filtered in a sensible way dependent to the context.

For example, figures 3.12 and 3.13 show an example of searching through system's users and phone book contacts in a coherent way, even if the underlying search details are different.

To clear the search, just click on the small button at the right end of the input field.

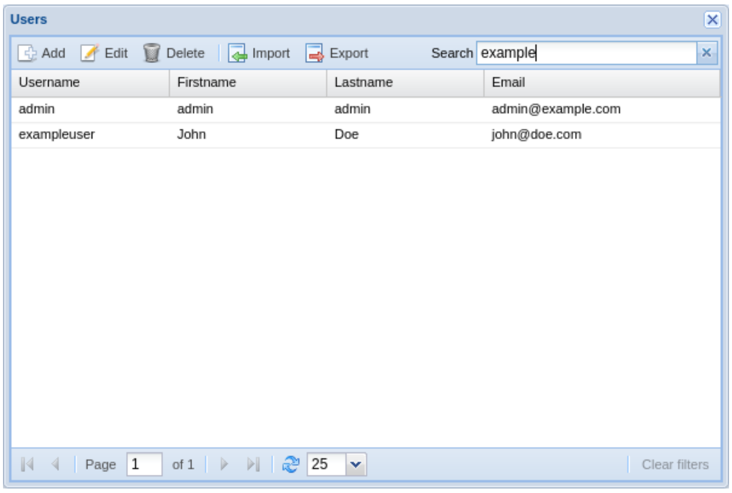


Figure 3.12: Live search on the users window.

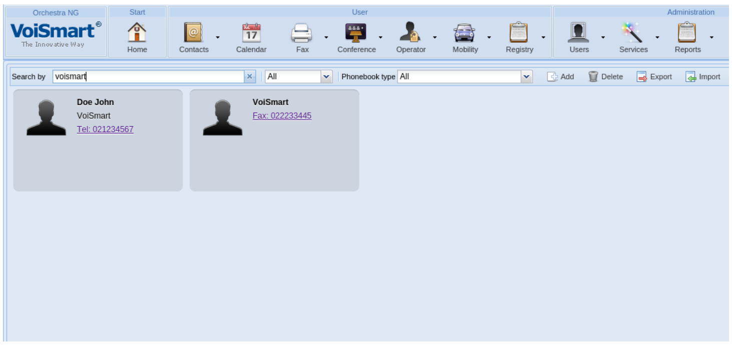


Figure 3.13: Live search on the phone book.

4

User services

Contents

4.1 Contacts

Orchestra NG has a full-featured phonebook system with all functions needed for proper usage in a company environment. It features a public phonebook, shared between all users; a groups-based one, to share contacts between people belonging to similar work areas and a personal phonebook, which is private to each user. It is also possible to quickly import and export all the dataset, for a quick provisioning.

In addition the backend can be an external directory server. See the Orchestra NG admin manual for information on how to integrate third party servers.

On chapter 7 on page 67 the phonebook feature is explained in detail.

4.2 Calendar

In this section you can create calendars and events used in user's mobility rules. For more details about calendars' configuration in Orchestra NG, refer to chapter 17 on page 145.

4.3 Fax

Each user of the Orchestra NG system can send and receive faxes from the web GUI, if permitted by the local administrator. For more details about the fax functionalities refer to chapter 8 on page 75.

4.4 Conference

The Orchestra NG Conference allows to create audio conference calls mainly in two ways: "Live" conferences or "Booked" conferences. For more details about conferences' configuration in Orchestra NG, refer to chapter 9 on page 87.

4.5 Operator panel

The Orchestra NG Operator Panel allows to monitor the status of local extensions, queues and agents. See chapter 12 on page 99 for a detailed explanation of all the features it provides.

4.6 Mobility

Mobility is an advanced call forwarding feature that allows to hunt a user with complex but easy to configure rules. No matter where the user is, the system is able to contact it across a range of devices. See the chapter 13 for an in deep explanation of this feature.

4.7 Registry

The registry section provides call and fax records for all the operation made by the currently logged in user.

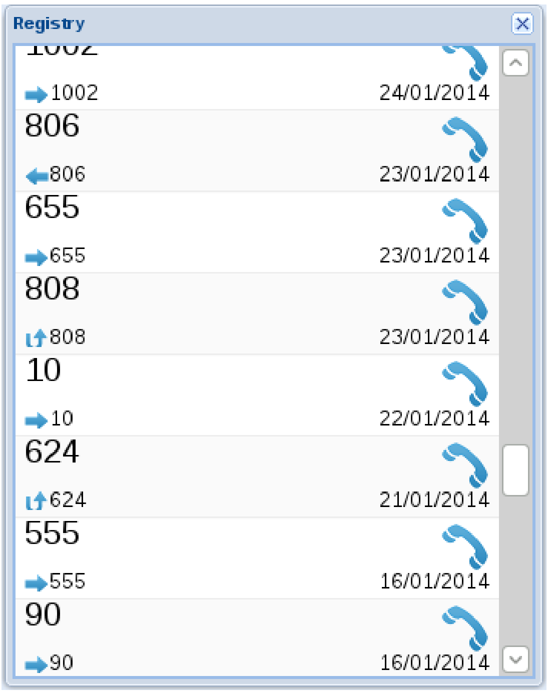


Figure 4.1: User phone interactions.

Call log

Selecting the Registry → Call log menu, the call log window will appear, figure 4.1, reporting all interactions with various numbers.

The log displays all the numbers to which the user has interacted, ordered by most recent interaction. Each entry reports the remote number, the date of last interaction and an icon displays the last interaction type, which can be:

- right arrow: outbound call towards the entry;
- left arrow: inbound, answered call from the entry;
- *U* shaped arrow: missed call from the entry.

By double-clicking on each entry, a full log of all interactions with the number will be shown, each one with the proper type cited above.

Fax log

Selecting the Registry → Fax log menu will activate the personal fax details records for the user, figure 4.2 on the following page, reporting all sent and received faxes, with the status of each transmission or reception. If the administrator has enabled the local fax storage, is also possible to view the fax inline by selecting the document icon or download it by selecting the check box on the appropriate row and pressing *Download Fax*.

A brief description of fields on the fax details records window are:

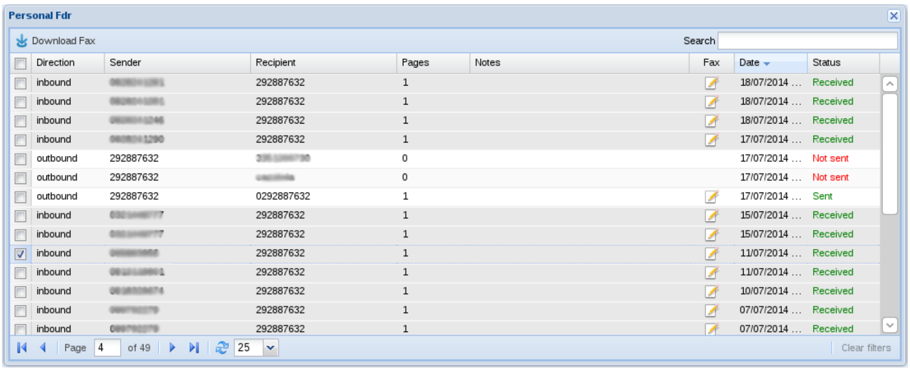


Figure 4.2: User personal fax details records.

- *Direction*: *outbound* if is a sent fax, *inbound* otherwise;
- *Sender*: number of the sender;
- *Recipient*: number of the recipient;
- *Pages*: number of pages transmitted or received;
- *Notes*: if the notes field has been filled up when sending the fax, it will be reported here;
- *Fax*: if a icon is shown, is possible to click on it to see the fax document;
- *Date*: date of the operation;
- *Status*: result status of the operation.

Recordings

Selecting the Registry → Recordings menu will activate the personal recordings details records for the user. The view is the same as described in section 18.3 on page 155, but filtered on the entries that belongs to the currently logged in user.

5

Administrator services

Contents

5.1 Users

Users

> This feature is licensed, you can create or enable items only if you have purchased a users license.

A user in Orchestra NG identifies uniquely every person who needs to interact with the system, and are used to authenticate access to the system, allow web access, assign different permissions, assign resources such as phones, and so on.

Every user has its own account, identified by a unique username. The list of active users can be opened by clicking on the Users → Users menu, figure 2.1 on page 4. On a newly created system, there is usually only one user per domain, the admin user, who has administrator's privileges, and can create other users. The default password for admin is *Voismart*, but it is strongly advised to change it to a more secure one.

> ✍ The administrator user created on the first domain is also the system administrator, i.e. he also has full access to the system administrator web interface.

To add a new user, click on the Add button. The user creation dialog will open, as shown in figure 2.2 on page 5. The parameter to be filled are the following:

***Username*:** the unique name to identify the user;

***Password*:** the user's password, used to authenticate on the system;

***Email*:** the user's email, used for, example, for system notifications;

***Language*:** the user's language, used to properly localize the web interface, the system notifications, . . . ;

***Firstname*:** the user's first name;

***Middlename*:** the user's middle name;

***Lastname*:** the user's last name;

***Timezone*:** the user's time zone;

***Role*:** the user's set of permissions, see section 5.1 on page 34;

***Lcr*:** the user's LCR, used to route outbound calls, see the Orchestra NG administrator manual for details;

***Session Timeout*:** the user's session timeout, in seconds, used to automatically expire the user's authentication to some services, for example the web interface, the timed service class authentication, and so on;

***Identification number*:** a unique user identifier used, for example, when announcing the agent answering a queue call (see section 14.2 on page 123 for more details);

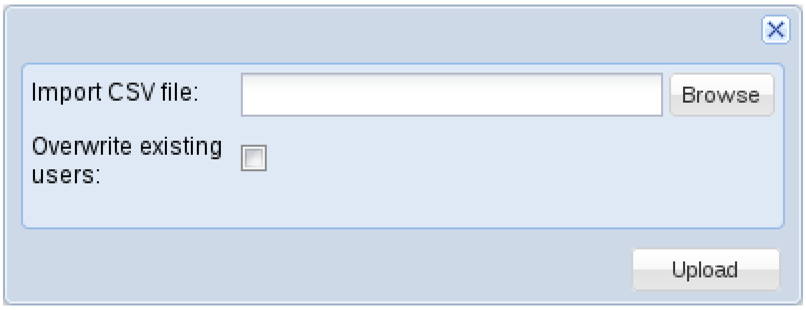


Figure 5.1: Import users dialog.

***Agent queue pause shared*:** if user joins into several queues, after answering in a queue, force to pause in all queues;

***Password validity*:** how long a password is valid (in days). A 0 value means never expires;

***Must change password*:** force a change password.

To modify an existing user, click the Edit button in the main toolbar, to open a window similar to the one shown in figure 2.3 on page 5. Clicking on the Details icon will open the *Edit user* window, which is exactly the same as described in the user creation paragraph.

To delete an existing user, just select the desired item, and click on the Delete button.

> ✍ It is not possible to delete the domain administrator user.

User import and export

To quickly create users in batch, or to restore a previously saved state, Orchestra NG provides import and export functionalities using CSV files.

To export all the domain users, click on the Export button and a CSV file will be downloaded. This file can be used as an import source directly and without further processing. An example file can be found in the appendix B on page 181.

> ✍ When exporting users, the password field is empty for all users, since, for security reasons, user's password are not stored anywhere.

To import users, click on the Import button to open the CSV selection dialog, figure 5.1. Then, select the import source file using the *Browse* button and click on *Upload* to start the import process. To replace eventual existing users with the one in the file, just check the *Overwrite existing users* option before starting the operation. If this option is not selected and a duplicate is found, an error is generated for that user, and the relevant data is skipped.

The input file format consists in a comma-separated-value file with 25 fields, some of them optional, and whose first line is an header describing the fields. The available field are the following, the ones marked with an asterisk are optional and can be left empty by using an empty string:

username: the desired username;

password: the desired password;

email: user's email address;

firstname: user's first name;

middlename*: user's middle name;

lastname: user's last name;

pin*: user's *PIN*, defaults to none;

timezone: user's timezone, in the *IANA Time Zone Database* format; defaults to "Europe/Rome";

> ✌ Common examples of valid timezones are:
>
> ```
> Europe/Berlin
> America/New_York
> Asia/Tokyo
> UTC
> ```

role user's Role, see section 5.1 on page 34 for details;

lcr*: user's LCR, defaults to none;

fax_enabled*: "true" to enable the fax permission to this user, "false" or empty to disable it; defaults to "false";

mobility_enabled*: "true" to enable the mobility permission to this user, "false" or empty to disable it; defaults to "false";

vmail_password*: user's voicemail password, defaults to none;

> ✍ To disable voicemail for an user, leave all voicemail parameters empty. If any field is set, it will be enabled.

vmail_quota*: maximum voicemail message length, in minutes;

vmail_sendemail*: "true" to send an email notification when a new voicemail message is received, "false" otherwise;

vmail_attachaudio*: "true" to attach the voicemail message to the notification email, "false" otherwise;

vmail_leaveonserver*: "true" to delete the voicemail message after sending the email notification, "false" otherwise;

exten_number_alias*: add an extension to the user with the given SIP number;

exten_username*: the extension's username, if not set, defaults to the extension number;

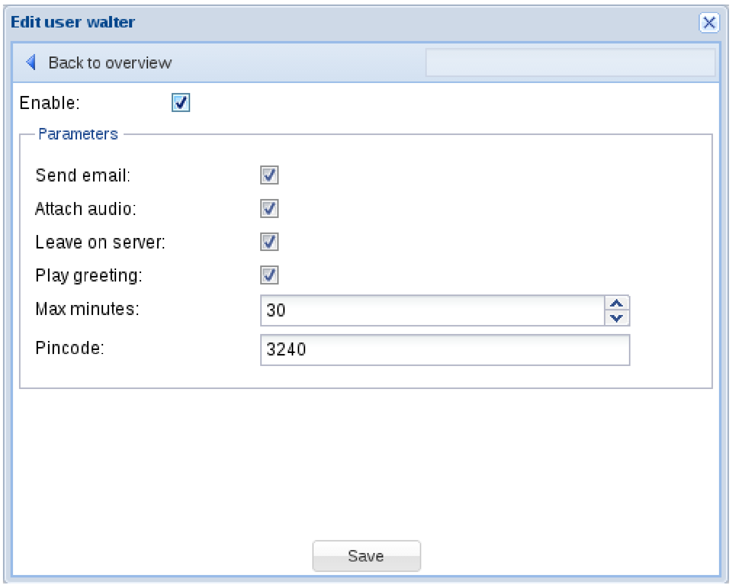


Figure 5.2: Voicemail configuration.

exten_password*: the extension's password, if not set, will be randomly generated;

exten_ringtime*: the extension's ring time;

exten_ringtime*: the extension's MAC address, used for autoprovisioning;

exten_mapping*: the extension's Mapping;

exten_aliasmap*: the extension's Alias Map;

An example of a valid file can be found in the appendix B on page 181.

When the import operation is completed, a GUI notification will be shown. In case of errors, like invalid input data and so on, an additional notification is shown and it is possible to download a new CSV file with the list of failed records, containing one line per error.

> ✌ The format of the error file is one row per error, with three fields: the line number relative to the import file, the username, and an error code and message.

Voicemail

For a general introduction to the voicemail system in Orchestra NG, see chapter 10 on page 93. In this section only the user's configuration parameters for voicemail will be described.

To enable and configure the voicemail for a user, just click the Voicemail icon in the *Edit user* window, figure 2.3 on page 5. A dialog like the one in figure 5.2 will open.

The first check box on that dialog, enables or disable the voicemail service for the selected user. The next parameters are:

Send email: check to enable email notifications upon receiving a voicemail message;

Attach audio: check to receive the voicemail message attached to the notification email;

Leave on server: uncheck to automatically delete the voicemail message after the notification has been sent;

Play greeting: check to allow users to record and use their own greeting message played when leaving a new message, see section 10.4 on page 96;

Max minutes: the maximum voicemail length that it is possible to leave, in minutes;

Pincode: the numeric code required to listen to and delete voicemail messages using the phone, see section 20.16 on page 171.

> ✍ If the *Leave on server* option is not checked, the audio message will be sent to user's email, but it will not be possible to listen to the message via Feature Codes, since the message will be immediately deleted.

> ☞ Do not uncheck both *Send email* and *Leave on server* options, or it will not be possible to listen to the voicemail and no notification will be sent.

Enabling fax service

> ℹ This feature is licensed, you can create or enable items only if you have purchased a fax users license.

To enable the fax service for a user, click the Fax icon in the *Edit user* window, figure 2.3 on page 5. To allow the selected user to send faxes, mark the *Enable* check box, and press the *Save*. For further details on the fax service, see chapter 8 on page 75.

Enabling mobility service

> ℹ This feature is licensed, you can create or enable items only if you have purchased a mobility users license.

To enable the mobility service for a user, click the Mobility icon in the *Edit user* window, figure 2.3 on page 5. To allow the selected user to configure his own mobility rules, mark the *Enable* check box, and press the *Save*. For further details on the mobility service, see chapter 13 on page 111.

Call forward

Call forward allows to setup up simple call forwarding rules for basic states resulting from calling a user. This is implemented as a subset of the mobility functions. To configure it, select Call forward icon. For further details consult section 13.2 on page 112.

Recall

To enable the recall service for a user, click the Recall icon in the *Edit user* window, figure 2.3 on page 5.

The recall service will allow the selected user to initiate a recall request to a local extension, if the call to the extension is not successful.

When not successful and the service is enabled, the system will playback a message reporting called user status and asks to hold on to initiate the recall request. If the call is closed now, the recall request will not start, otherwise, after 5 seconds the system will confirm that the recall request has been started.

It is possible to start a recall request to the last called local extension by dialing the recall feature code. See chapter 20 for details. The system will treat this request as the *no answer* case cited below.

The recall features takes in account different failed calls scenarios:

- busy: if the called user is busy, the system will monitor the called user and will initiate a call to the callee when the both callee and called are free; when the callee answers, the system will connect to the original called party. If the called party switch to busy state again, the recall request must be initiated again;
- no answer: if the called party does no answer, the system will monitor it for an outbound call. As soon as the called party outbound call is over, the system will initiate a recall to the callee (if free) and on answer connect it to the called party;
- unreachable: if all called party extensions are not reachable, the system will monitor for called user availability. As soon as the called party is online again, the system will initiate a recall to the callee (if free) and on answer connect it to the called party;

> ✍ The recall request has a time to live of 1 hour. All recall requests not yet processed within this time frame will be canceled and must be initiated again.

> ✍ Multiple recall requests to same user from same callee will be merged into one. Different recall requests to same user will be served with a first in first out queue.

> ☞ The recall feature works only between local extensions!

> ☞ If the called user forward the call to any external device or voicemail, the recall request cannot be initiated, since the system does not have the state of the external endpoints. Use the recall feature code cited in chapter 20 to initiate a recall request, that will be treated as the *no answer* case cited above.

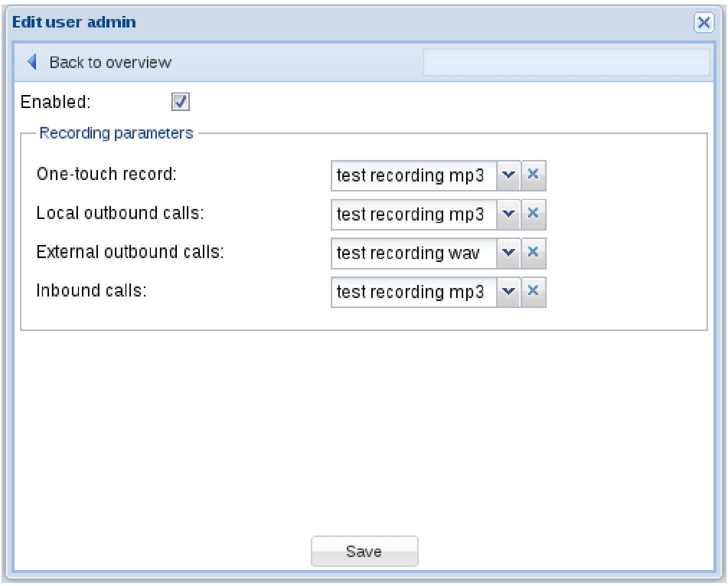


Figure 5.3: Call recording configuration.

Enabling Call Recording service

> This feature is licensed, you can create or enable items only if you have purchased a recording users and recording channels license.

To enable the Call Recording service for a user, click the Call recording icon in the *Edit user* window, figure 2.3 on page 5. This feature activates the call recorder for the selected user. To enable it mark the *Enable* check box, select which rule you want to assign to the user and press the *Save*. See picture 5.3. For further details on the service, see chapter 18 on page 151.

Users groups

A group is a way to reference a set of users using a descriptive name, and is a quick way to organize them for example to quickly assign them to queues, fax numbers, distribution lists, and so on.

To create a group, click on the Users → Users groups menu, to open the dialog shown in figure 5.4.

To create a new group, click on the Add button and insert a group name, used to uniquely identify it, and a generic description.

When created, users can be assigned to it by clicking on the Add users action next to it, which opens the user association window, figure 5.5. To associate users, either select them from the right pane and click on the *Add* button, or click on the *Add all* button to add every user, eventually matching the filter query entered in the above field. In a similar way, to disassociate users just select the desired items from the left pane and click the *Remove* button, or use the *Remove all* to disassociate all.

To delete a group, just select it, and click on the Delete button .

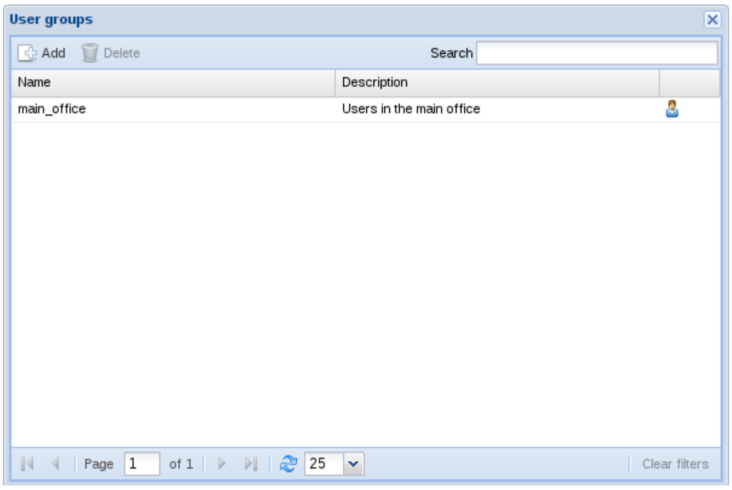


Figure 5.4: Group creation dialog.

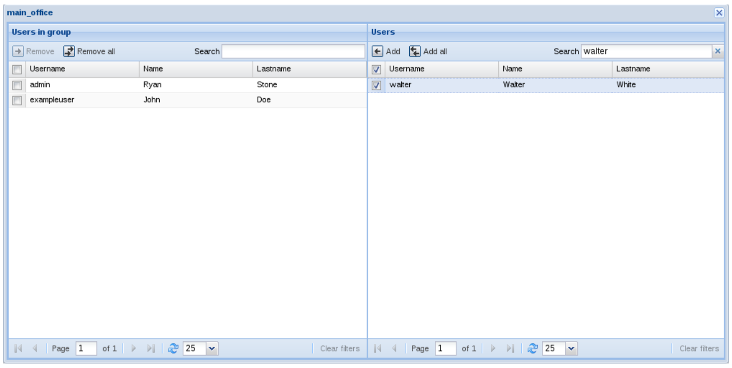


Figure 5.5: Dialog to associate users to a group.

> ✍ If a group is associated, for example, to a queue, it is not possible to delete it, unless it is disassociated first. In such case a GUI error message will appear to notify that it is currently used.

Roles

Orchestra NG provides features to limit access to specific functions on a per user basis, and allows things like granting some of the administrator functionalities to users, prohibiting features to them, differentiating user access, and so on.

In Orchestra NG, a *permission* is access to a specific function, such as *User Management*, *Operator panel user*, *Dialplan Management*, etc. A desired set of permission can be grouped together to create a *Role*, which defines a specific whitelist of granted permissions. A Role has a name which can be something descriptive such as *Limited administrator*, *Common user*, and so on. Every user must then be assigned to a proper role in the system, either upon creation, or when modifying it. In both cases, this can be done using the *Role* field in the *User details* window, as shown in figure 2.2.

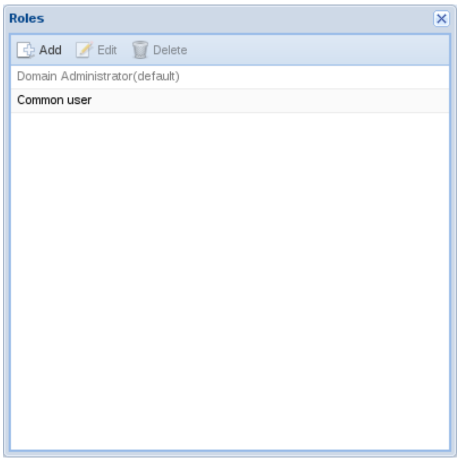


Figure 5.6: Roles creation dialog.

The role list window can be reached using the Users → Roles menu, figure 5.6.

Roles can be created, modified and deleted using the Add, Edit and Delete buttons respectively.

✍ The *Domain Administrator* role cannot be modified or deleted.

The creation role dialog (or edit), is shown in figure 5.7. The role name can be specified in the top input field, while the desired permissions can be selected by marking the appropriate check box below it.

A brief description of the role meaning follows:

Authentication on other phones: authenticate user on an extension via http api;

Bunch Management: read and write access to user's groups;

Bunch view: read access to user's groups;

CDR Management: can read and export CDR data;

CDR view: can only read CDR data;

Calendar Management: read and write access to system calendars;

Calendar view: read access to system calendars;

Conference Booking Management: can book a conference;

Conference Management: read and write access to conference rooms;

Conference view: read access to conference rooms;

Contact Management: import and export phone books, read and write access to external phone books;

Contact management group: write access to group contacts;

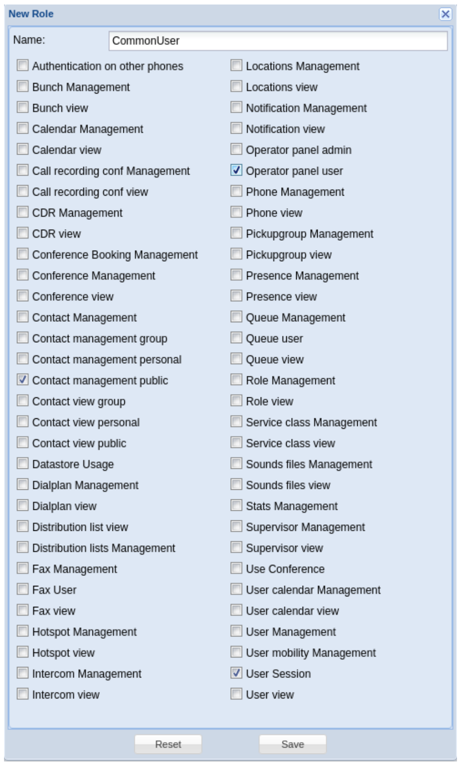


Figure 5.7: Roles creation dialog.

Contact management personal: write access to personal contacts;

Contact management public: write access to public contacts;

Contact view group: read access to group contacts;

Contact view personal: read access to personal contacts;

Contact view public: read access to public contacts;

Datastore Usage: read and write access to the custom data store;

Dialplan Management: read and write access to dialplan-related objects, like the main dialplan, emergency numbers, mappings, etc.

Dialplan view: read access to the dialplan;

Distribution list view: read access to distribution lists;

Distribution lists Management: read and write access to distribution lists;

Fax Management: read and write access to fax numbers and mail to fax configuration and can read FDR.

Fax User: can send faxes and read own fax registry;

Fax view: read access to fax numbers and mail to fax configuration;

Hotspot Management: read and write access to hotspot configuration and welcome page;

Hotspot view: read access to hotspot configuration and welcome page;

Intercom Management: read and write access to intercom service;

Intercom view: read access to intercom service;

Locations Management: read and write access to user location;

Locations view: read access to user location;

Notification Management: read and write access to notifications;

Notification view: read access to notifications;

Operator panel admin: can use the operator panel and modify layouts, queues, etc.

Operator panel user: can user the operator panel with no write operations allowed;

Phone Management: read and write access to extensions;

Phone view: read access to extensions;

Pickupgroup Management: read and write access to pickup groups;

Pickupgroup view: read access to pickup groups;

Presence Management: read access to presence informations;

Presence view: read access to presence informations and user and phone listings;

Queue Management: read and write access to queues;

Queue user: can add and remove self from a queue;

Queue view: read access to queues;

Role Management: read and write access to roles;

Role view: read access to roles;

Service class Management: read and write access to service classes;

Service class view: read access to service classes;

Sounds files Management: read and write access to sound files;

Sounds files view: read access to sound files;

Stats Management: can use the statistics panel;

Supervisor Management: read and write access to supervisors;

Supervisor view: read access to supervisors;

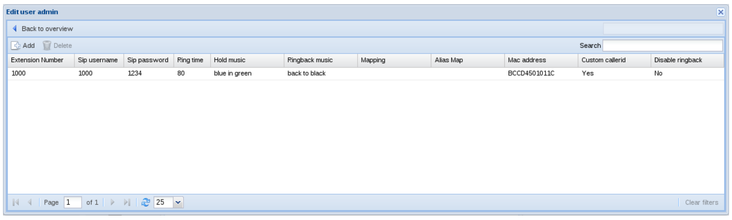


Figure 5.8: Dialog for user's extensions.

Use Conference: can use the live conference;

User Management: read and write access to users;

User Session: can use the web interface;

User calendar Management: read and write access to user's calendar;

User calendar view: read access to user's calendar;

User mobility Management: read and write access to own mobility;

User view: read access to users.

Extensions

> This feature is licensed, you can create or enable items only if you have purchased a extensions license.

An extension in Orchestra NG can only be configured if associated with a user. One user can have multiple extensions and every extension can be registered on multiple devices.

To add, modify or delete extensions, two main interfaces can be used.

The first one is used to access all extensions belonging to a particular user, and can be opened by clicking on the Extension icon in the *Edit user* window, figure 2.3 on page 5.

Figure 5.8 shows this dialog window. To add a new extension, click on the Add button and enter the extension details, according to the following description:

Extension number: the SIP extension's number;

SIP username: the extension's username;

SIP password: the extension's password;

Ring time: the extension's ring timeout;

Hold music: the extension's custom music on hold, see section 5.2 on page 41;

Ringback music: the extension's ringback music on ringing, see section 5.2 on page 41;

Mapping: the extension's outbound map, see section 5.2 on page 47;

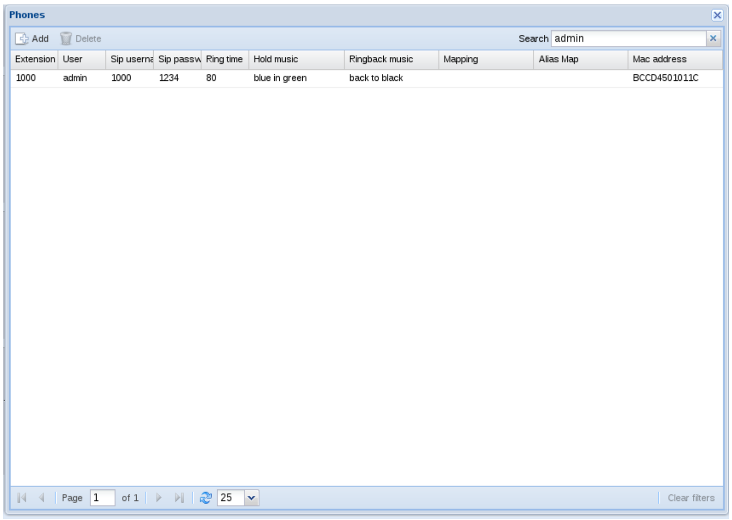


Figure 5.9: Dialog for domain's extensions.

Alias map: the extension's alias map, see section 5.2 on page 48;

Mac address: the extension's MAC address, used for autoprovisioning or remote phone book; see the Orchestra NG Administrator Manual;

Custom callerid: if you enable this field, caller id name will be the display name field of the phone's account;

Disable ringback: if you mark this checkbox, no ringback tone or music will be sent, delegating ring tone generation to the calling device; this is ignored if the call is being recorded;

To delete the selected extension, just click the Delete button.

> ✍ When an extension is deleted and immediately recreated, its presence information are lost until a new event is received (such as a register operation or a call). If this is not desired, just manually start a register request.

> ✍ If the extension is associated with some outbound recording rules, the *Disable ringback* value is ignored, and a ring back will always played back. This happens because the recording feature always needs media to be active, and disabling ring back disables also early media, which is not compatible with the recording functions.

The second interface can be reached by clicking on the Services → Phones menu, and provides at a glance all the extensions defined in the current domain, figure 5.9. The interface is very similar to the one described above, and shares all the configuration parameters, with only the user selection field added, to correctly associate extensions to a user.

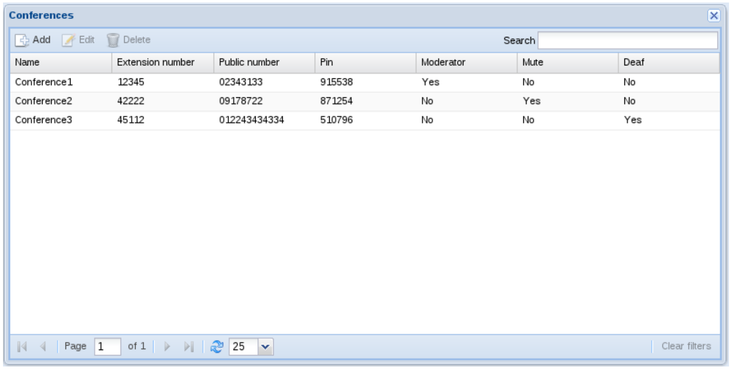


Figure 5.10: Conference list window.

5.2 Services

Conferences

This is the configuration section which allows to define the conference rooms this domain can use.

The main configuration window (figure 5.10) can be accessed by clicking on Services → Conferences. Follow a description of conference's parameter:

***Name*:** name of the conference, any label;

> ✌ You can create a conference that has multiple entries with same name so you can differentiate members who will join.

***Extension number*:** unique number, in domain, used to differentiate access to the conference with the same name. This number can be called using Feature Codes;

***Public number*:** number used to verify access by conference's service in Dialplan Editor;

> ✌ If multiple entries have same public number, when member joins the conference, needs to insert extension number.

***Pin*:** unique and not mandatory number generated manually or randomly used to verify access to conference;

***Moderator*:** enable or disable moderator feature. When moderator leaves conference, all other members are kicked out;

***Mute*:** enable or disable mute feature for member who joins conference with this extension number;

***Deaf*:** enable or disable deaf feature for member who joins conference with this extension number.

Queues

> ℹ This feature is licensed, you can create or enable items only if you have purchased a queues license.

For more details about queues, refer to chapter 14 on page 121

Dialplan Editor

The Orchestra NG Dialplan Editor allows to setup any kind of inbound call in a graphical way. See chapter 15 on page 129 for a detailed explanation of all the features it provides.

