Kerio Operator

Administrator's Guide

Kerio Technologies

Contents

Installing Kerio Operator

Product Editions

Software Appliance

Kerio Operator Software Appliance is an all-in-one package of Kerio Operator which also includes a special operating system.
Designed to be installed on a computer without an operating system, this edition is distributed as an installation disc. Software Appliance cannot be installed on a computer with another operating system and it does not allow to install other applications.

VMware Virtual Appliance

A virtual appliance designed for use in VMware products.
VMware Virtual Appliance is a Software Appliance edition pre-installed on a virtual host for VMware. The virtual appliance is distributed as OVF and VMX.

Kerio Operator Box

Hardware device ready for network connection. There are two types which differ in performance.

Kerio Operator Software Appliance

For Kerio Operator system requirements, refer to the Kerio Operator product pages.

You obtain Kerio Operator as a standard ISO image which you need to burn on a CD. Boot from this CD and install the Kerio Operator operating system. The Kerio Operator application is also installed during the process.

How to connect Kerio Operator to network

After booting the system, a console with the IP address for Kerio Operator is displayed.

If you use a DHCP service on your network, Kerio Operator will be assigned an IP address automatically and will connect to the network. If you do not use or do not wish to use DHCP for Kerio Operator, you have to set the IP address manually.

The current network configuration is displayed (and can be changed) in the Kerio Operator console in section `Network Configuration`. To set a static network address:

1. Select the `Assign static IP address` option in the console menu.
2. In the network interface on which the PBX should communicate, select the `Assign static IP address` option and enter the IP address, subnet mask and IP addresses of gateway and DNS server.

If you know the DNS name of the PBX, you can connect to it and configure it via the web interface.

Immediately after you connect Kerio Operator to the network, we recommend to read article concerning the security measures. Meeting security principles for Kerio Operator operation is extremely important. If the PBX is not protected by a firewall and supporting security rules, your internal telephone extension can be misused which may result in unexpected financial costs.

Kerio Operator VMware Appliance

For supported VMware product versions, check

http://www.kerio.com/operator/requirements/

Use an installation package in accordance with the type of your VMware product:

- For products VMware Server, Workstation, Player and Fusion, download the compressed VMX distribution file (`*.zip`), unpack it and open the file with extension `.vmx`.
- You can import a virtual appliance directly to VMware ESX/ESXi from the URL of the OVF file — for example:

  `http://download.kerio.com/dwn/operator/`

  `kerio-operator-appliance-2.3.0-2500-vmware.ovf`

  VMware ESX/ESXi automatically downloads the OVF configuration file and a corresponding disk image (`.vmdk`).

If you import virtual appliance in the OVF format, bear the following specifics in mind:

- In the imported virtual appliance, time synchronization between the host and the virtual appliance is disabled. However, Kerio Operator features a proprietary mechanism for synchronization of time with public Internet time servers. Therefore, it is not necessary to enable synchronization with the host.
- Tasks for shutdown or restart of the virtual machine will be set to default values after the import. These values can be set to “hard” shutdown or “hard” reset. However, this may cause a loss of data on the virtual appliance. Kerio Operator VMware Virtual Appliance supports so called Soft Power Operations which allow to shut down or restart hosted operating system properly. Therefore, it is recommended to set shutdown or restart of the hosted operating system as the value.

For more information, see section Network Connection.

Kerio Operator Box

For currently supported Kerio Operator Box configurations, refer to the Kerio Operator product pages.

For detailed information on connecting the device into the network, see the Kerio Operator Box 1000/3000 Series and Kerio Operator Box V300 installation guides.

How to connect box to network

Upon the first start, the appliance has a static IP address set to `10.10.10.1` on ethernet port 1. There are two ways to change the configuration:

- In the console — use an Ethernet cable to connect to the console. In the console menu, select the **Network Configuration** option and change the configuration.
- In the administration interface in section **System**.

  To connect to Kerio Operator, set the following TCP/IP parameters on your computer:

  - IP address: `10.10.10.2`
  - Subnet mask: `255.255.255.0`

To shut down the appliance:

1. Connect to Kerio Operator via the console and select the `Shutdown` command.
2. Kerio Operator series 1000 will shut down.

   Kerio Operator series 3000 will stop the server, however, the physical appliance stays switched on. Wait until you are not able to connect to Kerio Operator via Kerio Operator administration and turn the appliance off using the **pwr** button on the appliance.

Logging into Kerio Operator Administration

Overview

We recommend to use the supported browsers to connect to Kerio Operator Administration. For the list of the browsers, refer to the Kerio Operator product pages.

Kerio Operator Administration is currently localized into several languages. Select yours in the top right corner of the interface. The default language is set according to your browser language settings.

How to login

Before you login the first time, make sure you have:

- DNS name of the server with Kerio Operator.
- Supported browser

To login, enter the DNS name of the computer with Kerio Operator:

`kerio.operator.name/admin`

Administration runs solely via the HTTPS protocol on port 4021. The address is automatically redirected to:

`https://kerio.operator.name:4021/admin`

If the PBX is located behind firewall, HTTPS on port 4021 must be enabled.

If the URL is entered correctly, your browser displays a warning about a SSL certificate. After the installation, Kerio Operator creates a certificate which is not signed by a trusted certificate authority — it is a self-signed certificate (for more information, read article about the SSL certificates). Since you know the certificate can be trusted, you can add the security exception and continue to a login page.

First login

When you connect to the PBX for the first time, a configuration wizard is displayed where you:

1. Set the configuration wizard language.
2. Accept the Kerio Operator license agreement.

3. Set a password for the administration account (be sure to remember the password, you will need it to login to the PBX).

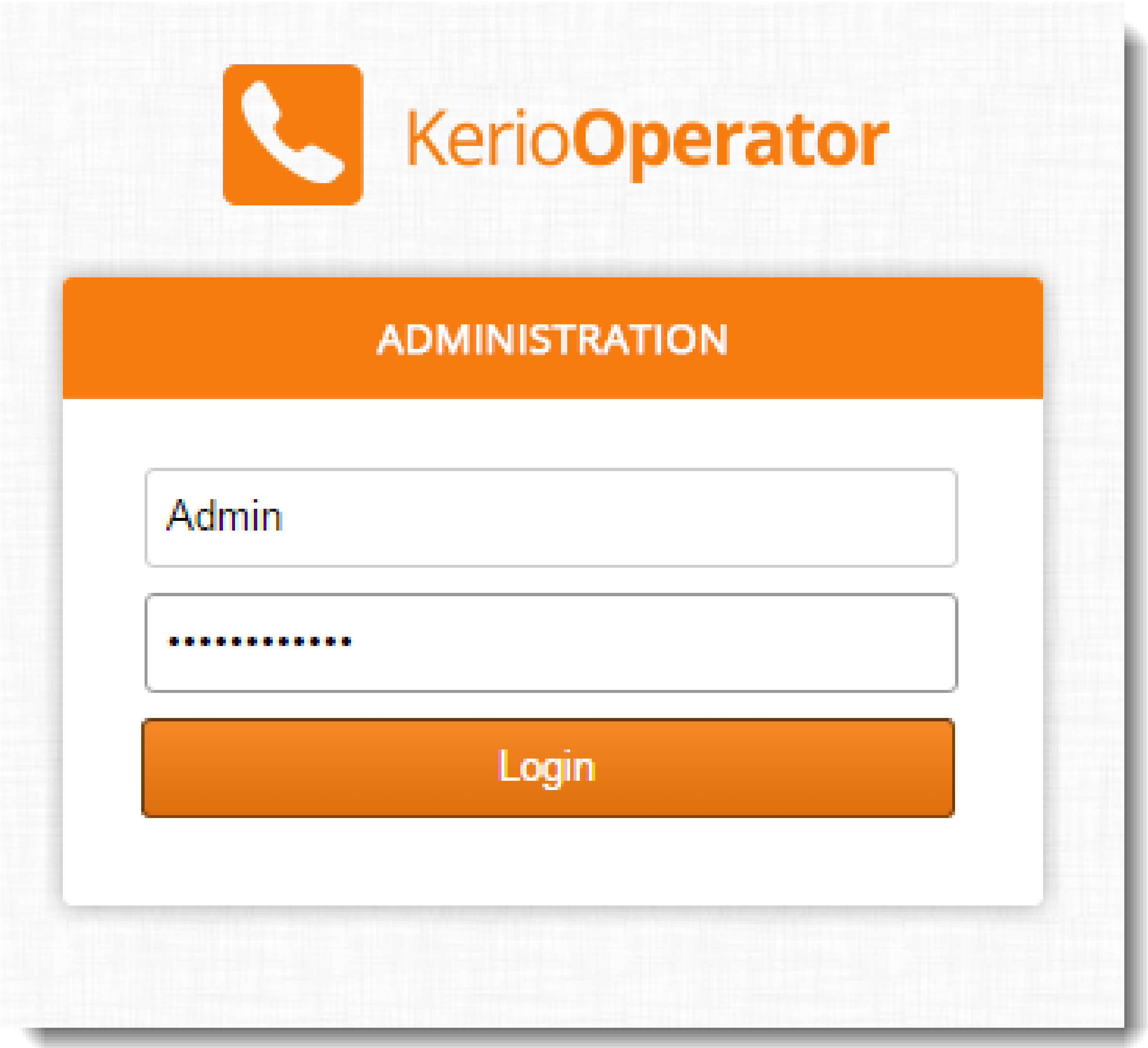
   This admin password is synchronized with password of user `root` in the operating system where Kerio Operator is installed (Kerio OS).

4. Set the time zone of Kerio Operator (requires a restart of the PBX).

5. Set the PBX language for communication with you and other users (warnings, auto attendant scripts, voicemail, etc.).

6. Configure the first extension number. If you use phone provisioning, extensions will be created automatically beginning with the number you enter here.

After successful configuration, the login page is displayed. Enter the username and password you created earlier.



To change the password, use the following steps:

1. Login to Kerio Operator using the HTTPS protocol
   (e.g. `https://operator.company.com/admin`)
2. Open the **Configuration** → **Users** section.
3. In the user list, select the administrator account you are logged in with and double-click on it.
4. Change the password on tab **General**.

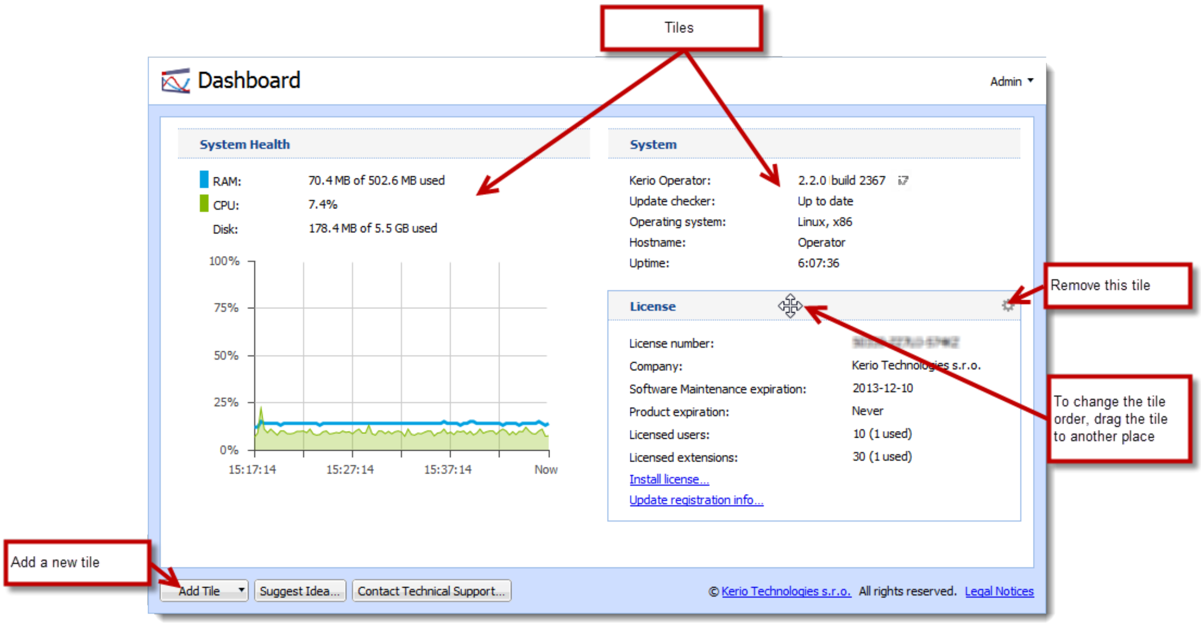
Using Dashboard in Kerio Operator

Dashboard overview

Kerio Operator includes a customizable Dashboard. Dashboard consists of tiles. Each tile displays a different type of information (graphs, statistics, etc.)

Dashboard is displayed in Kerio Operator after each login.

To display Dashboard later, go to **Configuration** → **Dashboard**.



Licenses and registrations

How to register Kerio Operator in the administration interface

You can register the product from the welcome page of the administration interface which is displayed after each login.

If Kerio Operator is protected by a firewall, it is necessary to allow outgoing HTTPS traffic for Kerio Operator at port 443. Unless HTTPS traffic is allowed, Kerio Operator cannot use the port to connect to the Kerio Technologies registration server.

When installed, the product can be registered as trial or as a full version.

Why to register the trial version

The trial version is intended to allow the customer to become familiar with the product's features and configuration. Once you register the trial version, you will be provided free Kerio Technologies technical support during the entire trial period (up to 30 days).

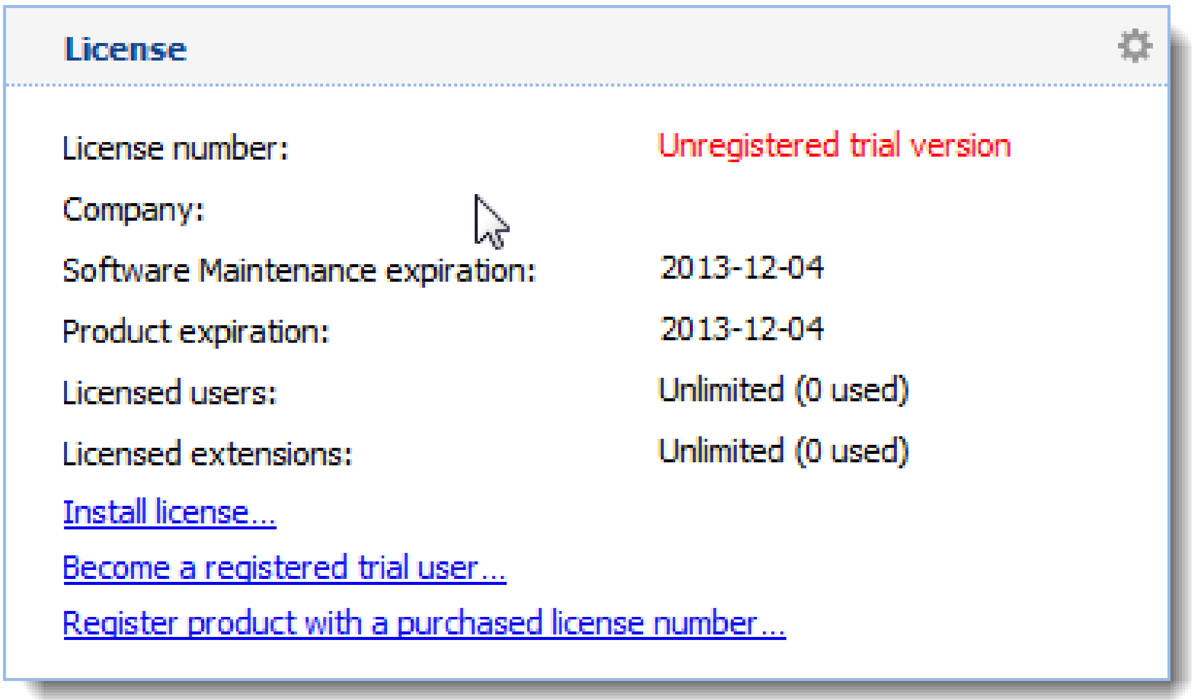


Figure 1 A Product Registration tile

The trial version can be registered by clicking on **Become a registered trial user** on Dashboard (see screenshot above). In the dialog box just opened, set the following parameters:

1. enter security code (CAPTCHA) from the image.
2. enter information about your company and agree with the privacy policy terms.
3. choose how many computers do you have in your company and how you learned of Kerio Operator.

Now, a special identification code called Trial ID gets generated. This ID is later required for contacting the technical support. After a successful registration, Trial ID can be found in the license information in the administration interface.

> Once you purchase the product, your Trial ID will become your license number (it will not change).

Registering full version

If your trial version is registered, the license key (`licence.key` file) is automatically imported to your product within 24 hours from your purchase. The Trial ID you entered in your product upon registration will be activated as a standard license number.

If you haven't registered your trial version:

1. Open the administration interface.
2. Click **Register product with a purchased license number** on Dashboard.
3. In the first step of the registration, enter the license number and enter the security code from the image.

   > The code is not case-sensitive.

4. Click **Next** to make Kerio Operator establish a connection to the registration server and check validity of the number entered.

   If the number is invalid, the registration cannot be completed.
5. Type the registration information about the company the product is registered to.
6. Kerio Operator connects to the registration server, checks whether the data inserted is correct and downloads automatically the license key (digital certificate).
7. Click **Finish** to close the wizard.

Manual import the license key

If you need to import a license key manually (for example from a backup), use the following steps:

1. Prepare the license key.
2. Login to Kerio Operator administration.
3. Click **Install license** on **Dashboard**.
4. In the **Install License** dialog, click **Browse**.
5. In the **Open** dialog, find the file `.key` with the license key and click **Open**.
6. In the **Install License** dialog, click **OK**.
7. Check the result in the **License** tile on **Dashboard**.

Kerio Operator installs the licence key.

Registering via a web browser

You purchased a license and your Kerio Operator cannot access the Internet? Follow these steps to register the product:

1. Go to https://secure.kerio.com/reg/
2. Register using your purchased license number.
3. By registering, you will receive a license key which must be imported to Kerio Operator.

The trial version of Kerio Operator cannot be registered via the website.

Securing Kerio Operator

Issues to address

- Restrict communication on firewall to necessary IP addresses and ports, especially if the PBX runs in the Internet.
- Restrict communication on the integrated firewall in Kerio Operator.
- Create strong SIP passwords.
- Restrict the number of attempts to enter SIP passwords.
- Using special rules, forbid international outgoing calls to countries you do not communicate with
- Restrict international outgoing calls to countries where you rarely call
- Encrypt your calls

The following sections describe these settings in detail.

Configuring firewall in local network

Kerio Operator is usually protected by firewall (in your local network or in the Internet). Certain ports need to be opened (or mapped) on firewall.