Sound Files

This is the configuration section which allows to define the sound files this domain can use. Sound file can be used in queues, in ivrs or as a dialplan block in Dialplan Editor (refer to section 15.2 on page 130). The main configuration window (figure 5.11) can be accessed by clicking on Services → Sound Files. In this window a list of all configurated sound files are shown. Clicking on the *Add* button a new window (figure 5.12) will open and you will be able to create a new sound file. It follows a description of parameters that you can configure, when you add or edit a soundfile:

***Name*:** name of the sound file, any label;

***Description*:** brief description of the sound file;

***Upload File*:** to upload a file, click on *Browse* and select a file that you prefer. The supported file types are WAV and MP3.

Clicking on *Upload* button in window 5.12 you will complete add or edit operation of a sound file.

If you click on icon, you can download file on your device.

You can edit inline a sound file by double clicking a record. In this case you can change only *Name* and *Description* parameters.

To remove a sound file, select a record and click on *Delete* button.

Fax numbers

This is the configuration section which allows to define the fax numbers this domain can use, and the their association with users. A user associated with a fax number (either directly or via one of the groups of users it belongs to), will be able to send faxes using it as the sender field, and will receive all inbound faxes sent to that number.

The main configuration window (figure 5.13) can be accessed by clicking on Services → Fax numbers.

Clicking on the *Add* button will create a new entry. The two needed values to input, are the number itself and the associated station ID.

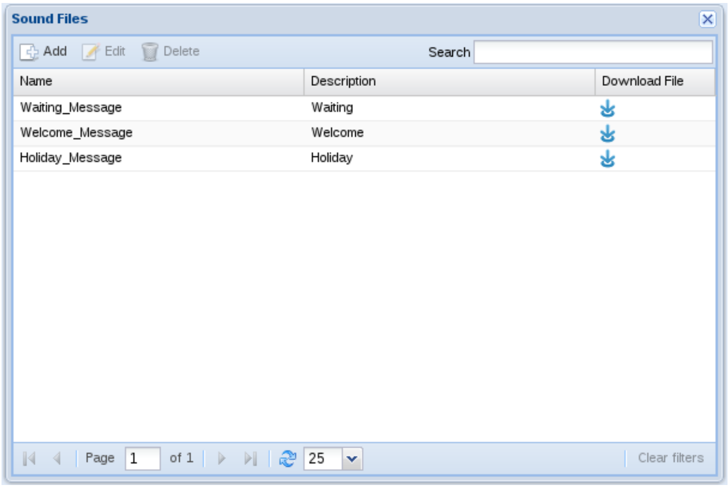


Figure 5.11: Soundfile list window.

Figure 5.12: Soundfiles upload window.

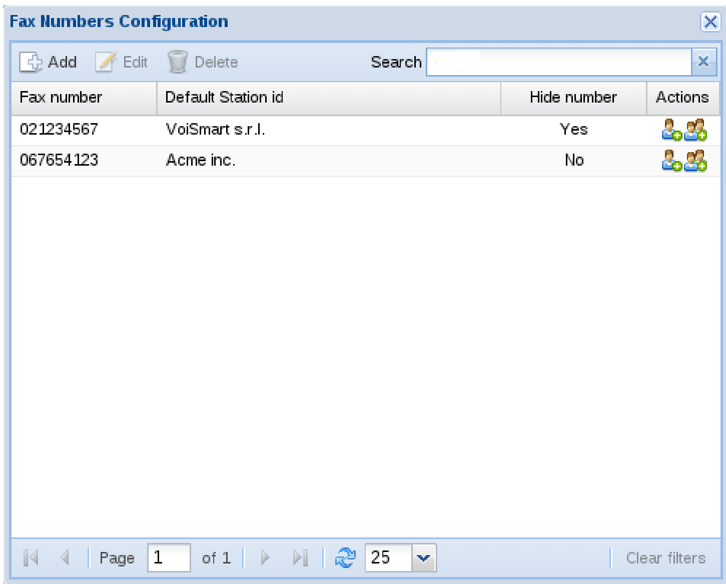


Figure 5.13: Fax numbers configuration dialog.

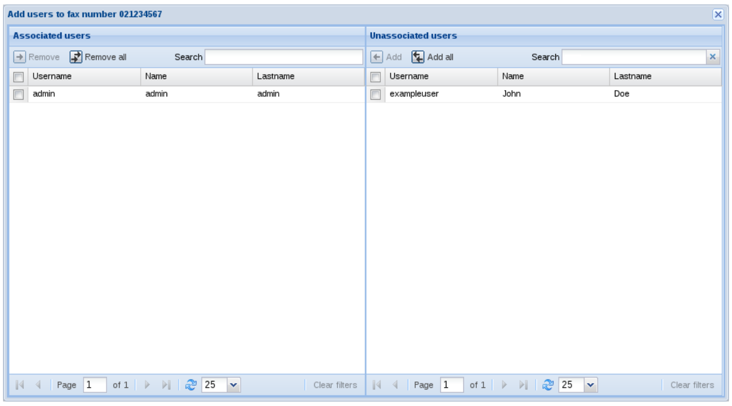


Figure 5.14: Configuration window for user's association with fax numbers.

✌ If no station ID is saved, it will be used the value configured in the E.164 section, in the system administrator's area.

✍ Only numbers matching the Inbound E.164 rules for the current domain can be added (see the Orchestra NG Administrator manual).

✍ When routing a call in the dialplan editor, always make sure that the number reaching the fax dialplan block will match the one defined here, or the inbound fax will not be dispatched to any user.

✍ The sender number is automatically added to the station ID of all sent faxes. To avoid that, it is possible to mark the option *Hide number* in the dialog shown in figure 5.13.

Users can be associated by selecting the Add users or Add groups actions. A window with the list of currently selected users or groups will open, which is modifiable by clicking the *Add* or *Delete* buttons. The delete action is straightforward, on the other hand, adding new users will open a window like the one in figure 5.14, which shows on the left pane all associated users, and on the right pane the remaining.

To remove users from that number, select the desired items and click the *Remove* button, or click the *Remove all* to disassociate all. Similarly, to add new users, use the *Add* and *Add all* buttons.

✍ If a filter is applied, the *Remove all* and *Add all* buttons will respect that filter, i.e. only items matching it will be added or removed.

Time Based Rules

In this section you can create calendars and events used as dialplan block in Dialplan Editor (refer to section 15.2 on page 130). If you create a calendar

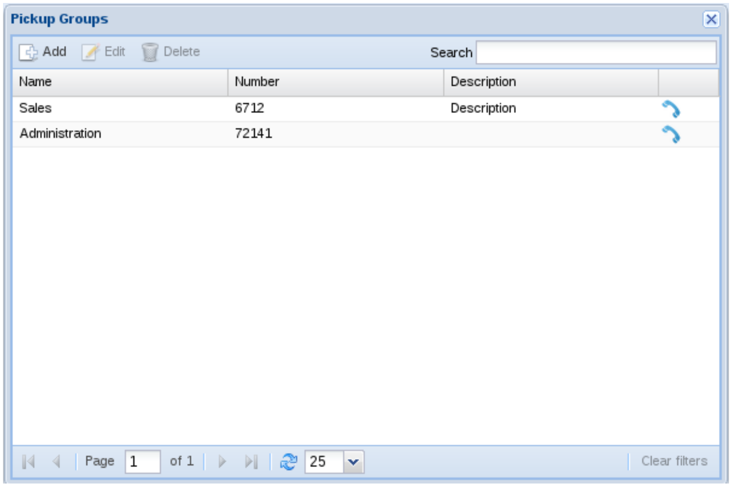


Figure 5.15: List of pickup groups window.

whose name is a number, you can enable or disable it using Feature Codes in section 20.3. For more details about calendars' configuration in Orchestra NG, refer to chapter 17 on page 145

Pickup Groups

This is the configuration section which allows to define the pickup groups this domain can use. Orchestra NG implements call pickup feature, so you can answer a call at your telephone for another extension in your call pickup group. Directed call pickup, if available, lets you pick up a call for a specific extension even if that extension is not part of your pickup group. The main configuration window (figure 5.15) can be accessed by clicking on Services → Pickup Groups. In this window a list of all configurated pickup groups are shown. To create a new pickup group, click on *Add* button and choose an unique *Number* in the domain, *Name* and a *Description* for this pickup group. The *Number* identifies pickup group and it will be used with feature codes defined in section 20.12. By clicking on icon a new window will open (figure 5.16 on the next page) and you will be able to associate extensions to this pickup group. To remove extensions from that pickup group, select the desired extensions and click the *Remove* button, or click the *Remove all* to disassociate all. Similarly, to add new extensions, use the *Add* and *Add all* buttons.

> ✍ If a filter is applied, the *Remove all* and *Add all* buttons will respect that filter, i.e. only items matching it will be added or removed, also no visible items due to pagination.

> ✍ An extension can belong to only one group. If you add an extension already associated with a pickup group, it will be removed automatically from previous group

To remove a pickup group, select a record and click on *Delete* button.

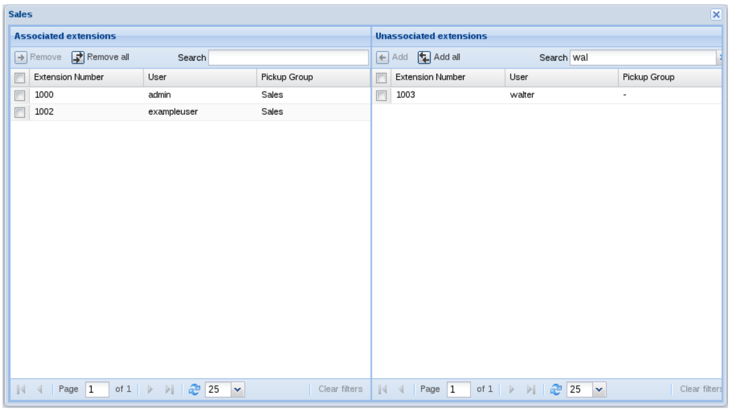


Figure 5.16: Associate extensions to a pickup group window.

Intercom Groups

Intercom, also known as paging, allows users to dial a group of extensions which will automatically answer the incoming call and recipients will hear the message from the speaker on their phones.

Intercom is a one-way audio session; it allows a single extension to communicate with a few people, also called unicast intercom/paging or a potentially large group of people multicast intercom/paging.

Unicast intercom

Unicast paging is a one-to-one call to each extension in the intercom group: the system establishes a regular call to each user through standard SIP calling and bridges them using the built-in conference engine. The audio will be setup as unidirectional from the calling user to the called ones.

> ✍ In large intercom groups, using unicast intercom can be a problem since it consume a large amount of CPU resources. An intercom group with 30 extensions in it is basically a 30 participants conference call. For large groups Multicast intercom should be preferred. The unicast intercom group size is CPU-dependent so the maximum number of members may vary in relation with the underlying available resources.

To create a new unicast intercom, open the main configuration window (figure 5.17 on the following page) by clicking on the Services → Intercom groups.

Description of parameters that you can configure for intercom groups:

***Name*:** short name for the group;

***Number*:** number to dial with the intercom feature code prefixed, see 20.11 on page 169 for reference;

***Description*:** brief description;

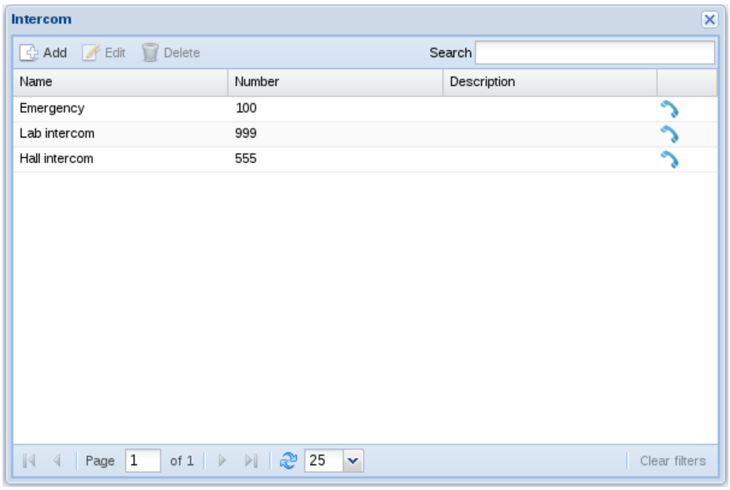


Figure 5.17: Main intercom groups window.

Using the action column on the right, by clicking on the phone icon, is possible to associate extensions to the intercom group. Each extension may belong to more than one intercom group.

Multicast intercom

Multicast intercom is a predefined one-to-many IP address that the phones are configured to listen to. The maximum listen addresses each phone can support depends on the phone model itself, but is common to have up to 10 listen addresses configurable. See also Administrator manual on auto provisioning this kind of configuration.

The advantage of a multicast intercom is that only one call is established between the caller and the Orchestra NG system. Then only one RTP stream is sent over the multicast group on which the phones are listening. If there are 100 phones listening on a multicast intercom, it is equivalent to a single call instead of a conference call with 100 participants. So multicast intercom is suitable for very large groups.

> ✍ Multicast is usually fine in local area networks, but will not work with remote phones, since specific routing must be setup to allow multicast traffic between different networks. Please check with the network administrator if multicast must be setup to allow to reach remote phones with intercom. Otherwise unicast intercom may be used, keeping an eye on resources usage if the group is big.

A multicast intercom call can be started using the appropriate feature code, see 20.10 on page 169. The group number represents the index number of the configured multicast address in the caller phone auto provisioning configuration. If the caller phone is not associated to an auto provisioning configuration the multicast intercom will not work.

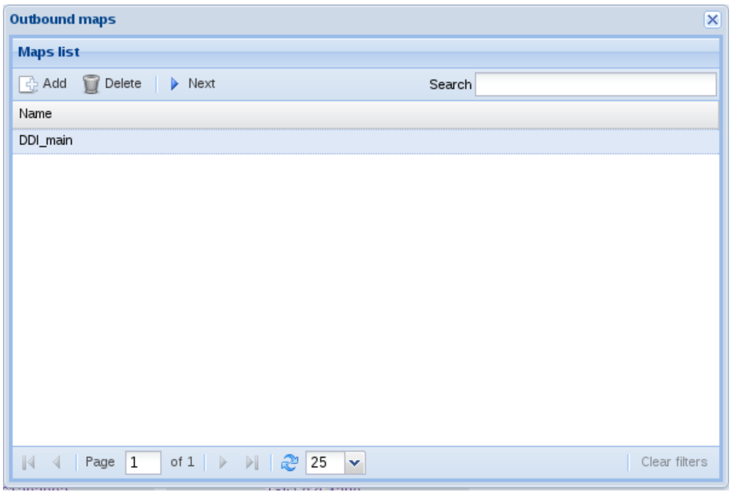


Figure 5.18: Main outbound maps window.

✍ Multicast intercom heavily depends on the type of phones used. Since many phones support maximum ten listen addresses, the number of multicast intercoms that can be used is ten.

✌ If the caller phone has an auto provisioning configuration which has 224.0.0.1:1234 multicast address in the 3rd position, to start a multicast call to 224.0.0.1:1234 3 must be used as <num> in the feature code. On the other hand, if 3 is our multicast group number, all phones that belongs to this group must have a common multicast listen address in their configuration, which must be the 3rd in the caller phone auto provisioning configuration.

Outbound maps

Outbound maps allow to remap caller numbers in a powerful way, using regexp substitution, see appendix D on page 185. It consists in a prioritized set of rules which can be applied to outbound calls, rewriting the caller number.

To create a new mapping, open the main configuration window (figure 5.18) by clicking on the Services → Outbound maps. In that dialog, click on the Add button and create a new mapping using a descriptive name. To add new rules, select it and click on the Next button to enter the outbound maps rules dialog, figure 5.19.

New rules can be created and destroyed using the Add and Delete buttons. A new rule consists in a descriptive name, and the match and replace patterns. The *Match rule* is the pattern against which the caller number is matched. If a match is found, the *Replace rule* pattern is used to rewrite the caller number, using the set of rules explained in appendix D on page 185.

Rules can be quick-tested by using the Test button and inserting the number to test and pressing *Ok*. The replaced number will be shown in a dialog, or a brief message will appear if no rules matches that number.

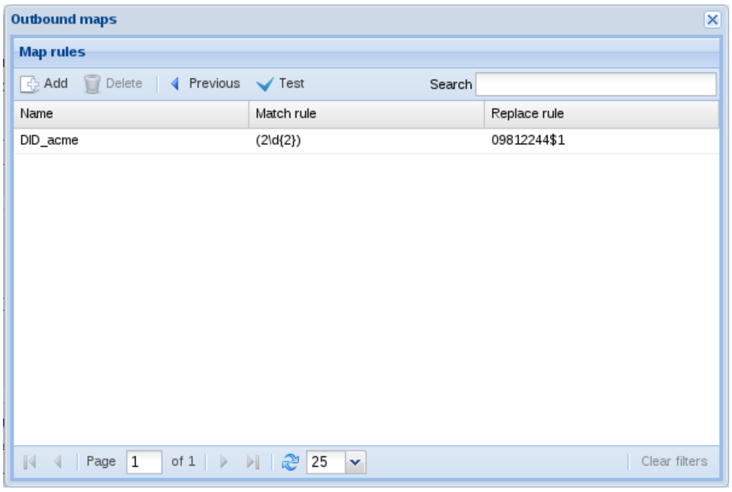


Figure 5.19: Outbound maps rules window.

Outbound maps are only used by the extensions associated to them using the *Outbound maps* field in the extension configuration dialog, section 5.1 on page 38.

An outbound map is not aware of the carrier that will be used. The same outbound map will be used on all carriers associated with the user's LCR.

If a DID service is available, using this outbound map, you can set caller number to the extension DID number:

Name	Match Rule	Replace Rule
DID_acme	`(2\d{2})`	`09812244$1`

This rule will expand every 3-digit number, that starts with 2, to the DID number.

Remind that caller numbers replaced by outbound map works only if it matches with inbound E.164 numbers in its domain. If there is no match, caller's number is marked as private.

To delete an outbound maps, just select it and click on the Delete button.

Alias maps

Alias maps allows to remap destination numbers in a powerful way, using regexp substitution, see appendix D on page 185. It consists in a prioritized set of rules which can be applied to outbound calls, rewriting the called number.

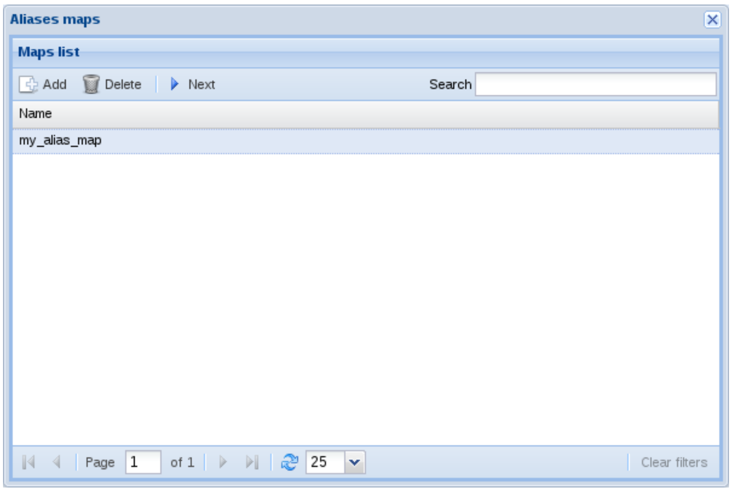


Figure 5.20: Main alias maps window.

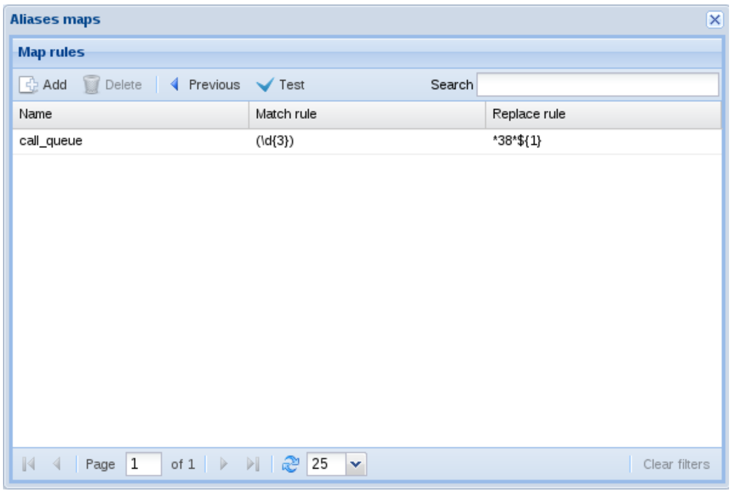


Figure 5.21: Alias maps rules window.

> ✍ Alias maps expansion happens very early in the call setup, but those rules will not be applied to the defined emergency numbers, if any.
>
> On the other hand, they will be applied before the check for feature codes, service classes and short numbers.

To create a new mapping, open the main configuration window (figure 5.20) by clicking on the Services → Alias maps. In that dialog, click on the Add button and create a new mapping using a descriptive name. To add new rules, select it and click on the Next button to enter the alias maps rules dialog, figure 5.21.

New rules can be created and destroyed using the Add and Delete buttons. A new rule consists in a descriptive name, and the match and replace patterns. The *Match rule* is the pattern against which the called number is matched. If a match is found, the *Replace rule* pattern is used to rewrite the called number, using the set of rules explained in appendix D on page 185.

Rules can be quick-tested by using the Test button ✓ and inserting the number to test and pressing *Ok*. The replaced number will be shown in a dialog, or a brief message will appear if no rules matches that number.

> ☞ Be extremely careful when writing alias rules and try to be as strict as possible with the match patterns, because it is possible to cut out unwanted numbers if using too general patterns.
>
> For example, since alias expansion is done even before user extensions lookup, with badly written rules it may be possible to even disallow calling other extensions or users.

Alias maps are only used by the extensions associated to them using the *Alias maps* field in the extension configuration dialog, section 5.1 on page 38.

> ✌ To call a queue from an extensions, the `*38*` feature code can be used (as explained in section 20.5 on page 167).
>
> To quickly call a queue without having to remember the feature code, the following alias map can be used:
>
> | Name | Match Rule | Replace Rule |
> |---|---|---|
> | Quick_Queue | `(\d{3})` | `*38*$1` |
>
> This rule will expand every 3-digit number to the feature code to call the appropriate queue.

To delete an alias maps, just select it and click on the Delete button.

Service classes

This is the configuration section which allows to define the service classes this domain can use. Orchestra NG is able to assign certain features and restrictions to each individual extension or user associated with a particular service class. A variety of features and/or restrictions are grouped together and can be assigned to more than one extension or user. The main configuration window (figure 5.22) can be accessed by clicking on Services → Service Classes. In this window a list of all configurated service classes are shown. To create a new service class, click on *Add* button and choose an unique *Name* in the domain for this service class. By clicking on icon a new window will open (figure 5.23) and you will be able to associate users to this service class. By clicking on icon a new window will open (figure 5.24 on page 52) and you will be able to associate extensions to this service class. In both figure 5.23 and figure 5.24 on page 52 to remove items (users or extensions) from that service class, select the desired items and click the *Remove* button, or click the *Remove all* to disassociate all. Similarly, to add new items, use the *Add* and *Add all* buttons.

> ✍ If a filter is applied, the *Remove all* and *Add all* buttons will respect that filter, i.e. only items matching it will be added or removed, also

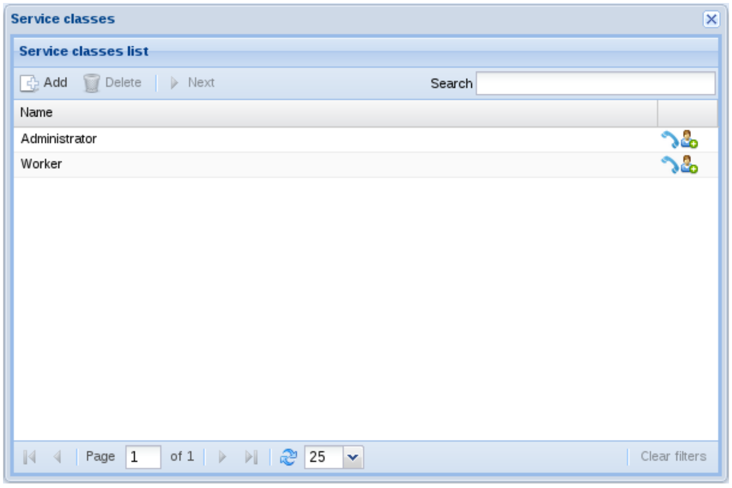


Figure 5.22: List of service classes window.

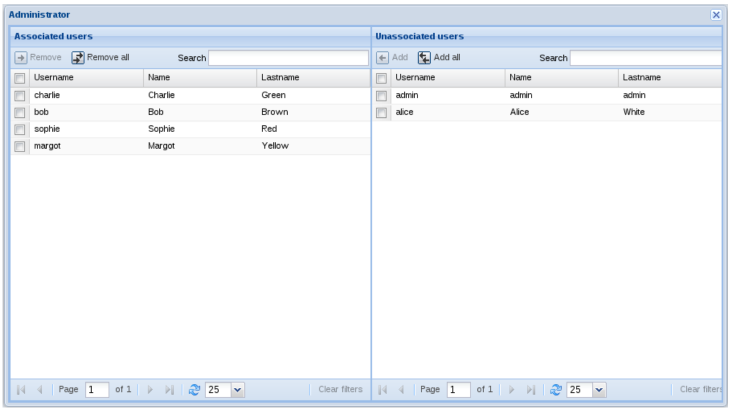


Figure 5.23: Associate users to a service class window.

no visible items due to pagination.

After selecting the service class and pressing the *Next* button it is possible to edit the service class rules, as shown figure 5.25 on the next page. For each rule you can configure these parameters:

***Priority*:** the rule priority, can be modified by dragging the rule and dropping it in the desired position;

***Description*:** a brief description for the rule;

***Match Rule*:** a regexp against which the dialed number will be matched;

***Authentication*:** if enabled, pin is asked to retrieve the user.

Each rule is a pattern match against the dialed number and this match is used in schema shown in figure 20.1 on page 174. The match priority can be adjusted by dragging the rule row in the desired position and dropping it.

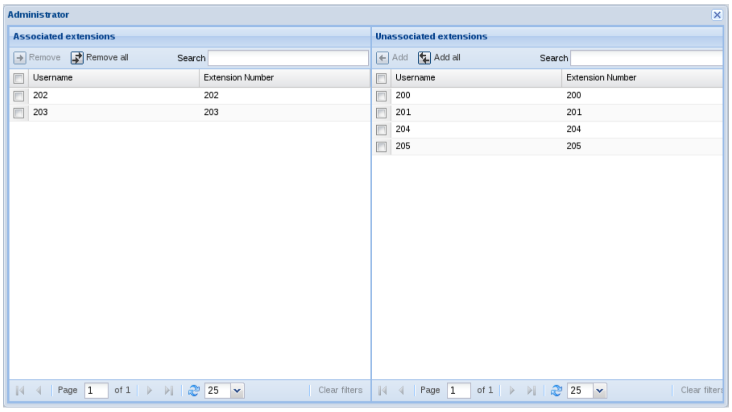


Figure 5.24: Associate extensions to a service class window.

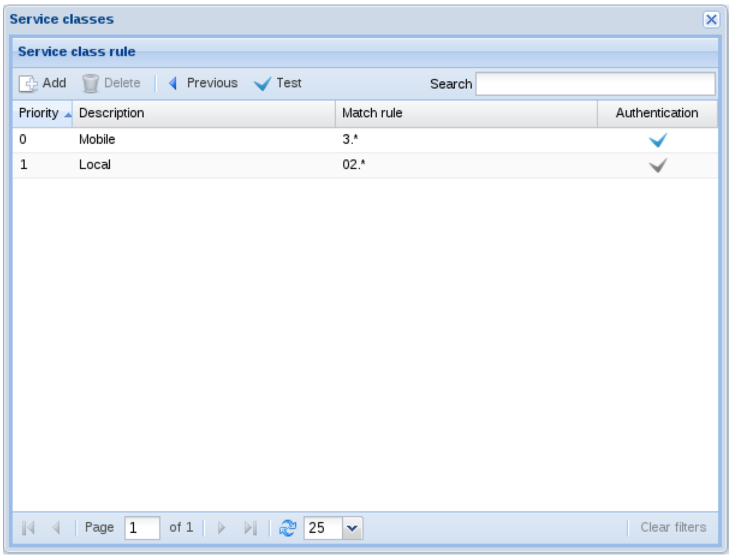


Figure 5.25: Service class rules window.

By using the *Test* button it is possible to check which rule is engaged for a specific number. Just write the number and press test. The selected rule will be highlighted in the service class rules window. See figure 5.26 on the next page for reference.

To remove a service class, select a record and click on *Delete* button.

Emergency numbers

This is the configuration section which allows to define the emergency numbers this domain can use. Emergency numbers are special numbers as they are called by Orchestra NG regardless of any service classes or alias maps enabled on the phones from which the call starts. The main configuration window (figure 5.27) can be accessed by clicking on Services → Emergency Numbers. In this window a list of all configurated emergency numbers are shown. Clicking

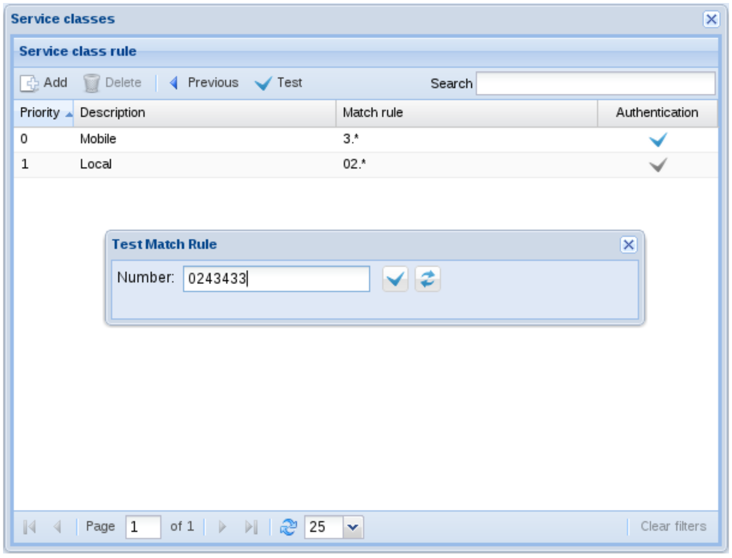


Figure 5.26: Service class rules test function window.

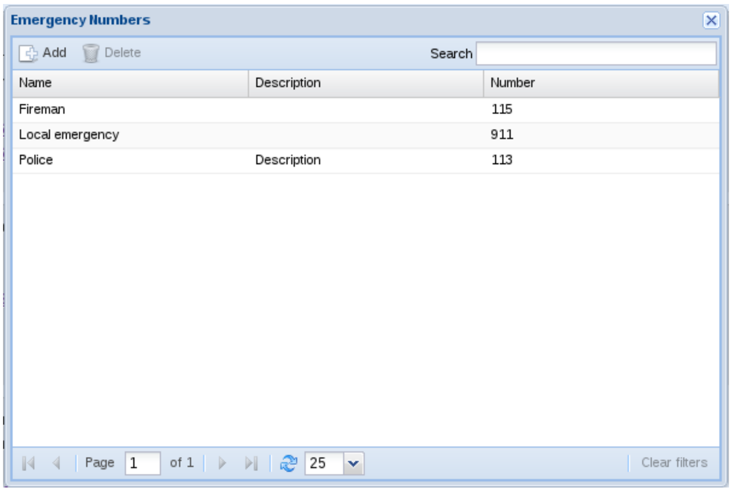


Figure 5.27: Emergency numbers configuration window.

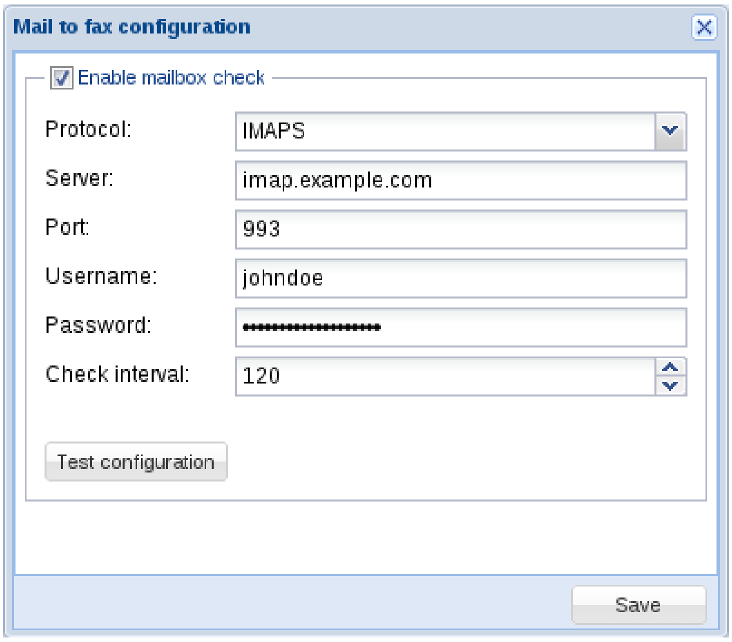


Figure 5.28: Mail to fax configuration window.

on the *Add* button you will be able to create a new emergency number. Follow a description of parameters that you can configure, when you add or edit an emergency number:

***Name*:** name for emergency number, any label;

***Description*:** brief description for the emergency number;

***Number*:** unique number in this domain for the emergency number.

> It is not possible to configure regular expressions for number field.

To remove an emergency number, select a record and click on *Delete* button.

Mail to Fax configuration

This section covers configuration of the mail to fax service in client mode, as described in section 8.2 on page 78.

To enable the service, mark the checkbox as shown in figure 5.28. After filling in all the details and saved the configuration, the system will periodically check the given mailbox for messages, and, after submitting the tasks to the fax server, will delete the messages.

The needed configuration parameters are as follows:

***Protocol*:** the retrieval protocol to be used to access the mailbox;

***Server* and *port*:** the server and port on which the mailbox resides. The standard port will be automatically filled in after selecting the protocol, but can be manually overridden;

***Username* and *password*:** mailbox credentials;

***Check interval*:** time interval after which the mailbox will be check again, in seconds.

> ✌ The supported protocols are:
>
> - POP or IMAP, to use the plain text protocol;
> - POPS or IMAPS, to use the secure protocols only;
> - POP or IMAP with TLS only, to use the plain text protocol, but only if an upgrade to encryption is possible.
>
> The supported authentication schemes are:
>
> **POP:** PLAIN, APOP and CRAM-MD5;
>
> **IMAP:** PLAIN, LOGIN and CRAM-MD5.

> ✍ It is highly recommended to use the secure or TLS only protocols to avoid sending the access credentials unencrypted over the network.

Call record configuration

The Orchestra NG system allows to setup a full featured recording server. See chapter 18 on page 151 for a detailed explanation of all the features it provides and how to configure and use it.

5.3 Reports

This section shows how calls and faxes, in domain-wide, are stored by Orchestra NG.

Cdrs

Orchestra NG stores a complete log of all inbound and outbound calls for logging purposes. Domain-wide call logs can be accessed by clicking on Reports → Call logs and shows all calls records originated or received from the user's domain.

The Call logs dialog looks like the one in figure 5.29 and consists of a list of call records with the following fields:

Direction: call direction, can only be "inbound" or "outbound". "Inbound" direction means that call is directed to pbx, while "outbound" direction means that call is generated by pbx;

From: the sender's caller id name;

To: the recipient's number;

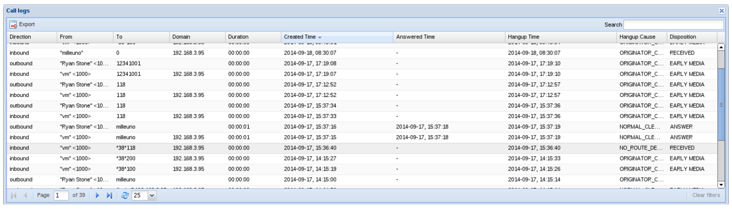


Figure 5.29: Cdrs list window.

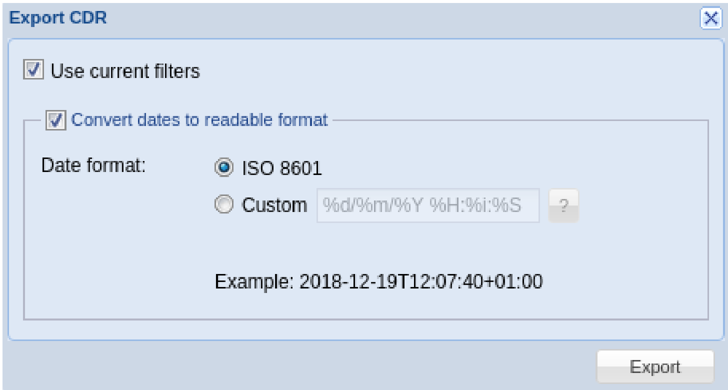


Figure 5.30: Cdrs export window.

Domain: (only available to the domain administrator), show call's domain;

Duration: the call's duration;

Created Time: date and time when the channel was created;

Answered Time: date and time when the channel was answered;

Hangup Time: date and time when the channel was closed;

Hangup Cause: hangup's cause;

Disposition: when available, it is hangup cause returned.

> ✍ The number of records for each call may vary depending on which service is called.

You can also export part or all call records as a CSV file. In the dialog shown in figure 5.29 by clicking on button, a new window will open as shown in figure 5.30.

In this dialog you can choose to export all records or to export filtered records. The records can be filtered by using an advanced search which will cause the export process to only select the rows matching the given filters. You can filter records according to several match rules and/or temporal range as shown figure 5.31 Using the system administration web interface you can also filter records by selecting domain using proper combo box.

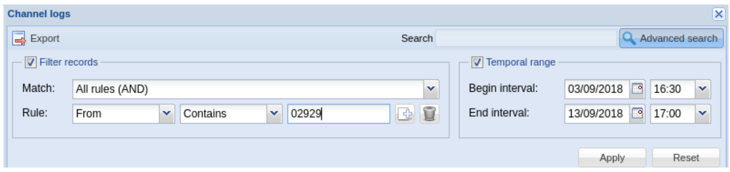


Figure 5.31: Cdrs filter window.

By clicking on the *Export* button, the export process will start and when done, you will be able to download a CSV file. An example of CSV file is shown in appendix C on page 183.

✍ Fields and values exported in the CSV file are not the same as shown in the GUI (i.e. date values are exported as Unix Timestamp microseconds).

To automatically convert dates in an ISO 8601 format using the user's timezone, the checkbox *Convert dates to readable format* can be enabled and use ISO 8601 format or a custom format.

☞ If there are many call records, the export process can take a long time, so please wait!

Fdrs

Orchestra NG stores a complete log of all inbound and outbound faxes for logging purposes. Refer to section 8.4 on page 80 for more details.

Recordings

Orchestra NG stores a complete log of all recorded calls with the relative files for logging purposes. Refer to section 18.3 on page 155 for more details.

5.4 Statistics

The statistics panel, accessible by selecting the Statistics button, provides comprehensive reports of all the inbound calls routed to the queues subsystem, giving the possibility to analyze queues, agents and global call center performances with several, informative charts. For further details refer to chapter 19 on page 159.

5.5 Notifications

All of the notifications users can receive, can be customized from the system administrator using the Notification menu.

Every notification subsystem has its own menu entry there, but they all share the same interface to create, modify, or delete custom notifications. If no custom messages are defined, the system defaults will be used.

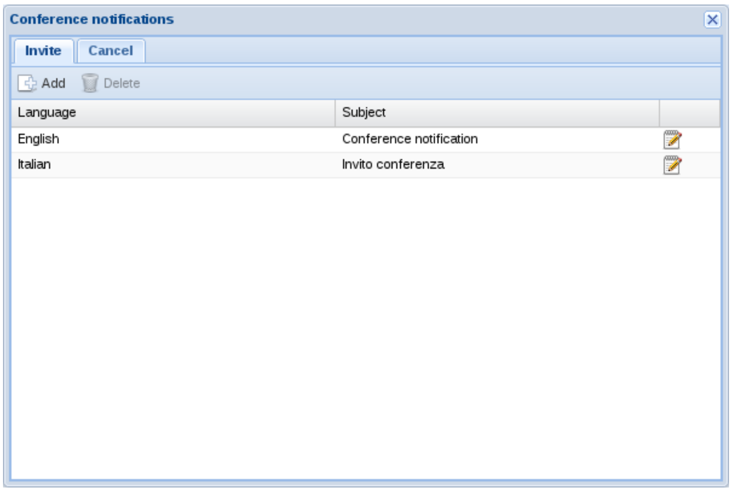


Figure 5.32: Dialog to create, modify and delete notifications.

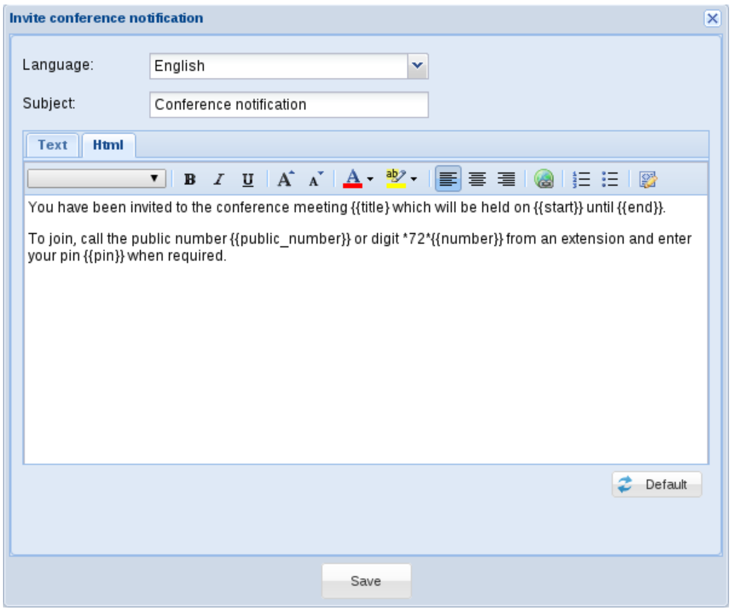


Figure 5.33: Customizing notification.

When clicking on one of the Notification menu entries, a window showing all custom notifications currently saved will open, figure 5.32.

To create a new message, click on the Add button, to open a dialog like the one in figure 5.33. On that dialog, the custom message can be entered either in plain text or using an html editor to have access to some formatting options. The subject and message language can be entered in the two top fields.

✌ Every user will receive the notification message in their own language, so, in order to have correct custom messages for all users on the system, it is advisable to add an entry for every language used in that domain.

The body and subject line use a simple templating system to insert some information which is not known at definition time, like, for example, the notification date. Before sending the message, every template term will be replaced with its correct value. The set of possible terms to be used is different depending on the context, so it will be introduced in the later sections. The general syntax, however, is to replace every occurrence of the terms enclosed in double curly braces with the corresponding value.

In the context of conference notifications, the special terms `title`, `start` are available (among some others). So, the text

```
You have been invited to the {{title}}, starting on {{start}}.
```

will be expanded as

```
You have been invited to the Weekly HelpDesk Meeting, starting on
2014-09-13 10:00:00.
```

Invalid terms won't be replaced, so for example

```
This will not be replaced: {{invalid term}}.
```

will be expanded exactly to the same text.

To modify an existing message, click on the Edit notification message button, and change the desired fields exactly as done for the creation.

To delete, click on the Delete button after having selected the desired item.

In Orchestra NG it is also possible to notify a set of related users on every notification received by a monitored user. This feature is called Supervisors, and it is useful to keep track of events not originally directed to the supervisors themselves. See section 5.5 on page 62 for details.

Conference

The conference notification dialog has two main tabs, for two kind of message, the invite notification, and the cancellation one.

The conference invite notification is sent when a user has been invited to attend a booked conference, or some details has been changed. The available template terms are:

Term	Substituted with
`title`	The title of the conference

Term	Substituted with
`start`	The date and time on which the conference will start
`end`	The date and time on which the conference will stop
`public_number`	The public number to call to join the conference
`number`	The conference number to call join the conference via Feature Codes
`pin`	The pin number to digit when required
`url`	The conference url, containing the event details

Web Conference

The web conference notification dialog has two main tabs, for two kind of message, the invite notification, and the cancellation one.

The web conference invite notification is sent when a user has been invited to attend a booked web conference or when he has been invited from *Social Chat* service. The available template terms are:

Term	Substituted with
`title`	The title of the conference
`public_number`	The public number to call to join the conference
`number`	The conference number to call join the conference
`url`	The conference url, containing the event details

Fax

There are four type of fax notifications that can be customized, each one reachable from its own tab.

The *Received* message is sent when some user receives a new fax. The available template terms are:

Term	Substituted with
`sender`	Sender number
`recipient`	Recipient number
`date`	Reception date
`users`	The list of users who received this fax
`supervisors`	The list of supervisors who received this fax
`fax_uid`	A unique string associated with this fax

The *Sent* notification is sent to the user who tried to send a fax to notify a successful operation, while the *Not sent* message is sent when the operation failed for some reason. In both cases, the valid template terms are:

Term	Substituted with
`sender`	Sender number
`recipient`	Recipient number
`date`	Reception date
`duration`	Fax sending duration, in seconds
`pages`	The number of pages sent
`notes`	The text of the (optional) notes field
`reason`	The failure reason (or `OK` if successful)
`users`	The list of users who received this fax
`supervisors`	The list of supervisors who received this fax
`fax_uid`	A unique string associated with this fax

The *Mail to fax* message is sent to the user who tried to send a fax using the mail to fax function, but the request was ignored due to missing informations or because of an invalid email was sent. The available template terms are:

Term	Substituted with
`reason`	The failure reason

Voicemail

A voicemail notification is sent when a new voicemail audio message is recorded. A new email will be sent, and the message can be customized here. The available template terms are:

Term	Substituted with
`username`	The username of the user who received the voicemail message
`domain_name`	The domain of the user who received the voicemail message
`date`	The date and time of reception
`name`	The name of the user who received the voicemail message
`surname`	The surname of the user who received the voicemail message

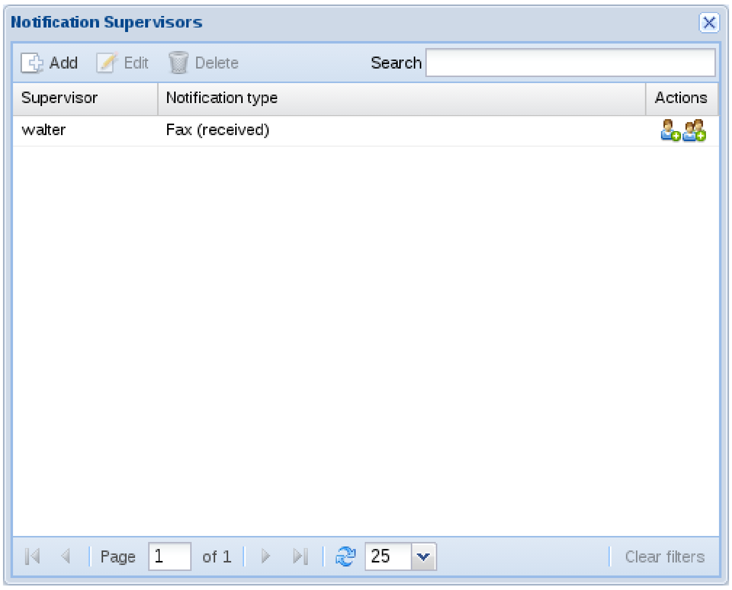


Figure 5.34: Supervisor creation dialog.

Term	Substituted with
`caller_id_name`	The caller ID name of the user who left the voicemail message
`caller_id_number`	The caller ID number of the user who left the voicemail message

Supervisors

Supervisors are system users with a set of configurable associated users under them. When one of this users receives a system notification, if a supervisor is defined for that kind of message, he will receive a copy of the notification.

To create a new supervisor, click on the Add button in the supervisors list window, figure 5.34, and choose the appropriate user in the first selection menu. Then choose the kind of message to be notified of, and click *Save*.

The currently available notification types are:

Fax: to receive copy of every fax notification;

Fax (received): to only receive copy for received faxes;

Fax (sent): to only receive copy for sent faxes.

To associate users to a supervisor, click on the Add users or Add groups buttons to open the users or group selection windows, figure 5.35. On that window, select the desired users (or groups of users) on the right pane and move them to the left pane by clicking on the *Add* button, which operates on the selected ones, or the *Add all* button, which adds all of them, with eventual filters applied.

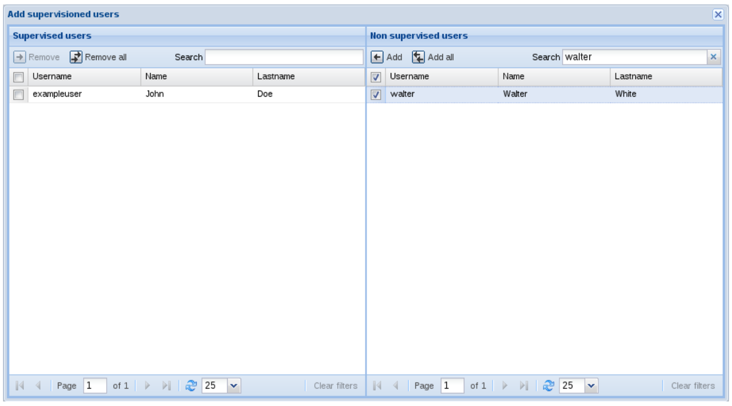


Figure 5.35: Dialog to add users to a supervisor.

6

Personal Area

Contents

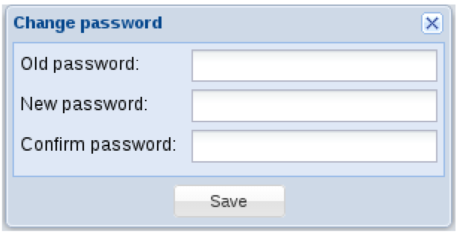


Figure 6.1: Password change form.

6.1 Pin

The PIN in Orchestra NG is a randomly generated number, unique among all domain's users. It is used in several services as an authentication token, like:

- in service classes, see section 5.2 on page 50 and section 20.15 on page 171;
- for fax authentication, see chapter 8 on page 75;
- to check the voicemail, by using it in place of the extension number; see section 20.16 on page 171;
- to enable and disable the hotdesking functions, see section 20.14 on page 170.

With Pin button is possible to read the current user's PIN, or to generate a new one by pressing the *Generate* button on the PIN window.

> The PIN cannot be chosen by the user for security purposes. Since is unique among all users, letting the user to choose it may reveal already set PIN, so opens the way to malicious use of the services authenticated with it.

6.2 Password

With the Password button opens a window, shown on figure 6.1, where a new password can be set, providing the current password. Password minimum length is 6 chars.

> Always choose strong passwords, with a mix of uppercase and lowercase letters, numbers and at least one punctuation mark.

6.3 Session

Using the Logout button terminate the current session, redirecting to the login page. If the session is not explicitly closed, the system will automatically log out the user after the session is idle for the timeout duration configured in the user details. See section 5.1 for reference on configuring the session timeout.

7

Contacts

Contents

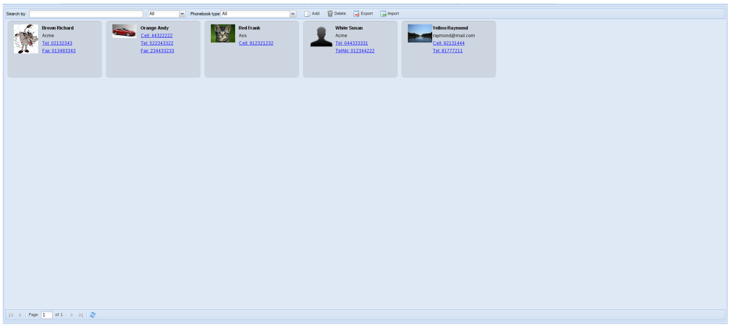


Figure 7.1: Contacts panel window.

7.1 Introduction

Orchestra NG's Contacts Panel, figure 7.1, allows to fully manage contacts saved in phonebooks configurated by administrator. In this panel you will be able to search, create or delete contacts according to your permissions. It is also available click and dial function for each contact phone number. If you click on a number saved as fax, window for send fax (figure 8.1) will open. It is possible to import or export a list of contacts using a CSV file. In section 7.3 on page 70 we will explain how to group contacts to create distribution lists.

7.2 Contacts panel

In main toolbar you can do almost all operations on contacts.

In the first textfield you can insert characters of words or number that you want to find. As soon as you stop typing, live search starts as explained in section 3.5 on page 19. Contact fields on which you can perform searches are: *All*, *Surname*, *First Name*, *Company*, *Number*, *Short Number*. If you choose *All*, it means that it will be used all fields just listed during research. In *Phonebook type* combo box you can select the types of contacts to search. Possible values are: *All*, *Public*, *Personal* and a list of groups to which the user, who's doing the research, belongs. If you choose *All*, all contacts, that user can see, will be returned.

To add a new contact click on *Add* button in main toolbar, while to edit a contact double click on it. In both cases a new window will open as shown in figure 7.2. This window is divided into sections. In section *Contact* you can add personal information of the contact. In section *Photo* you can upload an image of the contact. To add or change image, click on *Browse*, choose a file from your device and than click on *Upload*. The supported file types are JPEG, JPG, GIF and PNG.

> ✍ When you add a new contact *Upload* button acts as *Save* button while in edit it uploads or remove only image.

In section *Phones* by clicking on *Add phone number* button, you can add phone or fax number to this contact. For each number you can define a type: *Office Fax*, *Mobile*, *Office*, *Home Phone*, *Home Fax*, *Secondary Phone*, *Secondary*

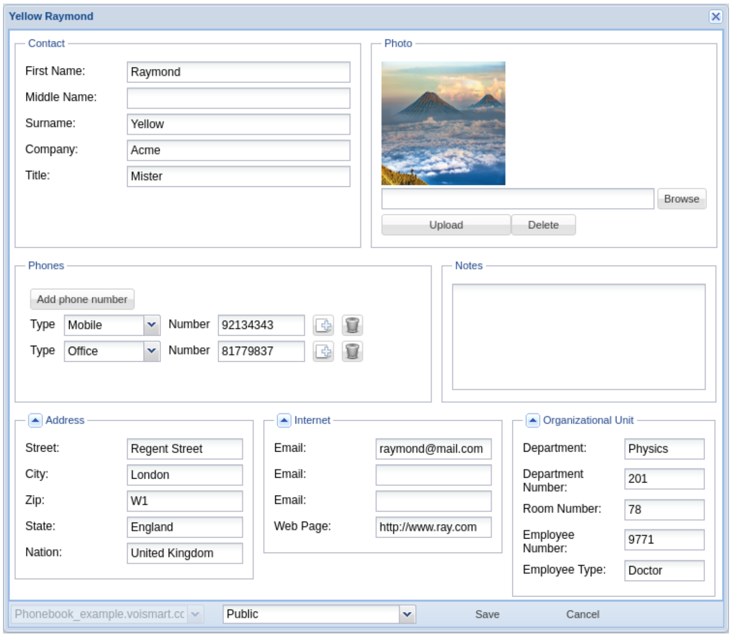


Figure 7.2: Contact details window.

Figure 7.3: Phone details window.

Fax. If you click on icon, a new window will open (figure 7.3). In this window you can configure two parameters:

***Description*:** brief description for this *Short Number*;

***Short Number*:** define a short and alternative number associated to this contact's number. This *Short Number* must be unique in the phonebook and LDAP server must support VsContactsNumber schema. Once you have created it, you can call this contact by dialing his *Short Number*.

In section *Notes* you can add notes about contact. In section *Address* you can fill address contact information. In section *Internet* you can add one or more emails for this contact and a link to a web site. In section *Organizational Unit*, if a contact is a part of an organization, you can fill organizational attributes. If you are adding a new contact, you can choose phonebook, among those configured, where to save the contact. Also you can choose if this contact will be a personal, public or group (among those to which the user belongs). If you

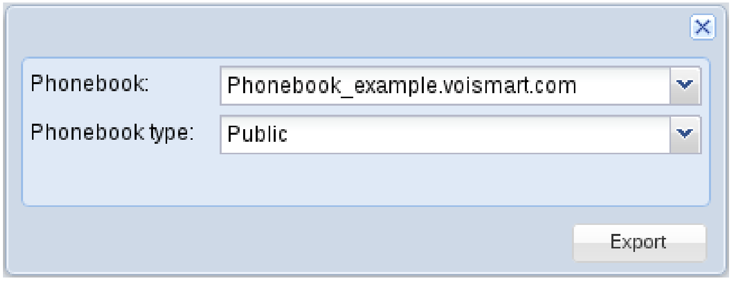


Figure 7.4: Export phonebook window.

select more than a group holding *CTRL* key button, a contact will belong to several groups. To save your changes, click on *Save* button, but remember that you can save contact only in accord to your permissions.

To delete a contact, select a contact and then click on Delete button in main toolbar.

To export a phonebook click on Export button. On click a new window will open (figure 7.4). In this window you can choose:

***Phonebook*:** select a phonebook, among those configurated, to export;

***Phonebook type*:** select which type of contacts to export. Values can be: *Public*, *Personal* or any groups name that belong to user logged in.

By clicking on *Save* button, the export process will begin and when finished, you will be able to download a CSV file with all exported contacts.

> ✍ Remind that contact's image will not be exported.

To import a phonebook click on Import button. On click a new window will open (figure 7.5). In this window you can configure these parameters:

***Import File*:** select a CSV file by clicking on *Browse* button. An example file can be found in the appendix A on page 179;

***Phonebook*:** select a phonebook, among those configurated, where to import new contacts;

***Remove Contacts*:** choose to remove all existing contacts before import new contacts;

***Header*:** enable this field if first row of CSV file contains column (field) names.

By clicking on *Upload* button, the import process will begin. When finished a pop up will tell you how many contacts were imported and, in case of some errors, you can download a file that contains contacts not imported.

7.3 Distribution lists

This is the configuration section which allows to define the distribution lists this domain can use. A distribution list is just a collection of contacts and it can be

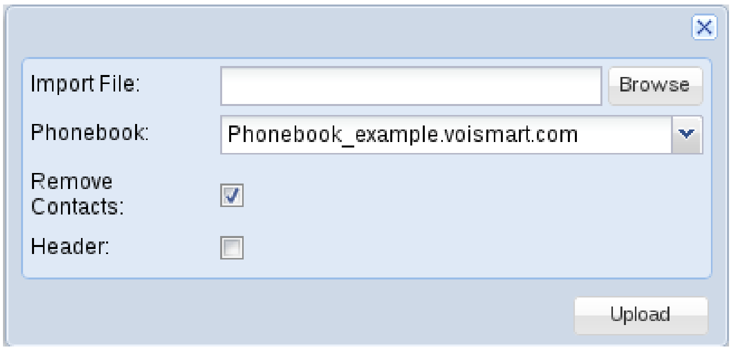


Figure 7.5: Import phonebook window.

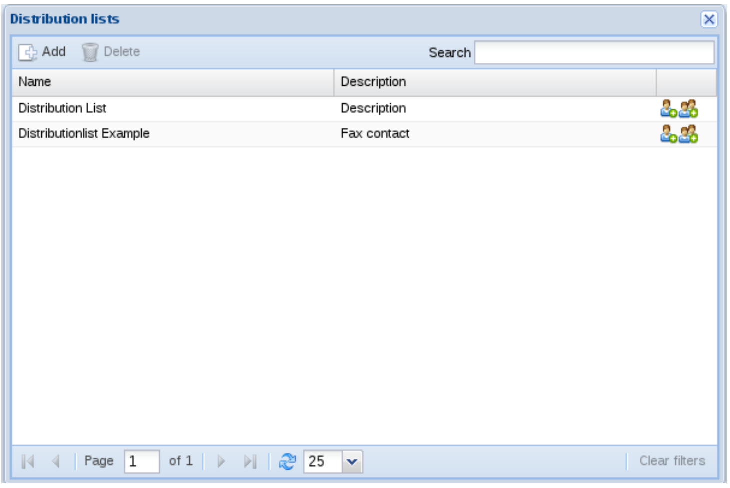


Figure 7.6: Distribution lists window.

used in mobility rules as explained in section 13.4 on page 115, in the Dialplan Editor as explained in section 15.2 on page 130, or to send faxes to multiple recipients, as explained in section 8.2 on page 76. The main configuration window (figure 7.6) can be accessed by clicking on Contacts → Distribution lists.

To add a new distribution list, click on the Add button. You can configure an unique *Name* and add a brief description for each distribution list. After a distribution list entry is created, clicking on the Add users or Add groups icon, a window (figure 7.7) with the list of currently selected users or groups will open. To remove users from that distribution list, select the desired items and click the *Remove* button, or click the *Remove all* to disassociate all. Similarly, to add new users, use the *Add* and *Add all* buttons. Once users are associated with a distribution list, you can choose sharing type for that users by clicking on icon or . A user with a *Viewer* permission can only view phonebook contacts associated with this distribution list. Vice-versa, a user with a *Administrator* permission will be able to add or remove users or phonebook contacts from that distribution list.

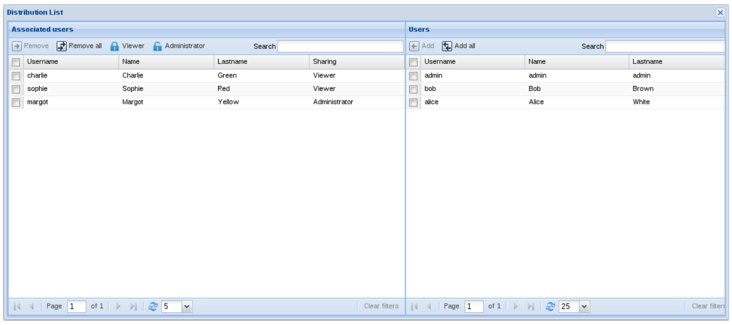


Figure 7.7: Distribution list users window.

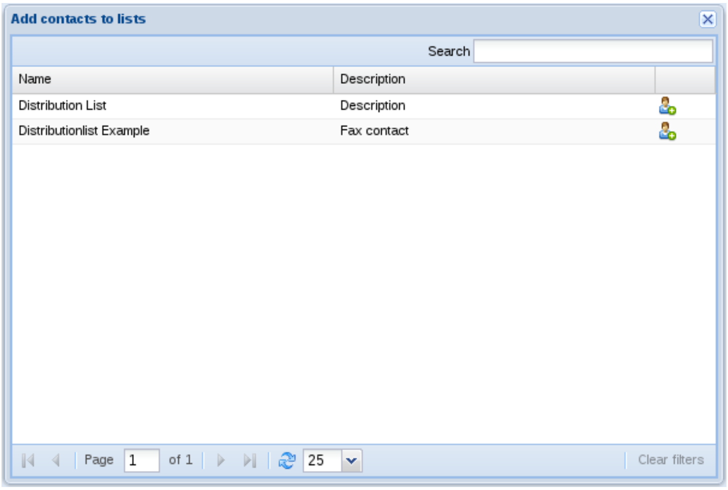


Figure 7.8: Add contacts to distribution lists window.

> ✍ If a filter is applied, the *Remove all*, *Add all*, *Viewer* and *Administrator* buttons will respect that filter, i.e. only items matching it will be added, removed or change sharing type, also no visible items due to pagination.

To delete an existing distribution list, just select the desired item, and click on the Delete button.

7.4 Add contacts to lists

This is the configuration section which allows to add or remove phonebook contacts to a distribution lists. This operation is possible only if you have *Administrator* permission for the distribution list, as explained in section 7.3 on page 70. Clicking on Contacts → Add contacts to list a new window (figure 7.8), with a list of distribution lists that you can manage, will open. Clicking on the icon, a window (figure 7.9) with the list of currently selected phonebook contacts will open. To remove phonebook contacts from that distribution list, select the desired items and click the *Remove* button, or click the *Remove all*

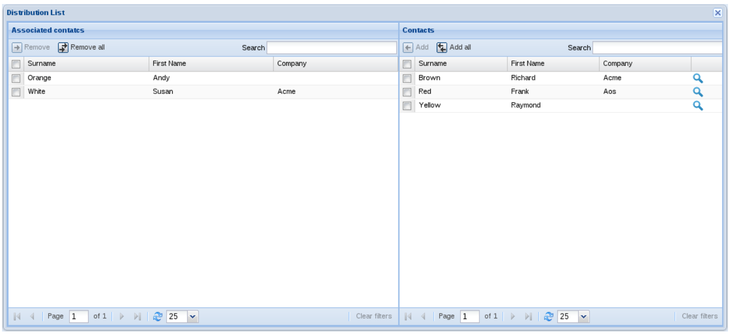


Figure 7.9: Associate contacts to distribution lists window.

to disassociate all. Similarly, to add new phonebook contacts, use the *Add* and *Add all* buttons. If you click on icon, detail contact window, as figure 7.2, will open.

> ✍ You can add only phonebook contacts that you can see, according to your phonebook permissions

> ✍ If a filter is applied, the *Remove all* and *Add all* buttons will respect that filter, i.e. only items matching it will be added, removed, also no visible items due to pagination.

8

Fax

Contents

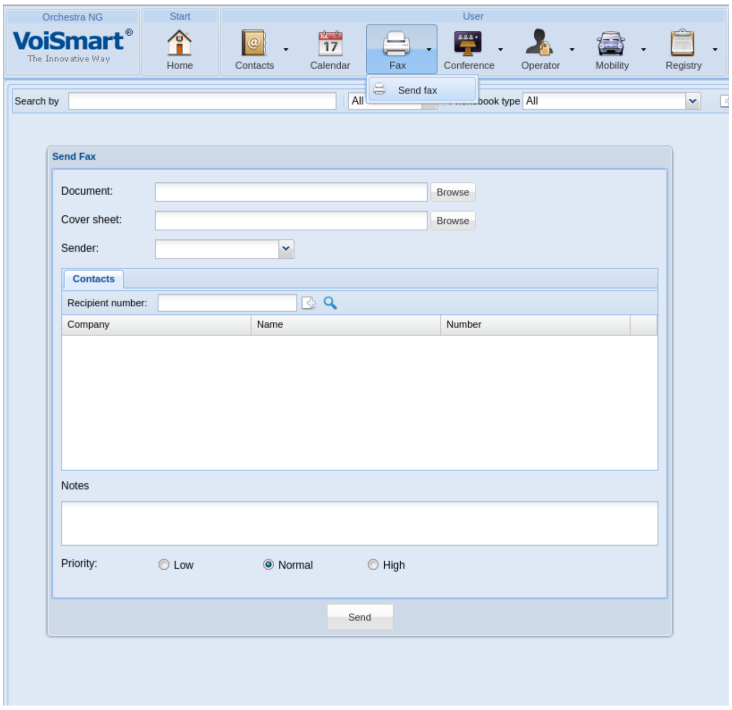


Figure 8.1: Send fax dialog.

8.1 Introduction

The Orchestra NG Fax Server is a complete solution for creating, sending, receiving and storing faxes in an easy and intuitive way. It provides a simple web interface to send and access document, and takes care of queueing, prioritizing, routing according to least cost rules and retrying transmission if needed. Inbound Faxes are dispatched to the configured users, and a notification system is extensively used to inform users on received documents or transmission result. Mail-to-fax and Fax-to-mail functions are available to make fax operations as easy as email management.

The Orchestra NG Fax Server natively supports PDF and Postscript documents, and has support for virtually any file format via the VoiSmart Fax Printer client.

8.2 Sending faxes

A new fax can be sent by clicking on the Fax → Send fax menu, which will open a dialog like the one in figure 8.1.

The document to send and an optional front page can be selected using the first two fields. The supported file types are PDF and PS.

In the *Sender field*, the sender number must be selected, choosing among the ones associated to the logged-in user. See section 5.2 on page 41 for details on how to change or add numbers for users, or the Orchestra NG administrator manual to configure fax numbers for the current domain.

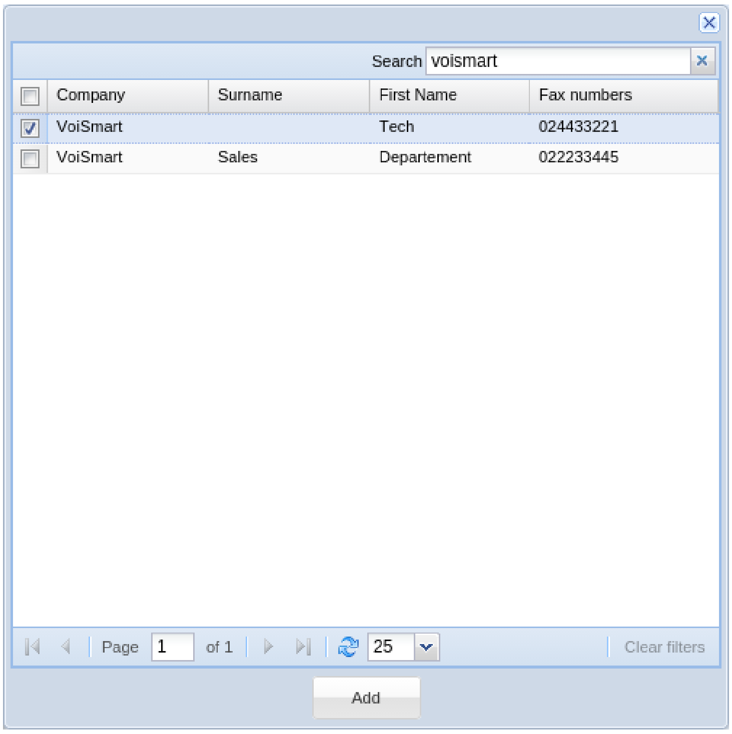


Figure 8.2: Dialog to add fax recipients by searching through contacts.

One or more recipients can be added by directly entering the number in the input field in the *Contacts* form and then clicking on the Add button, or by searching in the contacts. For the latter, click on the Search fax numbers button, a window like the one in figure 8.2 will appear with all the phone book contacts having a fax number; then select the desired items, and add them.