Service (default port)	Outbound connection	Inbound connection
SIP (5060)	allow	allow for SIP servers of your provider
IMAP (143)	allow if integration with Kerio Connect is enabled and there is a firewall between Kerio Connect and Kerio Operator.	deny
LDAP (389)	allow	deny
LDAPS (636)	allow	allow if you use mapping from Active Directory or Open Directory and there is a firewall between the directory service and Kerio Operator.
HTTP (80)	allow	deny
HTTPS (443)	allow	allow if you wish users to be able to connect to Kerio Phone from the Internet.
HTTPS (4021)	allow	allow if you wish users to be able to connect to the administration interface from the Internet.

Table 1 Services to be allowed on the firewall

Configuring firewall integrated in Kerio Operator

Prepare groups of IP addresses which you wish to allow for individual services (create them in **Definitions** → **IP Address Groups**).

You can configure the integrated firewall in section **Network** → **Firewall.**

Web server

If you want to restrict connections to Kerio Operator administration and softphone, check this option and select an IP group with addresses from which access will be allowed. Bear in mind that all the PBX users should be allowed to connect to Kerio Phone at least from their own workstation.

SIP

We recommend to restrict the SIP protocol solely to your internal network and external IP addresses of your SIP provider.

Phone provisioning

For security reasons, we recommend to restrict automatic phone provisioning solely to your internal network because TFTP sends configuration data as plain text.

CRM integration

For security reasons, we recommend to restrict communication solely to your internal network.

SNMP monitoring

For security reasons, we recommend to restrict communication solely to your internal network and IP adressess where monitoring servers are running.

If the options are unchecked, no restrictions are set.

Configuring protection against password guessing

Login data guessing is one of the most common attacks on a PBX. In Kerio Operator, attackers try to guess extension numbers and SIP passwords. This type of attack is defined by many unsuccessful attempts to enter extension number and SIP password during a login. Kerio Operator security settings enable you to limit the number of attempts of a phone (both software and hardware) to connect to the PBX. Apply settings as described below:

1. In the administration interface, go to **Security**.
2. Set the limit of unsuccessful attempts (usually 3 to 10 attempts) and set the time period during which attempts will be counted.

   Setting the time period protects real users who have forgotten their password or who have made mistakes during several logins. When the time limit expires, they can try to login to the PBX again.
3. Set the time during which Kerio Operator will block the source IP address.
4. You can also enter an email address that will be used for sending warnings about blocked IP addresses.

How to recognize there has been an attack attempt

In log **Security** look for the `Authentication failed` string. If there are many messages of this kind, somebody is trying to use a dictionary attack.

What to do in case of an attack

In case of an attack, apply the following instructions as soon as possible:

1. In section **Status** → **Calls** and in logs, look for information on which account has been abused.
2. Change the SIP password of this account.
3. Instruct users about handling their login details and secure behavior on the Internet.
4. The PBX is blocked, so it needs to be unlocked again.

Creating user accounts

User accounts overview

User accounts in Kerio Operator are used for:

- Login users to Kerio Phone
- Link users with an extension
- Set access rights to the system

Adding new accounts

You can create either a local user or map existing users from a directory service.

Adding local accounts

If you do not use directory services, create a local user in the Kerio Operator administration:

1. In the **Configuration** → **Users** section, click **Add**.
2. The **Add User** dialog box opens.
3. On the **General** tab, type username and password.

   The username must not contain spaces, diacritics and special symbols.
4. Click OK.

The user account appears in the **Users** section and the user can connect to Kerio Phone.

Adding accounts from directory service

Mapping differs according to the directory service used:

- Microsoft Active Directory
- Apple Open Directory

You need basic login credentials to connect directory service to Kerio Operator.

For more information, read Connecting Kerio Operator to directory service.

Assigning extensions to users

An extension is an internal telephone line. Each user can have assigned one or more extensions in Kerio Operator.

1. In the **Configuration** → **Users** section, select a user and click **Edit**.

   The **Edit User** dialog box opens.

2. On the **Extensions** tab, click **Add**.

   The **Select Extensions** dialog box opens.

3. In the **Select Extensions** dialog box, click **Add**.

   The **Add Extension** dialog box opens with predefined unused extension.

4. If the extension number meets your dial plan, click OK. If not, rewrite the extension number and then click OK.

5. Save the settings.

The users can use their Kerio Operator phone extension.

For more information about extensions, read the Creating extensions article.

Configuring ringing rules

For more information, read the Redirecting Calls article.

Redirecting calls

Overview

Redesigned in Kerio Operator 2.4!

Kerio Operator can route incoming calls to different internal extensions or external numbers.

You can configure ringing rules (call forwarding) for each user in the **Ringing Rules** section.

Users can also change their ringing rules in the Kerio Phone interface in the **Forwarding** section.

Configuring ringing rules in the administration

See the following example:

Bob has the internal extension 11 and a cell phone with the number5550155. He wants to receive calls on his cell phone. When he is busy, calls fallback to voicemail.

1. In the administration interface, go to **Configuration** → **Users.**
2. Select an account and click **Edit.**

   The **Edit User** dialog box opens.
3. Go to the **Extensions** tab.
4. Select an extension and click **Ringing Rules.**
5. Enable the **Ring extension** option.
6. Select a number for **Timeout.**

   When the specified time runs out, Kerio Operator forwards the call.
7. For **When busy**, select the **Continue** option.
8. Click Add and type the number **5550155** and a description (cell phone).
9. Select a number for **Timeout.**
10. Enable the **Fallback to voicemail** option.
11. Click **OK** to save your changes.

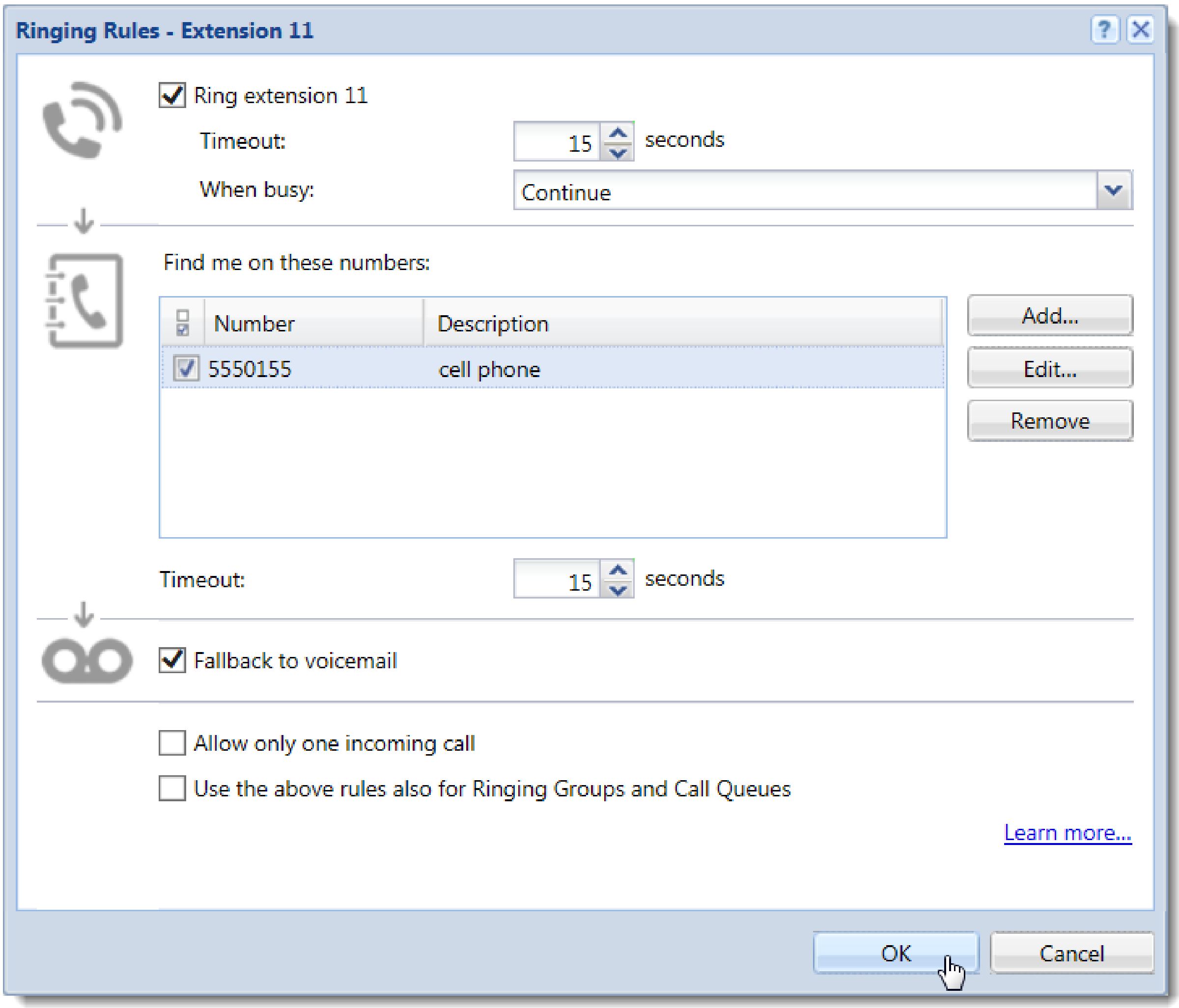


Figure 1

Additional configuration

New in Kerio Operator 2.4!

For ringing rules, you can configure additional settings:

- Configure extension to allow only one incoming call
- Apply ringing rules to calls coming from call queues and ringing groups

Configuring extensions to allow only one incoming call

If your phones support multiple calls, you can configure your extensions to reject or redirect additional incoming calls when an extension is already busy with a call.

To allow only one incoming call at a time:

1. In the administration interface, go to **Configuration** → **Users**.
2. Select an account and click **Edit**.

   The **Edit User** dialog box opens.
3. Go to the **Extensions** tab.
4. Select an extension and click **Ringing Rules**.
5. Enable the **Allow only one incoming call** option.
6. Click **OK**.

Kerio Operator now handles incoming calls using the configuration set in the **Ringing Rules** dialog box.

Applying ringing rules to calls coming from call queues and ringing groups

To configure ringing rules for calls from call queues and ringing groups:

1. In the administration interface, go to **Configuration** → **Users**.
2. Select an account and click **Edit**.

   The **Edit User** dialog box opens.
3. Go to the **Extensions** tab.
4. Select an extension and click **Ringing Rules**.
5. Enable the **Use the above rules also for Ringing Groups and Call Queues** option.
6. Click **OK**.

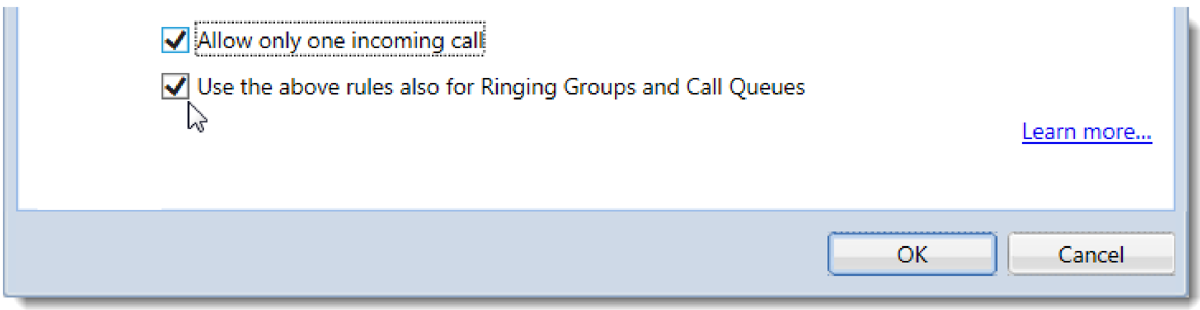


Figure 2

Configuring call forwarding in Kerio Phone

Users can also redirect calls to another number in their Kerio Phone. For more information, see Redirecting calls in Kerio Phone.

Creating extensions

Extension overview

An extension is an internal telephone line. Each user can have assigned one or more extensions in Kerio Operator.

The total number of extensions is limited to three times the number of licensed users.

Service extensions configured on the PBX services tab are not counted by the license file.

Adding new extensions

You have three options to add a new extension:

- An extension is created automatically when you connect a provisioned phone to the network.
- You can create an extension in **Configuration** → **Users** — the extension is assigned to a particular user.
- Create an extension in **Configuration** → **Extensions** — the extension is created as standalone (without being assigned to a user).

Creating a standalone extension

If you have a phone which is not used by any particular user, you can create a standalone extension for it.

1. In the administration interface, go to **Configuration** → **Extensions**.
2. Click **Add** → **Add Extension**.
3. Type an extension number.

   The field suggests an unused extension. You can change the extension number manually if necessary.
4. Save the settings.

SIP username and SIP password

Each extension has a SIP username and a SIP password. Kerio Operator uses SIP usernames and SIP passwords for authentication of phones to Kerio Operator. You use SIP username/password for connecting softphones or hardware phones to Kerio Operator (read more in the Configuring multiple registration of an extension article).

SIP usernames/passwords cannot be used to login into Kerio Operator or Kerio Phone.

Using SIP username/password

1. In the Kerio Operator administration interface, go to **Configuration** → **Extensions**.
2. Select an extension and click **Edit...**.
3. In the **Edit Extension** dialog, you can see fields **SIP username** and **SIP password**.
4. To display the SIP password, click the keys icon.

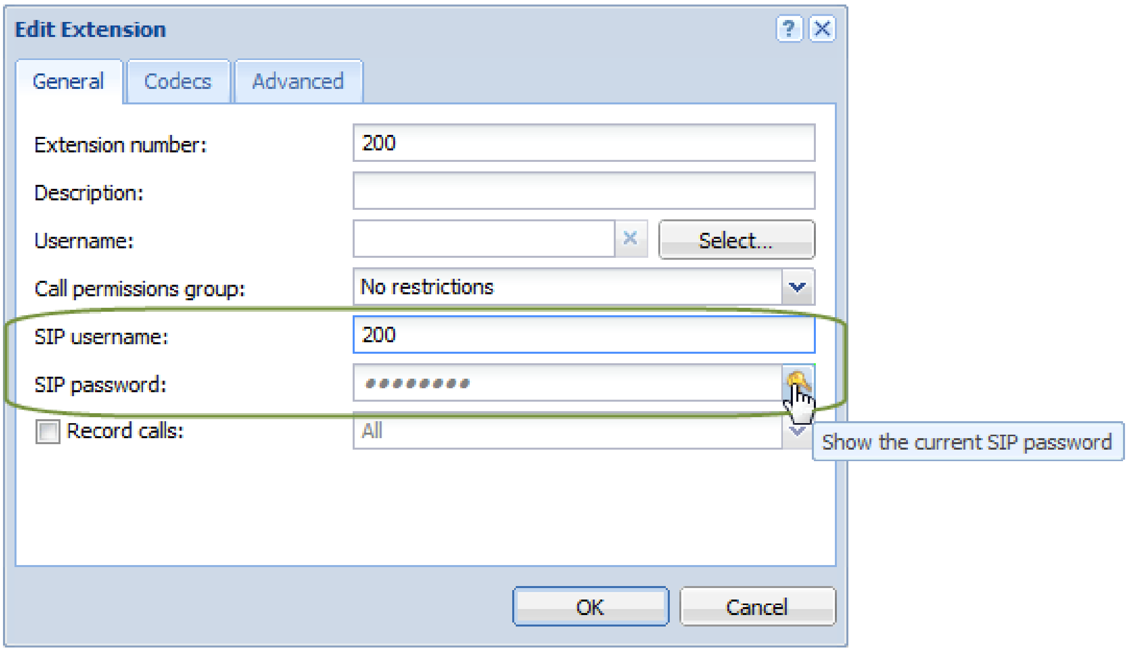


Figure 1 SIP username and SIP password

Now you can view the SIP username/password and use it for connecting a phone to Kerio Operator.

Encrypting calls

In Kerio Operator, you can encrypt your calls for any extensions.

1. In the Kerio Operator administration interface, go to **Configuration** → **Extensions**.
2. Select an extension and click **Edit...**
3. Click the **Advanced** tab and select **Encrypt communication (TLS and SRTP)**.
4. Click **OK**.

Now Kerio Operator encrypts all calls for the selected extension.

For more information about security, see Securing Kerio Operator.

Configuring multiple registration of an extension

Multiple registration overview

Do you want to use your extension with various phones? Softphone in your cell phone or IP phone in your smartphone? The solution is multiple registration.

Multiple registration (in contrary to assigning more extensions to one user) gives user the possibility to call from the same extension any time they make a call.

Example:
User Brenda Roar with username `broar` working at the Marketing department uses the extension 224. When necessary, she also works from home. She uses the following to communicate:

1. She has an automatically provisioned phone Cisco 7940 in his office.
2. She has X-Lite softphone on her home computer.
3. Occasionally, when connected via WiFi, she uses a SIP client on her mobile phone.

With correct settings of multiple registration that will be described in the following chapter she can use all the before-mentioned methods to authenticate.

Creating multiple registrations

1. Open section **Configuration** → **Extensions**.
2. Select Brenda Roar's extension (224). Click on Add → Add Another Registration.
3. A new registration is added to the user table. Add another registration. The result should be similar to the following image.