Alternatively, the document can be sent to all the fax contacts in a distribution list by clicking on the *Lists* form and by selecting and adding the desired item from the combo box. For details on distribution lists, please refer to section 7.3 on page 70.

To remove a number or a distribution list, just click on the Trash button.

The field *Notes* allows to attach a little description for the sent document. This text will be stored along with the fax, and can be accessed or searched by using the fax logs window (see section 8.4 on page 80).

> ✍ The text of the note will not be sent with the fax in any way and thus cannot be accessed or read by the receiver. It is only available to the sender.

Once all fields have been filled-in, the document can be sent by clicking on the *Send* button. A small notification will be shown upon successful transmission, or in case of errors. In any case, the sender will receive an email notification with the result of the operation. To customize this message, see section 5.5 on page 57.

Sending faxes using email

If mail-to-fax is enabled, faxes can be simply submitted to the system by sending a properly formatted email.

> ✌ There are two possible configurations for the mail-to-fax gateway: client mode and server mode. When the latter is used, Orchestra NG acts as an SMTP server and the email must be sent to the configured domain (see the Administrator manual for details). In client mode, an external mailbox is monitored and new messages are periodically retrieved and processed (see section 5.2 on page 54).
>
> Client and server mode can be used together without any problem.

In client mode, a new email must be sent to the configure mailbox, with the document attached (in PDF format) and the following fields in the email body using the format *parameter: value*:

pin: the user's pin, as explained in section 6.1 on page 66;

domain: the configured fax domain, as explained in the Orchestra NG Administrator Manual;

to: recipient number or distribution list name; can appear multiple times (in different lines) to have more recipient if needed;

from: optional sender number;

notes: optional notes field.

In server mode, a new email must be sent using `rcpt@faxdomain` as recipient, where `rcpt` is the recipient fax number or distribution list and `faxdomain` is the configured fax domain (see the Orchestra NG Administrator Manual for details).

> ✌ If your system administrator has configured your fax domain to `fax.voismart.com` and you want to send a fax to the numbers 021234567, 067654321 and to a list named `MyFaxContacts`, your email recipients will be
>
> `021234567@fax.voismart.com`
>
> `067654321@fax.voismart.com`
>
> `MyFaxContacts@fax.voismart.com`

The document to send must be attached (in PDF format) and the following fields in the email body using the format *parameter: value* must be present:

pin: the user's pin, as explained in section 6.1 on page 66;

from: optional sender number;

notes: optional notes field.

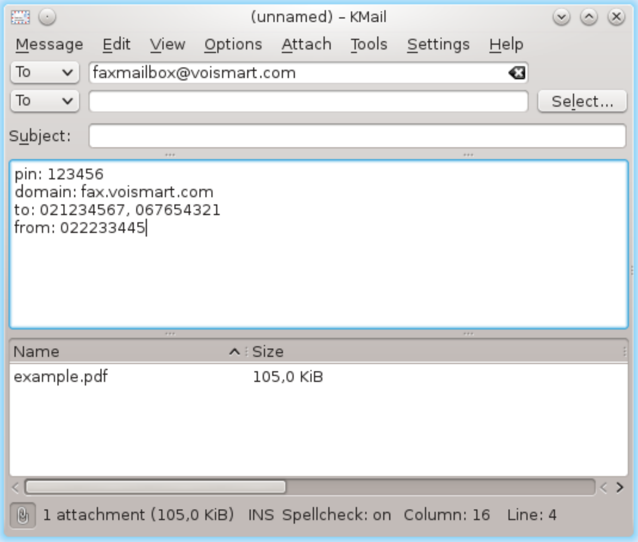


Figure 8.3: Example message in client mode.

> Eventual other recipients, not relevant to the configured fax domain or fax mailbox, will be delivered as normally expected, e.g. sending an email to
>
> `021234567@fax.voismart.com`
>
> `john@example.com`
>
> will send a fax to 01234567 and deliver a copy to `john@example.com`.

Figures 8.3 and 8.4 show an example email for client and server mode respectively.

8.3 Receiving faxes

When Orchestra NG is correctly configured, inbound faxes are automatically dispatched to the correct users based on the number. Every recipient, will then receive a notification email with various details, and the received fax as an attachment in PDF format.

The association between fax numbers and users or group of users is created or modified in the domain administration section, explained in section 5.2 on page 41. The email at which each user will receive the notification is configured by the system administrator when creating users. See section 5.1 on page 27 for further details.

Custom messages and informations about the fax notification email can also be configured by the system administrator. See section 5.5 on page 57 for details.

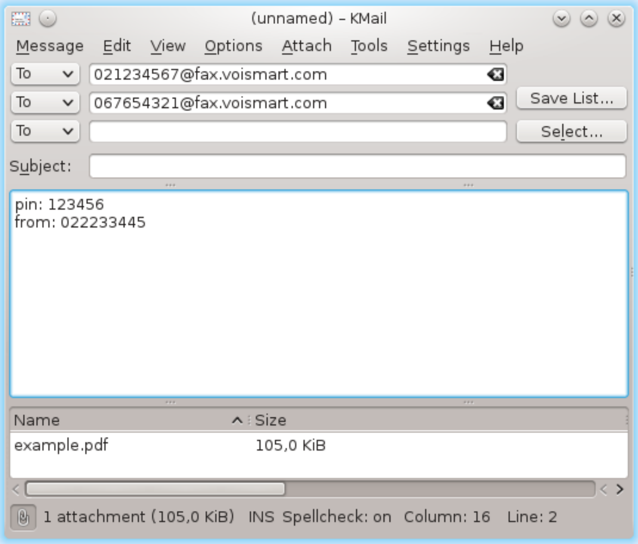


Figure 8.4: Example message in server mode.

8.4 Fax logs

Orchestra NG stores a complete log of all inbound and outbound faxes for logging purposes.

System-wide fax logs can only be accessed by the system administrator using the system administration web interface. Domain-wide fax logs can be accessed using two different dialogs, the first one can be reached by clicking on Reports → Fax logs and shows all fax records originated or received from the user's domain; the second one, by clicking on Registry → Fax log, which shows all records relevant to the user currently logged in. The access to these dialogs is protected by a different set of privileges, so generally only the latter can be opened by users with no administrator permissions (see section 5.1 on page 34 for details on user privileges).

The Fax log dialog looks like the one in figure 8.5 and consists in a list of fax records with the following fields:

Domain: (only available to the system administrator), shows the user's domain who sent or received the fax;

Direction: call direction, can only be "inbound" or "outbound";

Users: (only available to the domain administrator), shows the user who sent the fax, or the list of users who received it;

Sender: the sender's number;

Recipient: the recipient's number;

Pages: the number of pages of the document;

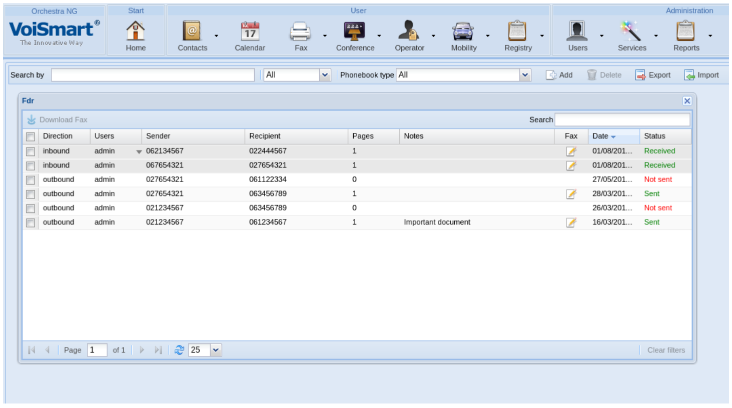


Figure 8.5: Fax details record.

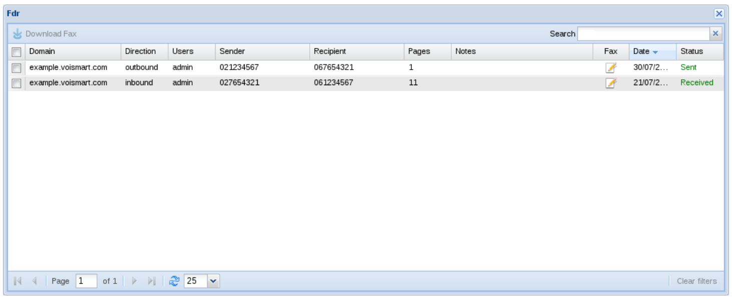


Figure 8.6: Fax admin details record.

Notes: optional description text added when sending the fax;

Fax: by clicking on this icon, the fax preview window will open (see section 8.4 for details); this icon will not be available if the document transmission didn't end successfully;

Date: date of reception or sending;

Status: result of the transmission.

An example of fax log dialog that a system administrator can open, it is shown in figure 8.6.

The list of records can be filtered and searched with respect to various criteria, see section 3.5 on page 16 for a description of the available features. Figures 8.7, 8.8 and 8.9 show some filtering examples applied to date, call direction and all fields, respectively.

Fax preview window

The fax preview window allows to quickly view a document, received or sent, and can be opened by clicking on the relevant icon in the fax log window, see section 8.4 on the preceding page.

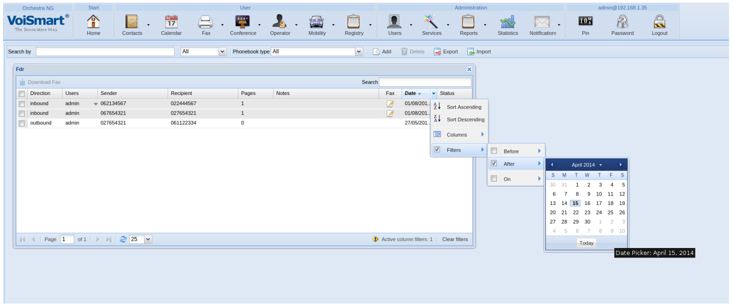


Figure 8.7: FDR with date filtering.

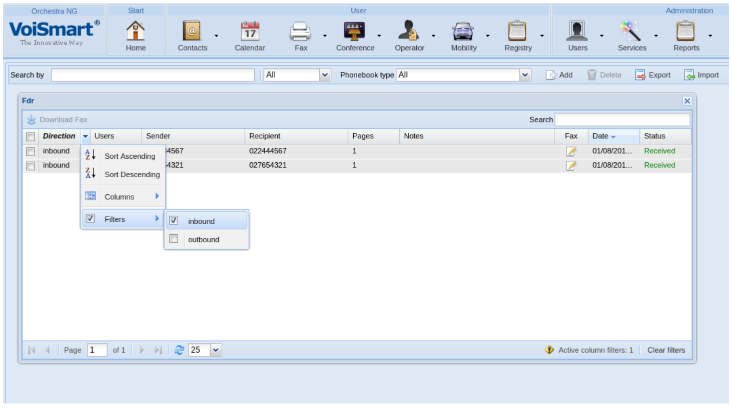


Figure 8.8: FDR with direction filtering.

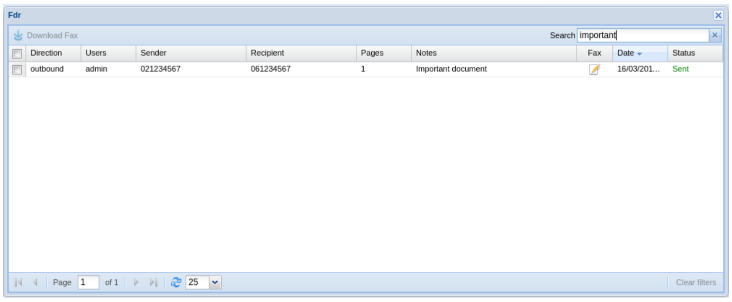


Figure 8.9: FDR with live search filtering.

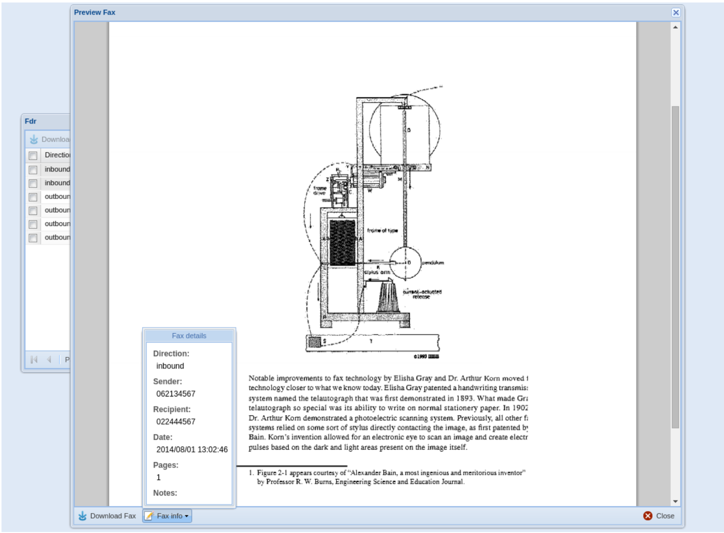


Figure 8.10: Fax document preview window.

✍ This feature requires browser support for PDF files preview, and a plugin may be needed. If no support is found, the preview window won't open, but instead it will be given the possibility to save the file. Consult your browser's manual for further details.

A window like the one in figure 8.10 will appear, allowing to browse through the pages of the fax. Details about the current document, and the possibility to save it to the local computer are available using the buttons at the bottom of the window. It is also possible to directly print it using a local printer with some browser-dependent controls usually available as a menu on the top the document or icons in the bottom-right corner.

Ingoing and outgoing fax status window

It is possible to inspect the status of outbound and inbound faxes by clicking on the Reports → Fax submenu and selecting the *Outbound fax status* or *Inbound fax status* respectively.

The first one, figure 8.11, shows every submitted document along with its current status and the following information:

Domain: (only shown in the system administrator GUI), shows the domain of the user who sent the fax;

Users: shows the user who submitted the document;

Sender: the sender's number;

Recipient: the recipient's number;

Notes: the optional text added when sending the fax;

Date: date of submission;

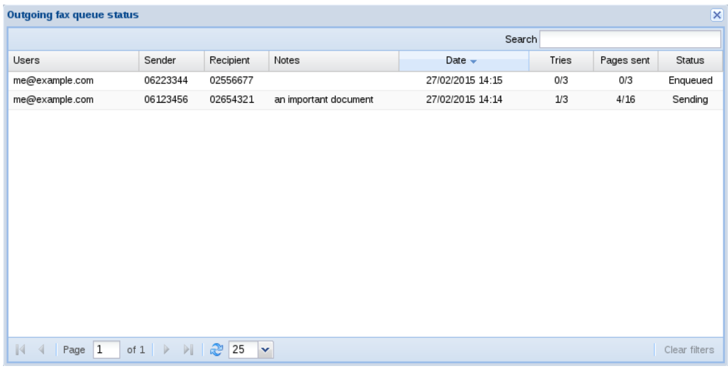


Figure 8.11: Status window for outgoing fax status.

Tries: the number of times the transmission has been attempted, along with the maximum number of allowed tries;

Pages sent: the number of currently transmitted pages along with the total number of pages of the document;

Status: current status of the job. Can be one of “Processing”, “Queued” or “Sending”.

> ✍ A single document sent to multiple destinations will appear as one row per recipient.

On the other hand, the Inbound status window, figure 8.12, details the current status of every ingoing fax, with the following further information:

Domain: (only shown in the system administrator GUI), shows the domain of receiving the fax;

Sender: the sender’s number;

Recipient: the recipient’s number;

Date: date of submission;

Pages received: the number of currently transmitted pages along with the total number of pages of the document;

Status: current status of the job. Can be one of “Processing” or “Receiving”.

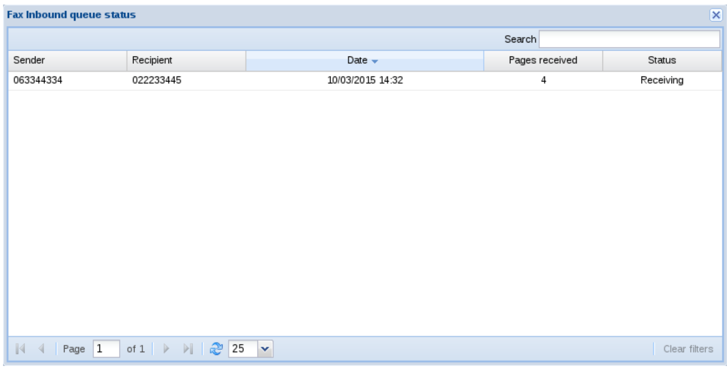


Figure 8.12: Status window for ingoing fax status.

9

Conference

Contents

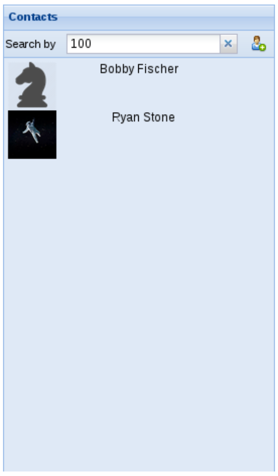


Figure 9.1: Contact search to add members to a conference.

9.1 Introduction

Orchestra NG provides a feature-rich solution for conferencing needs, and offers two main modes: "Live" conferences, and booked rooms.

The former allows you to access audio conference calls without the need of reservations or booking, and to start proactive calls towards the desired participants. It also allows to monitor members, recall them if needed, add (or exclude) participants on the fly and has powerful call-control actions, to enable or disable audio and microphone with just one click. It is the ideal solution for informal meetings or schedule-less conferences with at a moment's notice or needing member's supervision.

Booked conference rooms, on the other hand, offers a conference solution without the need of an operator, and allows the scheduling of events. Conference rooms are created by a privileged user according to the need and resources and users can schedule and book them, and add participants. Every member will receive a notification and will be invited to join the room with the details which are automatically sent to them.

9.2 Live conference

The live conference interface can be opened by clicking on the Conference → Conference panel menu entry and it consists in two main panes. The first one, on the left, is the main conference view and contains all current participants, one tile per member. To add people, the contacts pane on the right can be used.

Figure 9.1 shows the contact search interface. The main search field can be used to filter the contacts from the system's phone book. Upon entering some text, all contacts that matches the pattern in any of their fields (name, surname, phone or fax numbers, ...) will be shown.

When the desired entry has been found, it can be added by simply dragging it to the conference pane, as shown in figure 9.2. If the contact has more

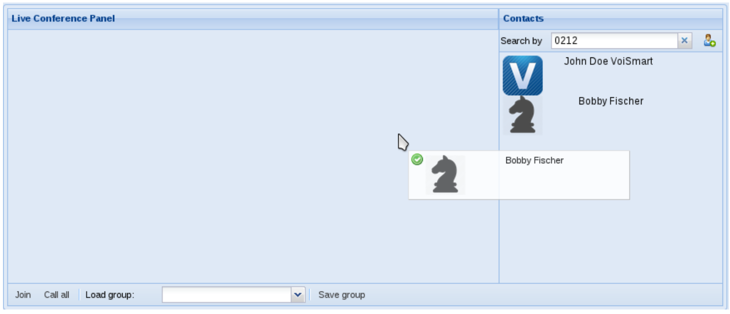


Figure 9.2: Adding a new member via drag&drop.

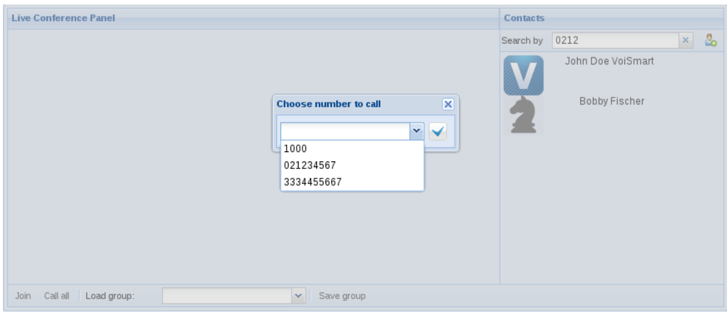


Figure 9.3: Choosing the number to call when multiple are defined.

Figure 9.4: Live conference toolbar.

than one number defined, a dialog will open to allow selecting the one to call, figure 9.3.

To add members not in the contact list, just enter the complete number in the search box and click on the Add button.

The operator can join the conference by clicking on the *Join* button in the bottom toolbar, shown in figure 9.4.

> ✍ Clicking on the *Join* button will immediately place a call towards the user, in contrast to regular members added via the contact pane, who needs an explicit call action, as explained later on in this section.

When one or more members are added, one tile per member is shown in the main conference pane, figure 9.5.

Every tile (figure 9.6), shows the member details, such as name, number and current status, and a group of action buttons. The initial status will be *Not invited*, meaning that it is currently added, but wasn't called yet. To call a participant, press the Call action. His status will change to *Ringing*, indicating a call in progress. When successfully connected, the status will change to *In conference*.

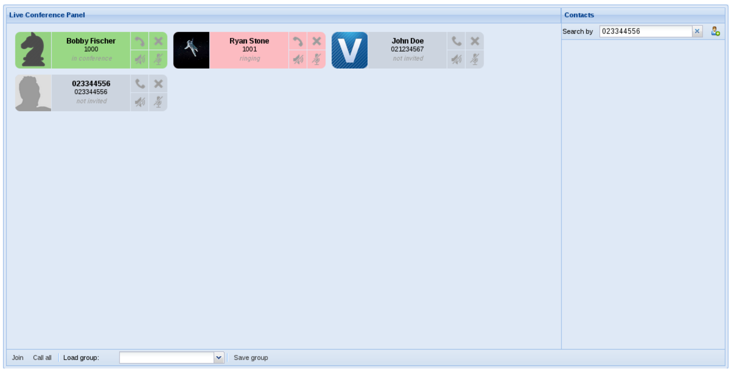


Figure 9.5: Live conference session with 4 members.

Figure 9.6: Conference member.

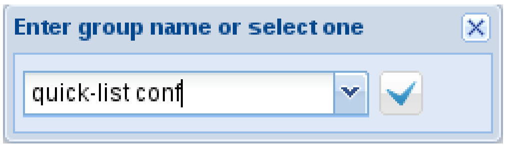


Figure 9.7: The save group dialog.

To quickly call all participants, the *Call all* toolbar button can be used.

An active member can be kicked by using the Hangup action . A short message will notify that user and the call will be then terminated. He can also be recalled by using the same procedure used above.

To remove a member from a conference, eventually kicking him out if currently active, the Kick action ×.

Every user can be muted and have their microphone turned on again by using the Mute action . A short notice will notify the user of the mute status change. To toggle the no-audio mode, the Disable audio action can be used.

Group management

To quickly create meetings with the same group of members, or to provide quick access to create a conference, Orchestra NG has the concept of conference groups.

A new conference group can be created using the *Save group* toolbar button which adds all the current members. The name of the group can be chosen in the creation dialog, figure 9.7.

When one or more groups are available, it can simply be loaded by choosing it in the selection menu in the bottom toolbar. All the members associated

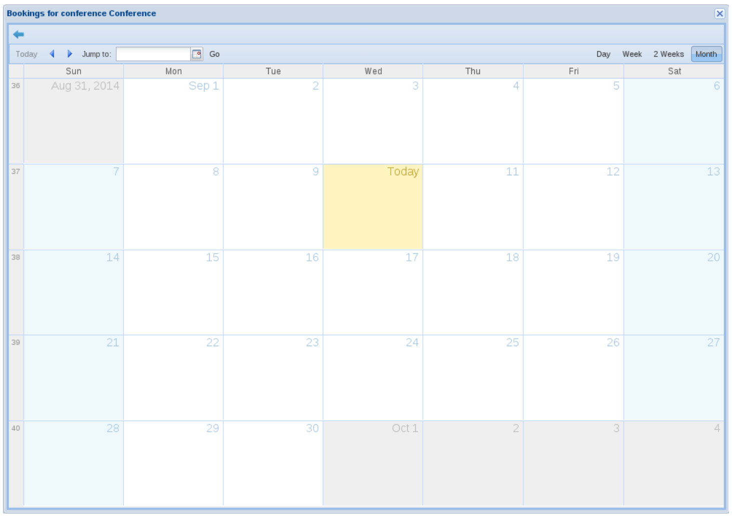


Figure 9.8: The conference booking dialog.

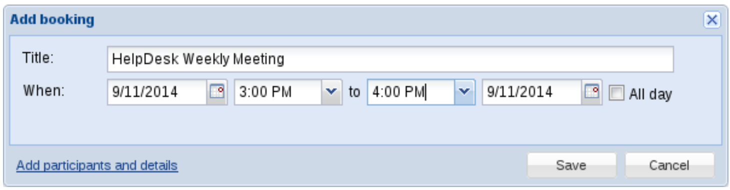


Figure 9.9: Booking conference event dialog.

with that group will be then added to the conference pane, ready to be called.

9.3 Conference room booking

Conference rooms creation is covered in section 5.2 on page 40.

To book a conference room, click on the Conference → Conference booking menu item, to open the list of available rooms, and then click on the Book conference icon relative to the desired room. A window like the one in figure 9.8 will appear.

To open the event creation window (figure 9.9), just click on the desired date, and fill-in the scheduled date and time, with a descriptive title. To add participants, click on the *Add participants and details* link on the bottom of the window. In the bottom form (figure 9.10), there are three ways of adding new members: by directly entering their email in the field and clicking the Add button, by searching through contacts using the Search contacts button or by selecting system users using the Search users button.

Conference members can be deleted by clicking on the Delete button relative to the desired participant.

For more details on calendar events, see chapter 17 on page 145.

When the booking is saved, an email will be sent to every participant to notify them of the invite. The access pin and the number to call to join the

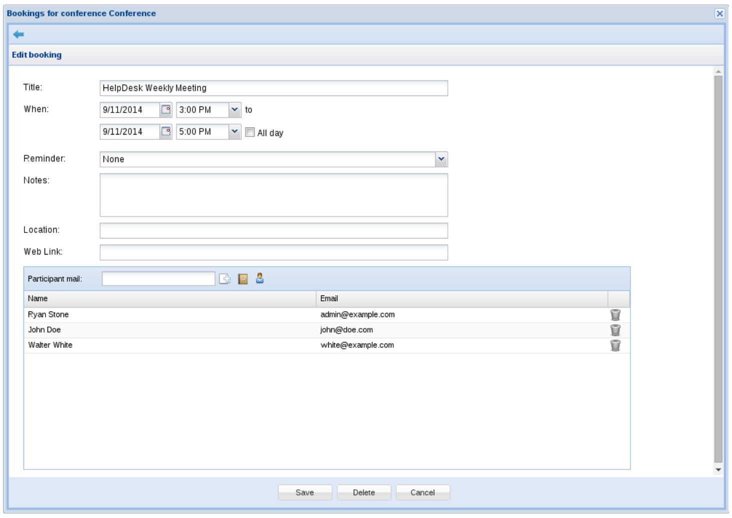


Figure 9.10: Adding participants to a conference.

conference will be in that message, which can be customized according to the needs, see section 5.5 on page 57.

To join a conference room, its public number can be called, if any, or the appropriate feature code can be dialed from a local extension using `*72* + num`, where *num* is the conference number (see section 20.9 on page 169). If the conference is booked, the PIN received in the notification invite will be required to be digited. An unbooked conference can also be joined if the PIN saved when creating the room is known, or no PIN was entered there.

10

Voicemail

Contents

10.1 Introduction

> ℹ This feature is licensed, you can create or enable items only if you have purchased a voicemail users license.

The voicemail system allows users to leave an audio message to the called party if not available, busy or depending on the mobility configuration. Orchestra NG also has support for receiving email notifications, listening to own messages, recording your own greeting message, and so on.

To customize the voicemail notification email, see section 5.5 on page 57.

10.2 Leaving a message

When the voicemail service will start, the greeting will be played to ask the user to leave a message. To save the message, just terminate the call, and the called party will receive it. If any key is pressed, a small voice-guided menu will be available with some options:

Key	Description
1	Listen to the recorded message
2	Save the recorded message
3	Re-record the message

When selected to save the message, two more options will be available:

Key	Description
*	Mark the message as urgent
#	Save the message and terminate the call

10.3 Listening to messages

To listen voicemail messages, just dial the `*100` or `*101` Feature Codes and digit your password when instructed, followed by a `#`.

> ✌ The `*100` feature code will also ask for the desired extension number, allowing to use it from other phones.
>
> On the other hand, `*101` will use the dialing extension to select the desired user.

After authentication, a voice will announce the number of unread and saved messages for that mailbox, and start playing eventual new messages.

The main menu has the following options:

Key	Description
`1`	Listen to new messages
`2`	Listen to saved messages
`5`	Enter the advanced options menu
`#`	Exit and terminate call

While listening to a message, the following options can be used:

Key	Description
`1`	Listen again to the current message from the start
`2`	Save the current message
`4`	Rewind
`6`	Fast forward
`7`	Delete message
`*`	Skip the message information (if playing)

After listening a new or saved message, the message menu will play, with the following options:

Key	Description
`0`	Listen again to the current message
`2`	Save the current message
`5`	Return the call to the user who left the message
`7`	Delete the current message
`8`	Not yet implemented

The advanced menu, accessible by pressing `5` on the main menu, has the following possible options:

Key	Description
`0`	Back to the main menu
`1`	Record a custom greeting
`2`	Choose greeting
`3`	Not yet implemented
`6`	Change your voicemail pin

See the following sections for a detailed description of these options.

10.4 Custom greetings

If a user has the *Play greeting* option enabled in his voicemail parameters (see section 5.1 on page 30), it is possible to record and use custom messages for the users leaving a voicemail audio mesage.

To record a new greeting, select `1` from the advanced menu and, when prompted for it, select a save slot for the new message from `1` to `9`. Having recorded it, the following options are available:

Key	Description
`1`	Listen to the newly recorded message
`2`	Save the message
`3`	Re-record the message

When done, it is possible to select and use it by selecting the `2` option in the advanced menu, and selecting the appropriate save slot.

10.5 Changing the voicemail pin

To change the pin number to access voicemail, the Orchestra NG GUI can be used, see section 5.1 on page 30.

The pin can also be changed by accessing the advanced menu and pressing `6`. When instructed, it is possible to digit the new desired pin, followed by `#`.

In either ways, after the change the old pin will be not valid anymore, and only the new one can be used.

11

Presence concepts

Contents

11.1 Introduction

Orchestra NG has a native and unified concept of presence information which is used extensively throughout the system. This allows smart routing decisions presence-based (e.g. the Mobility service), but also improves the user experience encouraging presence awareness to view other user's availability, reducing communication delays and giving an hint about the correct communication tool to use depending on the current status and his readiness to communicate.

A presence service provides information about the ability a user have, or his willingness, to communicate, depending on the status he sets or the information that the system automatically gathers.

Essentially, every user publishes a presence status either with a direct action on one of his devices and clients (e.g. by setting a DND status on his phone or his Instant Messaging client) or automatically, for example by logging in with a new device or stopping any activity on a device with an active IM client. Orchestra NG collects this stream of information from all available sources and processes it, merging and aggregating it in a single global status and finally, stores it. Every subscriber will then receive the resulting data, providing a coherent presence status to all users and to every consumer client and service.

11.2 Presence status

Every user at every point in time has a presence status attached, which is the aggregated info from all the presence sources. This can be one of:

Available: the user is available and willing to communicate;

Busy: the user is busy in a conversation or in an activity;

DND: the user is in a "Do not disturb" mode;

Away: the user hasn't performed any action in a while, and is considered to be away from any device;

Offline: the user hasn't any connected or reachable device.

Example of presence consumers which use this information are the Mobility service (chapter 13 on page 111), the Operator panel (chapter 12 on the facing page) and the IM service (chapter 16 on page 141).

12

Operator panel

Contents

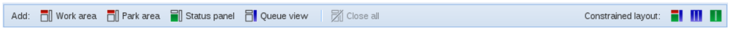


Figure 12.1: Operator panel toolbar.

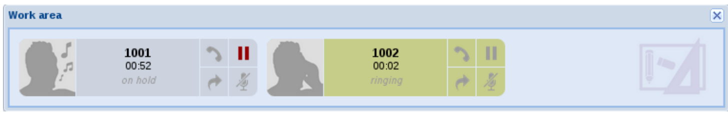


Figure 12.2: Work area with two active calls, on hold and ringing.

12.1 Introduction

> This feature is licensed, you can create or enable items only if you have purchased a operator panel license.

Orchestra NG's operator panel is a complete solution for operators and users to monitor, supervise and manage calls and users.

It is perfectly integrated with presence and contacts, and can show user's and agent's current status.

It allows supervision and management of own calls and queue calls, and helps operators to see active queue calls, transfer them to agents and users, and improve efficiency by adding and removing agents on the fly via a simple drag&drop.

It also shows queue statistics to help operators to better manage resources and signals when too many calls are waiting.

The information shown can be easily customized and view settings saved on a per user basis. Operators can monitor all users, or a subset of them and different views are available, according to different needs.

12.2 Panel types

The Operator panel interface is organized in windows and panels that can be automatically or manually positioned on screen. There are four main panel types that can be opened by clicking on the appropriate button in the main toolbar, shown in figure 12.1.

The Work area

The first one is called *Work area* and it is a panel type which cannot be opened more than once.

This window contains all the active calls initiated or received from the currently logged-in user. Figure 12.2 shows an example with two calls. Each call is visually rendered with an element called "tile", which shows various details.

Every work area tile is divided in two main parts: on the left, the call status is shown, while on the right there are four actions buttons.

The call status shows the caller or called number, the duration and the status (*on hold* and *ringing* in the example), and has a background color of red for active calls, blue for parked, green for ringing and gray for held calls.

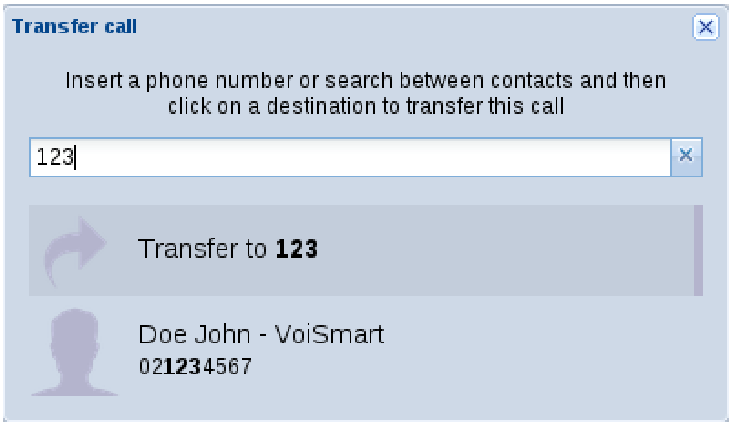


Figure 12.3: Call transfer dialog.

The Hangup action, can be used to terminate the call, and is available for every call status.

The Hold action, can be used to put the call on hold or release it, and is available only when the call is in the active or hold states.

The Mute action, can be used to quickly mute and unmute the call, and can only be used for active calls.

The Transfer action, can be used to transfer the active call to another user, contact, or extension. When clicked, a window will open (figure 12.3) which can be used to select the recipient of the blind transfer action, either by searching among contacts, or by directly entering the number. The call will be redirected as soon as the user clicks on the desired item. This action is only available for active, held and inbound ringing calls. When right clicked, a window will open (figure 12.4) which can be used to select the recipient of the supervised transfer action, either by searching among contacts, or by directly entering the number.