Configuring multiple registration of an extension

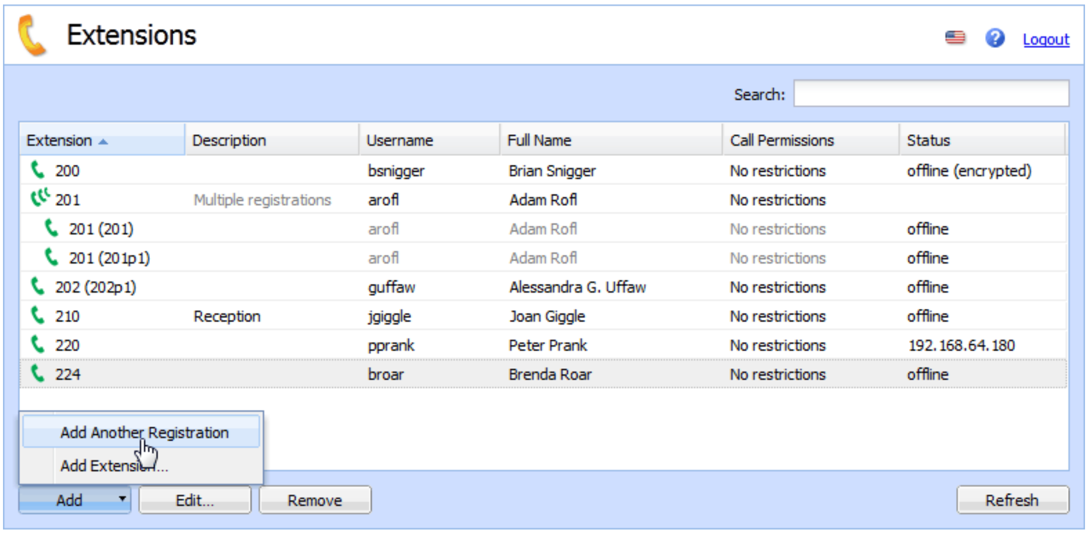


Figure 1 Extensions → Add Another Registration

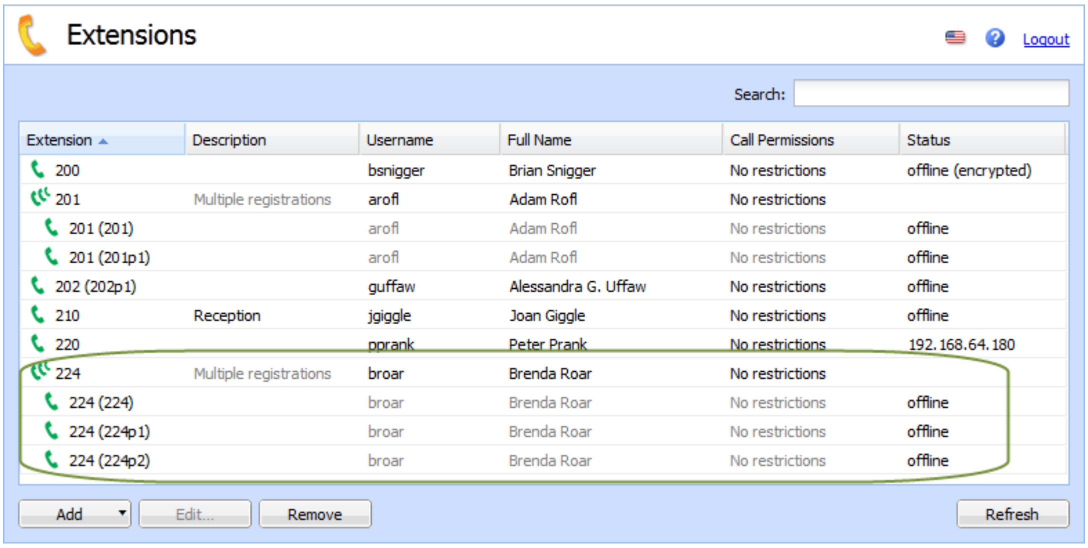


Figure 2 Extensions → Multiple registration

4. Double-click the 224p1 registration and note the SIP username and SIP password from the opened dialog.

5. Click OK.

6. In the X-Lite settings (detailed info for installation can be found in article Configuring the X-Lite software phone), enter the newly generated string into **User ID** and the SIP password into **Password**.

7. Repeat steps 4 to 6 for the second registration for the SIP client on a mobile phone.

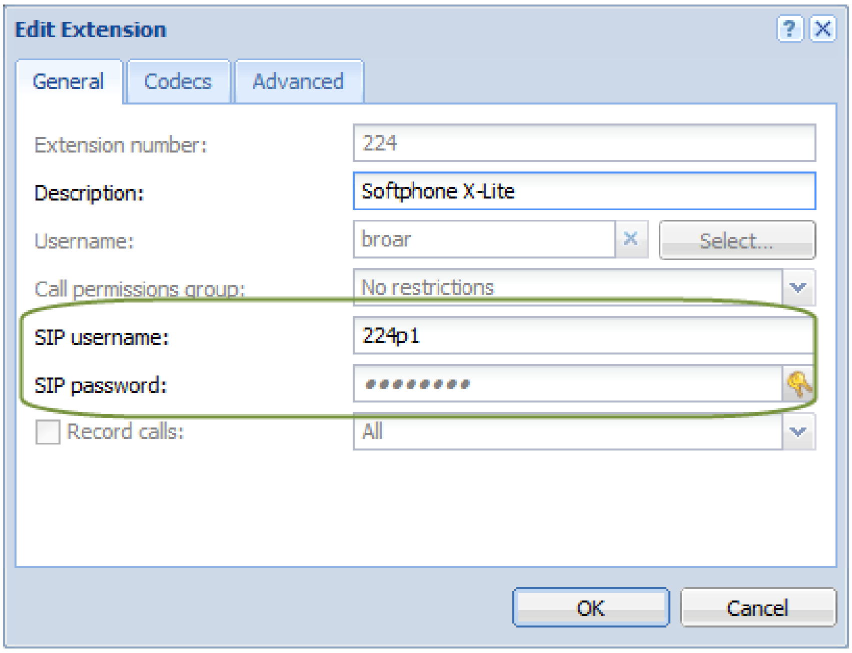


Figure 3 Edit Extension → Login information for X-Lite

Displaying, hiding and overriding phone numbers

Hiding users´ phone number

Redesigned in Kerio Operator 2.4!

To hide users´ phone numbers for outgoing calls:

1. In the administration interface, go to the **Configuration** → **Call Routing** → **Routing of outgoing calls** section, select a prefix and click **Edit**.

   The **Edit Outgoing Route** dialog box opens.

2. Go to the **Exceptions** tab.

3. **Add** an extension.

4. Select the box in the **Hide Caller ID** column.

5. Click **OK**.

Some VoIP service providers do not allow hiding of phone numbers. If you use one of these providers, this settings do not work. See article Connecting to VoIP service provider.

Changing phone number to a name

For outgoing calls, you can change the phone number to display a name:

1. In the administration interface, go to the **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls** section, select an interface and click **Edit**.

   The **Edit External Interface** dialog box opens.

2. Go to the **Advanced** tab.

3. In the **Outgoing calls** section, select the **Override display name with** option, and type a new name.

4. Click **OK**.

Extending display names for incoming calls

New in Kerio Operator 2.4!

In Kerio Operator, you can extend the display name of incoming calls. The configuration works for all numbers that reach the interface and Kerio Operator adds the configured text to the beginning of the number or the caller's ID.

For example, a call center provides a technical support for several companies (for example, **Workplace**). Administrator wants to extend a display name of incoming calls with the company name, so the call center employees know from where comes the call:

1. In the administration interface, go to the **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls** section.
2. Select an interface and click **Edit**.

   The **Edit External Interface** dialog box opens.
3. Go to the **Advanced** tab.
4. In the **Incoming calls** section, select the **Prepend display name with** option, and type **Workplace -**.
5. Click **OK**.

After this configuration, Kerio Operator extends all incoming calls to this interface with **Workplace -** (for example, **Workplace - 555 0155**).

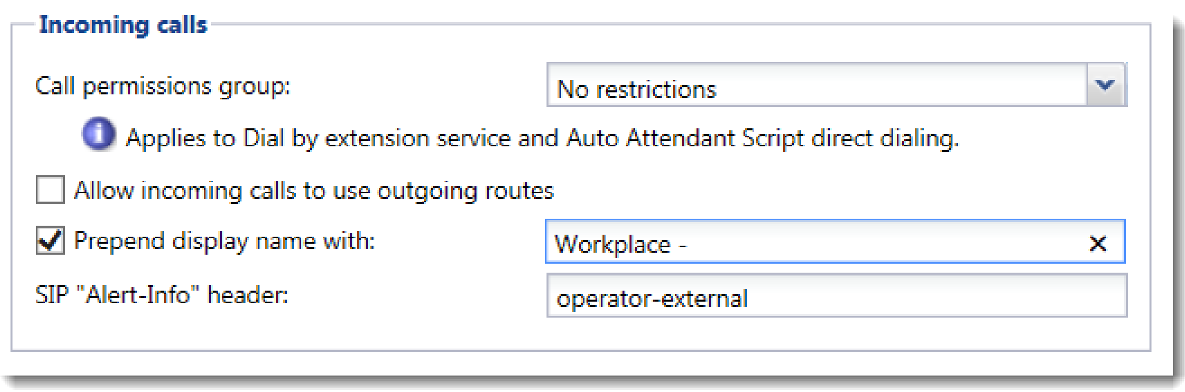


Figure 1

Connecting Kerio Operator to directory service

Which directory services are supported in Kerio Operator

Kerio Operator supports the following directory services:

- Microsoft Active Directory
- Apple Open Directory

What is the connection used for

In practice, mapping accounts from a directory service provides the following benefits:

Easy account administration
: Apart from the internal database of user accounts, Kerio Operator can also import accounts and groups from an LDAP database. Using LDAP, user accounts can be managed from a single location. This reduces possible errors and simplifies administration.

Online cooperation of Kerio Operator and directory service
: Additions, modifications or removals of user accounts/groups in the LDAP database are applied to Kerio Operator immediately.

Using domain name and password for login
: Users may use the same credentials for Kerio Phone login and domain login.

- Mapping is one-way only, data are synchronized from directory service to Kerio Operator. Adding a new user in Kerio Operator creates a local account — it will not be duplicated into the directory service database.
- When creating user accounts in a directory service, ASCII must be used to specify usernames. If the username includes special characters or symbols, user may not be able to login to Kerio Phone or the administration interface.
- If you disable users in Microsoft Active Directory, they are also disabled in Kerio Operator (they will not be able to login to Kerio Phone, make or receive calls with their extensions).
- If you disable users in Apple Open Directory, they stay enabled in Kerio Operator.

Phone extensions can be managed in a directory service (if available) or locally in Kerio Operator. Select the most convenient option.

Connecting to a directory service

To map users from a directory service:

- Connect to directory service in section **Integration** → **Directory Service**.
- Activate users.

All information about directory services can be found in the **Config** log.

Microsoft Active Directory

In the administration interface, go to **Integration** → **Directory Service**.

1. Check the **Map user accounts from a directory service** option and select your directory service type.
2. In the **Domain name** field, enter the name of your Microsoft Active Directory domain — the domain name is then copied in other necessary fields.
3. In the **Hostname** field, enter the DNS name or IP address of the Microsoft Active Directory server. If you have a backup server, enter its name in the **Secondary hostname** filed.
4. In the **Username** and **Password** fields, enter the authentication data of a user with at least read rights for Microsoft Active Directory database. Username format is `user@domain`.
5. Within the communication of the Microsoft Active Directory database with the PBX, sensitive data may be transmitted (such as user passwords). For this reason, it is recommended to secure such traffic by using SSL. To enable LDAPS in Microsoft Active Directory, it is necessary to run a certification authority on the domain controller that is considered as trustworthy by Kerio Operator.
6. The rest of the items in the dialog are completed automatically. Do not change them unless you have a special reason to do so. These items are Microsoft Apple Open Directory domain name and Kerberos Realm which has to match the Microsoft Active Directory domain name, written in capital letters.

Apple Open Directory

In the administration interface, go to **Integration** → **Directory Service**.

1. Check the **Map user accounts from a directory service** option and select your directory service type.
2. In the **Domain name** field, enter the name of your Apple Open Directory domain — the domain name is then copied in other necessary fields.

3. In the **Hostname** field, enter the DNS name or IP address of the Apple Open Directory server. If you have a backup server, enter its name in the **Secondary hostname** filed.

4. In the **Username** and **Password** fields, enter the authentication data of a user with at least read rights for Apple Open Directory database. Username format is: `uid=root,cn=users,dc=domain,dc=tld`.

5. Within the communication of the Apple Open Directory database with the PBX, sensitive data may be transmitted (such as user passwords). For this reason, it is recommended to secure such traffic by using SSL. To enable LDAPS in Apple Open Directory, it is necessary to run a certification authority on the domain controller that is considered as trustworthy by Kerio Operator.

6. The rest of the items in the dialog are completed automatically. Do not change them unless you have a special reason to do so. These items are Apple Open Directory domain name and Kerberos Realm which has to match the Apple Open Directory domain name, written in capital letters.

Activating users from a directory service

Once the mapping is set, select individual users and map them to the PBX. This is how to map users:

1. Open the **Configuration** → **Users** section.

2. Click **Import** → **Import from a Directory Service.**

3. In the dialog, select all users you wish to map (you can also add users later) and click **Next.**

4. If users in the directory service have phone extensions assigned, you can either keep them or disable them. If you disable them, you have to assign new extensions. You can do it, for example, while changing your dial plan.

5. Click on **Finish.** Activated users are displayed in section **Configuration** → **Users.**

Only extensions in attributes `telephoneNumber` (Microsoft Active Directory, Apple Open Directory) and `otherTelephone` (Microsoft Active Directory) can be mapped (are displayed). If you create special attributes in a directory service for your phone numbers, you will not be able to map such extensions.

Configuring automatic phone provisioning

Phone provisioning overview

Watch the Configuring automatic phone provisioning in Kerio Operator video.

Phone provisioning is used for automatic configurations of selected hardware SIP phones. Phone provisioning means:

- phone automatically connects to the PBX after booting and is assigned a phone extension,
- extensions are managed in the administration interface,
- if you confirm or plan it, the system will perform an automatic restart of provisioned phones if needed,
- phone firmware is automatically updated,
- displaying a company logo on hardware phones supported by Kerio Operator
- accessing company contacts through LDAP

Automatic firmware update is not supported for the Polycom phones and the original Cisco phones (Cisco SPA is supported). However, there is a possibility to update the firmware. You can upload all necessary files to folder `/var/tftp` in Kerio Operator manually. For detailed information see article Uploading configuration files to Kerio Operator TFTP server.

Use of phone provisioning is not always suitable. If Kerio Operator is located and runs in the Internet, for security reasons we do not recommend to use automatic phone provisioning.

What you need

1. In your local network, you need a DHCP server supporting parameter 66 (TFTP server address). Enter the address of Kerio Operator in this parameter.

   DHCP server integrated in Kerio Control supports parameter 66.

2. Only selected phones support automatic phone provisioning.
3. Appropriate settings need to be done in Kerio Operator.

If you wish to connect a phone which is not currently supported in Kerio Operator, you cannot use automatic provisioning. The configuration must be done on the hardware phone.

How to add a phone

1. In the administration interface, go to **Provisioned Phones** → **Hardware Phones**.
2. Click **Provisioning Settings**. The configuration dialog windows is opened.
3. Check the **Enable provisioning** option. The option must be checked.
4. Check option **Create new extension for newly registered phones** in case you create users locally (do not map them from a directory service).

   The **Create new extension for newly registered phones** option is checked by default. If you uncheck it, you cannot use automatic remote phone restart — you will have to restart phones manually if needed.

5. Each telephone must be authenticated when connecting to the PBX. Extension number and password are used for SIP authentication (Master Password in this case). Option **Master password for phones is enabled** enables to create one password for all provisioned phones. The password is saved in the configuration file which is sent to the phone upon the first connection to the network and the phone will use this password to authenticate at Kerio Operator.

   If you disable option **Master password for phones is enabled**, all phones will have their own passwords (it can be viewed in the configuration dialog of each phone).

Now the general environment for the provisioned phones is configured. Once a phone is connected to your network, it will be listed in section **Provisioned Phones**.

Adding phones manually

Phones which are not connected to the network can also be provisioned. You may do so manually — you need the phone's hardware address and the type of the phone. The procedure is described below:

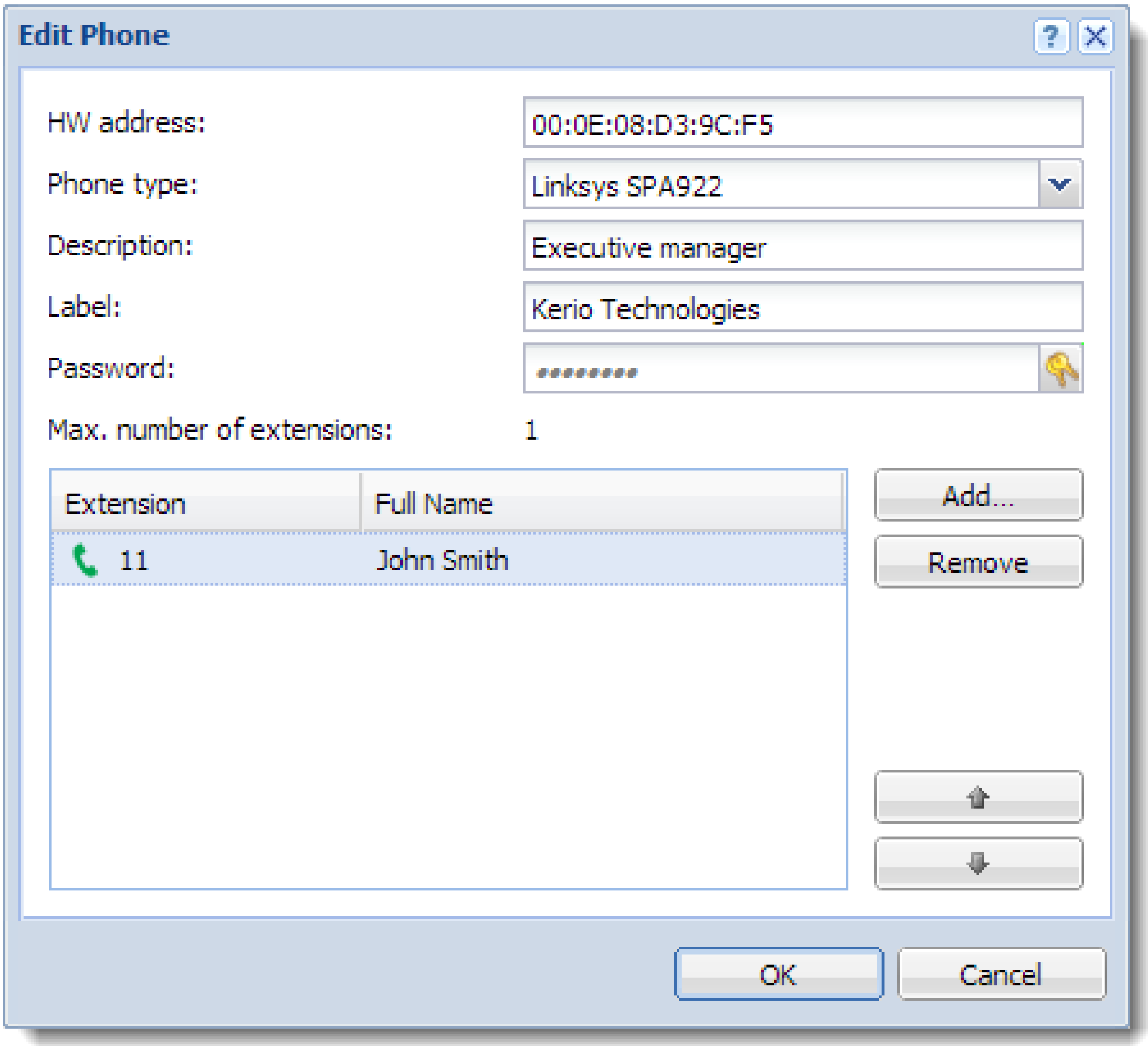


Figure 1 Connecting a phone manually

1. In section **Provisioned Phones**, click **Add**.
2. This opens a dialog which requires the hardware address of the phone (MAC address of the network card in the phone). The address may lack the colons. Once you save it, the colons will be added automatically.
3. Select the correct type of the hardware phone (special configuration scripts are created according to the phone type).
4. (Optional) Set a label of the phone (for example the name of your company).