When you start a supervised transfer on an active call, work area tile changes its call status and action buttons as shown in figure 12.5.

The Transfer action, can be used to complete the supervised transfer only when desired party answers.

The Close action ×, can be used to cancel the supervised transfer (also before desired party answers) and retrieve caller put in hold previously.

> ✍ If desired party doesn't answer, is unreachable or rejects the call, supervised transfer is canceled and transferrer is connected again with the original party.

The Park area

The second panel type is called *Park area* and it is a panel type which cannot be opened more than once.

This window contains all the calls parked by the currently logged-in user. Figure 12.6 shows an example with two parked calls. Each call is visually rendered with an element called "tile", which shows various details.

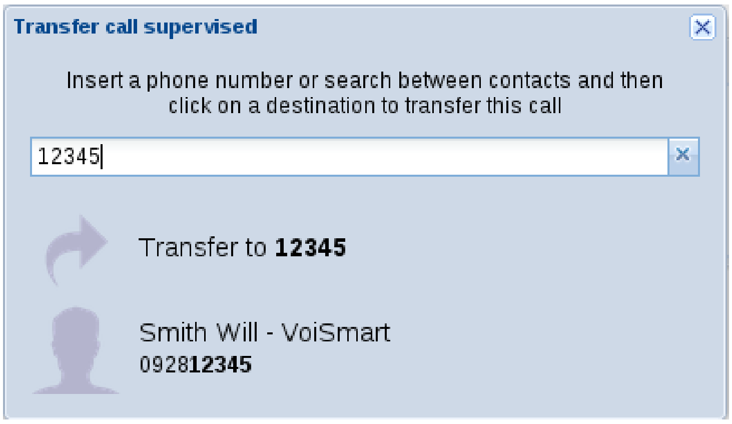


Figure 12.4: Call supervised transfer dialog.

Figure 12.5: Supervised transfer call example.

Figure 12.6: Park area with two parked calls.

Every park area tile, with a background color of blue, is divided in two main parts: on the left, the call status is shown, while on the right there are two actions buttons.

The call status shows the caller or called number, how long a call is parked and a *parking lot* number to retrieve the call using Feature Codes as explained in section 20.6.

The Hangup action, can be used to terminate the call.

The Transfer action, can be used to perform a blind transfer of the parked call to another user, contact, or extension. When clicked, a window will open (figure 12.3) which can be used to select the recipient of the transfer action, either by searching among contacts, or by directly entering the number. The call will be redirected as soon as the user clicks on the desired item.

The Status panel

The third panel type available, is the *Status panel*, figure 12.7, which can be used to monitor and supervise users and to perform several other actions explained later in this section.

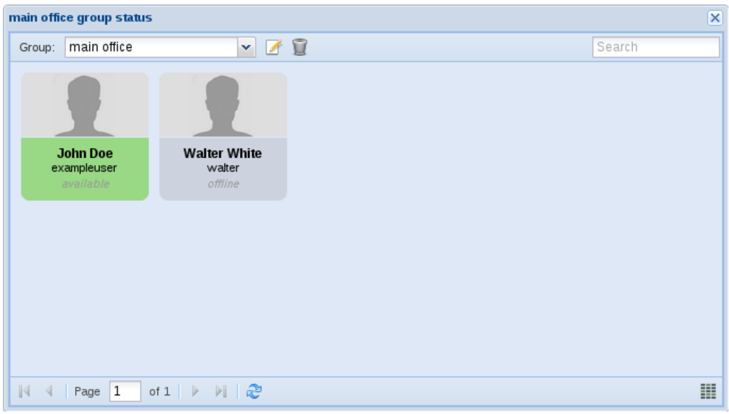


Figure 12.7: Status panel with two users, in available and offline status.

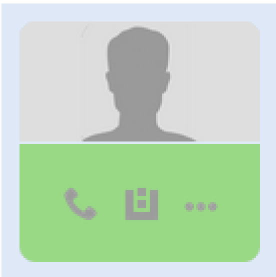


Figure 12.8: Action buttons on the status tile appears upon hovering.

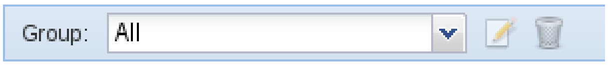


Figure 12.9: Groups toolbar for Status panel.

The main tile in this case shows user's details and their presence status: green for available, red for busy or DND, yellow for away and gray for offline. Hovering on a tile, will show some action buttons (figure 12.8): the Call action will call that user; the Open queue action will open a *Queue view* on the first queue to which that user belongs (if any) and the Show info action will open a dialog with some extended information on that user.

The tile size can be changed using the Change tile size button, which toggles between the small and normal sizes, which is particularly useful when monitoring a large group of users.

It is possible to filter or show only a subset of users by using the search field in the toolbar, or by using groups. Two types of group can be used, system groups, which are only used as read-only here, and the Operator panel's group, which can be created, modified and deleted. Both types can be accessed from the toolbar, and it is possible to switch type by clicking on the *Group* or *System group* label, as shown in figure 12.9 and 12.10.

System groups can be selected in the menu shown in figure 12.10. For more details, see section 5.1 on page 33.

Figure 12.10: System groups toolbar for Status panel.

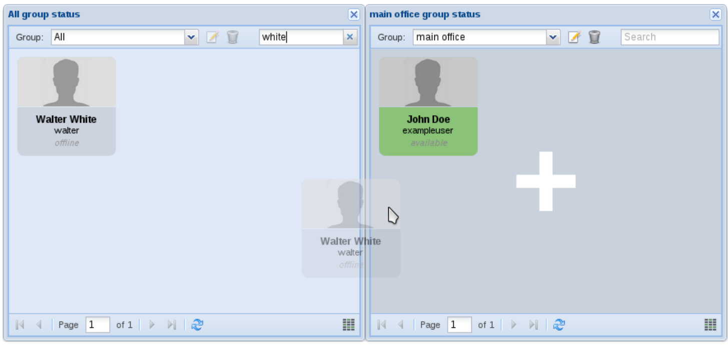


Figure 12.11: Adding a new user to a group, via drag&drop.

Operator panel's groups can be also be selected using the same menu, which in this case shows two extra elements, an *All* group, which is not modifiable and contains all users, and a *Create a new group...* item, which will open a new dialog asking for the name of the group to create.

Having created a new group, new users can be added to it via drag&drop. To do so, open a new *Status panel*, drag the desired user, and drop it on the panel with the newly created group, figure 12.11.

Similarly, to remove a user, just drag the user to the trash icon that will appear at the top of the panel, figure 12.12.

To delete a group, or to modify its name, the two buttons on the panel's toolbar can be used, as shown in figure 12.9.

There is one special kind of *Status panel*, which shows all agents belonging to a queue and can be opened using the *Queue view*. The main peculiarities are that there is no concept of user's groups here, and the drag and drop operations will trigger a different kind of action. This and more will be described in section 12.2.

The Queue view

The last panel type, is the *Queue view*, which shows a list of incoming calls for a particular queue and some relevant statistics.

In figure 12.13 a *Queue view is showed* on the right, and the corresponding *Agent view* on the left.

The particular queue to monitor can be chosen in the dropdown menu on the top of the window. After selection, all incoming calls not yet answered, will be shown in the central pane. In figure 12.13, two active calls are waiting for an operator, and the call tile shows some relevant information, like the caller name and the current waiting time. Also, the Answer action allows the logged user to answer that call.

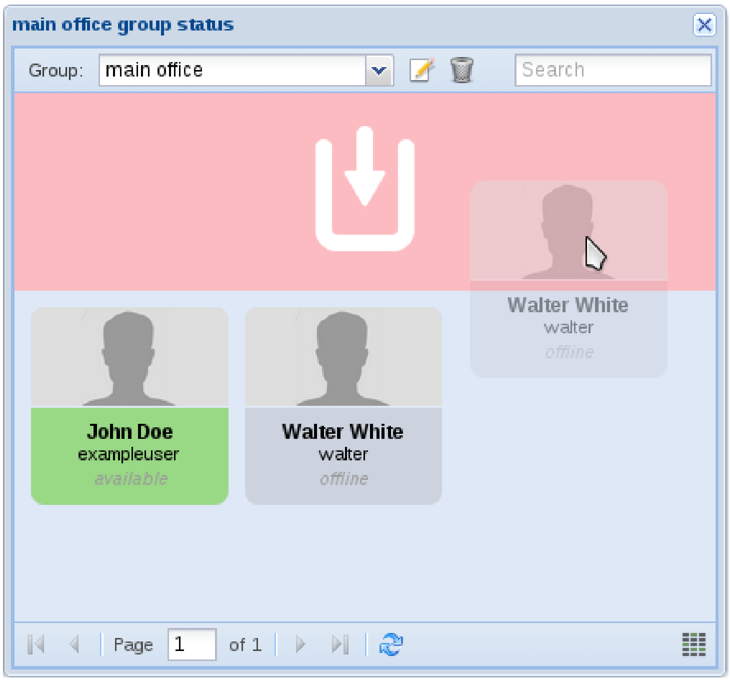


Figure 12.12: Removing a new user from a group, via drag&drop.

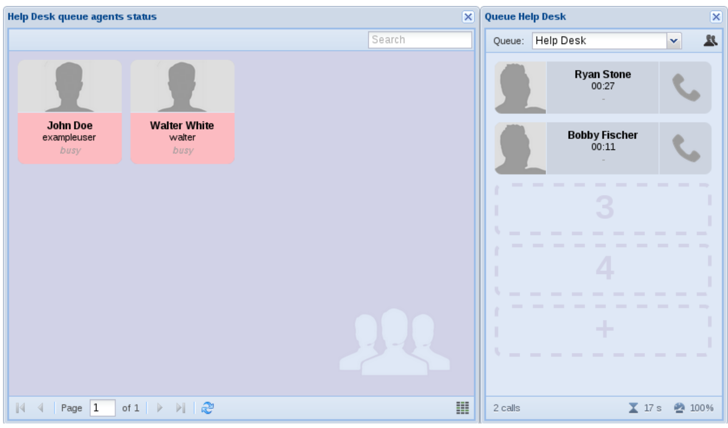


Figure 12.13: Queue and Agents views with two active calls.

Figure 12.14: Statistics bar in Queue view.

> ✍ Clicking on the answer call, will generate a new call to all user's phones, and only upon answering it, the call will be effectively established.

The *Agents view* for the selected queue can be opened by clicking on the Show agents button and shows a special *Status panel* filtered by the relevant agents. In figure 12.13, two agents are displayed, both in a busy state. This view, although similar to the *Status panel*, has some notable differences. First of all, it hasn't a toolbar for group selection, since it already has a well defined users filter. Also, dropping new users or deleting them, while done exactly as shown in figure 12.11 and 12.12, causes a new agent being added or removed to that queue, respectively. This is very handy to efficiently managing resources on the fly while monitoring the queue.

> ✍ Adding a new agent to a queue using the operator panel will use the default settings, as defined at queue creation. To modify that, use the queue configuration window, as described in section 14.2.

The queue statistics bar is located at the bottom of the panel (figure 12.14), and shows three values.

The first one is the number of calls currently waiting for an operator. This field will have a warning symbol ⚠ if this number is equal or greater than the configured value for *Visual alert* set when creating the queue.

The second field shows the average waiting time, while the last one is the percentage of calls answered within the configured queue's service level agreement (see section 14.2 on page 123 for details). This last two values can be reset by clicking on them.

12.3 Transferring calls

Transferring calls using the Operator panel is usually done dragging a call to the desired user. The transferee can be selected either in a *Status view* or in an *Agent view*, with search or group filters applied or not.

The call to be transferred can be dragged from the *Work area*, to select one of own calls, or from a *Queue view* or from *Park area*. An example is displayed in figure 12.15.

While dragging a call, upon reaching the user tile, an indicator will be shown to clarify the action being performed as soon as the call is released on the user tile. In figure 12.16a the overlay transfer indicator is shown when the operation is possible, while figure 12.16b shows a transfer operation not allowed, e.g. when trying to use as transferee an offline user.

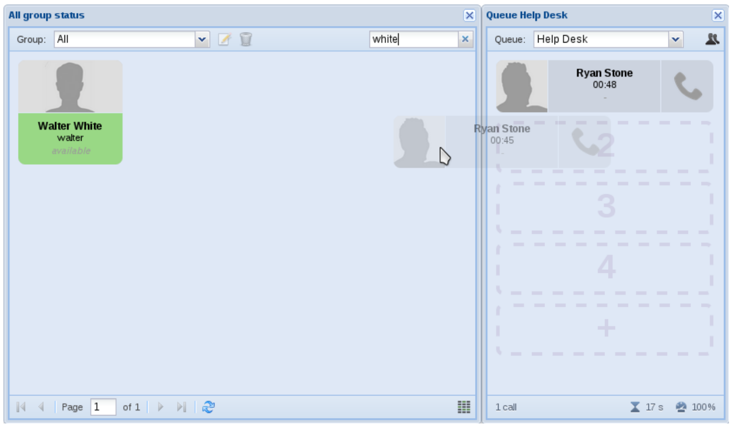
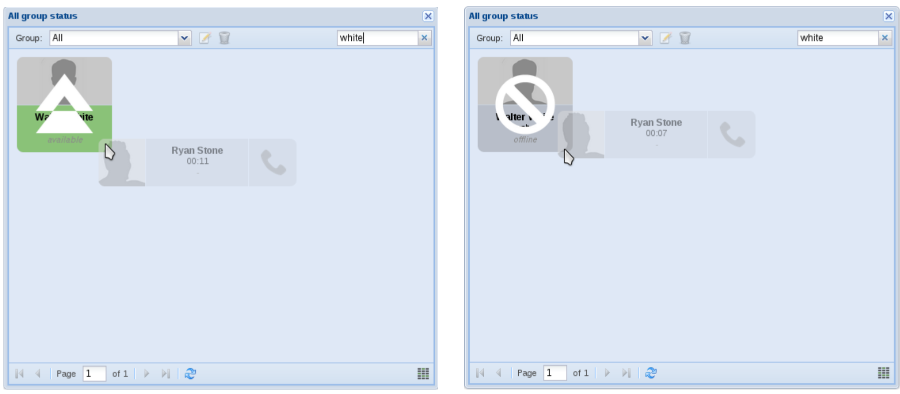


Figure 12.15: Transfer a queue call to a user via drag&drop.



(a) Transfer allowed.

(b) Transfer not allowed.

Figure 12.16: Transfer call to user via drag&drop.

Calls in the *Work area*

Call tiles in the *Work area* can be transferred to any user with only one caveat: calls originated from the transferrer cannot be redirected while in the ringing state. As soon as the call is released, the transferrer will be immediately disconnected, and the call will be redirected to the transferee.

This kind of call can also be dropped on a *Queue view* to transfer it to all of the related agents or in the *Park area* to park the call.

Calls in the *Queue view*

A call waiting in a queue for an operator can also be transferred to *Park area*, to a user or agent by simple drag&drop.

Answering it can be done by clicking on the Answer action on the call tile, or by dropping it on the *Work area*.

Also, it can be moved to another queue by simply dragging it to another *Queue view*.

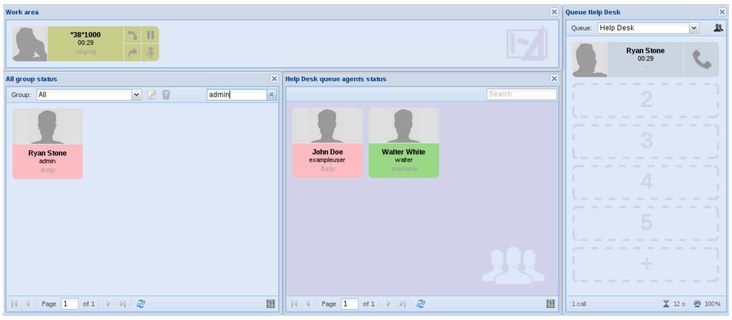


Figure 12.17: Operator layout with two Status panels and only one Queue view.

Calls in the *Park area*

Call tiles in the *Park area* can be transferred to any user or any *Queue view* by simple drag&drop.

Dropping a parked call on the the *Work area* let you to retrieve the call from the parking status.

12.4 Layouts

Orchestra NG Operator panel has advanced support for layout management and supports automatically saving and restoring the desired window layout. The layout management buttons can be accessed from the right side of the main toolbar, see figure 12.1 on page 100.

The layout can either be constrained, or custom, referring to having a predefined and automatic window placement or a free positioning and resizing of windows respectively. The two modes can be toggled by clicking on the *Constrained layout*/*Generic layout* label in the toolbar.

The layout is automatically saved and will be restored on the next login.

Constrained layout

In constrained layout, windows are automatically placed and resized to occupy the maximum screen size.

The first available setup is called the "Operator layout" and can be activated by clicking on the Operator layout button in the main toolbar. It consists in a top *Work area*, below which are placed the *Park area* and *Status panels*, with the *Queue views* on the right area of the screen, stacked one on each other.

This layout, figure 12.17, is particularly useful to have an overview on one or more queues and some group of users or agents, while also being able to manage own calls.

The DSS layout button, applies the "DSS layout", which is useful to quickly control the user's status, eventually grouped in one or more windows, figure 12.18. It consists in a group of *Status panels* and *Agents views*, stacked horizontally on the screen, until all the available space is used.

The last layout type, can be activated using the Queue layout button in the toolbar, which places on screen only the *Queue views*, stacked horizontally (figure 12.19). This layout is useful to monitor a set of queues.

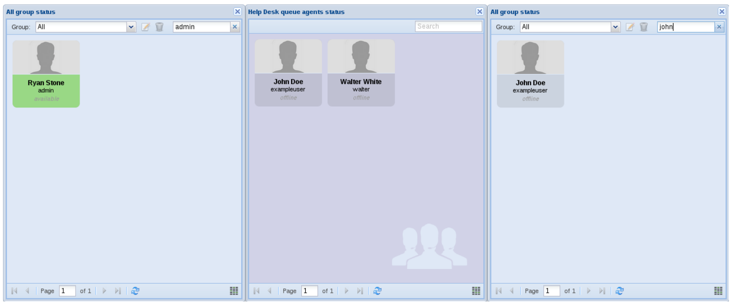


Figure 12.18: DSS layout with two Status panels and one Agents view.

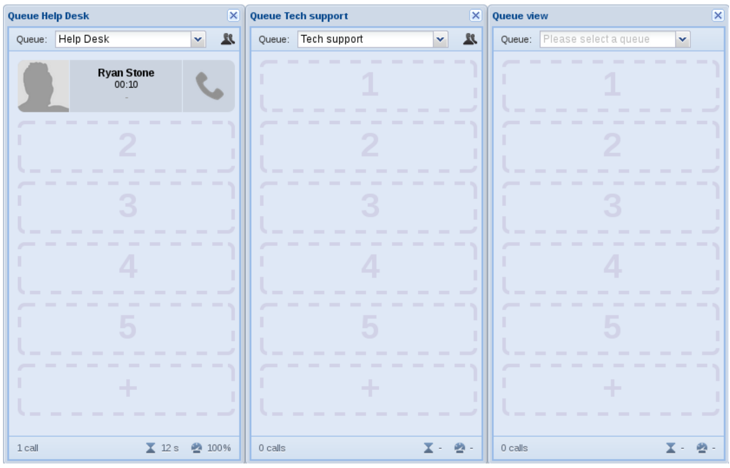


Figure 12.19: Queue layout with three Queue views opened, and only one waiting call.

Generic layout

The generic layout can be used to have a custom placement and size of windows on the screen and can be enabled either by clicking on the toolbar, as explained above, either by trying to resize or to move a window. In the latter case, since the user started customizing the layout, the mode will automatically switch to the generic mode.

When the desired window configuration has been applied, the system will automatically save it using the *Personal* item in the layout toolbar.

If multiple layouts must be saved, or to use a more descriptive name for it, the Save layout button can be used. A dialog will open where it is possible to use a custom name, and the layout will be saved.

To switch to a different saved layout, just select it using the combo box in the toolbar.

Using the Delete layout button, the current layout can be deleted.

Figure 12.20 shows an example of custom layout, where two queues are being monitored, and their agents status is displayed.

All custom layouts, excluding the *Personal* one, are shared among users. For this reason, *Status panels* using user-defined groups cannot be saved in a

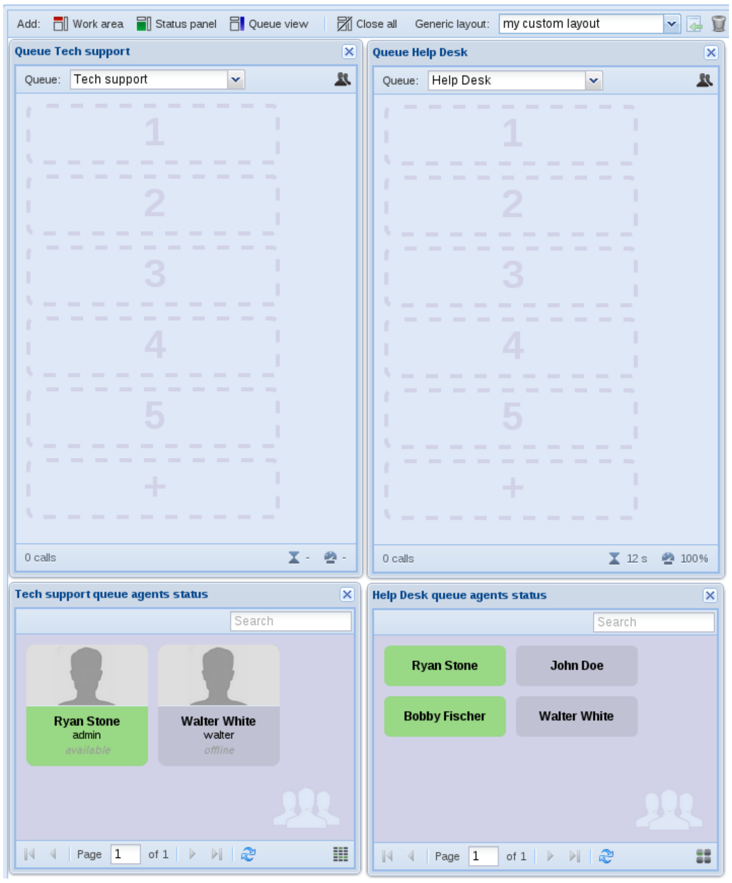


Figure 12.20: Custom layout, with two Queue views and their relative agents.

shared layout (since the group is not shared).

13

Mobility

Contents

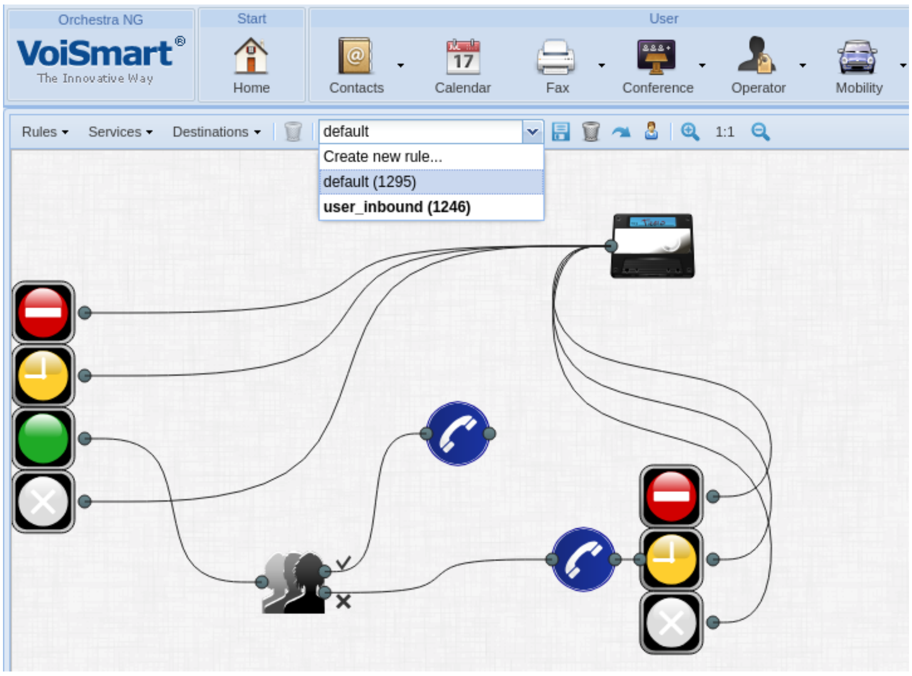


Figure 13.1: Mobility panel overview (cropped).

13.1 Introduction

Mobility is an advanced call forwarding feature that allows to create custom hunting rules for each user in order to accommodate both simple and complex user-hunting scenarios. No matter what device is called or where the user is, the mobility features allows to have customer communications always delivered.

Call forwarding, as configurable using features codes described in section 20.3, is just a predefined set of system rules automatically created in the mobility system.

An overview of the configuration panel with the default rule is shown on figure 13.1, accessible by selecting the Mobility → Configure menu.

13.2 Call forward

Call forward allows to setup up simple call forwarding rules for basic states resulting from calling a user. This is the same as the call forwarding features of any mobile phone.

There are four basic states used by call forwarding:

- *Always*: always divert the incoming call to the configured number;
- *On busy*: if the user is busy, divert the call to the configured number;
- *On no answer*: call the configured number when the user does not respond to the incoming call;

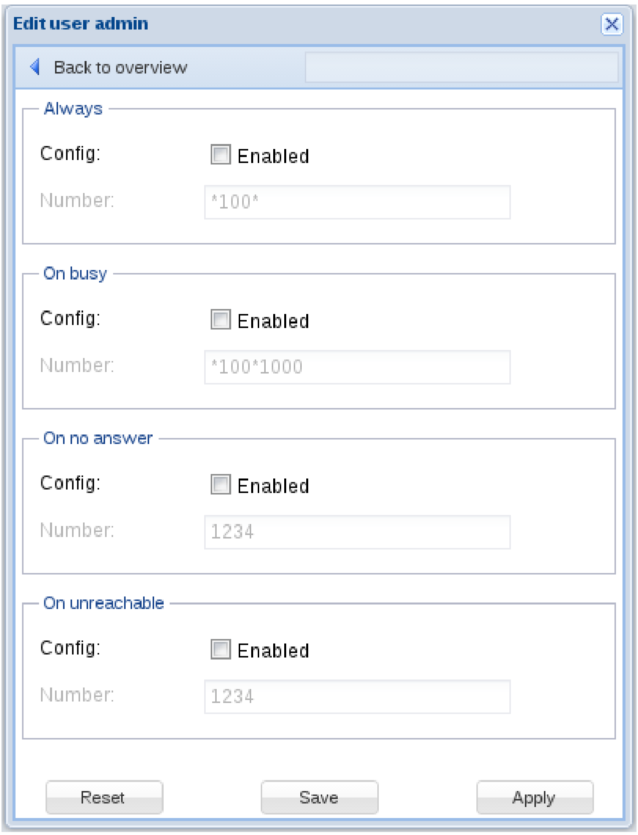


Figure 13.2: User call forward setup.

- *On unrechable*: if all user extensions are not reachable, call the configured number.

The call forward can be configured by an administrator using the users configuration panel, as show in section 5.1 on page 27 and figure 13.2, or from the user phone using the appropriate feature codes provided in chapter 20 on page 165.

If configured from the users configuration panel, figure 13.2, remember to select *Save* to save it or *Apply* to save and apply. Just saving it will not make it active.

> ✍ As stated above, call forward uses the mobility engine. When activated it creates a custom mobility rule and activates it, disabling any current active mobility rule. It is possible to switch it back by using the mobility configuration panel.

13.3 Concepts

The entire setup is graphical, just select the blocks and connect them as needed, by just using point-and-click actions. Configurable blocks can be edited by double clicking on them. The mobility panel has a menu bar, with the following functions, described in order:

- *Rules* see below;
- *Services* see below;
- *Destinations* see below;
- the select box allows to switch between rules and permit to create new ones. If *Create new rule...* is selected, a window will ask for the name, which must be unique; a rule with the name *call forward* is automatically created when using the call forward features codes and cannot be deleted: only switching to another rule is possible;
- the *Save* icon saves the modifications made to the current, selected rule;
- the *Delete rule* icon deletes the current rule. If the rule is marked as active (written in bold in the select list) cannot be deleted, since it is in use. To delete it, create a new rule and mark it active or mark active an existing ruleset, then switch back to the one that was active and delete it;
- the *Reset* icon aborts all unsaved modifications and reloads the rule;
- the *Make active* icon make the shown rule as active;
- the *Make active for users* icon make the shown rule as active for selected users;
- zoom icons are used to zoom-in, zoom-out and reset the zoom level;
- the *Delete item* icon is activated when one element is selected in the panel and if pressed will delete the block and remove all its connections.

The first block on the left is always present and cannot be deleted. It is called *Presence Router* and is the first block engaged when a call tries to reach a local user. It checks the presence status of the called user, aggregating both SIP and IM presence status and send the call to the right exit point. A brief description of the exit points follows.

- the red exit means that the user is busy, where busy can be SIP busy (on phone) or the IM client is set in do not disturb state;
- the yellow exit means that the IM client reports the user is away;
- the green exit means that the either the SIP device(s) and the IM client(s) reports the user as available;
- the gray exit means that the user's SIP device(s) are not reachable.

See chapter 11 on page 97 for more detailed information on the Orchestra NG presence system.

> ✍ If the IM client is not connected, its status is not computed.

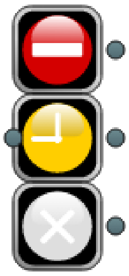
☞ Remember to click on the *Save* icon to save the modifications before leaving the panel, or any change will get lost!

13.4 Mobility main components

The mobility configuration has three main sections where components are present and can be added to the main panel.

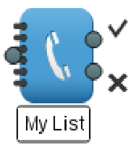
Rules

Rules are decision-making blocks, or routers: with this blocks the incoming call can be routed with several, different rules. A description of all decision-making blocks follows.



Checks the result of a call and route accordingly. Red exit means called party busy, yellow that the call has not been answered within timeout, gray that the remote endpoint is unreachable. This block can be connected only after a *Destination* block

Checks if the time and date of the incoming call match some events defined in the selected user calendar. If a match is found, follows the exit on top, otherwise the bottom one. This is useful to do different things on different scheduled events or to have different flows for different days/hours of the day. The calendar can be selected between user-defined calendars by double-cliking on the block. Check section 4.2 on page 22 on how to manage calendars



Checks if the calling party belongs to the selected contacts list. Follows the exit on top if the calling user is in the list, the bottom one otherwise. This is useful to route the call differently for different calling users. The list can be selected between user defined lists by double-cliking on the block. Check section 7.3 on page 70 on how to manage contact lists

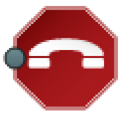
This block allows to follow a different path if the caller is external to the pbx. Follows the exit on top if is external, the bottom otherwise. Useful to build different hunting rules for local calls and external ones

This block checks if the call is coming from a queue. Follows the exit on top if it is originated by a queue, the bottom otherwise. Useful to build different hunting rules for queue calls and normal ones; for example, in most cases is not needed to hunt a user in a complex way if the call is coming from a queue. For an explanation on the queue, see chapter 14 on page 121 and their usage in the dialplan, section 5.2 on page 41

Services

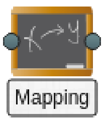
Services are pbx services where the call can be connected as final destination of his path or to change some properties. When a call ends into a service without output ports, no real extension is connected anymore and the flow stops. A description of all services blocks follows.

Connect the call to the voicemail service



Hangup the call

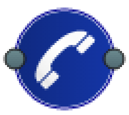
Set distinctive ringtone for the incoming calls, configuring proper SIP *Alert-Info* header.



The mapping block can be used to modify call properties like the calling and called party numbers according to the dialplan design requirements. Multiple blocks can be chained to modify several call properties at once.

Destinations

Destinations are services used to place a call to a user. From a destination block the call can continue to follow other paths if the connection with the real user failed. A description of all destinations blocks follows.



This destination just calls the user extension. The exit can be connected to other blocks for further processing.

Hunt the user by calling the numbers defined in the selected list. The hunt list can be selected between user defined hunt lists by double-cliking on the block. See section 13.4 on the next page for further information.

Call the selected user. The user can be selected by double-cliking on the block.

Forward the call to the number specified. The number can be specified by double-cliking on the block.

This destination calls the user extension and if user is not available put on hold the caller for a specified timeout. If there is not a response within timeout, call proceeds to other blocks for further processing.

Forward the call to the queue specified.

Hunt me

The hunt-me block has a special configuration that is shown on figure 13.3 on the following page. It has three fields, explained below:

- in the first list field, the user-defined hunt list can be selected. All the numbers defined in this list will be called;
- the *Parallel calls* flag sets how calling the list is handled: if it is flagged all numbers of the list will be called in parallel, otherwise each number of the list will be called in the defined order;
- the *Call screening* flag enables an IVR that will be played by to the *called* user as soon as he answers the call. The IVR will playback the calling number to the user and prompt for action. Possible interactions during the presentation are described in table 13.4.

Table 13.4: Mobility hunting DTMF interactions

DTMF	Menu	Description
1	Screening	Answers to the announced, incoming call
2	Screening	Refuse the announced call

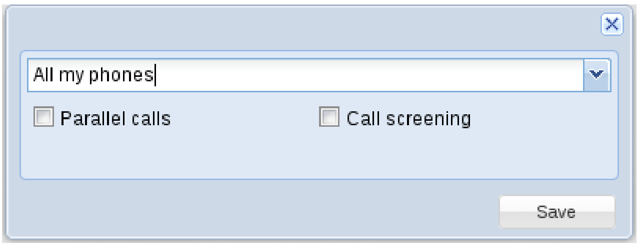


Figure 13.3: Hunt block configuration window.

DTMF	IVR Menu	Description
3	Screening	Forwards the announced call to voicemail. If the called party does not hangup, he can listen to calling party while it leaves the voice message
3	Voicemail	Answer the calling party while is leaving a voice message
1	During call	Dial all other numbers of the list, in a parallel fashion. When one of the numbers answers the on-going call will be connected to it
*7	During call	Start a supervised transfer
*	Attended Xfer	Finalize the supervised transfer when the third party has answered
#	Attended Xfer	Abort the supervised transfer by connect back to the calling party. It can be used when the third party is ringing or has already answered
0	Attended Xfer	Connect all three parties to a three-way conference

✌ If the TTS has been enabled in the system and the caller is present in the system phonebooks, the name and surname will be played back, instead of the number.

13.5 My numbers

From the Mobility → My numbers menu is possible to configure one or more numbers that can be associated to hunt lists. For each number is possible to

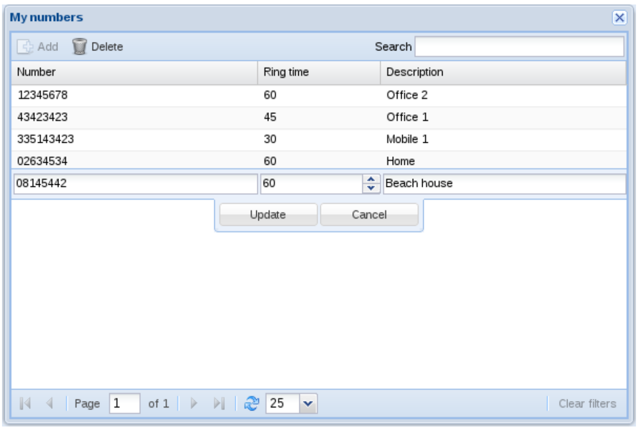


Figure 13.4: Personal numbers configuration window.

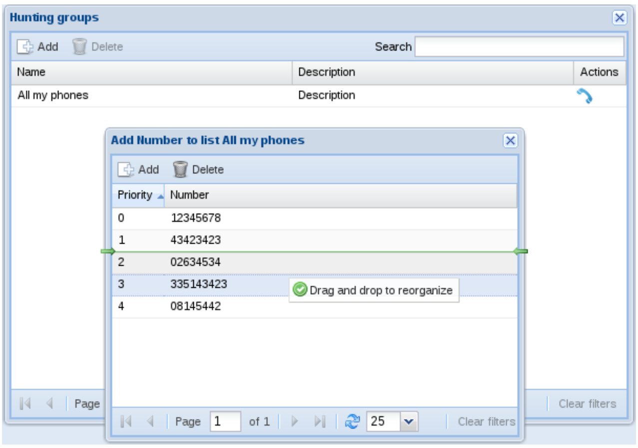


Figure 13.5: Hunt lists configuration window.

set the ring time, in seconds. Refer to figure 13.4 for an example.

13.6 Hunting groups

From the Mobility → Hunting groups menu is possible to configure one or more hunting groups (lists) to be used in the mobility rules. Just create the list and with the action icon associate one or more numbers defined in the section 13.5 on the preceding page. It is possible to define an hunt priority by dragging the numbers and dropping it into the desired position. See figure 13.5 for reference.

✍ If the hunt group is used in a parallel fashion, the priority is ignored.

✍ If a list contains numbers with different ring times, and the list is selected for a parallel hunt group, the lowest ring time will be used.

14

Queues

Contents

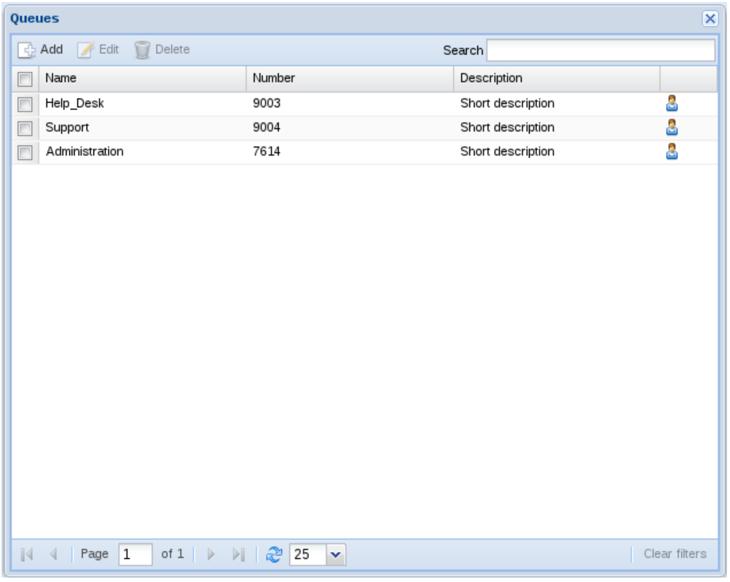


Figure 14.1: Queue list window.

14.1 Introduction

This is the configuration section which allows to define the queues this domain can use, and the their association with agents. A user associated with a queue (either directly or via one of the groups of users it belongs to), will be able to receive calls routed in that queue. The queue list window (figure 14.1) can be accessed by clicking on Services → Queues. Clicking on the *Add* button a new window (figure 14.3) will open and on save will create a new entry. For more details about queue parameters refer to section 14.2 on the next page.

After a queue entry is created, clicking on the icon, a window (figure 14.4) with the list of currently selected users or groups will open. Users can be associated by selecting the Add users or Add groups actions. Adding new users or groups will open a window like the one in figure 14.5, which shows on the left pane all associated users, and on the right pane the remaining.

To remove users from that queue, select the desired items and click the *Remove* button, or click the *Remove all* to disassociate all. Similarly, to add new users, use the *Add* and *Add all* buttons.

> ✍ If a filter is applied, the *Remove all* and *Add all* buttons will respect that filter, i.e. only items matching it will be added or removed, also no visible items due to pagination.

You can configure agent's parameters inline, one by one, or by clicking on icon. In the latter case a window (figure 14.6) will open and you can apply same parameters to selected or all agents. For more details about agent's parameters, refer to 14.3 on page 125 .

For queues that have a strategy where it is important the position of the agents, you can select an agent and change his position using or arrows.

Using you can remove selected agents from queue.

14.2 Queue parameters

***Name*:** name of the queue, any label;

***Number*:** unique number, in this domain, associates with this queue. This number can be used with queues' Feature Codes;

***Wait timeout*:** sets the time in seconds that a call will wait in the queue before it is routed to the next block in the dialplan. A 0 value means that call will stay in queue forever;

***Music on hold*:** sound file to reproduce while caller is waiting;

***Strategy*:** how a call is dispatched to agents. Possible strategies are:

Ring all: ring all available agents until one answers;

Round robin: take turns ringing each available agent;

Round robin with memory: round robin with memory, remember where we left off last agent answered;

Fewest Calls: ring the one with fewest answered calls from this queue;

***Service Level Agreement*:** used for service level statistics. Calls answered within service level time sets. A 0 value means no statistics;

***Max number calls in queue*:** maximum number of people waiting in the queue. A 0 value means unlimited;

***Weigth*:** when compared to other queues, higher weight values get first shot at available agents when the same agent is included in more than one queue. A 0 value disables this function;

***Visual alert*:** number of calls that causes the display of an alert in operator panel as shown in section 12.2 on page 104;

***Announce frequency*:** frequency (in seconds) of queue announces, if any option of *Announce position* or *Announce holdtime* or *Custom announce soundfile* is checked;

***Announce position*:** regularly play a message to inform the caller about his current position in queue;

***Announce holdtime*:** regularly play a message to inform the caller about the estimated hold time (in minutes) for the queue. This value can be reset using the appropriate button in the operator panel view, as explained in section 12.2 on page 104. No message will be played if the hold time is less than a minute;

***Custom announce soundfile*:** regularly play a custom sound file;

***Announce identification number*:** when the agent answers the call, a message will be played to announce the identification number of the answering operator. The identification number must be assigned in the user's detail dialog, as explained in section 5.1 on page 27;

***Continue after answer*:** whether or not to continue following the call flow defined on the dialplan editor when a queue call is terminated after being answered by an agent. To use this feature a Router diaplan element should be connected to the queue's exit port to check the variable *queue-answered* against the two possible values: *true* or *false*, as shown in figure 14.2;

***Continue in case of maximum call waiting*:** whether or not to continue following the call flow defined on the dialplan editor when a call joins a queue and the max numbers of calls configurated is reached. To use this feature a Router diaplan element should be connected to the queue's exit port to check the variable *queue_max_waiters* against the two possible values: *true* of *false*;

***Escape using DTMF*:** whether or not to let caller to leave queue pressing 0 key and to continue following the call flow defined on the dialplan editor. To use this feature a Router diaplan element should be connected to the queue's exit port to check the variable *queue-exit-key* against the two possible values: *true* or *false*;

***Continue without logged agents*:** whether or not to skip queue, following the call flow defined on the dialplan editor, when a call joins a queue without agents logged. To use this feature a Router diaplan element should be connected to the queue's exit port to check the variable *queue-skip-empty* against the two possible values: *true* or *false*;

***Default agent parameters*:** default agent parameters used when an agent is added to queue by GUI or by Feature Codes;

> ☞ These values affect only when you add agents, parameters for agent already associated with the queue, won't change!

***Ignore offline*:** ignore unreachable status of the agents. This will place call to the agent even if their devices are offline, useful to divert call to another service like mobility or call forward and not lose queue calls;

***Periodic reset*:** enable the daily reset for the real time statistics;

***Time*:** time of the day when to reset real time statistics;

***Callback enabled*:** enable callback service;

***Time based rule*:** valid calendar events when booked queue callbacks can be run. In case there is no match with any event, the call is moved to last position. By default booked callbacks are cancelled after two days;

***Delay between retries*:** time in seconds before retry a callback in case previously it had not been successful;

***Retries*:** max number of a retry for a callback. In case of unsuccess dial, callback is put in last position;

***Ring time*:** how many seconds to ring for a queue callback;

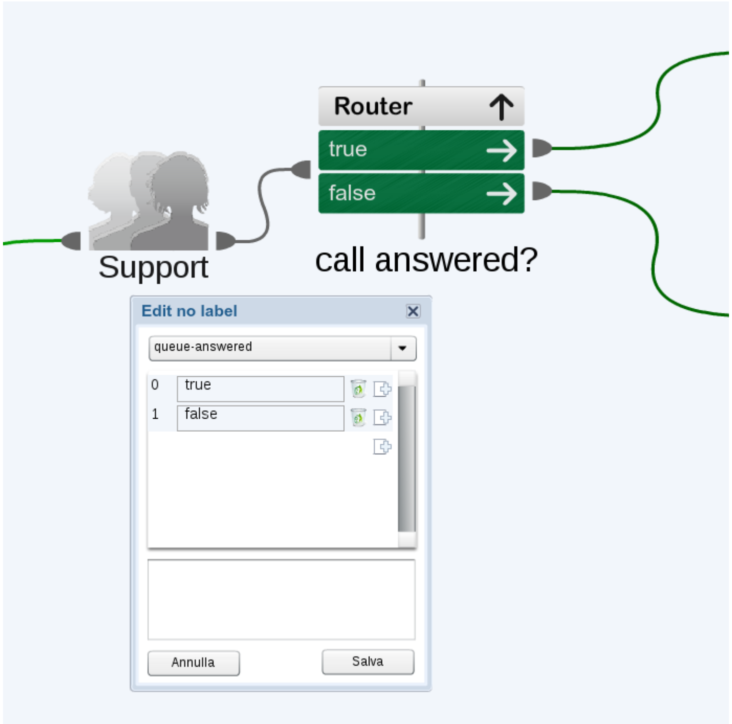


Figure 14.2: Configuring possible queue exit callflows.

Wait time before enable callback: after this wait time in queue, caller can now pressing 1 key and book a callback. Callback announce is enabled and played to caller;

Username: username used to make the outgoing call;

Caller id numer: caller id number used to make the outgoing call;

Regex for callback number: regex used to valid alternative callback number dialed by caller;

Announce callback: announce soundfile to warn caller that callback is enabled;

Announce callback booked: optional announce soundfile to warn caller that callback is booked;

Announce callback answered: optional announce soundfile played to caller when answers to a callback and before join again the queue;

Description: brief description for the queue.

14.3 Agent parameters

Call timeout: number of seconds that an agent rings. This value overrides ring time configured in section phones;

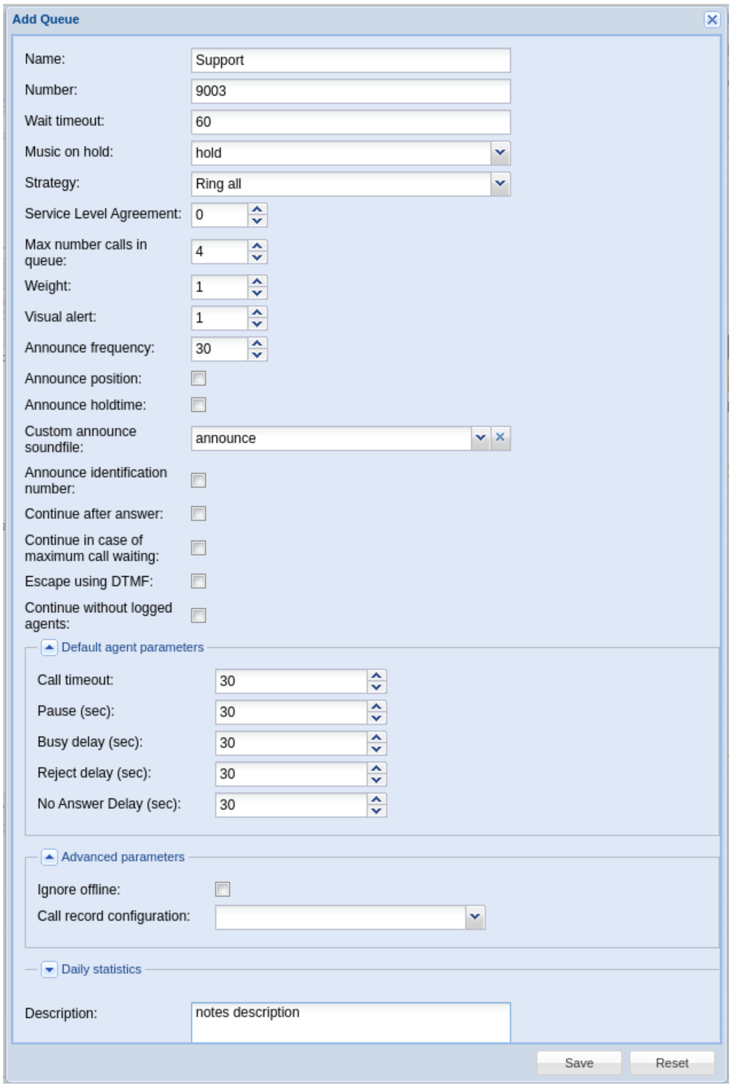


Figure 14.3: Queue configuration parameters window.

***Pause*:** number of seconds to wait before a call is routed to an agent after an answered call;

***Busy delay*:** number of seconds to wait before a call is routed to an agent if a busy tone is detected from agent's phone;

***Reject delay*:** number of seconds to wait before a call is routed to an agent if a reject tone is detected from agent's phone;

***No Answer Delay*:** number of seconds to wait before a call is routed to an agent if he doesn't answer.

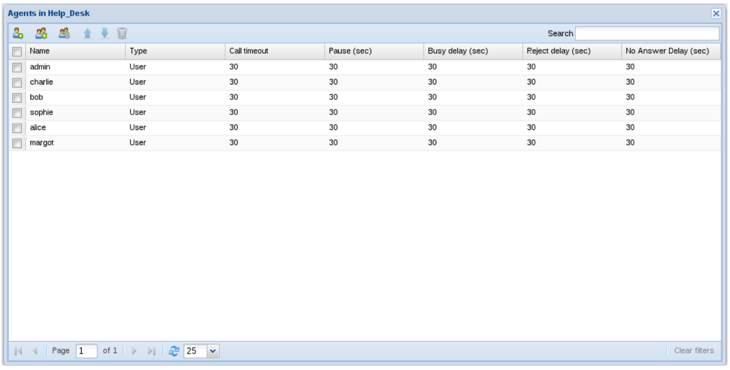


Figure 14.4: Agents in queue window.

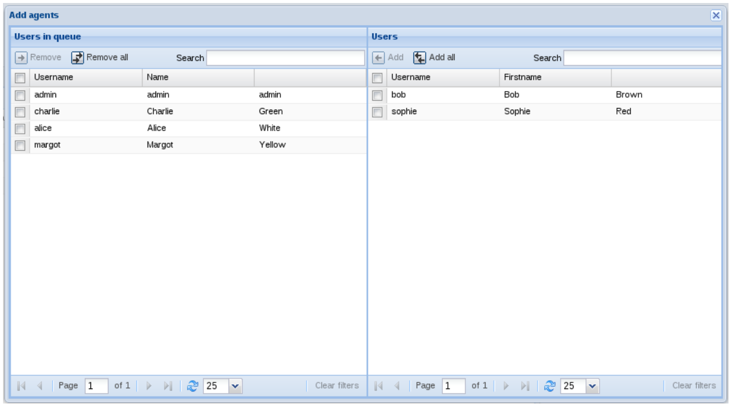


Figure 14.5: Configuration window for user's association with queue.

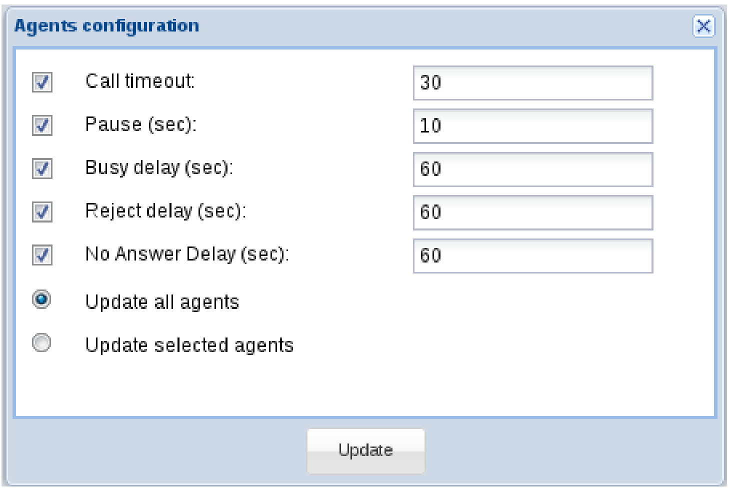


Figure 14.6: Agents configuration parameters window.

15

Dialplan Editor

Contents

15.1 Introduction

The Orchestra NG dialplan editor is a powerful feature that allows to setup any kind of inbound call routing and IVR trees one company may need. The dialplan editor is a graphical gui where all building blocks can be arranged with drag and drop, and call flows connected using point and click.

No programming skills are needed, the call flow is completely visible in a single view for a quick design and easy following of the call routing.

To access the dialplan editor panel, select the Services → Dialplan Editor menu. The panel is split into two main areas:

- on the top there are several drop down menus with all building blocks and a tools area, used to perform several operation within the main work area, see section 15.7 on page 138 for reference;
- on the center, there's the main work area where the dialplan is built.

The main work area can be panned, enabling *Panning mode*, and moving around while keeping the click pressed. This is useful to move around in an extended tree without having to keep the zoom level too low.

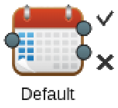
> ☞ The dialplan allows you to solve practically any call routing and call property manipulation requirement that you may have. The dialplan editor is very flexible in allowing the construction of decision trees based on linked routing and mapping tables. However you should take care not to use too many tables and an over-elaborate structure. The configuration may become large and difficult to manage. Always try to keep it as simple as possible and try to group routing and mapping blocks using proper pattern matching in order to reduce the used blocks and entries in each single block.

15.2 Building blocks

To use each block select it from the drop down menu and it will be dropped on the work area. Some blocks are editable inline: the parameter needed can be searched by deleting the text below the icon and the full list of valid values will appear, or searched by starting typing into the field itself.

When the block is red, means that the block has no valid values or is incorrectly configured.

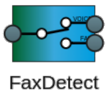
Answers a call passing through it. If not use the call will be left in ringing state, which can be useful to play prompts in-band without billing the caller



Routes the call based on events on the configured system calendar. If there's a match, follows the top exit, otherwise the bottom one. Useful to build different paths based on hourly/daily or periodic events, like playing back closed hours prompts or connecting different extensions/ivr/services on different times. This block is editable inline. To manage system calendars, refer to section 5.2 on page 43

Connects the call to the conference system. The current called number is passed to the service and rules defined in section 5.2 on page 40 will apply

Connects the call to the fax server service. The current called number is passed to the fax service and routed to final recipients following the rules defined in section 5.2 on page 41



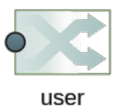
Enables fax detect on the call. The call will immediately follow the *Voice* exit but the system will listen for fax tones for 60 seconds while the call is be routed to other services. If during the 60 seconds period a fax is detect the call is routed back to the *Fax* exit and will follow the new path. Normally used to enable a listener for faxes during an IVR by connecting the *Voice* exit to the IVR service and the *Fax* exit to the fax server service

Hangups (closes) the call

See section 15.6 on page 138

See section 15.3 on page 133

See section 15.5 on page 136



Place an external call and connect to the incoming one. To route the outbound call an LCR is used; the call will use the selected user's LCR and the will appear as if the user originated it. This block is editable inline. See section 5.1 on page 27 for users configuration and administrator manual for an in-deep LCR configuration explanation

See section 15.4 on page 134

Connect the call to a local user by searching the called number between the locally defined extensions. For example, if the called number is 1234 this service will search the local extension 1234 and ring it. In most cases a mapping before this service is needed

Routes the call by searching the calling number into the selected distribution list. If the number is present into the list, follows the top exit, otherwise the bottom one. Useful to create different paths for different callers, like priority customers or blacklists. This block is editable inline. See section 7.3 on page 70 for managing distribution lists.

Connects the call to the selected queue. If the call has not yet been answered, the queue music on hold will not get played back, unless the telco allows it. This block is editable inline. See section 5.2 on page 41 for queues configuration.

Connects the call to an automatic queue. It starts the queue defined by the called number, searching the attribute number of the queue. For example, if the called number is 1234 the service will search and start the queue with the attribute number equal to 1234. If the queue does not exist, the call gets dropped. If the call has not yet been answered, the queue music on hold will not get played back, unless the telco allows it. This block is editable inline. See section 5.2 on page 41 for queues configuration.

See section 15.3 on the facing page

This service allows to transfer the call control to an external server, for third party applications integration. Since this is a special feature that requires developer-level insights, please refer to a sales representative for further information.

Playbacks the selected sound file and proceed to next step when the playback finishes. This block is editable inline. See also section 5.2 on page 41

Connects the call to the selected user, without checking the called number. If the user has multiple extensions, the system will ring them all. This block is editable inline.

Connects to the voicemail service in order to leave a voice message to the selected user. This block is editable inline.

Give priority to a call before join a queue. Calls with a higher priority will be served sooner. Disables position announcement, in queue configuration, to prevent lower priority calls from receiving a position backward announcement. This block is editable inline.

15.3 Router

The call router is a very efficient and flexible tool for routing calls in the dialplan. Each routing block is responsible for a specific routing criterion such as the called or calling party number or any other call property. Multiple blocks can be linked together to form a decision tree.

Orchestra NG router block supports matching the call properties defined in table 15.2.

Table 15.2: Available call properties.

Call property	Description
calling-redir-e164	redirected Number, the number to which the call was last presented
called-e164	called party number
calling-e164	name of the caller, provided by the user agent that has called us
calling-e164	calling party number provided by the network, can be masked (hidden)
calling-e164-ani	Automatic Number Identification, the number of the calling party (caller) cannot be masked, but not all networks provides it
calling-party-category	the type of device placing the call, also known as ANI2
calling-ip	IP address of the signaling source for a VoIP call
queue-answered	queue call is answered. Possible values are *true* or *false*
queue-max-waiters	max number of waiting calls in a queue reached. Possible values are *true* or *false*
queue-exit-key	user press exit key while in queue. Possible value are *true* or *false*
queue-skip-empty	call skipped queue without logged agents. Possible value are *true* or *false*

Table 15.2: Available call properties.

Call property	Description
year	calendar year, 0-9999
day-of-year	day of year, 1-366
month	month, 1-12 (Jan = 1, etc.)
day-of-month	day of month, 1-31
week-of-year	week of year, 1-53
week-of-month	week of month, 1-6
day-of-week	day of week, 1-7 (Sun = 1, Mon = 2, etc.) or "sun", "mon", "tue", etc.
hour	hour, 0-23
minute	minute (of the hour), 0-59
minute-of-day	minute of the day, (1-1440) (midnight = 1, 1am = 60, noon = 720, etc.)
time-of-day	time range formatted: hh:mm[:ss]-hh:mm[:ss] (seconds optional). Example: "08:00-17:00"
date-time	a date/time range, tilde-separated, formatted as YYYY-MM-DD hh:mm[:ss], with seconds optional. Example: 2010-10-01 00:00:01~2010-10-15 23:59:59

To add entries on the router double-click on the router block and the figure shown in 15.1 on the next page will appear. On the top drop-down list select the call property that the router should match and add entries using the add symbol. The match priority can be adjusted by dragging each single entry and dropping it in the desired position. Each entry can be a fixed match or a regular expression. For regexp usage, see appendix D.

Remember to press *save* before closing the window. The label of the router can be changed by double clicking to the text below the icon.

> ✍ Always put more strict matches on the bottom of the list and more generic on the top.

15.4 Mapping

The mapping blocks can be used to modify call properties like the calling and called party numbers according to the dialplan design requirements. Multiple blocks can be chained to modify several call properties at once.

The mapping dialog, as shown in figure 15.2 on the facing page permits to select which call property to match, which are the same of the router block,

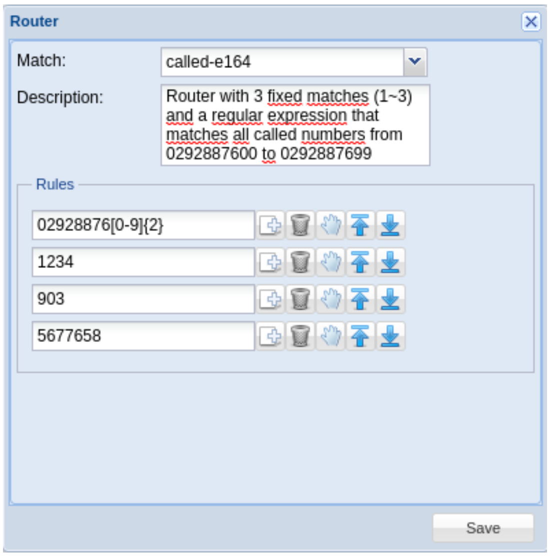


Figure 15.1: Router dialog window.

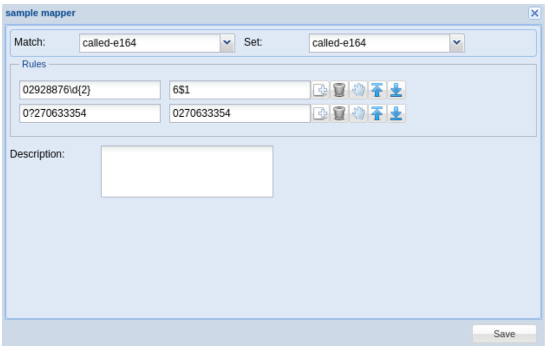


Figure 15.2: Mapping dialog window.

see table 15.2 on page 133, and which property the mapper should modify, as shown in table 15.3. Matching and substituting follows the regular expressions rules, described appendix D.

Table 15.3: Call properties that can be modified.

Call property	Description
calling-redir-e164	redirected Number, the number to which the call was last presented
called-e164	called party number
calling-e164	name of the caller, provided by the user agent that has called us

Table 15.3: Call properties that can be modified.

Call property	Description
calling-e164	calling party number provided by the network, can be masked (hidden)
calling-e164-ani	Automatic Number Identification, the number of the calling party (caller) cannot be masked, but not all networks provides it
calling-party-category	the type of device placing the call, also known as ANI2

✌ Mappers are commonly used to manipulate called and calling numbers, to perform translations between public numbers and local extension, like mapping a sequence of public numbers to extensions numbers before connecting the call to the *Phone* block in the editor.

To add entries on the mapper double-click on the mapping block. Remember to press *save* before closing the dialog. The label of the mapper may be changed by double clicking to the text below the icon.

15.5 IVR

ℹ This feature is licensed, you can create or enable items only if you have purchased a ivrs license.

The IVR block permits to create a voice-driven menu, where the calling user can interact using DTMF. The IVR dialog window, opened by double-clicking on the IVR block and shown on figure 15.3 on the next page, permits to quickly create the menu and to setup the wanted behavior.

Before creating the IVR some sound files must be uploaded into the Orchestra NG system, as explained in section 5.2 on page 41.

On the left of the dialog is possible to configure the prompts and timeouts to be used during playback:

- *Menu*: the name of the sound file to be used as main menu for the IVR. The box can be searched by starting to type in the textfield;
- *Timeout*: name of the sound file to be played back when there's a selection timeout. The value is expressed in milliseconds. There's a timeout when the calling user is idle and does not input any entry within the configured time span;
- *Error*: name of the sound file to be played back when the caller selects a wrong entry within timeout;
- *Timeout*: how many second to wait for any user input, before playing back the timeout prompt;

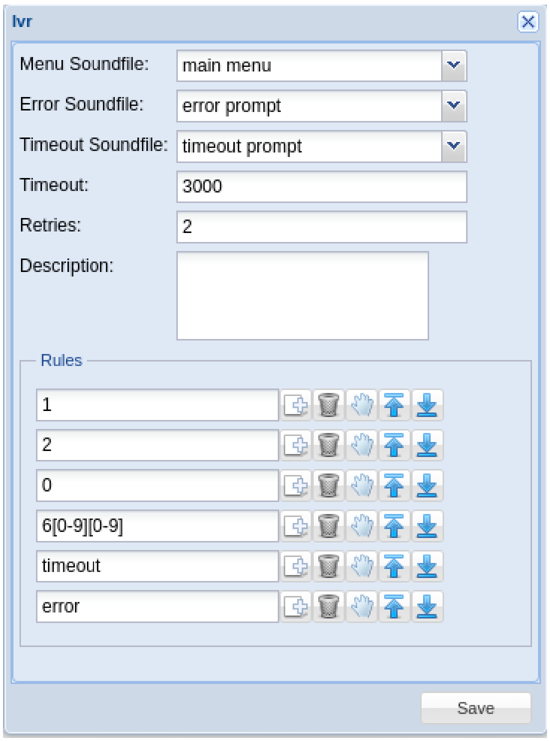


Figure 15.3: IVR dialog window.

- *Retries*: how many retries to allow in case of error.

On the right of the dialog the IVR entries are configured and represent what exit the user can select using DTMFs sequences.

Each entry can be numeric or a regular expression, in order to match several possible entries.

On exit, the called-e164 number is set to the selection done by the user (i.e. if the user selectes 1234 on the IVR prompt and 1234 is a valid exit, the called-e164 will be set to 1234).

There are two special, alphanumeric matches, as shown in figure 15.3:

- *timeout*: if set, follows this exit on timeout, after playing the timeout prompt;
- *error*: if set, follows this exit on error and after all possible retries, after playing the error prompt;

If the above special exits are not set, the call will get closed when the condition happens.

> ☞ A selection matching multiple entries is considered to be an error. In this case the *Error* message will be played and the *timeout* entry, if defined, will be followed.

15.6 Helpdesk

> This feature is licensed, you can create or enable items only if you have purchased a helpdesk license.

The helpdesk block is a specialized IVR with additional fields that is used for delegating part of the IVR control to an external server, which must implement the business logic and must indicate with exit the call must follow when the decision has been made.

Since this feature needs developer-level insights, please refer to a sales representative for further information.

15.7 Tools

The toolbar on the top part of the dialplan editor panel is used to perform several operations on the configuration and on the graphical view. Each item is described below.

	Saves and applies the current layout. Modifications not saved are lost when closing the dialplan editor
	Deletes one or more blocks. Select one or more blocks with *Selection mode* enabled. The selected items will be highlighted. Then select the trash icon and the selected items will be deleted
	Increase zoom factor
	Automatically arrange zoom level to fit all blocks
	Decrease zoom factor
	Duplicate selected item(s)
	Enable selection mode to select more than one at the same time
	Enable panning mode to scroll diaplan
	When the dialplan is extended with a lot of blocks, it is convenient to group them and hide from the overview. Select one or more blocks with *Selection mode* enabled. Press the group icon and the selected items will be grouped into a box icon. To edit what is inside the box, double-click on the box. To exit from box editing, click on save grouping icon or back to main dialplan without saving changes
	Unbox a group icon. Click on the grouping box than needs to be ungrouped to release blocks inside

	When editing a grouping box, add output for current box
	When editing a grouping box, back to main dialplan without saving changes in current grouping box
	Saves current changes in grouping box and back to main dialplan
	Loads a dialplan backup file, in old xml format or in json format, from local computer
	Saves a dialplan backup file, in json format

16

IM service

Contents

16.1 Introduction

> This feature is licensed, you can create or enable items only if you have purchased a instant messaging users license.

Orchestra NG is also a IM server. The service is provided using the open standard XMPP protocol. It is fully integrated with the presence subsystem, section 11 on page 97, to provide unified experience between chat and voice features.

16.2 Clients

Being a standard XMPP server, any client supporting it can be used. Common parameters are:

- Account: also called Jabber ID, is the fully qualified user name, with domain: e.g. bob@example.com. The same credentials used to log into the Orchestra NG web GUI. Some clients may have the tuple user name and domain separated, so fill them in the appropriate fields;
- Password: fill the Orchestra NG user password;
- Resource: XMPP servers are able to log in several client programs simultaneously using the same user name and password. The resource is used to distinguish among different clients. Commonly is already filled with a default value, but can be changed to represent what client is, for example by setting the resource to *work* or *home*. When other people look at *you* in their rosters, they usually see a list of resources you are available through. They could send a message to your particular resource, using these JIDs: you@jabber.org/Home or you@jabber.org/Work;
- Hostname: XMPP is able to find the real hostname and port of the server using the DNS service, but DNS setup is needed, as discussed in section 16.3 on the next page. Otherwise using the advanced client parameter hostname is possible to specify to which server the client must connect without querying the DNS;
- Port: the default server port is 5222/TCP for plain (no encryption) or TLS secured connection and 5223/TCP for old-style SSL connection. Using a TLS capable client is recommended for privacy and security;

> ✍ XMPP clients looks up the server in the following order:
>
> - look up _xmpp-client._tcp service for the domain part of the Jabber ID;
> - look up the A record for the domain part of the Jabber ID;
>
> If the domain part of the Jabber ID is an IP address or the hostname is forced using client advanced parameters, the client will directly connect to the server, as specified.

✍ Since the encrypted connection uses certificates signed by the Orchestra NG built-in Certification Authority, the client may provide an alert about the certificate that is not trusted. Just confirm the warning by accepting the certificate or install the Orchestra NG Certification Authority certificate, as described in section 21.2 on page 176.

16.3 DNS SRV records

The best way to provide XMPP services is to publish the Orchestra NG hostname on a DNS service using SRV records. This makes possible for clients to discover automatically the IP address of the server based on the domain part of the Jabber ID.

Record format

The SRV record has the following format:

```
_service._proto.name TTL class SRV priority weight port target
```

- service: the symbolic name of the desired service;
- proto: the transport protocol of the desired service; this is usually either TCP or UDP;
- name: the domain name for which this record is valid;
- TTL: standard DNS time to live field;
- class: standard DNS class field (this is always IN);
- priority: the priority of the target host, lower value means more preferred;
- weight: a relative weight for records with the same priority;
- port: the TCP or UDP port on which the service is to be found;
- target: the canonical hostname of the machine providing the service.

XMPP SRV record format

Since XMPP is a TCP only service, the record format will be:

```
_xmpp-client._tcp.example.com. TTL IN SRV priority weight port target
_xmpp-server._tcp.example.com. TTL IN SRV priority weight port target
```

The first service, *_xmpp-client*, is for client to server communications, it is the one looked up by clients.

The second one, *_xmpp-server*, is for server to server communications, which makes possible to other XMPP servers serving other domains to federate with the Orchestra NG one.