   The upper label on the phone display.
5. Assign the phone user or users who will use it.

> If you do not know to which person the extension will be assigned, check option **Generate new extension number** and the extension will be assigned automatically. Phones without extensions assigned cannot be provisioned.

Importing from CSV file

Phones can be imported from a CSV file. Data in the file must follow certain rules:

- hwAddress — hardware address of the phone,
- phoneManufacturer — name of the phone's manufacturer,
- phoneType — phone type,
- extension1; extension2; ... — extensions assigned to the phone. The maximum number of extensions depends on the phone type.

Each phone uses one line and all items are separated by a semicolon.

The file may look as follows:

```
00:1a:a0:be:1e:cd;Cisco;7940;111;112
00:1b:b0:cd:e1:ca;Cisco;7960;115
00:1c:c0:ab:a2:24;Linksys;SPA942;113;114
```

Import data from a CSV file as described below:

1. In the **Provisioned Phones** section, click on **Advanced** → **Import from a CSV file**.
2. This opens dialog **Import from a CSV file** — click on **Upload CSV file**.
3. If the data in the file are correct, a list of all the phones and extensions is displayed. Check those you want to import.

4. Click **OK.**

5. The imported phones are displayed in the **Provisioned Phones** table.

Restarting provisioned phones

When you change configuration which affects provisioned phones, the phones need to be restarted (for example, when you create a new call route). When you do so, a dialog window recommending phone restart is displayed. You can do it immediately or wait for a more convenient time (for example to an off-peak time). To restart phones later:

1. Open the **Provisioned Phones** section.

2. Click **Advanced** → **Restart All Phones.**

Some Cisco telephones from newer series are not able to restart automatically. In case of configuration changes you have to check the result. If anything is wrong, restart the phones manually.
This warning doesn't relate to Cisco SPA phones.

Firmware

Kerio Operator allows easy installation of phone firmware which are managed through the phone provisioning:

1. Go to section **Provisioned Phones** and click on the **Advanced** → **Firmwares** button.

2. In the **Firmwares and Logos** dialog, select a firmware and click **Edit.**

3. In the **Edit firmware** dialog, select Verify the firmware.

   Kerio Operator vrifies if the firmware includes all important files and information.

4. Click **Upload File.**

5. This opens a dialog where you select a firmware file and confirm the selection.

6. In the **New firmware** dialog, select the appropriate phone.

7. Click **OK.**

The new firmware is installed and after the restart will be installed to phones.

Uploading a phone provisioning module

If you want to change or create a provisioning module (archived templates + PHP scripts which can change phones behavior), download Provisioning Developer Documentation and read it carefully.

When the provisioning module is prepared and archived, upload it to Kerio Operator:

1. Go to administration interface.
2. In section **Provisioned Phones**, click **Advanced** → **Provisioning Modules.**.
3. Click **Upload.**
4. Restart your phones.

Overriding templates

If you want to change a provisioning template (remote changing BLF, speed dials, etc.), go to the Editing provisioning templates article.

What to do if you want to know the password of your phone

If any of your users needs to know the password of their phone, we do not recommend to provide them with the Master Password. We have a specific solution:

1. In the administration interface, go to **Provisioned Phones** → **Hardware Phones.**
2. Click **Provisioning Settings.**
3. Disable master password.

Once you disable it, each phone will have their own password which can be shared with individual users.

Configuring inter-digit timeout

Inter-digit timeout sets the time between dialing the last digit and automatic dial. If your users complains that it is too long or too short, you can adjust it:

1. Go to the administration interface.
2. In section **Provisioned Phones**, go to **Provisioning Settings.**
3. In the **Phone Provisioning Settings**, set the **Inter-digit timeout.**

How phone provisioning works

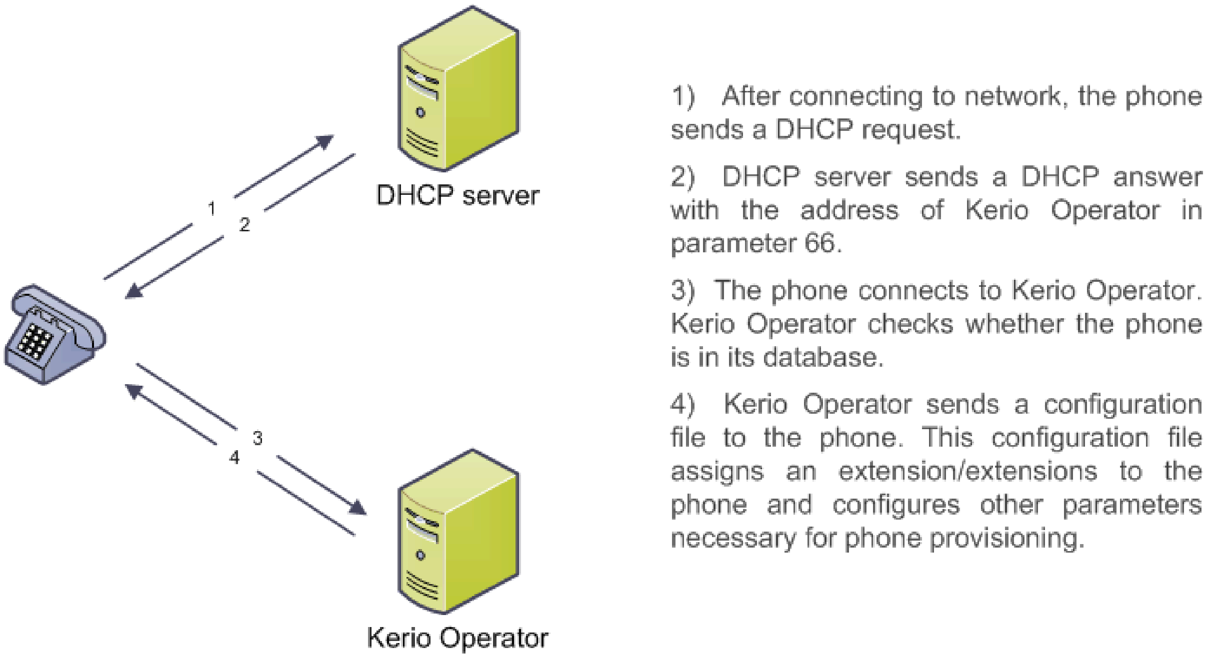


Figure 2 Automatic HW phone provisioning

This is how the automatic phone provisioning works:

- The telephone boots in the network and sends a DHCP request for an IP address.
- DHCP server accepts the request, assigns an IP address and sends it back in a DHCP reply. Besides the IP address, the message also contains TFTP (Trivial File Transfer Protocol) server address — Kerio Operator, in our case.
- SIP phone connects to TFTP server integrated in Kerio Operator.
- Kerio Operator checks whether the phone is new:
    - if it is new, Kerio Operator generates a new phone extension for the phone;
    - if it is not new, Kerio Operator finds the extension which the phone has used.
- Kerio Operator generates a configuration file suitable for the particular phone type and sends it via the TFTP protocol.
- The phone is configured using the values it has acquired in the configuration file and is ready to be used.

Some phones perform an automatic restart during the configuration.

Using provisioning tools

Provisioning tools overview

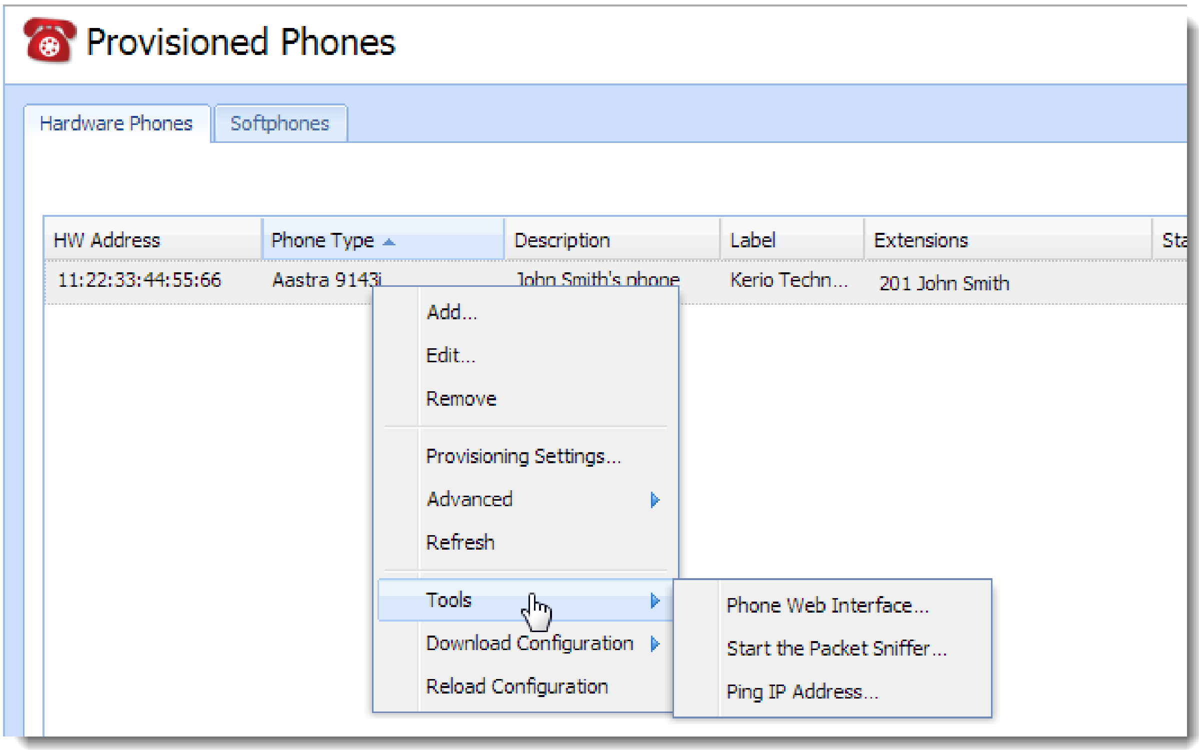
New in Kerio Operator 2.3!

Kerio Operator includes tools for phone administration. These tools can:

- display the phone web interface.
- open a packet sniffer for a communication between the phone and Kerio Operator.
- ping IP address of the phone.

Using provisioning tools

1. In the administration interface, go to **Provisioned Phones**.
2. Right-click a provisioned phone and in the context menu select **Tools**.
3. Select a tool and use it.



Accessing company contacts through LDAP on provisioned phones

LDAP configuration overview

Kerio Operator offers searching in your LDAP directory from your provisioned phones.

Cisco79xx phones are not supported.
Polycom phones are not supported with Kerio Connect LDAP.

Connecting to Kerio Connect LDAP/Microsoft Active Directory

1. In the administration interface, go to **Provisioned Phones**.
2. Click the **Provisioning Settings** button.
3. In the **Phone Provisioning Settings** dialog, select option **Directory configuration is enabled.**
4. Click **Configure.**
5. Click **Configuration Wizard.**
6. Select type of a service:
    - Kerio Connect LDAP — type Kerio Connect hostname, username and password.
    - Active Directory — type domain name and hostname of your Active Directory and credentials of account with at least read-only access to Active Directory,

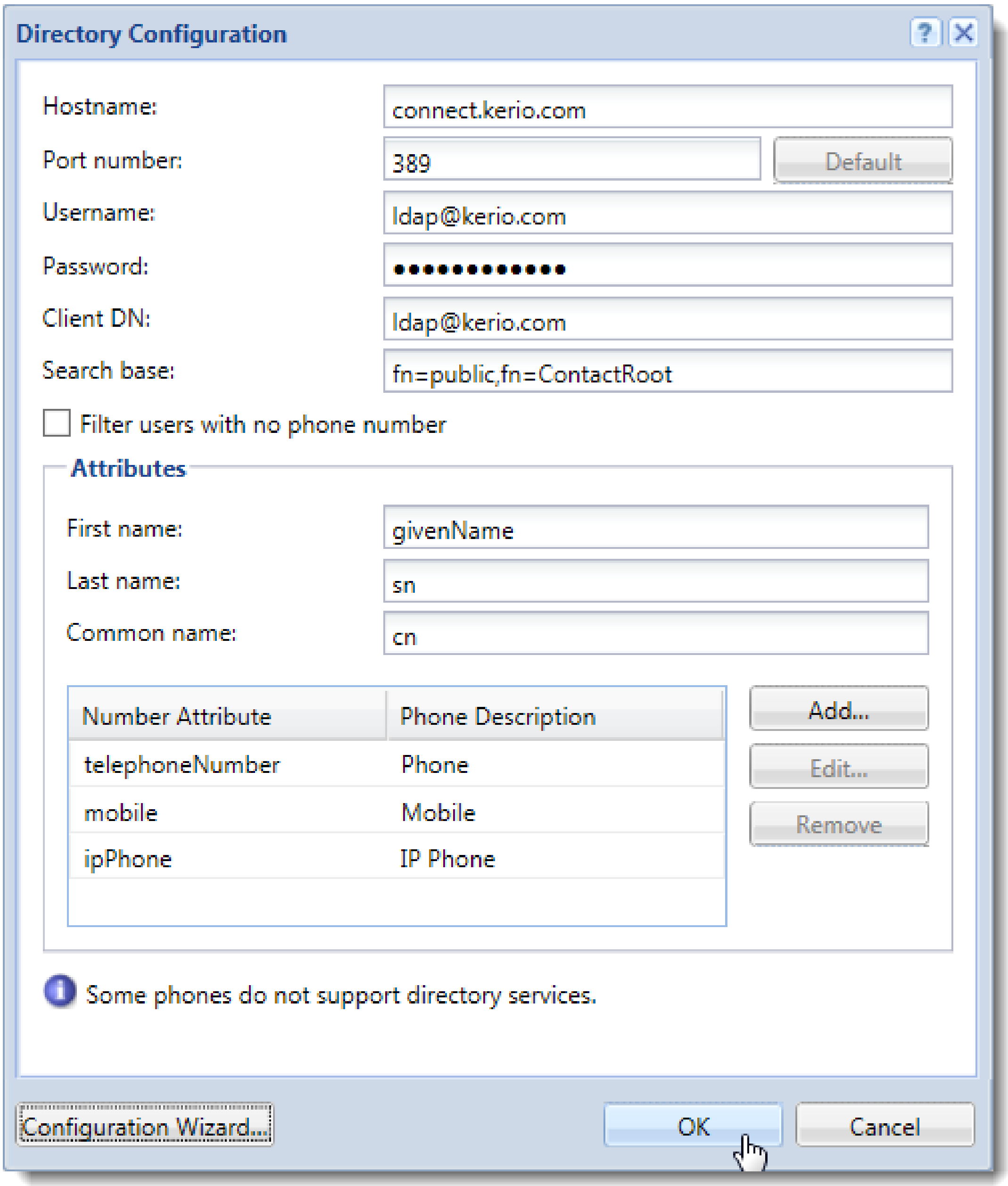


Figure 1 The Directory Configuration dialog after finishing Kerio Connect LDAP configuration

We recommend to create a special account with read-only access and use credentials of this account.

7. Save the settings.
8. In **Provisioned Phones**, click **Advanced** and restart all provisioned phones.

   Phones read the new configuration and start to communicate directly with the LDAP server.

Try this feature on your phone. Find a directory on the phone and check the contact list.

For information on how to use your phone directory, read the manual of your phone.

Connecting to LDAP in general

1. In the administration interface, go to **Provisioned Phones**.
2. Click the **Provisioning Settings** button.
3. In the **Phone Provisioning Settings** dialog, select option **Directory configuration is enabled.**
4. Click **Configure**.
5. Fill the **Directory Configuration** dialog.
6. Save the settings.
7. In **Provisioned Phones**, click **Advanced** and restart all provisioned phones.

   Phones read the new configuration and start to communicate directly with LDAP server.

Try this feature on your phone. Find a directory on the phone and check the contact list.

For information on how to use your phone directory, read the manual of your phone.

Displaying your company logo on the provisioned phones

Summary

You can display your company logo on hardware phones supported by Kerio Operator.

What you need

- Logo — each phone firmware needs a logo in a different format.
- Phones must be provisioned.

Which type of logo do you need

1. In the administration interface, go to **Provisioned Phones**.
2. Click the **Advanced** → **Logos** button.
3. In the **Firmwares and Logos** dialog, go to tab **Logos**.
4. Find the firmware type installed on your phones and click **Edit**.

   In **Notes**, you can find the logo parameters.