XMPP SRV record example

If our domain is *example.com*, the SRV records will be:

```
_xmpp-client._tcp.example.com. 86400 IN SRV 5 0 5222 example.com.
_xmpp-server._tcp.example.com. 86400 IN SRV 5 0 5269 example.com.
```

Within this example, the Orchestra NG users are in the form user@example.com.

17

Calendar

Contents

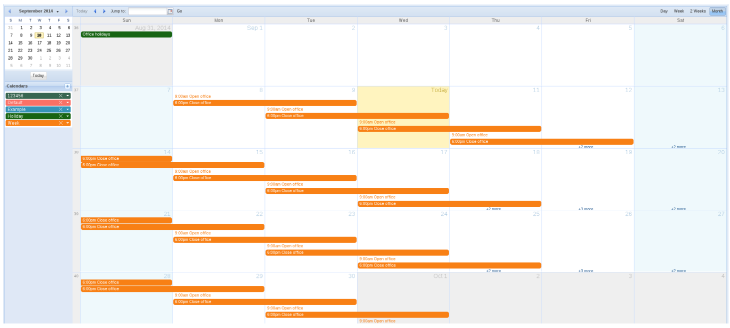


Figure 17.1: Calendar panel.

17.1 Introduction

Orchestra NG's calendar panel (figure 17.1) is a complete solution to create single event or recurrent events using an easy and intuitive web interface. The main use of calendars is therefore to create rules based on events associated with them. Calendars, and events created in them, can be used as services in Dialplan Editor or by users in their mobility rules.

17.2 Calendar configuration

By default Orchestra NG's calendar panel has a calendar named default. If you click on + near *Calendars*, in left side panel, a new window (figure 17.2) will be opened and you will be able to create a new calendar. For each calendar, you can configure two parameters:

***Name*:** unique name in this domain, any label;

***Description*:** brief description of the calendar.

To complete calendar's creation, click on *Save* button.

You can change calendar's color by clicking on ▽ icon in correspondence of each calendar created. A new window (figure 17.3) with a color palette will appear, and you will be able to choose a new color for calendar. When you change calendar's color, all events, associate with it, will change color. If you double click on a calendar, you can show or hide its events from panel.

> ☞ Hidden events from the panel does not mean they are disabled from the calendar used in Dialplan Editor or in mobility rules!

To delete a calendar, and automatically all its events, click on × icon.

17.3 Event configuration

To create an event you can simply click in calendar panel and it's not important calendar view (*Day*, *Week*, 2 *Weeks*, *Month*) used. On click a new window, as figure 17.4, will open. When you add a new event, you can configure these parameters:

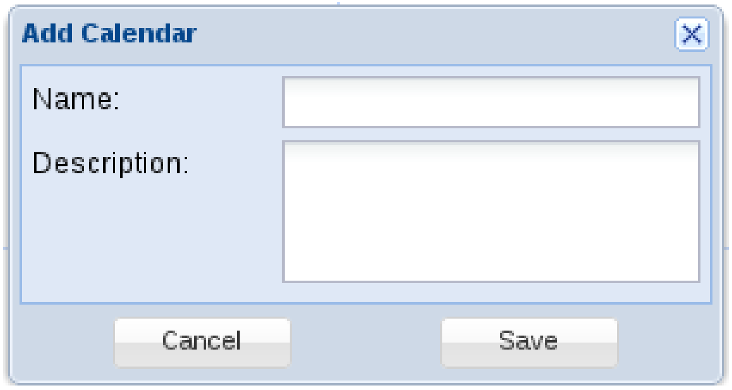


Figure 17.2: Add calendar window.

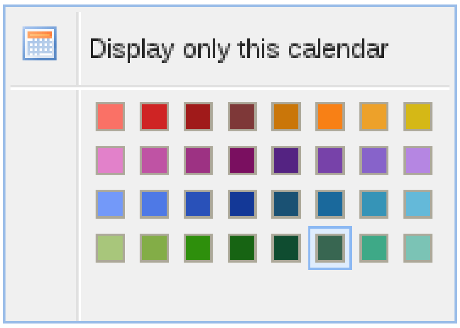


Figure 17.3: Calendar color palette window.

***Title*:** a label that describes this event;

***When*:** set the dates and hours of validity for this event. Start date and end date can be different. If you enable *All day* hours are irrelevant;

***Calendar*:** choose calendar that will be associated with this event.

To create this event, click on *Save* or if you want to configure more complex options, click on *Edit Details...* button. In the latter case a new window (figure 17.5) will open with additional parameters. These parameters are:

***Repeats*:** choose if this event will be a recurrent event. Use this drop-down to specify the frequency of the event. The remaining fields in this dialog change based on the repeat frequency selected. Possible values are:

- ***Does not repeat*:** no repeats;
- ***Daily*:** customize daily repeats;
- ***Every weekday (Mon-Fri)*:** customize weekday repeats;
- ***Weekly*:** customize weekly repeats;
- ***Yearly*:** customize yearly repeats.

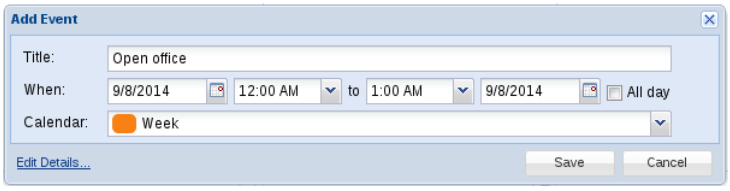


Figure 17.4: Add event to calendar window.

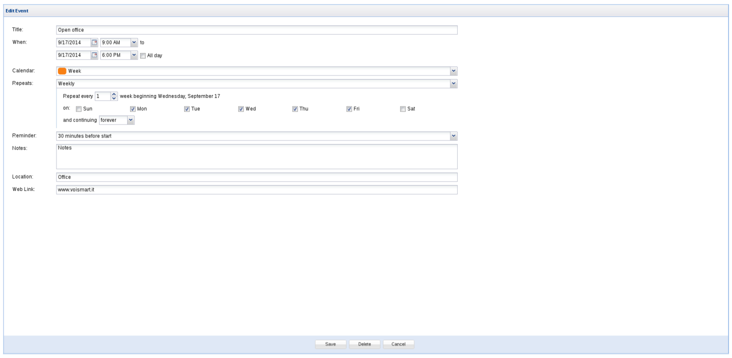


Figure 17.5: Edit details add event window.

> ✍ If you choose *forever* as end date of your recurrent event, keep in mind that this value doesn't mean really forever but at most, events will be created for the next two years!

***Reminder*:** set an alert time for this event. Not yet implemented;

***Notes*:** add some notes for this event;

***Location*:** add a location for this event. Not yet implemented;

***Web Link*:** add a web address for this event. Not yet implemented.

If you right click on a created event, a window like figure 17.6 will open. Actions that can be performed are:

***Edit Details*:** open same window show in figure 17.5;

***Delete*:** delete this event;

***Move to...*:** move this event to selected new date;

***Copy to...*:** duplicate this event to selected new date.

> ✍ Events are shown and saved using the current browser's timezone.

If event is a recurrent event when you click on *Delete* a window as figure 17.7 will open. In this window you can choose to delete this single event of the series, future events of the series with respect to this event or all events of the series.

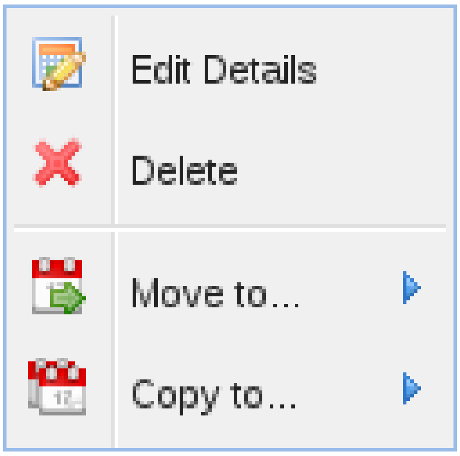


Figure 17.6: Edit details window by right-click.

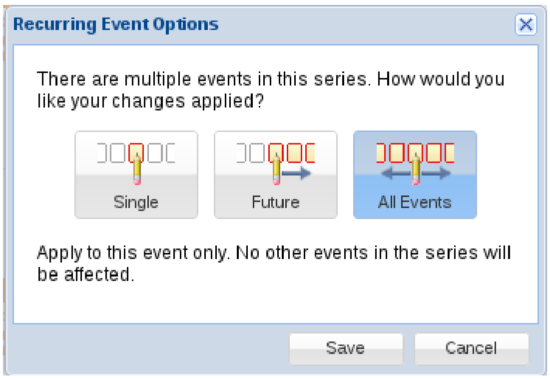


Figure 17.7: Delete recurrent events window.

18

Call recording

Contents

18.1 Introduction

The Call Recording feature allows to setup a versatile call recording system that can be used to improve agents' performance in call center applications, security, increase sales, reduce liability and help to comply with regulations.

Main features are:

- blindly record all calls or select what to record;
- initiate recording using a DTMF sequence or a specific key on the phone;
- store audio files using a template for file name format;
- optionally insert record tones to be played while call recording is active;
- optionally playback an announcement before starting the recording;
- WAV or MP3 audio format;
- encrypt audio files using industry standard strong cryptography (AES-256), using two passwords for increased security;
- cryptographically sign each audio file to check consistency and detect tampering;
- record each call leg on different channels of the stereo file, to isolate noise and chatter in order to provide better understanding of the conversation;
- web audio player, leveraging HTML5 technology;

> ℹ This feature is licensed, you can create or enable items only if you have purchased a recording users and recording channels license.

18.2 Configuration

The main configuration window (figure 18.1 on page 154) can be accessed by clicking on Services → Call record configurations. From here it is possible to create, edit, delete rules or assign them to specific users.

When editing or creating a rule the figure 18.2 on page 155 will appear. The fields are explained below.

- *Name*: short name for the rule;
- *Description*: a brief description, if needed;
- *Filename Template*: template to use to create the file name of the audio file. Each term must be enclosed into a double curly brace. See table 18.1 on page 154 for further details;
- *File format*: output recording format, can be WAV or MP3. For better results, use WAV, for reducing storage space, use MP3;

- *Hash file*: compute a cryptographically secure hash of the audio file and store it on the internal database. From the recording report it is possible to check if the stored hash matches with the file, in order to check file integrity and detect tamper;
- *Enabled*: enable audio file encryption;
- *Allow recovery*: permit the support services to recover the encrypted files if the passwords are lost;
- *First password*: first password for the encryption;
- *Confirm password*: confirm the above field;
- *Second password*: second password for the encryption;
- *Confirm password*: confirm the above field;
- *Record tone*: play a record tone in background during the whole recording;
- *Announcement*: playback a system sound file once, as soon as the recording starts. The sound file can be selected from the drop-down select box;

✌ Two passwords are used for encryption in order to allow two different administrators to use their own and require the agreement of both to decrypt or playback the recording. This configuration is used in environments where top privacy is important, avoiding to give control to only one person.

☞ If the encryption passwords are lost or cannot be remembered, and *Allow recovery* is not selected, there's no way to recover the recorded calls, since AES-256 is used for the encryption, which is strong enough to be used for protecting national top secret documents. Even if *Allow recovery* is selected, but the system lose it's internal database and no backup is present, it is not possible to recover the encrypted files. So a good strategy is to always store passwords in a secure place.

✍ A random string is always appended to the file name and cannot be disabled or changed. This is meant to protect against wrong template setups which can overlap filenames resulting in overwriting an existing recording. For example, if the template just uses the `caller_e164` tag, not using a proper random string on the filename results in generating the same file for recorded calls from the same caller.

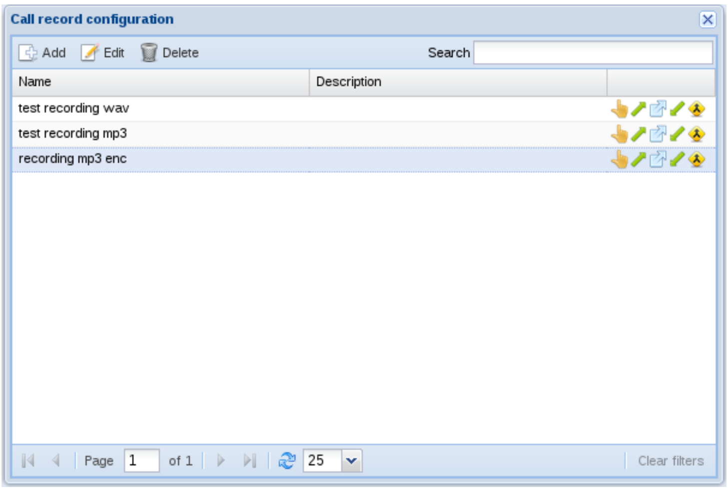


Figure 18.1: Call recording configuration main window.

Table 18.1: Call recording filename tags.

Term	Substituted with
`uuid`	Unique ID of the call
`username`	username to which the recording belongs
`domain_name`	domain to which the call belongs
`date`	date and time of the recording, in Y-m-d-H:M:S format
`accountcode`	the fully qualified accountcode of the recording
`caller_ani`	ANI number of the caller
`caller_e164`	the calling number
`called_e164`	the called number

Each configuration can be associated in four different ways to extensions or queues, by selecting the icons in the action column, explained below:

- : allows user to start a new recording or to stop an existing one by pressing *2 during the call;
- : records calls generated from the selected users towards local extensions;
- : records calls generated from the selected users towards remote parties, usually via some trunk;
- : records calls received by the selected users;
- : records calls entering the selected queue service;

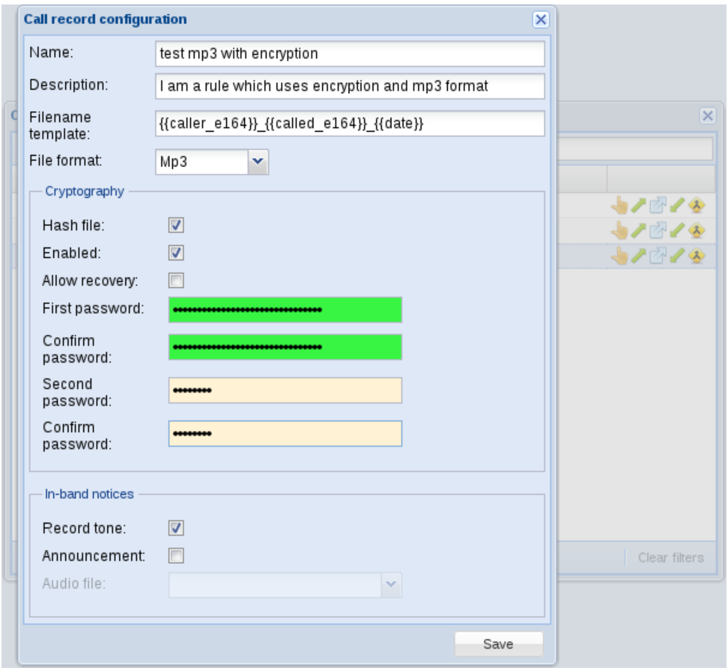


Figure 18.2: Call recording configuration edit window.

✌ The association between a single user and the rules, or to check which rules has a user, can be done from users configuration area, see 5.1 on page 33 for reference.

Since rules assignment can overlap, the precedence between rules is:

- queue: if a call enters a queue and then is connected to a user with inbound call recording or one touch enabled, the queue rule will win;
- outbound: if a user has local outbound recording enabled and calls towards a user with one touch record or inbound call recording set, the outbound rule will win;
- inbound: if a user has inbound call recording set, and he is called by a user with one touch recording enabled, the inbound rule will win;
- outbound: if a user has both one touch record and outbound rules, outbound will win;
- one touch: this rule will work only if no other method is involved.

18.3 Reporting

By selecting the Reports → Recordings it is possible to view the logs of all recorded calls and listen to them or download the files. From the Registry → Recordings user menu the same view will appear, but showing only the recordings associated with the currently logged in user. This view also shows the live calls that are currently active and being recorded. See figure 18.3 on the next page for reference.

The window will show the following fields:

- *Caller E.164*: calling number of the call;
- *Called E.164*: called number;
- *Called ANI*: called ANI;
- *Account*: accountcode or username to which the call belongs. Calls recorded from a queue does not belong to anyone, so are visible only from the administrators;
- *Duration*: duration of the recording, can be different from the duration of the call;
- *Date*: date and time of the start of the recording;
- *Format*: audio format used for the file;
- *Status*: status of the call file in the recording chain, described below.

The call passes through several states during the recording process:

- *RECORDING*: the call is still active and being recorded. At this time duration is not available;
- *RECORDED*: the call has been terminated, so recording is complete but it is waiting to be processed (for signing, encryption, etc);
- *PROCESSING*: the call is currently being processed;
- *READY*: the call is ready to be listened and stored at the final destination;
- *FAILED*: the processing failed for some reason. Processing can be retried using the icon in the action column;
- *NOFILE*: the call duration is too short an no audio file has been generated;

The action column has several icon used to perform different activities on the record. Not all icons will be available every time, since they depends on the recording status.

- : playback the recording using the web audio player. If the recording is encrypted will prompt for the passwords;
- : download the file. If the recording is encrypted will prompt for the passwords;
- : check the signature of the file, if enabled by the respective rule;
- : if audio processing failed, try to resubmit it again;

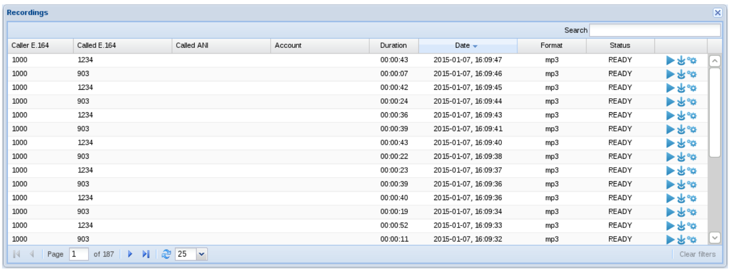


Figure 18.3: Call recording details records

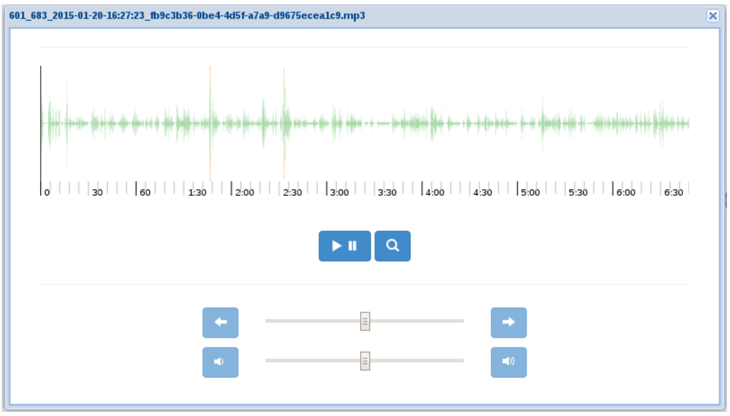


Figure 18.4: Call recording web playeri1

Web Audio Player

The recording system has a complete web audio player that allows to listen and navigate the audio file without the need of any external player. The play/pause button switches between playback and pause.

The player displays the waveform in order to navigate in the audio by simply selecting the point in time to reproduce. If the recording is long, it will be shrinked to fit the window, but can be zoomed in using the zoom button. Pressing it again, will zoom out.

With the first slider is possible to pan between DX and SX channels, to listen to one party of the call at once. By setting it back to center both will be played.

The second slider is used to set the playback volume. See figure 18.4 for reference.

> ✍ The web audio player uses the Web Audio API, which are not supported by Internet Explorer, so it works only with Chrome, Firefox and Safari. Attempting to play the audio file with Internet Explorer will result in downloading the file.

19

Statistics

Contents

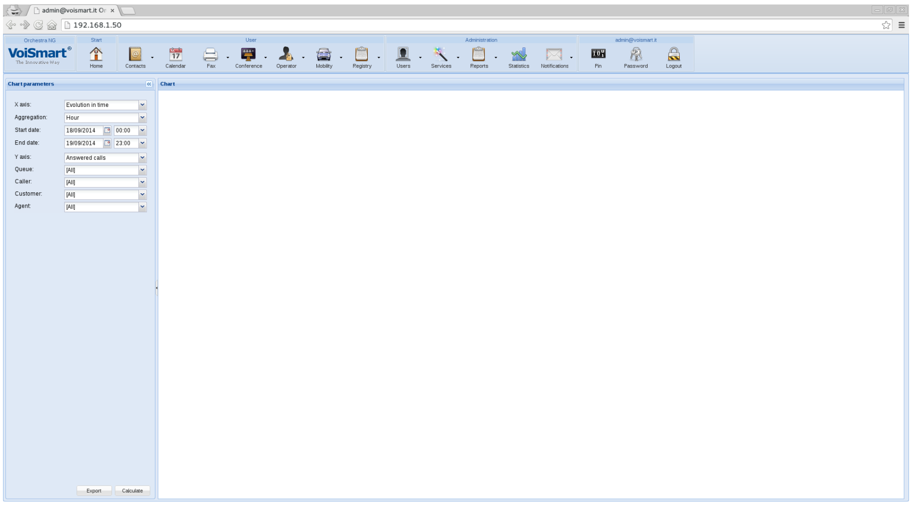


Figure 19.1: Statistics panel.

19.1 Introduction

This feature is licensed, you can create or enable items only if you have purchased a statistics license.

The statistics engine of Orchestra NG provides comprehensive reports of all inbound calls routed to the queues. It provides in-deep analysis for queues and agents performance, either global and broken down to single caller. Everything can also be analyzed for different time aggregations.

The engine leverages on state of the art NoSQL database in order to guarantee performance and scalability, while data aggregation provides quick display of all information without the need to calculate views every time, which translates in lower CPU usage.

The statistics engine can be only activated on x86_64 architecture. Disk space is also important and proper planning must be done in order to estimate the calls volume which gets translated into space needed for the statistics database.

19.2 Statistics panel

By selecting the Statistics button, the statistics panel will show up, figure 19.1. The bigger area, *Chart*, on the right will display the chart that is selected on the left area, *Chart parameters*. The *Chart parameters* panel can be collapsed with using the icon on the top-right its toolbar.

By default, the statistics are computed every hour. So new results from current calls will not show up until the next analysis. The term *hour* is not the exact value like 7, 8, 2 pm and so on, but is a 3600 seconds time interval from the start of the Orchestra NG

> system. So, if the system was booted up at 3:17 pm, the statistics engine will run every 17th minute of the hour.

19.3 Chart parameters

- *X axis*: Two kind of charts can be provided: line charts for evolution of a specific data type and column charts for comparing the magnitude of different items of the same data type. See table 19.1 on the following page;
- *Aggregation*: what time frame to use for aggregating data; depends on the type of data shown on X axis;
- *Start date*: shows only data newer than the selected value. Always select the starting point of each aggregation period, i.e. if the aggregation period is weekly, the beginning of week (Monday) must be chosen;
- *End date*: shows only data older than the selected value; Always select the starting point of the element of each aggregation period, i.e. if the aggregation period is monthly, the beginning of month (1st day) must be chosen;
- *Y axis*: Y axis parameter depends on what has been chosen on X axis. Refer to table 19.2 on the next page for all possible values, keeping in mind that not all may be shown at the same time;
- *Queue*: filter by the selected queue;
- *Caller*: filter by the selected caller;
- *Customer*: filter by the selected customer. A customer is an entry of the phonebook, which can have more numbers (callers) associated to it;
- *Agent*: filter by the selected agent;
- *Group*: filter by the selected users group.

When the parameters and filters have been chosen, press the button *Calculate* to show the graph. By rolling over each point for line graphs or the column for column charts, the value of the item will be displayed. See figures 19.2 and 19.3 for example outputs.

The data set can be also exported by pressing the *Export* button. A CSV file will be downloaded with all the points of the graph.

> ✍ If the selected period needs to display more data points than what can be displayed (e.g. it makes no sense to display more points than the resolution of the browser), an error will be displayed. To overcome it, just select a shorter time frame or a different aggregation.

Table 19.1: Statistics X axis parameters.

X axis	Description
Evolution in time	Show a line chart representing the magnitude of Y axis parameter during time. Aggregation can be hourly, weekly, daily and yearly
Agent	Column chart showing agents on each column
Group	Column chart showing every users group belonging to a queue
Customer	Column chart showing a customer on each column; customer can correspond to a caller if not inserted into the phonebook, otherwise if present on the phonebook all numbers under the contact entry will be grouped as customer
Queue	Column chart showing a queues on each column
Caller	Column chart showing all single callers on each column
Distribution in time	Column chart showing Y parameter distributed across each hour if aggregated hourly, each day for daily aggregation or each month for montly aggregation
Unanswer cause	Column chart showing all causes of calls that have not been answered

Table 19.2: Statistics Y axis parameters.

Y axis	Description
Answered calls	Number of calls that have been answered
Average queue wait	Averaged wait time in the queue, expressed in seconds
Total calls	Total calls that have been received
Transferred calls	Total calls that have been transferred
Average queue call duration	Averaged duration of the call, if answered
Unanswered calls	Number of calls that have not been answered

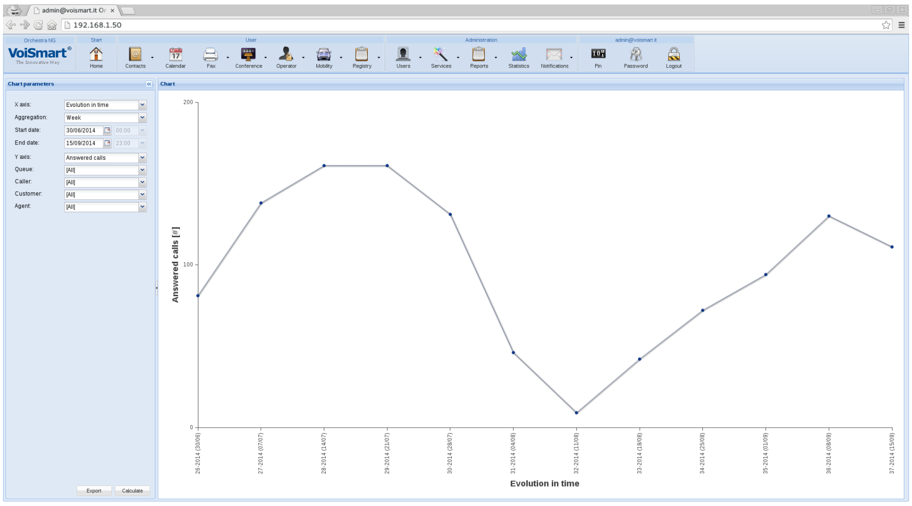


Figure 19.2: Line graph representing weekly evolution of answered calls.

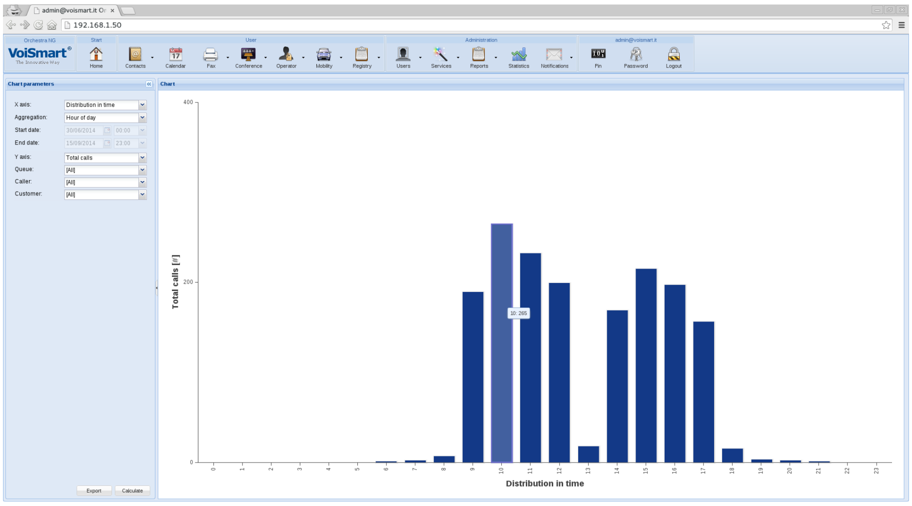


Figure 19.3: Column graph representing hourly distribution of all calls received by all queues.

20

Feature Codes

Contents

20.1 Call Recording

Number	Name	Description
*2	Recording start/stop	Starts to record the current call or stop if already recording. Can be used only during a call

20.2 Recall

Number	Name	Description
*5	Recall activation	Start a recall request to the last internal extension dialed

20.3 Calendars

Number	Name	Description
20 + num	Calendar Always On	Calendar <num> always enabled, so matches always in the dialplan
21 + num	Calendar Always Off	Calendar <num> always disabled, so never matches in the dialplan
22 + num	Calendar Always Auto	Calendar always follows the defined events. This is the default
23 + num	Calendar Toggle	Toggle between previous Calendar modes

It is possible to monitor a calendar status by configuring a BLF on the phone with the extension *23* + <calendar num>, where <calendar num> is the number of the calendar being monitored.

The calendar status is mapped on BLF in the following way:

solid red Calendar is set in Auto mode;

solid green Calendar is set in Always Off;

blinking red Calendar is set in Always On.

> ✍ BLF colors and behavior may vary on different phones brands.

20.4 Privacy

Number	Name	Description
31 + num	Privacy On	Call number <num> by setting privacy headers in order to hide the caller identity. Must be supported by remote site
#31* + num	Privacy Off	Call number <num> by setting privacy headers in order to show the caller identity. Must be supported by remote site

20.5 Queues

Number	Name	Description
#38	Queue remove all	Log the caller out from all the queues
38 + num	Queue call	Call the queue <num>
*38*0* + num	Queue Unjoin	Logs the caller out from the queue <num>
*38*1* + num	Queue Join	Logs the caller into the queue <num>
*38*2* + num	Queue Toggle	Toggle the queue <num> membership of the caller

It is possible to monitor a user's queue membership by configuring a BLF on the phone with the extension *38*2* + <queue num> + * <username>, where <username> is the username part of the user being monitored on queue having number <queue num>.

The membership status is mapped on BLF in the following way:

solid red user is logged into the queue;

solid green user is logged out of the queue.

> ✍ BLF colors and behavior may vary on different phones brands.

20.6 Call Park

Number	Name	Description
*36	Call park auto	Park the call, announcing the park lot number which is automatically choosen. If used with a call transfer, perform a supervised transfer in order to hear the park lot number, otherwise it will be announced to the transferred party
37 + num	Call park	Park the call in park lot number <num>. If the position is already taken, return busy. Is possible to park a call in this way using either blind or supervised transfer
#37* + num	Call unpark	Pickup a call parked in position <num>. Returns not available if no call is parked in the <num> slot

Is possible to monitor a park lot position by configuring a BLF on the phone with the extension park+<num>.

20.7 Call Forwards

Number	Name	Description
61 + num	CF Always	Forward all incoming calls to <num>
#61	CF Always Off	Disable CFA
62 + num	CF on Busy	Forward all incoming calls to <num> if the called party is busy
#62	CF on Busy Off	Disable CFB
63 + num	CF on No Answer	If the called extension does not answers, forward the call to <num>
#63	CF on No Answer Off	Disable CFNA
64 + num	CF on Unreachable	If the called party is not reachable, forward all incoming calls to <num>
#64	CF on Unreachable Off	Disable CFU

Number	Name	Description
#65	All Call forwards Off	Disable all CF services

Call forwarding service replaces current mobility rule with a system one, which can be shown with the web GUI but cannot be modified. If other mobility rules are configured, is possible from the web panel to switch to another rule from the system one. When call forward feature codes are used, the system rule gets applied automatically. See chapter 13 on page 111 and section 13.2 on page 112.

20.8 Apply Mobility Rule

Number	Name	Description
66 + pin user * + mobility id	Apply a mobilty rule	Apply a mobilty rule with <id> for user with <pin user>

To retrieve mobility id, use number listed next to mobility rule name as shown on figure 13.1 on page 112.

20.9 Conference

Number	Name	Description
72 + num	Conference	Join conference room <num>

20.10 Multicast intercom

Number	Name	Description
78 + num	Multicast intercom	Call multicast intercom group <num>, which is the n^{th} multicast address in the caller phone auto provisioning configuration.

20.11 Intercom

Number	Name	Description
79 + num	Intercom	Call intercom group <num>

20.12 Call Pickup

Number	Name	Description
*80	Pickup Any	Pickup the call on any ringing extension
*81	Pickup Group	Pickup the call on any ringing extension of the group where the calling extensions belongs
81 + num	Directed Pickup Group	Pickup the call on any ringing extension of the group <num>
82 + num	Pickup Extension	Pickup the call on the ringing extension <num>

If multiple extensions are ringing, pickup will steal the one that started first.

20.13 DND

Number	Name	Description
*85	DND on	Enable DND
#85	DND off	Disable DND

Many SIP devices allows to configure an extension to call when the DND button is pressed. In order to avoid misalignment between phone DND status and Orchestra NG DND status is strongly recommended to make use of this feature on the device.

20.14 Hotdesking

Number	Name	Description
*90	Enable hotdesking	An IVR will ask for the user pin number in order to connect the phone from where the feature code is dialed to the user
*91	Disable hotdesking	Disable hotdesking. Also, a “Thank you” message will be played back if the phone was connected to another user, otherwise the call will be closed

Number	Name	Description
*92	Enable timed hotdesking	Like *90 but after the user session timeout the phone will be freed and reconnected to the original owner

20.15 Service Classes

Number	Name	Description
*930	Service class cached authentication	An IVR will request the user pin in order to enable the service class on the phone, without asking the pin on every call
*931	Service class cache disable	Disable the cached authorization
*932	Service class timed authentication	Like *930, but the cached authorization will expire after the user session timeout, which can be configured from the web gui

Service classes logic can have several use cases: refer to figure 20.1 on page 174 for an overview of various scenarios.

20.16 Voicemail

Number	Name	Description
*100	Voicemail	Connect to voicemail service, which will ask for extension number and pin for authentication; the extension can be replaced with the user PIN number
100	User Voicemail	Leave a voicemail to the current <accountcode> of the call. For example, if a call forward to voicemail is needed, just set *100* as forward to number
100 + num	Direct Voicemail Leave	Leave a voicemail to the user which owns the extension <num>

Number	Name	Description
*101	Voicemail	Connect to voicemail service by just asking for the pin. The extension will be the phone from which the call is originated

20.17 Eavesdrop/Call barge

Number	Name	Description
842 + num	Eavesdrop / Call barge	Barge into the call of the extension <num>

During the listening, is possible to interact with the service using DTMF:

- 1: to speak to the called party;
- 2: to speak to the calling party;
- 3: to speak to both parties;
- 0: to return to listen-only mode.

20.18 Transfers

Number	Name	Description
39 + num	Transfer to call router	Can be used to transfer the current call to the dialplan call router with label <num> (if there is more than one matching the label, there's no guarantee on which one of them will be chosen).