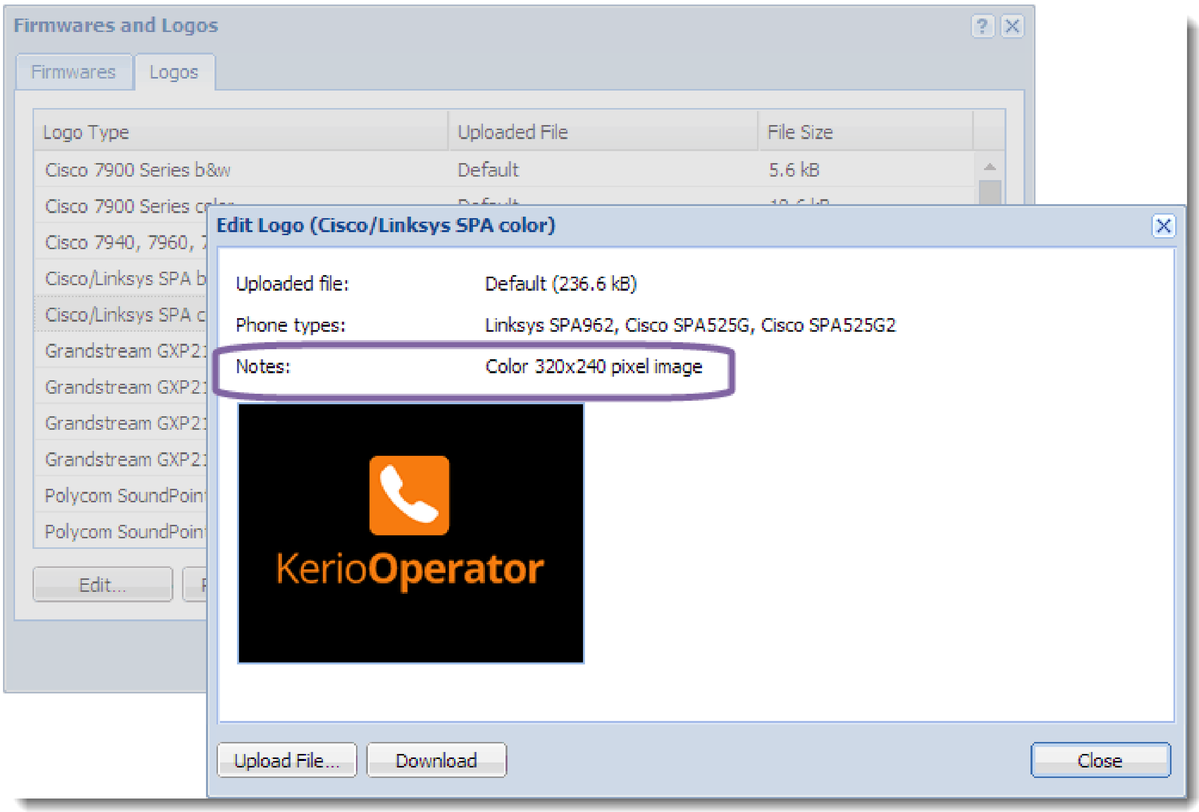


Figure 1 Logo parameters in the Edit Logo dialog

Adding your logo to phones

The Kerio Operator logo is set by default and you have to change it:

1. In the administration interface, go to **Provisioned Phones**.
2. Click the **Advanced** → **Logos** button.
3. Find the logo type for your phone and click **Edit.**
4. Click **Upload File** and upload your logo.
5. Close the dialog.
6. In **Provisioned Phones**, click the **Provisioning Settings** button.
7. In the **Phone Provisioning Settings** dialog, select **Display logo on the screen**.
8. Save the settings.
9. Restart all phones manually.

Provisioning of Kerio Operator Softphone for mobile devices

Softphone provisioning overview

Auto-provisioning and its functionality is described in a special article — Configuring automatic phone provisioning.

This article describes auto-provisioning of Kerio Operator Softphone for mobile devices.

Prerequisites

- Kerio Operator must have a DNS name. Type the DNS name in the **Configuration** → **Network** section.

  To secure your Kerio Operator Softphones on Android devices, you must have a fully qualified domain name in the SSL certificate of the Kerio Operator server.

- Kerio Operator must use a valid SSL certificate. The certificate name must correspond with the Kerio Operator DNS name.

  For more information, see Securing Kerio Operator Softphone with SSL certificates.

Configuring provisioning for Kerio Operator Softphone

First, to users who want to use Kerio Operator Softphone, add a new extension or a new registration of their existing extension.

Second, add users to provisioning:

1. In the administration interface, go to **Configuration** → **Provisioned Phones** → **Softphones**
2. Click **Add**.
3. In the **Select User** dialog, select the user who wants to use Kerio Operator Softphone.
4. Save the settings.

Third, users must configure their mobile devices and connect to Kerio Operator.

Securing Kerio Operator Softphone with SSL certificates

To secure your Kerio Operator Softphones, you must have one of the following SSL certificates:

- A paid SSL certificate signed by a certification authority.

  These certificates do not require any further configuration.

  > ⚠ Do not use wildcard certificates. Kerio Operator Softphone follows the RFC 5922 standard.

- A self-signed certificate created by your Kerio Operator server.

  If you use a self-signed certificate, users must download and install the certificate manually. For more information, see Using the self-signed certificate from your Kerio Operator server section in the **Configuring Kerio Operator Softphone** article.

Configuring a dial plan

Users with Kerio Operator Softphone want to use their contact list, where phone numbers are stored in different formats. The Dial Plan translates phone numbers from the format used in a user's contact lists to the format that can be dialed via your Kerio Operator PBX:

1. In the administration interface, go to **Provisioned Phones** → **Softphones**.
2. Click **Dial Plan Configuration**.
3. Click **Add** to create a new rule.
4. Save the rule and click **Test** in the **Dial Plan Configuration** dialog.
5. If you need more rules, create another one.
6. Sort rules from specific to general.

   Rules are applied from top to bottom.
7. Save the settings.

Creating rules

You can use the following characters when creating new rules.

Character	Description
0 to 9	digits
x	a single wildcard
*#+	Keyboard symbols
[ ]	A collection that can include a range. For example [6-9] means 6 7 8 9. Or [136-9] means 1 3 6 7 8 9.
.	Repeat the last element 0 or more times. For example, with the pattern "12." the following input will match: 1 (The "2" is repeated zero times) 12, 122, 1222 and so on

Table 1 Characters for your dial plan

Example 1: International calls from USA

Match number: `+x.`

Remove prefix: +

Add prefix: 011

The following image describes a scenario when you wan to call from the USA (prefix 011) to GB (prefix +44) and outgoing prefix of your company is 9.

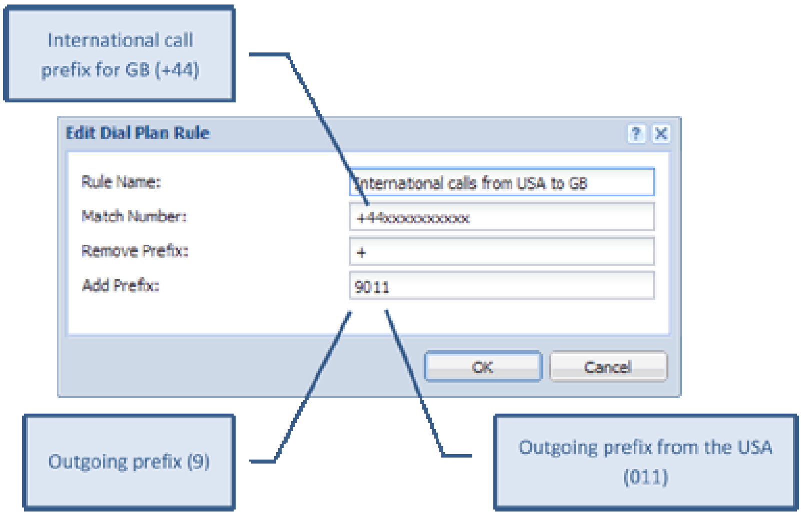


Figure 1 International calls from USA to GB

Example 2: Outgoing prefix 9

Match number: `x.`

Remove prefix: leave empty.

Add prefix: 9

Example 3: International calls in Europe (replacing "+" by "00")

Match number: `+x.`

Remove prefix: +

Add prefix: 00

Configuring parameter 66 in DHCP server in Kerio Control

What is parameter 66 in a DHCP server?

The DHCP protocol assigns IP addresses. Apart from these addresses you can also send additional parameters via the DHCP protocol. Parameter 66 configures the TFTP server address.

How to set parameter 66 in Kerio Control

1. In the administration interface, go to section **DHCP server**.
2. If you use the automatically generated scopes, use **Click to configure scopes manually**.
3. Select a scope and open its settings (the **Edit Scope** dialog).
4. Click **Add**.
5. Add parameter 66.
6. Type an IP address through which Kerio Operator communicates.

Uploading configuration files to Kerio Operator TFTP server

Why to use phone or other device configuration file

- phone provisioning of unsupported devices (hardware phones or other devices with a TFTP client)
- phone firmware upgrade
- BLF configuration, ring tones (different ring tones for different phones)
- password change for all extension assigned to one phone

Obtaining the configuration file

The following instructions will come in handy, if you wish to change the configuration file of a provisioned phone:

1. In the administration interface, go to **Configuration** → **Provisioned Phones**.
2. Right-click the phone whose configuration file you wish to download.

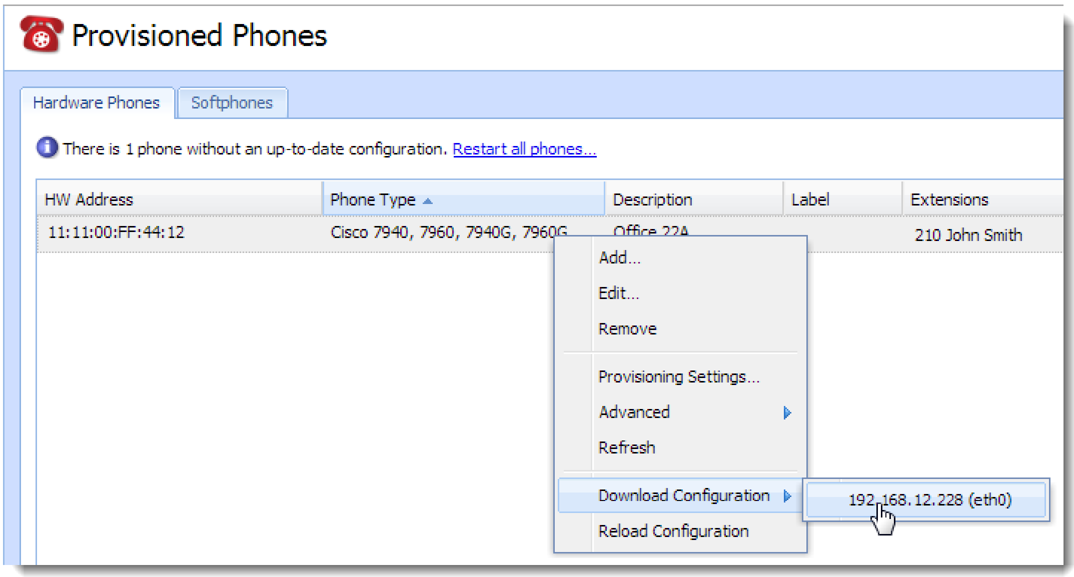


Figure 1 Downloading the configuration

3. Click **Download Configuration** and select the interface.

   Each interface has a different configuration — different IP addresses.

4. The ZIP file with the current configuration will be automatically saved on your computer.

Uploading new or changed configuration files to Kerio Operator

What you need

The file must be uploaded via SSH using SCP.

Locate configuration files to `/var/tftp`

How to enable SSH in Kerio Operator

Follow these instructions:

1. In the administration interface, go to section **Status** → **System Health.**

2. Click **Tasks** while pressing the **Shift** key.

3. Select **Enable SSH**.

4. Connect to Kerio Operator via SCP (use for example WinSCP for Windows) and upload the file via SSH using SCP.

   For access use username `root` and password of a Kerio Operator administrator.

Connecting to VoIP service providers

Overview

Redesigned in Kerio Operator 2.4!

You can connect Kerio Operator to your VoIP service provider's SIP server or to a standard phone network. This article deals with the first option: connecting to a VoIP service provider.

Prerequisites

Before you configure an interface, you need to know:

- Telephone number (or numbers) from your SIP provider
- Domain/hostname of SIP server
- Username and password for authentication
- At least one internal extension defined in Kerio Operator — preferably the extension of an employee who redirects the calls

Adding an interface

To configure an interface, you must first configure call routing. Once you configure incoming call routing, a configuration wizard configures outgoing call routing automatically.

1. In the administration interface, go to **Configuration** → **Call Routing** and click **Add SIP interface**. This opens the configuration wizard.
2. Type a name for the interface (for example, the provider's name).

   The name must not contain spaces or special characters and must be unique.
3. Select **New provider**.

   The configuration differs for settings with one number or multiple numbers and for a SIP trunk with an interval of phone numbers.

One or multiple numbers

1. If you acquire one or multiple phone numbers from your provider, type the numbers in the **New provider** → **With external number** field. You can:

    - Separate individual numbers with commas (for example, 555450, 555451, 555452, and so on)

    - Type the whole range using a dash (for example, 555450-555459)

2. Click **Next**.

3. Select an extension that receives all calls from the provider.

4. (Optional) In the **Prefix to dial out** field, type a prefix for outgoing calls and click **Next**.

    Kerio Operator uses the prefix to route calls to your provider's SIP server. This prefix can be the same for other providers. See Working with prefixes for outgoing calls.

5. Type the domain name or the IP address acquired from your provider.

    Type the username and password if the server requires authentication.

6. Select **Required to register** (the majority of providers require registration to a SIP server) and click **Next**.

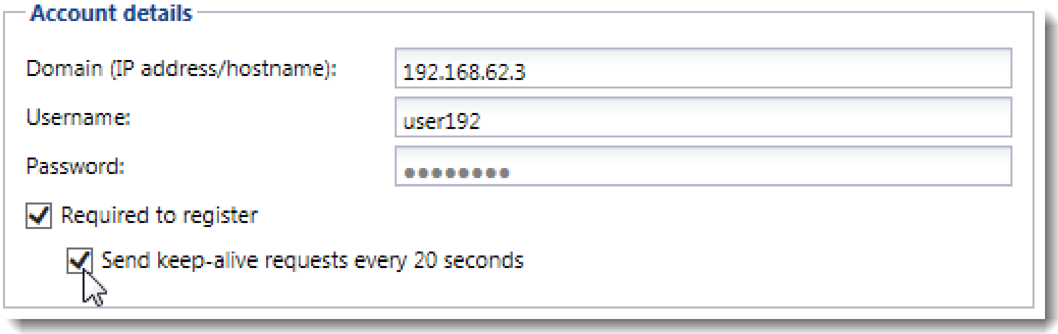
7. Verify the information in the **Summary** section.

    If you need to add more information from your provider (for example, outbound proxy, inbound proxy, registrar, and so on), select the **Edit details of the created interface** option. For more information, see the Configuring additional SIP details section.

8. Click **Finish**.

9. (Optional) Double-click the interface and enable the **Send keep-alive requests every 20 seconds** option.

    If your SIP provider does not send keep-alive packets, or your firewall or router has short and unchangeable NAT timeout for UDP connections, use this option to keep the UDP session open.



10. Click **OK** to save your changes.

11. Create a rewriting rule to correctly map numbers to internal user extensions.

Interval of numbers

1. If you acquire a SIP trunk with an interval of numbers from your provider, type an x in place of the digits that vary (for example, 555xxx).

2. Click **Next**.

3. Select the extension to which you want Kerio Operator to redirect all calls to unassigned (unused) extensions.

4. (Optional) In the **Prefix to dial out** field, type a prefix for outgoing calls.

   Kerio Operator uses the prefix to route calls to your provider's SIP server. This prefix can be the same for other providers. See Working with prefixes for outgoing calls.

5. Click **Next**.

6. Type the domain name or the IP address acquired from your provider.

   Type the username and password if the server requires authentication.

7. Select the **Required to register** option if the provider requires registration. With large number intervals, some providers do not require registration. Instead they use the IP address of your Kerio Operator. The address must be static and the provider needs to know about any changes that may occur.

8. Verify the information in the **Summary** section.

   If you need to add more information from your provider (for example, outbound proxy, inbound proxy, registrar, and so on), select the **Edit SIP details of created interface** option. For more information, see the Configuring additional SIP details section.

9. Click **Finish**.

10. Create a rewriting rule to correctly map numbers to internal user extensions.

Configuring additional SIP details

To set additional settings in your interface for incoming and outgoing calls:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**, select a SIP interface and click **Edit.**

2. On the **SIP Details** tab, you can:

   - Type addresses to outbound proxy, inbound proxy and registrar (Kerio Operator uses domain by default)

   - Change the transport protocol

   - Change the DTMF method

   - Type an authentication username (Kerio Operator uses the SIP username by default)

   - Change outgoing headers

3. Click **OK** to save your changes.

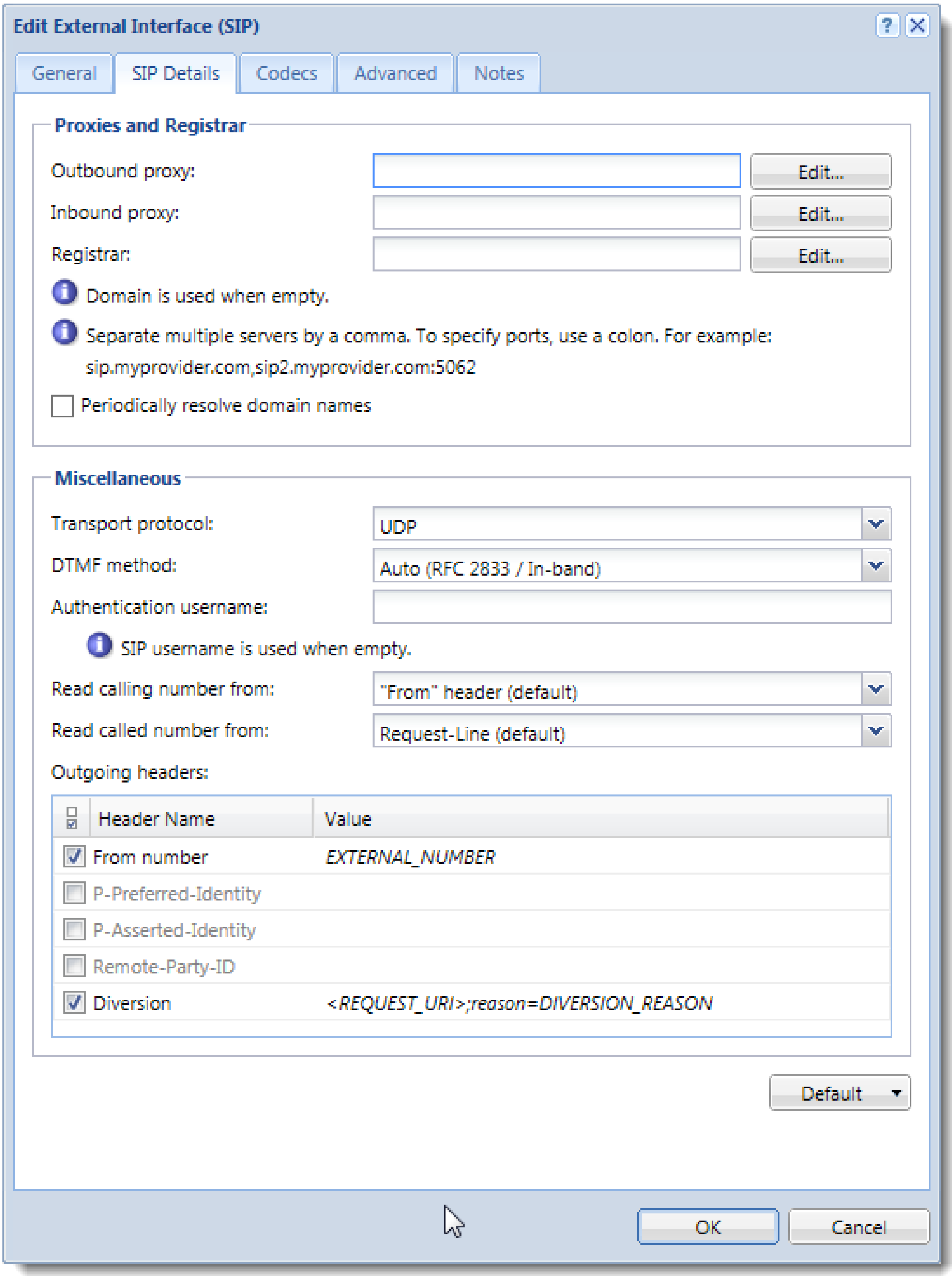

Edit External Interface (SIP)
General
SIP Details
Codecs
Advanced
Notes
Proxies and Registrar
Outbound proxy:
Edit...
Inbound proxy:
Edit...
Registrar:
Edit...
Domain is used when empty.
Separate multiple servers by a comma. To specify ports, use a colon. For example:
sip.myprovider.com,sip2.myprovider.com:5062
Periodically resolve domain names
Miscellaneous
Transport protocol:
UDP
DTMF method:
Auto (RFC 2833 / In-band)
Authentication username:
SIP username is used when empty.
Read calling number from:
"From" header (default)
Read called number from:
Request-Line (default)
Outgoing headers:
Header Name
Value
From number
EXTERNAL_NUMBER
P-Preferred-Identity
P-Asserted-Identity
Remote-Party-ID
Diversion
<REQUEST_URI>;reason=DIVERSION_REASON
Default
OK
Cancel

Figure 1

Configuring DTMF method

New in Kerio Operator 2.4!

For some SIP providers, the default configuration of DTMF detection, **Auto (RFC 2833 / In-band)**, does not work. You must find out the correct method from your SIP provider and configure it manually, as follows:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.
3. Go to the **SIP Details** tab.
4. Select the correct **DTMF method**.
5. Click **OK**.

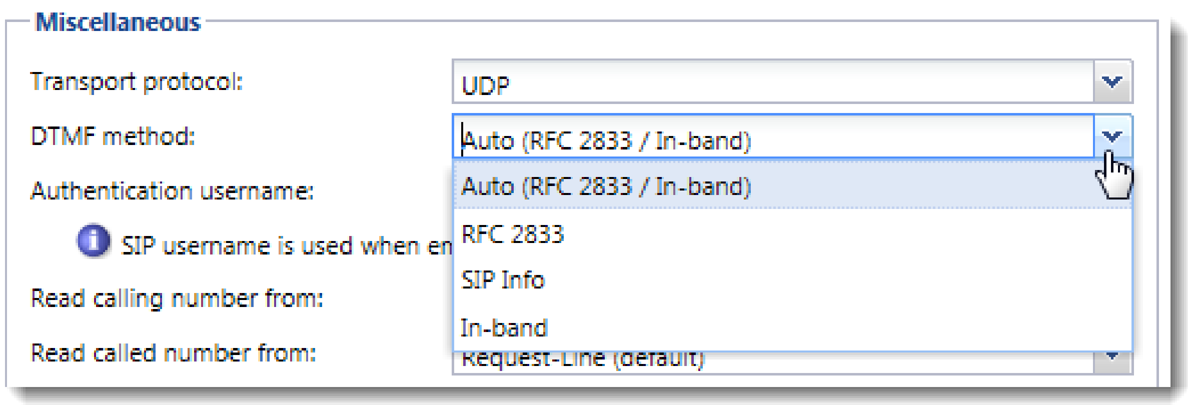


Figure 2

Configuring outgoing headers

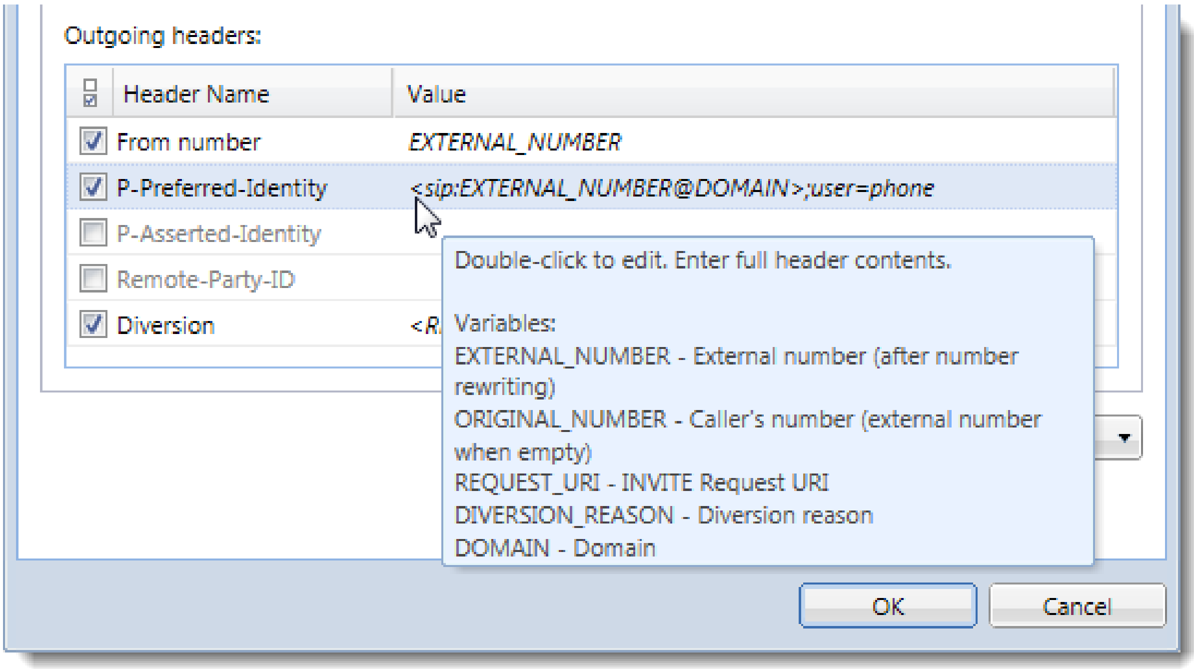
New in Kerio Operator 2.4!

For some providers, you must add additional configuration to the SIP headers they provide.

To configure outgoing headers:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.

3. Go to the **SIP Details** tab.
4. Enable the outgoing header (see the list of supported headers below).
5. Double-click in the **Value** column and type the header content (see the list of supported variables below).
6. Click **OK**.



Kerio Operator supports these headers:

- From number
- P-Preferred-Identity
- P-Asserted-Identity
- Remote-Party-ID
- Diversion

To edit headers, use these variables:

- `EXTERNAL_NUMBER` shows the external number after number rewriting
- `ORIGINAL_NUMBER` shows the number of the caller
- `REQUEST_URI` requests the information from the header of the forwarded call

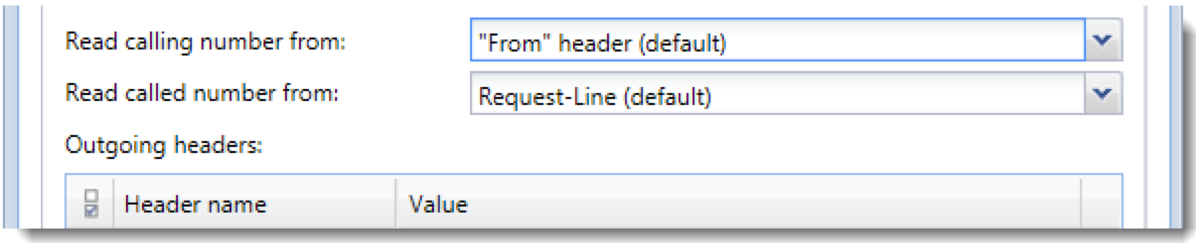
- `DIVERSION_REASON` sends the reason of the call forwarding
- `DOMAIN` shows the domain of the interface

Reading the Caller ID from outgoing headers

New in Kerio Operator 2.4.4!

If your provider does not send the information about calling or called numbers in default headers (**From** for calling number and **Request-Line** for called numbers), you can configure Kerio Operator to read this information from different headers (for example, **P-Asserted-Identity**):

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.
3. Go to the **SIP Details** tab.
4. For the fields **Read calling number from** and **Read called number from**, select a new header.
5. Click **OK** to save your settings.



Displaying the caller's number when transferring and redirecting calls

For more information, see Displaying the caller number when transferring and redirecting calls.

Resolving domain names of SIP providers

Your SIP providers may change their IP address for your registration without prior notice. To avoid inaccessibility, configure Kerio Operator to periodically resolve domain names and renew the registration:

1. In the administration interface, go to **Call Routing** → **Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.
3. Go to the **SIP Details** tab.
4. Select the **Periodically resolve domain names** option.
5. Click **OK**.

Kerio Operator now periodically resolves domain names of your SIP provider and renews your registration whenever the IP address changes.

Mapping of numbers

See the Mapping external and internal numbers article for more information.

Mapping external and internal numbers

Overview

Redesigned in Kerio Operator 2.4!

In Kerio Operator you can map external numbers to internal extensions. You can:

- Strip the first `0-n` digits from the number, including reducing the number to an empty string
- Add other digits to the beginning of the number

Routing incoming calls

In Kerio Operator, you can use rewriting rules to map numbers for SIP and standard phone interfaces. Depending on your provider's requirements, you may need to strip out or change numbers

Example:

- A company has 100 phone numbers from a telephone provider.
- For incoming calls, the provider sends the whole number.
- For outgoing calls, the provider requires the whole number.
- Internal extensions have the format `2xx`.
- The prefix for outgoing calls is `9`.

When external **Phone A** (with the number **5550399**) calls internal **Phone B** (with the number **5550101** and the internal extension **201**):

1. **Phone A** dials **Phone B's** number and a signal goes to the provider.
2. The provider sends the number to Kerio Operator.
3. The rewriting rule strips five digits from the left and adds the prefix `2`.
4. The call connects.

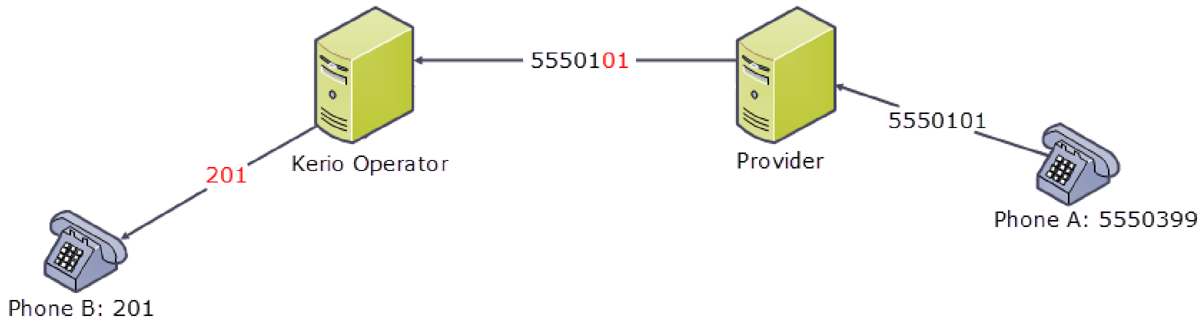


Figure 1

Mapping a trunk of numbers

To set the interface for an interval of numbers (55501xx in this example):

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of outgoing calls**.
2. Select the routing rule for the provider interface and click **Edit**.

   The **Edit Incoming Call** dialog box opens.

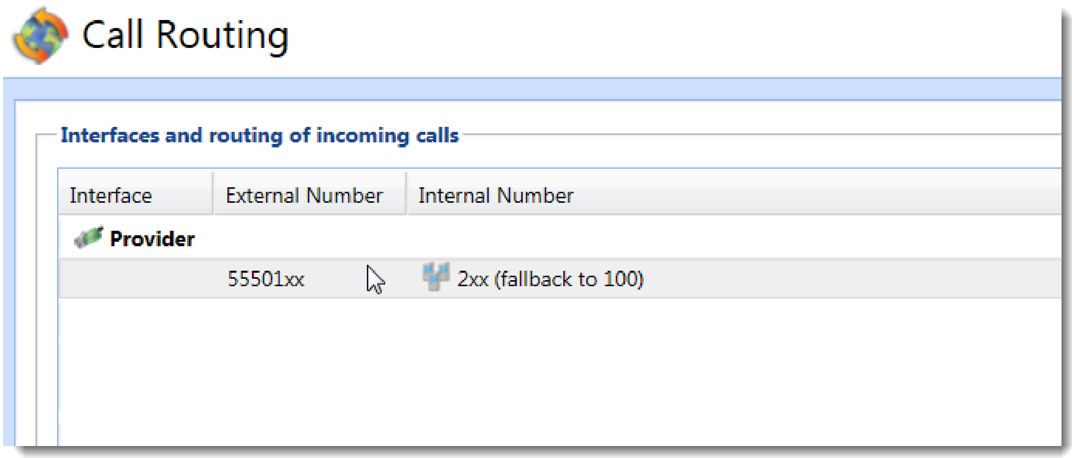


Figure 2

3. In the **Called number** section, strip the first five digits from the left, and add the prefix 2.

   This modifies the number to the final format of the extension (2xx).
4. In the **Calling number** section, do not strip out any digits, and add the prefix 9.

   This is useful when you want to call back the external number.
5. Click **OK**.

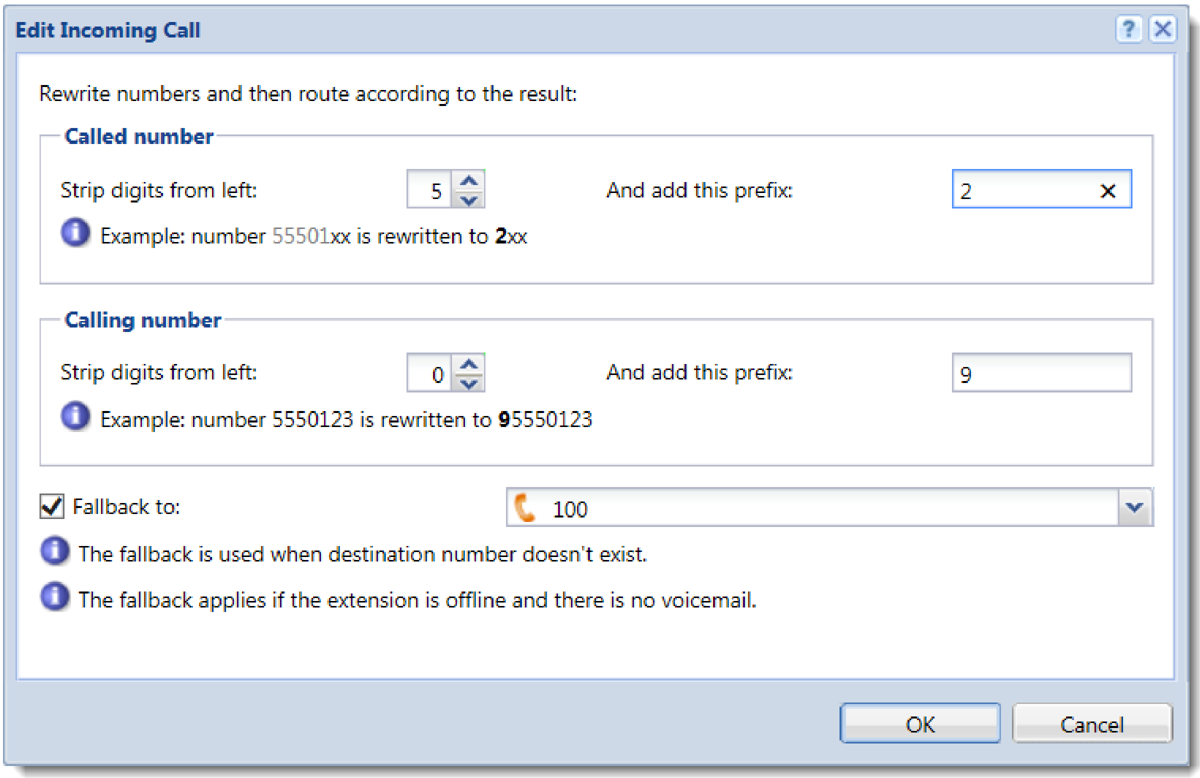


Figure 3

Mapping a single number or multiple numbers

To set the interface for single or multiple numbers (`5550100 to 5550199` in this example):

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls.**

2. Select the routing rule for the provider interface and click **Edit**.

   The **Edit Incoming Call** dialog box opens.

3. Double-click a line in the **Extension** column and assign an extension to the external number.

4. In the **Called number** section, strip the first two digits from the left, and add the prefix 2.

   This modifies the number to the final format of the extension (`2xx`).

5. Click **OK**.

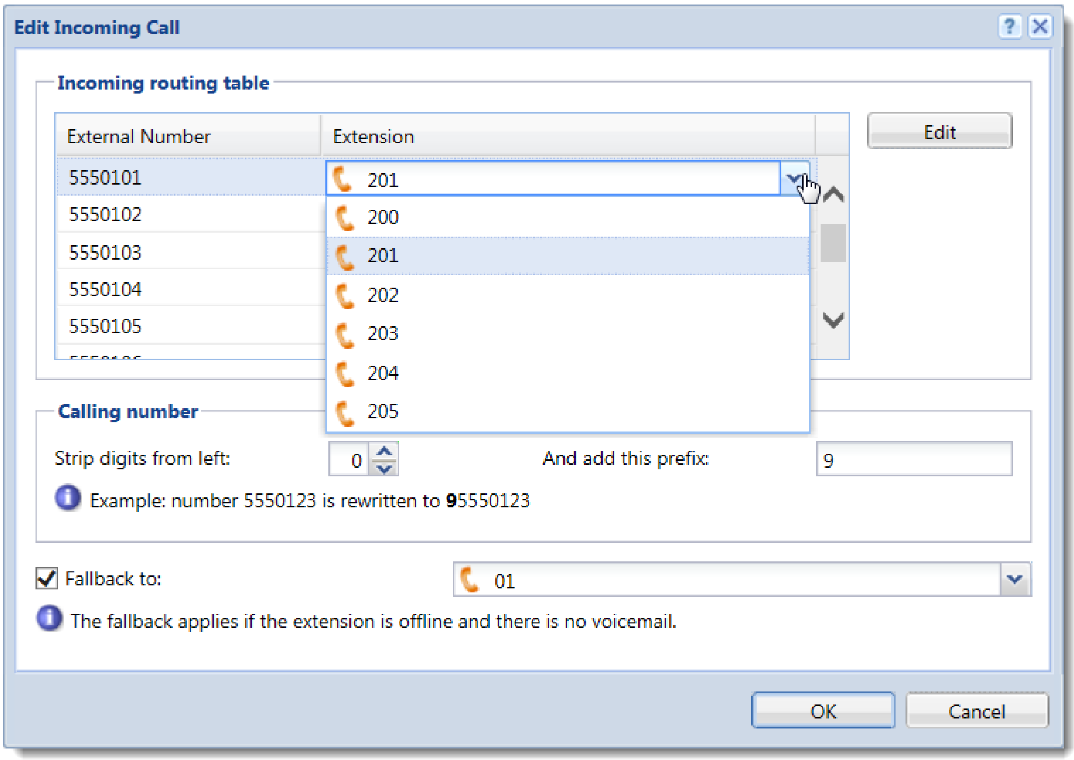


Figure 4

Routing outgoing calls

You can configure outgoing calls when creating an interface, either SIP or hardware cards. For rewriting the numbers, you need additional configuration.