20.19 Testing Codes

Number	Name	Description
*9000	Echo Test	After a tone, the system will send back the received audio stream. Useful to check if the audio path is correct

Number	Name	Description
9001 + num	Dialplan Test	Test the dialplan by connecting the call to the inbound number <num>, as if the call is coming from the external trunks
*9002*1	TTS Test	Check the English text for the speech system
*9002*2	TTS Test	Check the Italian text for the speech system

☞ Testing codes may create erroneous CDR records, so must be ignored for data analysis

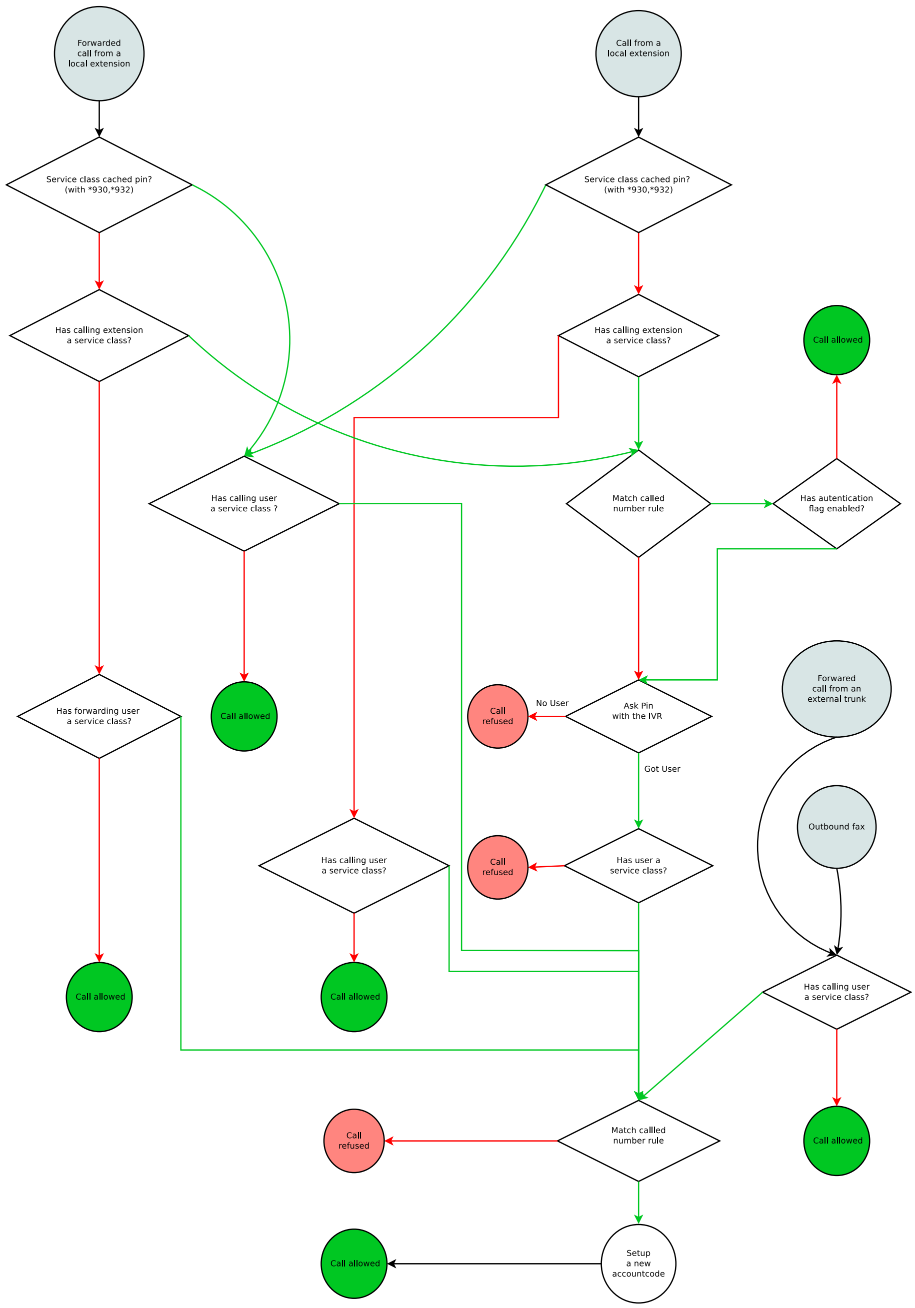


Figure 20.1: Service classes logical flow.

21

Security

Contents

21.1 Introduction

The Orchestra NG system is not meant to be a security appliance. While the development of the product takes in account possible misuse and misconfiguration it cannot be 100% secure if proper actions are not done. This chapter wants to indicate common security best practices, but is too far to be exhaustive.

> ☞ Security is not just a set of configurations. Is also a proper knowledge of the deployment, of the technologies involved and of the possible problems. First of all, security is a way of thinking: no manual will ever give you a 100% secure product, if you are not thinking with security in mind.

21.2 Certificates

All Orchestra NG services are available with plain and encrypted connections. To make this possible, the system has an internal Certification Authority which is used to generate certificates for all the services, for all domains (tenants).

Since is not an official Certification Authority, many clients can complain when using the encrypted connections, and commonly is possible to simply *accept* the certificate, by white listing it or ignoring the warning.

To have a more secure and strict certificate check, is possible to export the Certification Authority public certificate in order to add it to the various clients and avoid further warnings.

To export the Certification Authority certificate, visit the URL:

```
http(s)://_ng_hostname_/CA/cacert
```

An X.509 PEM CA root certificate will be downloaded. Refer to the client or operating system manual to figure how to import it as a new, trusted CA .

21.3 Web GUI

The Orchestra NG web gui can be also used with HTTPS. This method is preferred in order to protect the communication between the clients and the server. Since user password are sent in clear text when logging into the system, using SSL prevents information leakage.

21.4 Firewall

The system provides an embedded firewall that is completely managed by the application. If configured from the standard linux commands, it will get reset upon application start. Is always strongly recommended to configure and enable the firewall function. For further information on how to manage it, refer to the administrator manual. If the built-in firewall is not configured, the system will be completely open.

If an external firewall is preferred, the following list describe the ports used by Orchestra NG that must be opened to use the corresponding service.

- 22/TCP: for accessing the local shell, SSL encrypted;

- 25/TCP: for using the internal mail server for mail to fax and related functions;
- 80/TCP: for accessing web gui or provisioning the phones, in clear text;
- 443/TCP: for accessing web gui or provisioning the phones, SSL encrypted. Used also for provisioning of VoiSmart softphones;
- 5222/TCP: used by instant messaging clients;
- 5269/TCP: used by instant messaging server to federate with other XMPP servers;
- 7777/TCP: used by the Hotspot service;
- 16384 to 32768, UDP: for RTP media streams;
- any port defined in the *SIP profiles*, either TCP and UDP.

> ☞ Not all ports must be opened to the public, but only the needed ones. If a service is not to be exposed, do not open the corresponding ports.

21.5 Accounts

One of the common mistakes is to use weak account password. There are two kind of account password in Orchestra NG:

- user accounts: credentials used to connect to the web gui and to provision VoiSmart softphones;
- SIP accounts: used to authenticate SIP phones in order to make and receive calls;
- system accounts: also Linux accounts, the ones that can be used to perform operation from the system shell.

Using weak user password means that a malicious user can connect to the web interface and use or configure service on behalf of another user. If the leaked password belongs to an administrator, the service can be disrupted.

> ☞ One common mistake is to have a weak admin user password, frequently being "admin", as the default.

Using weak SIP password can lead to use the system as a calling platform for malicious users, resulting in high billing costs. The system already creates a random, secure password for SIP extensions and is suggested to not change it.

> ☞ One common mistake is to have the SIP password equal to the extension which is equal to the SIP username: e.g. extension 200 configured with SIP username 200 and password 200. Common scanners over the public Internet, when an open SIP server is found, try to brute force the SIP accounts by trying all numeric combinations. If one valid account is found, the server is almost

immediately used to place international calls.

Weak system (Linux) accounts can lead to unwanted remote accesses which can transform the system to a remote *zombie* for executing commands; commonly transforming the system into a member of a bigger group of computers using for executing a range of illegal operations (distributed denial of service, IRC bots, storage for illegal files and so on...)

21.6 Backup

The system offers a backup function from the administrative web gui. Always use it and archive in a secure place the exported file. Having it means that is possible to replace a broken unit or a hacked system without incurring into days of downtime. This is more important as the configuration is complex.

✍ Always do a backup before and after a configuration change!

A

CSV Phonebook file

When you want to import contacts in a phonebook, you need to upload a CSV file like this:

```
id,title,sn,givenName,initials,o,st,street,postalCode,l,co,description,mail,labeledURI,
    ↪ vsctgroup,number,typenumber,vsnumtype,vsforward,vsfast,vsnumdesc
1,,Brown,Richard,,Acme,,,,,,,,,vs-public,02132343,telephoneNumber,TelUfficio,,,
1,,Brown,Richard,,Acme,,,,,,,,,vs-public,013483343,facsimileTelephoneNumber,FaxAbitazione
    ↪ ,,,
2,,Red,Frank,,Aos,,,,,,,,,vs-personal,912321232,mobile,Cellulare,,,
3,,White,Susan,,Acme,,,,,,,,,vs-public,044333331,telephoneNumber,TelUfficio,,,
3,,White,Susan,,Acme,,,,,,,,,vs-public,012344222,homePhone,TelAbitazione,,,
4,,Yellow,Raymond,,,England,Regent Street,W1,London,United Kingdom,Notes,raymond@mail.com
    ↪ ,http://www.ray.com,vs-public,92131444,mobile,Cellulare,,,
4,,Yellow,Raymond,,,England,Regent Street,W1,London,United Kingdom,Notes,raymond@mail.com
    ↪ ,http://www.ray.com,vs-public,81777211,telephoneNumber,TelUfficio,,,
5,Astro,Orange,Andy,,,,,,,,,,,vs-public,522343322,telephoneNumber,TelUfficio,,,
5,Astro,Orange,Andy,,,,,,,,,,,vs-public,44322222,mobile,Cellulare,,,
5,Astro,Orange,Andy,,,,,,,,,,,vs-public,234433233,facsimileTelephoneNumber,FaxUfficio,,,
```

First row contains column or field names and values are separated by comma. On import you can omit this header and in this case default values will be used. Mandatory values for a contact are: *id*, *sn*, *givenName*, *vsctgroup*. Contacts with multiple numbers have one line for number and each row has same *id*. If you add a *number* in a contact row, you must add also a *typenumber* and a *vsnumtype*.

Type of contact depends on *vsctgroup* field and possible values are: *vs-public* for a public contact, *vs-personal* for a personal contact, any label to identify a group contact. Type of number depends on *typenumber* and *vsnumtype*. Possible values for *typenumber* are: *telephoneNumber*, *mobile*, *facsimileTelephoneNumber*, *homePhone*. Possible values for *vsnumtype* are: *FaxUfficio*, *Cellulare*, *TelUfficio*, *TelAbitazione*, *FaxAbitazione*, *TelAltro*, *FaxAltro*.

Field *vsfast* is used to identify a short number and *vsnumdesc* contains a brief description of short number.

Field *vsforward* is currently not used.

B

CSV User import

Example CSV file for user import:

```
username,password,email,language,firstname,middlename,lastname,pin,timezone,role,lcr,
    ↪ fax_enabled,mobility_enabled,vmail_password,vmail_quota,vmail_sendemail,
    ↪ vmail_attachaudio,vmail_leaveonserver,exten_number_alias,exten_username,
    ↪ exten_password,exten_ringtime,exten_macaddr,exten_mapping,exten_aliasmap,
    ↪ exten_soundfile
walter,secretpassword,white@example.com,en,Walter,,White,74361,Europe/Rome,
    ↪ DomainAdmin,default_lcr,true,true,3431,30,on,on,on,1003,1003,123456,30,
    ↪ CC11FF2233BB,my_outbound_map,my_alias_map,my_music_on_hold
bobby,,bobby@voismart.it,en,Bobby,,Fischer,29246,Europe/Rome,DomainAdmin,default_lcr,
    ↪ true,true,1285,30,on,on,on,1001,milleuno,123456,10,DD11FF2233BB,,,
exampleuser,,john@doe.com,en,John,K.,Doe,20272,Europe/Rome,User,default_lcr,true,true
    ↪ ,7442,30,on,on,on,1002,milledue,123456,30,AA11FF2233BB,,,
```

The above file is also a possible output for a user export operation, with the important distinction that the password field is always empty in this case.

C

CSV Export cdr

Example CSV file for export cdr:

```
id,direction,name,user_name,caller_id,domain_name,billusec,created_time,answered_time,
    ↪ hangup_time,hangup_cause,destination_number,endpoint_disposition
976,inbound,,milleuno,"""milleuno"" <milleuno
    ↪ >",192.168.3.95,6539454,1411388240992190,1411388243612343,1411388250151797,
    ↪ NORMAL_CLEARING,1000,ANSWER
977,inbound,,1000,"""vm""
    ↪ <1000>",192.168.3.95,2039596,1411389653772154,1411389653812144,1411389857771820,
    ↪ NORMAL_CLEARING,*100,ANSWER
978,outbound,,,"""Bobby Fischer"" <milleuno>",,0,1411389903411310,0,1411389916291738,
    ↪ USER_BUSY,1000,
979,inbound,,milleuno,"""milleuno"" <milleuno
    ↪ >",192.168.3.95,39819510,1411389902571226,1411389916451745,1411389956271255,
    ↪ NORMAL_CLEARING,1000,ANSWER
980,outbound,,,"""Bobby Fischer"" <milleuno>",,0,1411389997832090,0,1411390001011638,
    ↪ USER_BUSY,1000,
981,inbound,,milleuno,"""milleuno"" <milleuno
    ↪ >",192.168.3.95,9520212,1411389997291282,1411390001191648,1411390010711860,
    ↪ NORMAL_CLEARING,1000,ANSWER
982,inbound,,1000,"""vm""
    ↪ <1000>",192.168.3.95,2608108,1411390042651076,1411390042711095,1411390068792175,
    ↪ NORMAL_CLEARING,*100,ANSWER
983,outbound,,,"""Bobby Fischer"" <milleuno>",,0,1411390092991757,0,1411390095290965,
    ↪ USER_BUSY,1000,
984,inbound,,milleuno,"""milleuno"" <milleuno
    ↪ >",192.168.3.95,31699733,1411390092551383,1411390095591366,1411390127291099,
    ↪ NORMAL_CLEARING,1000,ANSWER
```

Domain_name field is available only when you export records using system administration web interface.

D

Regular Expressions

This section is not meant as a definitive guide for *Regular Expressions*, but as small introduction with commonly used patterns.

A *Regular Expression*, called also *regexp*, is a string expression that describe a set of strings that matches it. With a regexp it is possible to create a rule that matches several input strings, like numbers, thus avoiding the definition of many match rules. A single match task can frequently be specified by different expressions, so there's not a single way to obtain the same result.

> ✌ If it is needed to match all numbers in the range `123400-123499`, instead of writing 100 single matches (`123400`, `123401`, ..., `123499`) it is possible to define a rule like `1234..` that matches all the range.

If a literal match against a symbol is needed, escape it with \. For example to match an asterisk (*) in a number, the expression * must be used.

On table D.1 all supported symbols are shown. On table D.2 some application examples are reported and refer to table D.3 for some substitution examples.

Table D.1: Regular expressions valid symbols.

Symbol	Description
.	Indicates a single-digit placeholder. For example, `1234...` matches any dialed number beginning with `1234`, plus three additional digits.
*	Indicates that the preceding digit or pattern occurred zero or more times.
+	Indicates that the preceding digit or pattern occurred one or more times.
?	Indicates that the preceding digit or pattern occurred one or zero times.
|	*OR* operator. If `A` and `B` are regular expressions, `A|B` will match any string that matches either `A` or `B`.

Table D.1: Regular expressions valid symbols.

Symbol	Description
`{m,n}`	Indicates that the preceding digit or pattern occurred at least `m` and at most `n` times. `n` can be omitted and is assumed to be infinity if using a trailing comma, `m` otherwise.
`( )`	Indicates a group of patterns, also can be repeated with a repeating qualifier, such as `*`, `+`, `?`, or `{m,n}` For example, `(ab)*` will match zero or more repetitions of `ab`. Groups are also captured and can be used in replace rules by referring to them with the `$idx` or `{idx}` syntax, where `idx` is the index of the matched group, starting from `1`.
`[ ]`	Indicates a character class, which is a set of characters that you wish to match. Characters can be listed individually, or a range of characters can be indicated by giving two characters and separating them by a `-`. For example, `[abc]` will match any of the characters `a`, `b`, or `c`; this is the same as `[a-c]`, which uses a range to express the same set of characters. `[09]` matches only `0` or `9` while `[0-9]` matches all ten digits from `0` to `9`. To match only all lowercase letters, your expression would be `[a-z]` Symbols are not active inside classes, `[a2*]` matches `a`, `2` and `*` characters. `^` as first character in a class is a complementing set, which indicates to match everything except the set. For example `[^0]` matches everything except `0`.
`\d`	Matches any decimal digit; this is equivalent to the class `[0-9]`
`\D`	Matches any non-digit character; this is equivalent to the class `[^0-9]`
`\s`	Matches any whitespace character; this is equivalent to the class `[\t\n\r\f\v]`
`\S`	Matches any non-whitespace character; this is equivalent to the class `[^\t\n\r\f\v]`
`\w`	Matches any alphanumeric character; this is equivalent to the class `[a-zA-Z0-9_]`
`\W`	Matches any non-alphanumeric character; this is equivalent to the class `[^a-zA-Z0-9_]`

Table D.2: Regular expressions examples.

Expression	Description
`0?12345678`	Matches the number `12345678` with an optional `0` at the beginning
`021234..`	Matches the numbers from `02123400` to `02123499`
`021234[0-9]{2}`	Same as above
`0212[3-6]\d+`	Matches from `02123` to `02126` followed by one or more digits, so `021231`, `02123544`, `0212400` all are possible matches
`0212[3-6].+`	Similar to above, but will match also all characters after `[3-6]`, not only digits
`445566+`	Matches `44556` followed by one or more `6`, like `445566`, `4455666`, `4455666666`, ...

☞ Avoid match-all rules like `.*` because it can lead to unwanted side effects. Always use as strict as possible matching patterns. Laziness is not a good reason to use match all expressions.

Table D.3: Regular expressions substitutions.

Expression	Replacement	Description
`0?12345678`	`1234`	`12345678` with an optional `0` at the beginning is blindly rewritten to `1234`
`021234(..)`	`456$1`	numbers from `02123400` to `02123499` are rewritten to `45600` to `45699`
`021234(..)`	`456$10`	Error! Group `10` does not exists!
`021234(..)`	`456${1}0`	numbers from `02123400` to `02123499` are rewritten to `456000` to `456990`
`02(\d{3})([0-9][56])`	`${2}55${1}`	numbers like `02 123 45` are rewritten to `45 55 123` (note where the groups expansion is used)

E

Protocols and standards

> ✍ The following list of supported protocols and standards is provided for information purposes only and is not meant to be complete or guarantee of compatibility with other devices or implementations. Some protocols or standards may not be fully implemented.

E.1 VoIP

Signalling

- UDP, TCP and TLS transports
- RFC 2617: HTTP Digest Authentication
- RFC 3261: SIP v2.0
- RFC 3262: PRACK and 100rel
- RFC 3263: Locating SIP Servers
- RFC 3264: SDP Offer/Answer Negotiation
- RFC 3265: SIP Event Notifications
- RFC 3323: Privacy
- RFC 3325: Asserted Identity
- RFC 3327: Path
- RFC 3515: REFER
- RFC 3551: RTP/AVP
- RFC 3711: SRTP
- RFC 3842: Message waiting event
- RFC 3856: Presence
- RFC 3892: Referred-By

- RFC 3891: Replaces
- RFC 4028: Session Timers
- RFC 4566: SDP Session Description Protocol

Media

- G.711u, G.711a
- G.722, G.722.1
- G.729: requires a separate channels license
- iLBC
- Speex
- H.263 (pass through only)
- H.263-1998 (pass through only)
- H.263-2000 (pass through only)
- H.264 (pass through only)
- T.38: fax over IP networks in real time

E.2 Fax

- V.21
- V.27ter
- V.29
- V.17
- ECM (error correcting mode)
- T.4 1D, T.4 2D, and T.6 image compression

E.3 PSTN Telephony

- FXS, FXO: analog telephone signalling and interfaces
- BRI, PRI: basic rate and primary rate digital interfaces
- ISDN: digital transmission over PSTN
- Q.921
- Q.931: signalling for DSS1

E.4 Others

- HTTP, HTTPS: web GUI and API access
- RFC 6455: Websockets
- SSH: remote management
- XMPP: instant messaging
- SSL: encryption of several communication channels, like HTTP, VPN, IM, SIP, ...

Glossary

ACD In telephony, an automatic call distributor (ACD) or automated call distribution system, is a device or system that distributes incoming calls to a specific group of agents based on customer need, type, and agent skill set.

ALG Application-level gateway. In the context of computer networking, an application-level gateway consists of a security component that augments a firewall or NAT employed in a computer network. It allows customized NAT traversal filters to be plugged into the gateway to support address and port translation for certain application layer "control/data" protocols such as SIP and others.

ATA An analog telephony adapter or analog telephone adapter (ATA) is a device used to connect one or more standard analog telephones to a digital telephone system, such as voice over IP.

BRI A Basic Rate Interface, is one of the kinds of ISDNaccess interfaces. A BRI provides 2 B-Channels, for transporting data, and a single D-Channel, for signalling and control.

CDR Call detail record, is a collection of data records produced as a result of phone calls and including details such as dialed number, call duration, call creation date and time, ... Used for billing or logging purposes.

DSS1 Digital Subscriber Signalling System No. 1 (DSS1) is a digital signalling protocol (D channel protocol) used for the ISDN. It is defined by ITU-T I.411 (ETS 300 102). It supports Bearer Capability, Low Level Compatibility and High Level Compatibility, ANI, DNIS and redirected number signaling in both directions. A standard developed by ETSI for Europe is known as Euro-ISDN or E-DSS1 or simply EDSS1 (European DSS1). See also ISDN. , 196

E1 An ITU-T and ETSI standard designed to carry digital signals over telephony trunks or lines at a bit rate of 2048 Mbit/s. Commonly used in Europe.

E.164 E.164 is an ITU-T standard which defines a numbering plan and general format of numbers for telephony systems.

Euro-ISDN see DSS1.

FDR Fax detail record, is a collection of data records produced as a result of fax calls and including details such as dialed number, call duration, transmission result, . . . Used for billing or logging purposes.

FXO A Foreign eXchange Office is an interface port that receives an analog line from the PSTN. An FXO device is a device with an FXO port attached, such as a phone or a fax. An FXO gateway is a device providing interconnection between an analog line and an IP-PBX.

FXS A Foreign eXchange Subscriber is an interface port that delivers an analog line to the subscriber. An FXS gateway is a device providing interconnection between an IP-PBX and an FXO port.

ISDN Integrated Services for Digital Network, is an ITU-T and ETSI standard to allow transmission of data and voice over a PSTN network. It is also common to refer to ISDN as the circuit-switched physical network itself. See also DSS1. , 195

span A span represents a single physical port of any TDM interface.

LCR LCR, or Least-cost routing, is the functionality of selecting the route for an outbound call based on some configurable settings, generally based on cost, but also on time and date, dialed number and so on. Can also mean Least-cost router, a single system component which routes the call according to a least-cost rule. 29, 48, 131

LDAP Lightweight Directory Access Protocol, is a protocol implemented by applications providing directory services, i.e. a software which provides to clients informations about users and services throughout the network.

NAT Network Address Translation is the task of rewriting IPdata packets headers when leaving a network element such as a router or gateway, with the purpose of IP masquerading or more generally mapping a set of IP addresses into another (e.g. for hiding a private LAN address space behind a single address, usually in the public space).

PRI A Primary Rate Interface, is one of the kinds of ISDN access interfaces. A PRI provides 30 E1 B-Channels or 23 T1 B-Channels.

PSTN The Public Switched Telephone Network, is the infrastructure providing public telephony and telecommunications, and is the collection of the interconnected circuit-switching networks around the globe and operated by telephony providers.

queue A queue is a simply ordered list of calls to be dispatched to agents. The algorithm by which calls are dispatched is called the queue strategy. 41, 201

RTP The Real-time Transport Protocol is a protocol used to carry media streams over IP networks and it is often used in conjunction with the SIP signalling protocol. 177

SBC A Session Boarder Controller is an entity used in VoIP networks which acts as a controller over the signalling and media of VoIP calls, usually placed on the edges of different networks to provide different services such as securing the internal network from the outside, media transcoding, signalling protocol translations, topology hiding, applying QOS policies, . . .

SIP The Session Initiation Protocol is an IETF-standardized signalling protocol used to control media sessions like voice or video calls.

SIP proxy A SIP proxy server is a SIP client and server software which acts as an intermediary between other SIP entities providing features such as routing, rewriting, applying policies and interpreting received messages before forwarding to a SIP server.

SIP trunk The term "trunk" derives from its use within circuit-switched telephony systems and, in the context of SIP, usually means a virtual sip entity on a server which process a request according to a predefined set of polices and rules. This is usually regarded as the VoIP equivalent of a trunk in the TDM world and can be used as a connection beetween SIP servers to provide inter-domain communication, to provide PSTN termination by connecting to a gateway service or an ITSP, and so on.

SIP gateway A *SIP gateway* in Orchestra NG is a SIP trunk which provides a connection to an ITSP, another Orchestra NG instance or a generic SIP-compliant IP-PBX.

SIP profile A *SIP profile* in Orchestra NG, is a set of common configurations to be applied to media calls, used to be able treat differently devices connected to different network segments. A *SIP profile* is commonly identified by a unique combination of the IP-port pair.

TCP The Transmission Control Protocol is a transport layer protocol providing an ordered, reliable and error-checked transmission of data packets over a network. 176, 177

TDM TDM, or Time-division-multiplexing, is a form of signal multiplexing (i.e. a way of conveying multiple signals or bit-streams on a shared communication medium) used for call interleaving in telephony systems. , 196

AES The Advanced Encryption Standard (AES) is a specification for the encryption of electronic data established by the U.S. National Institute of Standards and Technology (NIST) in 2001. 152

B-channel A *bearer* channel in an ISDN network providing data and voice transport at a *full-duplex bit rate* of 64 Kbit/s.

BLF a Busy Lamp Field is a visible indicator, usually a light or led, which shows the status of another terminal connected to the same PBX. 166, 167

CAC The Call Admission Control is a preventive procedure to control traffic congestion by limiting and possibly rejecting a call, according to some configured rules.

CSV A Comma Separated Values (also sometimes called Character-Separated Values, because the separator character does not have to be a comma) file stores tabular data (numbers and text) in plain-text form. 28, 68, 70, 161, 179, 181, 183

Carrier A carrier in Orchestra NG is a logical group of trunks that can be associated to an LCR rule.

Codec Short for Coder-Decoder, a codec is a software or hardware-based program designed to encode and decode a signal.

D-channel A *delta* channel in an ISDN network providing signalling and control information at a 16 Kbit/s rate for BRI, and 64 Kbit/s for PRI.

DHCP DHCP is an IP network protocol used to automatically assign network configurations such as DNS server, the default gateway, . . . to clients.

DNS DNS is a service which translates domain names into IP addresses. 142

DSS Direct Station Select, is the feature of having a group of keys on a terminal to select other terminals or stations to call. Often associated with a BLF indicator. 108

DTMF Dual-tone multi-frequency signaling (DTMF) is an in-band telecommunication signaling system using the voice-frequency band over telephone lines between telephone equipment and other communications devices and switching centers. 152

Extension An extension refers to a phone (physical or software-based) connected and configured on an IP-PBX. 4, 38

Feature Codes Feature Codes are codes allowing you to use the dial-pad on your telephone to access, activate, deactivate special features on an IP-PBX.

HTML5 HTML5 is a core technology markup language of the Internet used for structuring and presenting content for the World Wide Web. As of October 2014 [update] this is the final and complete fifth revision of the HTML standard of the World Wide Web Consortium (W3C). 152

Hotspot A hotspot is a site that offers Internet access over a wireless local area network (WLAN) through the use of a router connected to a link to an Internet service provider. Hotspots typically use Wi-Fitechnology.

IMAP The Internet Message Access Protocol is an IP protocol used for email retrieval. When used over SSL-secured connections, it is often used the term IMAPS. 55

IM Instant messaging (IM) is a type of online chat which offers real-time text transmission over the Internet or any other IP network. Short messages are typically transmitted bi-directionally between two parties, when each user chooses to complete a thought and select "send". Some IM applications can use push technology to provide real-time text, which transmits messages character by character, as they are composed. More advanced instant messaging can add file transfer, clickable hyperlinks, Voice over IP, or video chat. 142

IVR In telecommunications, IVR allows customers to interact with a company's host system via a telephone keypad or by speech recognition, after which they can service their own inquiries by following the IVR dialogue. IVR systems can respond with prerecorded or dynamically generated audio to further direct users on how to proceed. IVR applications can be used to control almost any function where the interface can be broken down into a series of simple interactions. 2, 131, 136

MP3 MPEG-1 or MPEG-2 Audio Layer III, more commonly referred to as MP3, is an audio coding format for digital audio which uses a form of lossy data compression. 152

MWI A message-waiting indicator (MWI) is a telephone feature that illuminates a generic indicator like a LED or icon on the LCD display, to notify the user of waiting voicemail messages on the IP-PBX.

Multicast In computer networking, multicast (one-to-many or many-to-many distribution) is group communication where information is addressed to a group of destination computers simultaneously. 45, 46

NAS A network access server (NAS) is a single point of access to a remote resource. It is meant to act as a gateway to guard access to a protected resource. This can be anything from a telephone network, to printers, to the Internet.

NTP Network Time Protocol (NTP) is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency data networks.

POP The Post Office Protocol is an IP protocol used for email retrieval. When used over SSL-secured connections, it is often used the term POPS. 55

QOS The Quality of Service is used to guarantee a certain level of performance in telephony and computer networks by providing the ability to limit or assign different priorities to single applications, users, type of traffic and so on. Common requirements are often fixed bit rates, response delays, jitter...

RADIUS Remote Authentication Dial In User Service (RADIUS) is a IETF standarized networking protocol that provides centralized Authentication, Authorization, and Accounting (AAA) management for users that connect and use a network service.

Realm A Realm is the set of authentication resources, usually consisting of an authentication server and its security policies.

Roster In XMPP, one's contact list is called a roster, which consists of any number of specific roster items, each roster item being identified by a unique JID (usually of the form <contact@domain>). A user's roster is stored by the user's server on the user's behalf so that the user may access roster information from any resource. 142

SMTP The Simple Mail Transfer Protocol is an IP protocol used for email transmission. When used over SSL-secured connections, it is often used the term SMTPS. 78

SRV record An SRV record is a DNS record type used to define the location on the network of specific services such as SIP or XMPP servers. 143

SSL The Secure Sockets Layer (SSL) is a cryptographic protocol for secure communication and providing strong encryption of network traffic at application level protocol. 142, 200

TLS The Transport Layer Security is a cryptographic protocol for secure communication and providing strong encryption of network traffic for application level protocols like HTTP, SMTP, ... 55, 142

TON TON or type of number, indicates the scope of the address value, such as whether it is an international number, a national number, unknown or other formats.

Trunk In the TDM world, a trunk is a physical line or circuit connecting telephony switches, providing a transmission channel between two elements. In this document, the term TDM will be used interchangeably to denote a TDM trunk or a SIP trunk, as it is common practice in the telecommunication field.

VLAN In computer networking, a single layer-2 network may be partitioned to create multiple distinct broadcast domains, which are mutually isolated so that packets can only pass between them via one or more routers; such a domain is referred to as a virtual local area network, virtual LAN or VLAN.

VPN A Virtual Private Network is a *point-to-point* connection used to link two private networks over a public one (such as the Internet), using a secure, encrypted tunnel.

WAV Waveform Audio File Format (WAVE, or more commonly known as WAV due to its filename extension) (rarely, Audio for Windows) is a Microsoft and IBM audio file format standard for storing an audio bitstream on computers. 152

WSS WSS stands for secured WebSocket, a WebSocket over SSL.

WebRTC WebRTC (Web Real-Time Communication) is an API definition drafted by the World Wide Web Consortium (W3C) that supports browser-to-browser applications for voice calling, video chat, and P2P file sharing without plugins.

WebSocket WebSocket (WS) is a protocol providing full-duplex communications channels over a single TCP connection. The WebSocket protocol was standardized by the IETF as RFC 6455 in 2011. , 200

XMPP Extensible Messaging and Presence Protocol (XMPP) is a communications protocol for message-oriented middleware based on XML (Extensible Markup Language). The protocol was originally named Jabber, and was developed for near real-time, instant messaging (IM), presence information, and contact list maintenance. Designed to be extensible, the protocol has also been used for publish-subscribe systems, signalling for VoIP, video, file transfer, gaming, Internet of Things (IoT) applications such as the smart grid, and social networking services. Unlike most instant messaging protocols, XMPP is defined in an open standard and uses an open systems approach of development and application, by which anyone may implement an XMPP service and interoperate with other organizations' implementations. 142, 143

bricked The word "brick", when used in reference to consumer electronics, describes an electronic device such as a phone, router, router, or tablet computer that, due to a serious misconfiguration, corrupted firmware, or a hardware problem, can no longer function, hence, is as useful as a "brick".

dscp Differentiated services or DiffServ is a computer networking architecture that specifies a simple, scalable and coarse-grained mechanism for classifying and managing network traffic and providing quality of service (QoS) on modern IP networks. DiffServ can, for example, be used to provide low-latency to critical network traffic such as voice or streaming media while providing simple best-effort service to non-critical services such as web traffic or file transfers.

T1 An ITU-T and ETSI standard designed to carry digital signals over telephony trunks or lines at a bit rate of 1544 Mbit/s. Commonly used in North America and Japan.

UDP The User Datagram Protocol is a transport layer protocol providing a very simple model for packet transmission over a network without guarantees on order, reliability or retransmission of lost data. 177

Wi-Fi Wi-Fi, also spelled Wifi or WiFi, is a local area wireless technology that allows an electronic device to exchange data or connect to the internet using 2.4 GHz UHF and 5 GHz SHF radio waves.

Agent In Orchestra NG, an agent is a user who is a member of one or more queues.

Autoprovisioning Autoprovisioning is the process of auto configuring IP-phones via a central configuration server like Orchestra NG.

Domain In Orchestra NG, a domain is a single tenant which organizes togheter related user and extensions in a way that its configuration (dialplan, extensions, call routing, service classes, ...) is completely isolated from other users on the system. That level of separation is granted by SIP domains. Usually in Orchestra NG, a different domain is created for every company o tenant which uses the system. 76

RPM Red Hat Package Manager or RPM Package Manager (RPM) is a package management system. The name RPM variously refers to the .rpm file format, files in this format, software packaged in such files, and the package manager itself. RPM was intended primarily for Linux distributions; the file format is the baseline package format of the Linux Standard Base.

Station ID A station ID is a short string (typically less than forty characters) which identifies the fax machine, and is printed as an header on received and sent documents. Usually denoted as "Called subscriber identification" or "Transmitting subscriber identification" when differentiating the receiving and trasmitting end. 41

yum The Yellowdog Updater, Modified (yum) is an open-source command-line package-management utility for Linux operating systems using the RPM Package Manager.