Example:

- External **Phone A** has the number **5550199**.
- Internal **Phone B** has the number **5550101** and the internal extension **201**.
- For outgoing calls, Kerio Operator uses the prefix **9**.
- The provider needs the whole number for outgoing calls.

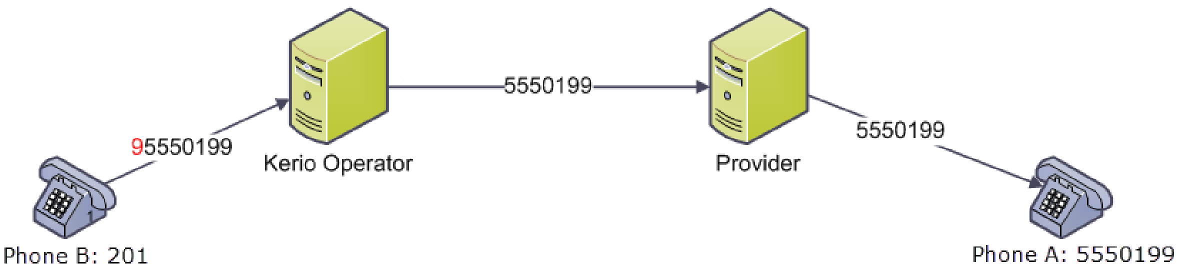


Figure 5

When **Phone B** calls **Phone A**:

1. **Phone B** dials the number with the **9** prefix (**95550199**).
2. Kerio Operator uses rewriting rules and strips out the first digit (**9**).

   The number Kerio Operator sends to the provider is **5550199**.
3. The provider connects to **Phone A**.

To achieve this configuration:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Routing of outgoing calls**.
2. Select an interface and click **Edit**.

   The **Edit Outgoing Route** dialog box opens.
3. In the **Called number** section, strip one digit from left and do not add a prefix.
4. Click **OK**.

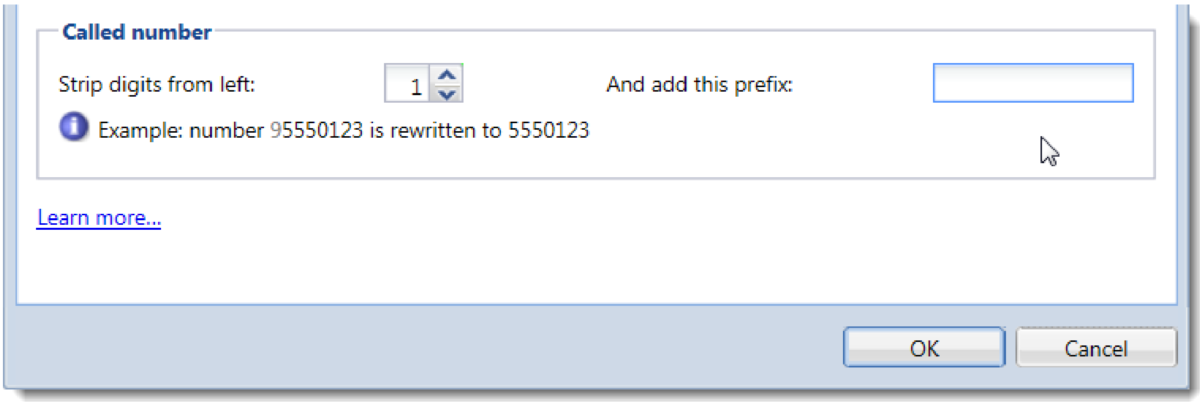


Figure 6

Rules for outgoing calls

You can configure rules for outgoing calls:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Routing of outgoing calls**.
2. Select an interface and click **Edit**.
3. In the **Calling number (Caller ID)** section, select one of these options:

- **Map extensions to external numbers based on routing of incoming calls** if you want to use a table of external numbers configured for the provider
- **Assign the default number to all extensions** if you want to use a default number for all extensions
- **Rewrite extension numbers (default number not used)** if you want to rewrite numbers in a specific way

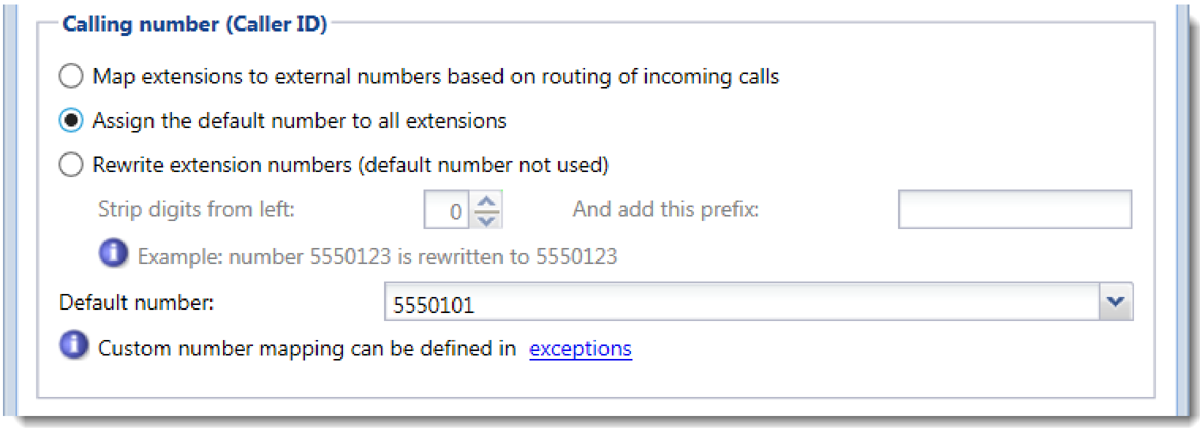


Figure 7

Exceptions to the outgoing routes

To create an exception:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Routing of outgoing calls**.
2. Select an interface and click **Edit**.
3. Enable the **Use route only for numbers defined in exceptions** option.
4. Click the **Exceptions** tab and click **Add**.
5. To change the internal number, double-click the displayed extension and select a new extension.
6. To change the external number, double-click the displayed number and select a new number.
7. If you want to hide this extension's number so the call recipient cannot see it, select the box in the **Hide Caller ID** column (see Displaying, hiding and overriding phone numbers for more details).
8. Click **OK**.

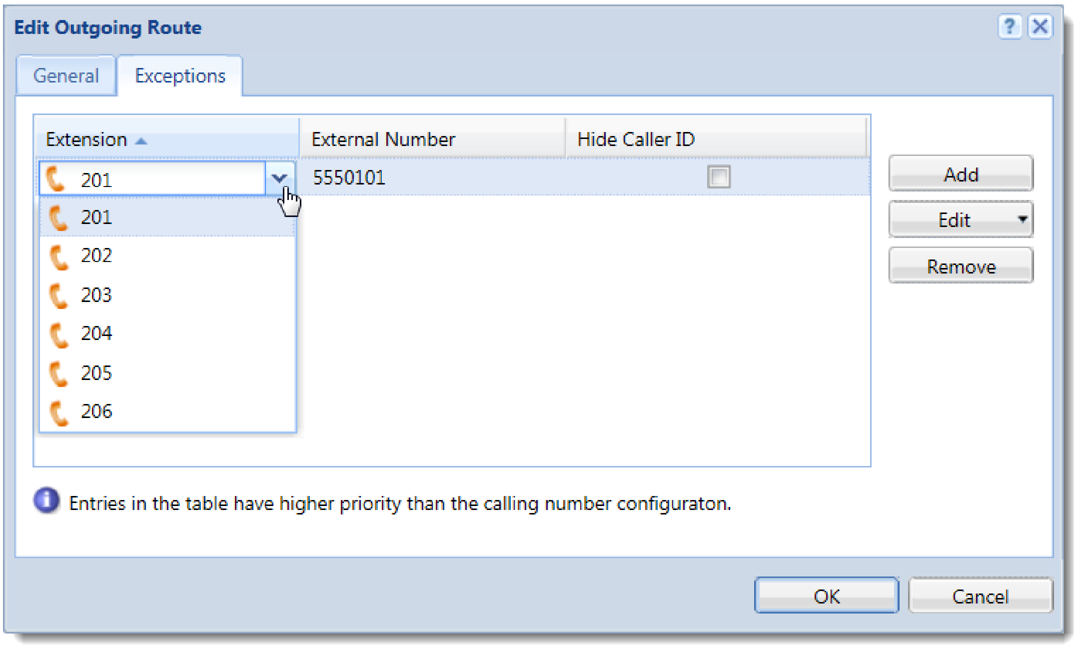


Figure 8

Working with prefixes for outgoing calls

New in Kerio Operator 2.4!

Kerio Operator works with prefixes for outgoing calls in a specific schema and you can use one prefix for multiple providers. Kerio Operator uses the longest prefix matching the dialed number. If that dial attempt fails, Kerio Operator tries the next route with the same prefix.

Example

- Use the prefix **011** for two providers (**provider1** and **provider2**) and the prefix **0** for outgoing calls.
- Dials the number **011 234 567**.

After dialing this number:

1. Kerio Operator goes through the **Routing of outgoing calls** table and tries to match the prefix.
2. Kerio Operator finds two matching prefixes, **0** and **011**, and uses the longest prefix.
3. Kerio Operator tries the **011** prefix to connect to **provider2**.

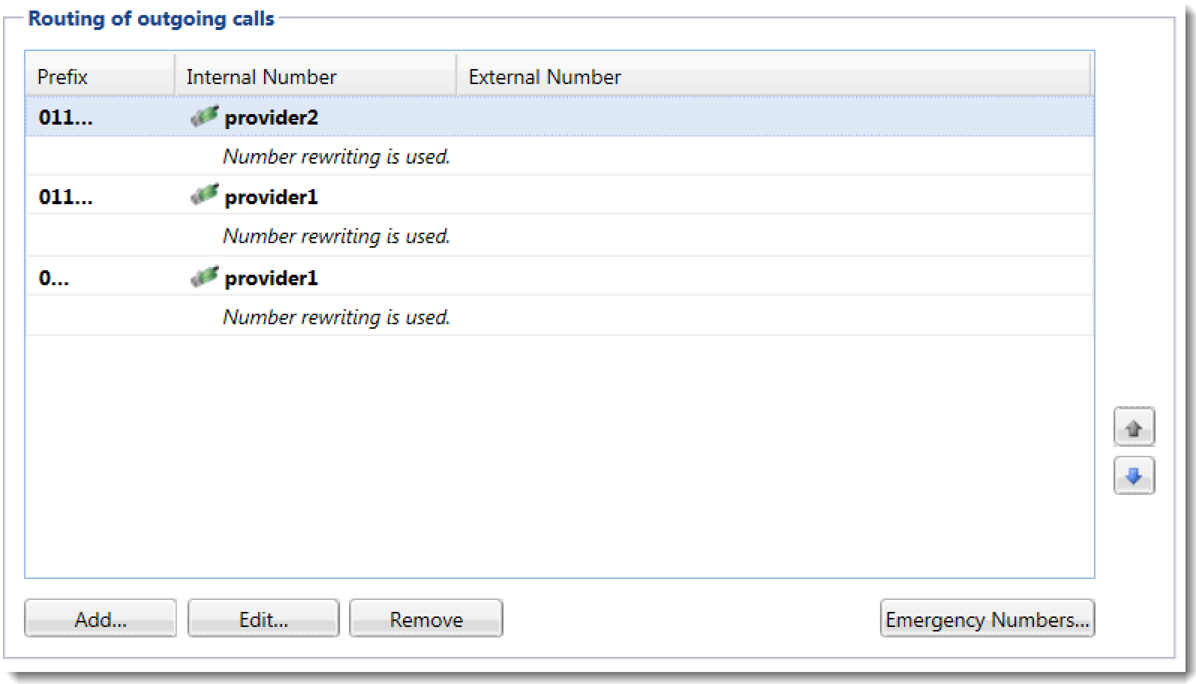


Figure 9

4. If the connection does not work, Kerio Operator uses the same prefix to connect to **provider1**.

5. If the connection still does not work, Kerio Operator does not try to use the last prefix (in this example, the **0** prefix), and the call fails.

Changing the order of prefixes

New in Kerio Operator 2.4!

Kerio Operator works with providers for the same prefix in order from top to bottom. You can change that order by the using arrows on the right side of the administration interface to move it up or down.

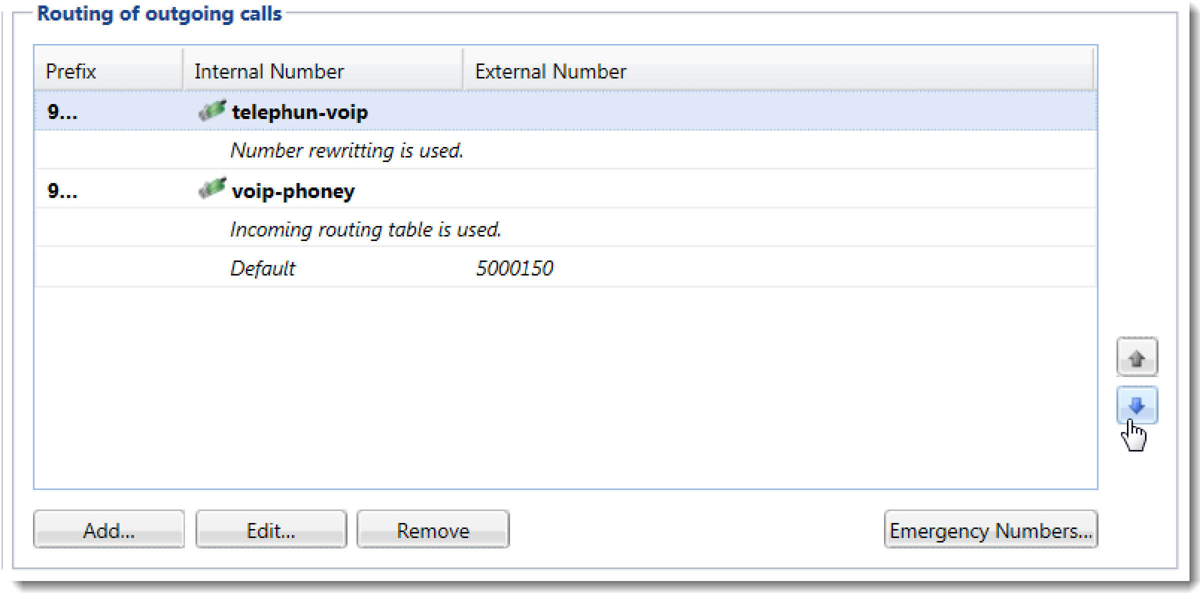

Routing of outgoing calls
Prefix
Internal Number
External Number
9...
telephun-voip
Number rewritting is used.
9...
voip-phoney
Incoming routing table is used.
Default
5000150
Add...
Edit...
Remove
Emergency Numbers...

Figure 10

Using Opus codec for Kerio Phone

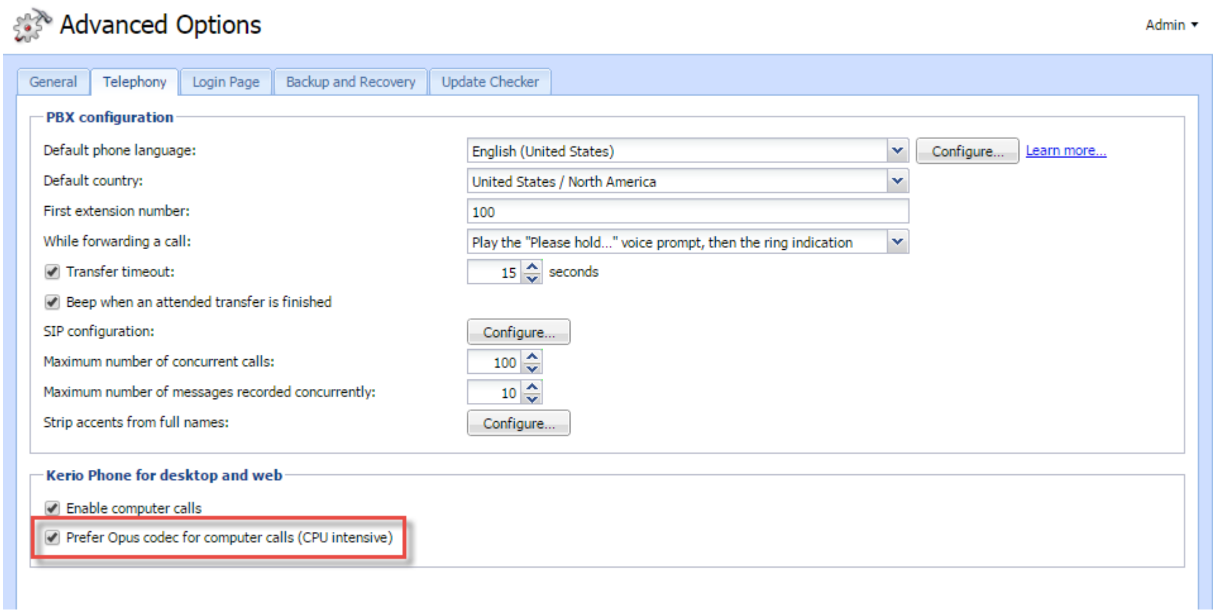
Overview

New in Kerio Operator 2.5!

Kerio Operator allows you to use the Opus codec for calls via Kerio Phone for desktop and web. To use Opus for all your calls:

1. In the Kerio Operator administration interface, go to **Advanced Options** → **Telephony**.
2. In the **Codec configuration** section, select the **Prefer Opus codec** option.
3. Click **Apply**.

Kerio Operator transcodes Opus to another codec every time the other caller doesn't use it. Transcoding calls increases the CPU usage. If you expect larger amount of concurrent calls, disable this option.



Disabling computer calls for Kerio Phone

Overview

New in Kerio Operator 2.5.2!

Kerio Operator enables you to make calls via Kerio Phone for desktop and web using WebRTC.

You can disable the WebRTC support in the administration, so that users cannot see and use the **Computer** extension in their applications.

You can disable computer calls for:

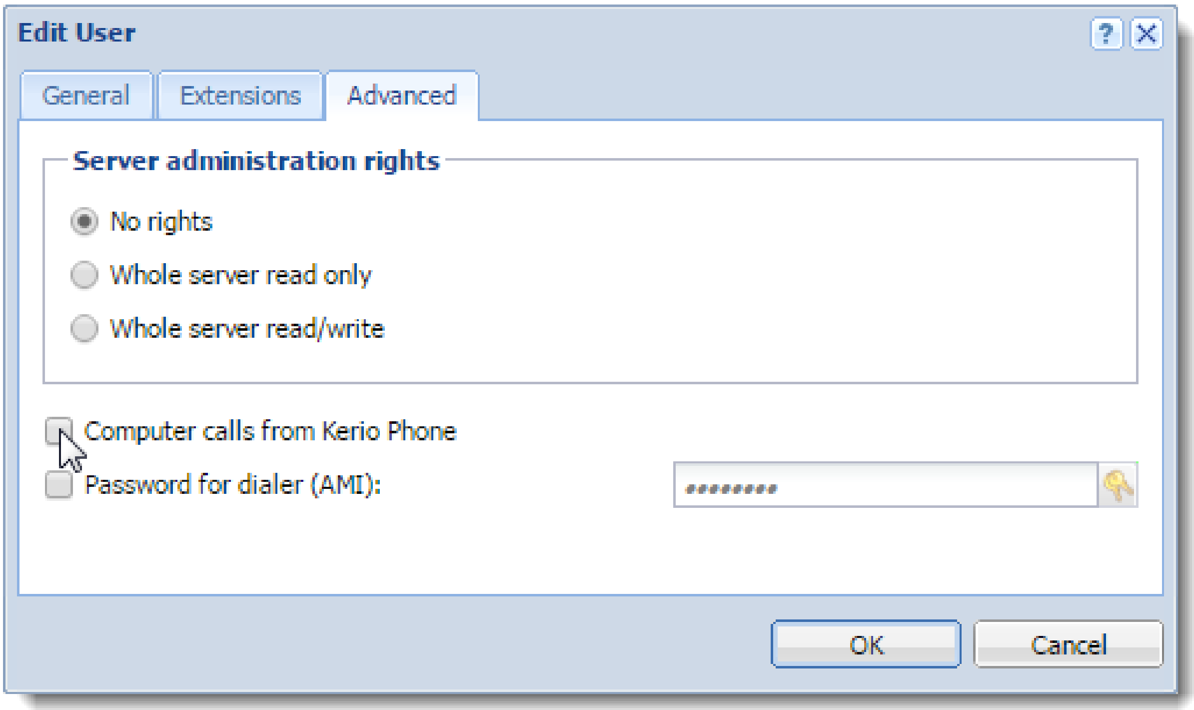
- A single user
- Multiple users
- All users on your Kerio Operator Server

Disabling computer calls for a single user

1. In the administration interface, go to **Configuration** → **Users**.
2. Select a user and click **Edit**.
3. Switch to the **Advanced** tab.
4. Deselect the **Computer calls from Kerio Phone** option.
5. Click **OK**.

At this point, a selected user can no longer use the **Computer** extension for making calls.

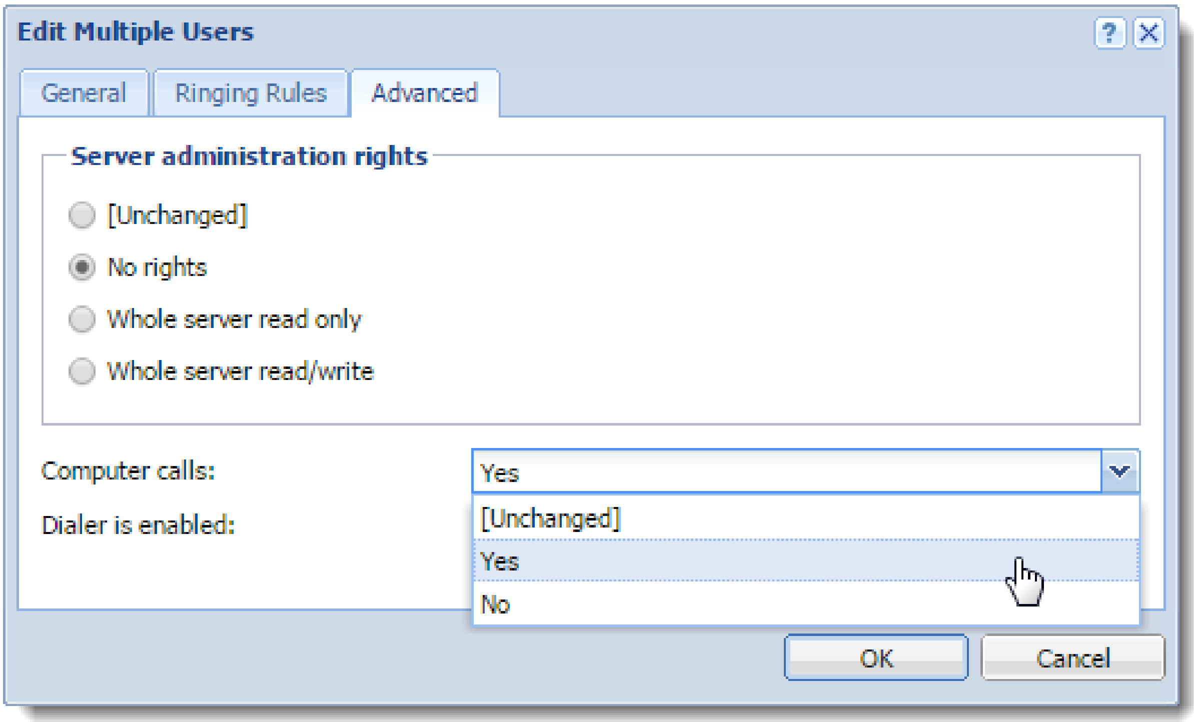
Disabling computer calls for Kerio Phone



Disabling computer calls for multiple users

1. In the administration interface, go to **Configuration** → **Users**.
2. Select multiple users and click Edit.
3. Switch to the **Advanced** tab.
4. In the **Computer calls** field, select **No**.
5. Click **OK**.

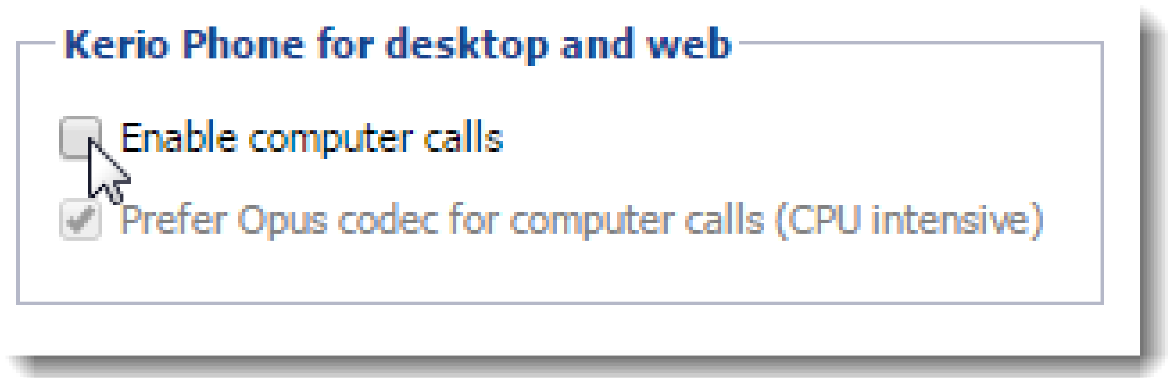
At this point, selected users can no longer use the **Computer** extension for making calls.



Disabling calls for all users

1. In the administration interface, go to **Configuration** → **Advanced Options**.
2. Switch to the **Telephony** tab.
3. In the **Kerio Phone for desktop and web** section, deselect the **Enable computer calls** option.
4. Click **Apply**.

At this point, users on your Kerio Operator server can no longer use the **Computer** extension for making calls.



Connecting multiple Kerio Operators

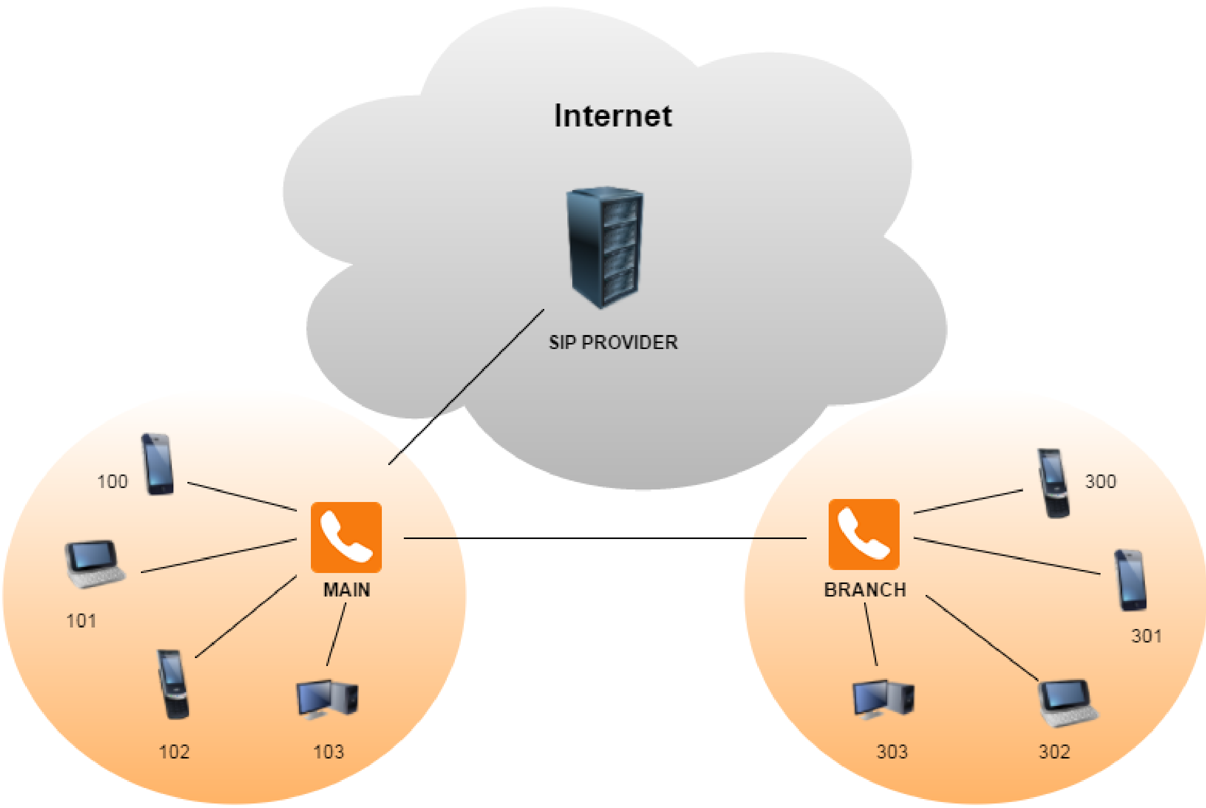
Overview

Redesigned in Kerio Operator 2.4!

In Kerio Operator, you can connect multiple Kerio Operator servers. This enables you to directly reach remote phones by their extensions for free and send or receive external calls through a relay server.

The section below describes how to connect these two servers:

- The main server, which has internal extensions 100 — 199
- The branch server, which has internal extensions 300 — 399



For more information about routing of calls between Kerio Operator servers and the PSTN, see Routing calls between multiple Kerio Operators and the PSTN.

Prerequisites

Before the start of the configuration, you need:

- Two Kerio Operator servers up and running
- Extension schemes for both phone networks, each of which has a unique set of extensions
- Both servers with public IP addresses or connected to the same network with a VPN tunnel

Connecting servers

On each Kerio Operator server, add a SIP interface for the other server.

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**.
2. Click **Add SIP interface**.

   The **Add SIP Interface** dialog box opens.
3. Type a name for the interface and select **Link to another PBX (without an external number)**.
4. Click **Next**.
5. In the **Prefix to reach the other PBX** field, type the appropriate number:
   - On the main server, type the prefix 3 (the first digit of each extension on the branch server)
   - On the branch server, type the prefix 1 (the first digit of each extension on the main server)
6. Click **Next**.
7. In the **Domain (IP address/hostname)** field, type the domain or the IP address:
   - On the main server, type the IP address of the branch server.
   - On the branch server, type the IP address of the main server.

8. Disable the **Required to register** option.

9. Click **Next**.

10. Verify the information in the **Summary** section.

11. Click **Finish**.

After the configuration of interfaces, Kerio Operator creates incoming and outgoing routes that use configured prefixes. These routes do not rewrite any numbers. Make test calls between the connected servers to reach their extensions.

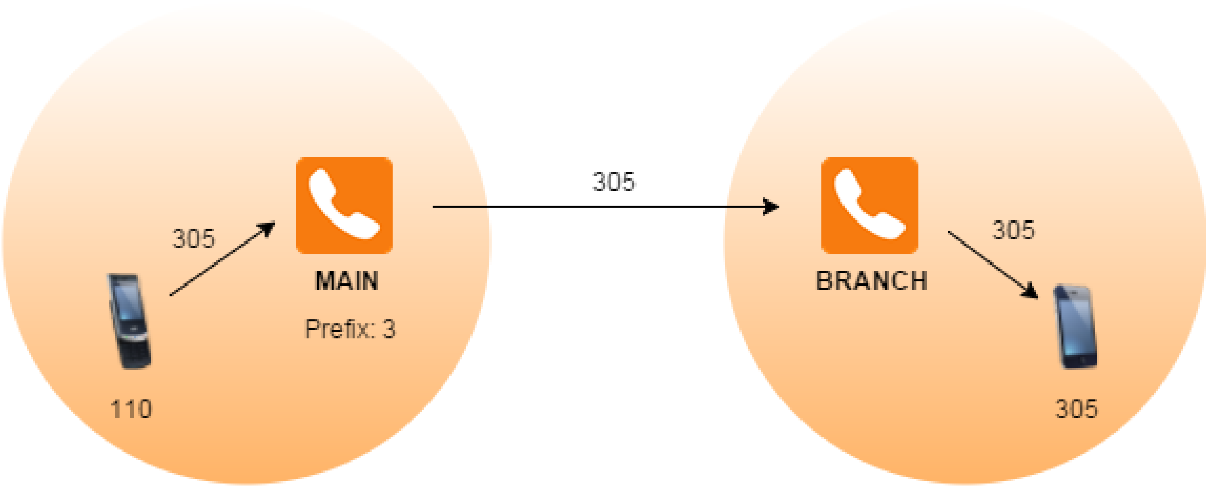
Example of a test call

Call number 305 from extension 110 on the main server:

1. The user with an extension 110 dials number 305.

2. Kerio Operator on the main server recognizes the prefix 3 and routes the call to the branch server.

3. The call arrives at the branch server and rings on the 305 extension.



Routing calls between multiple Kerio Operators and the PSTN

Overview

Redesigned in Kerio Operator 2.4!

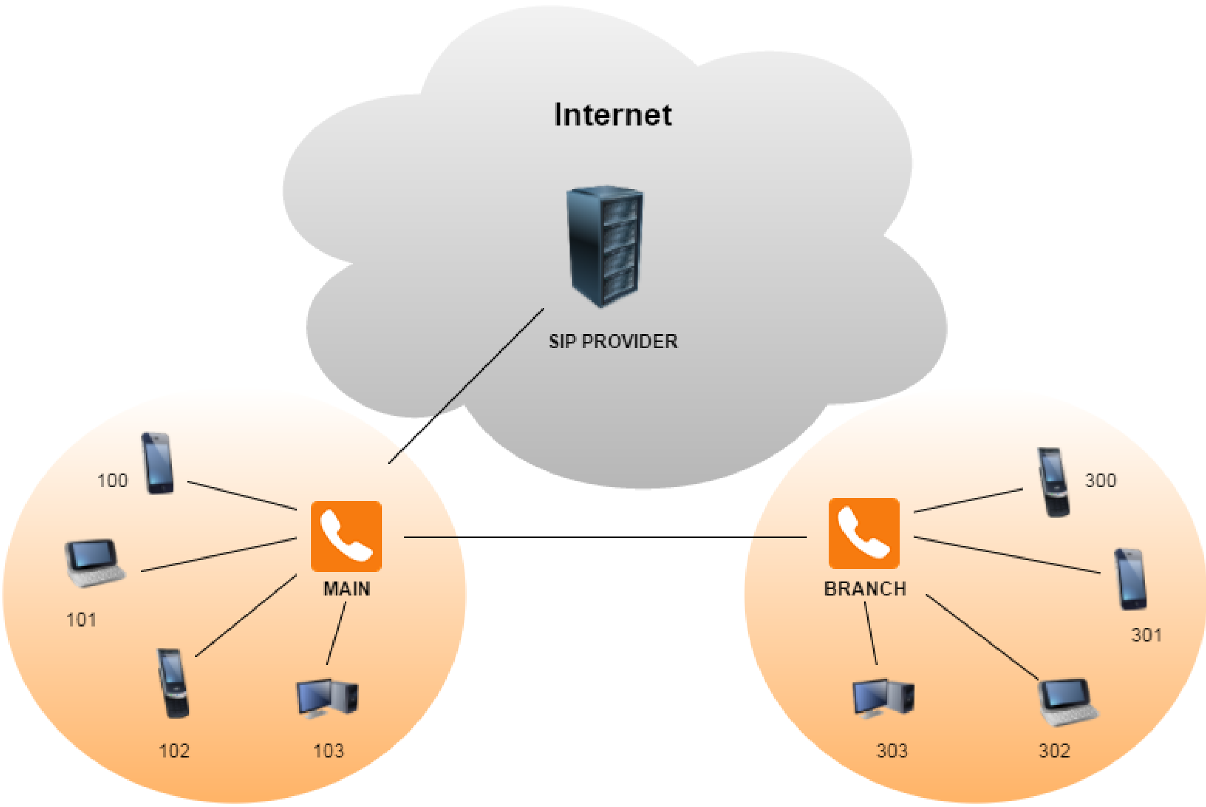
This article describes how to:

- Reach the public switched telephone network (PSTN) from your connected Kerio Operator servers
- Route incoming calls from the PSTN to your branch servers

For more information about connecting multiple Kerio Operator servers, see Connecting multiple Kerio Operators.

The sections below use the following example:

- Two connected Kerio Operator servers up and running:
  - The main server, which has internal extensions 100 — 199
  - The branch server, which has internal extensions 300 — 399
- Outgoing calls from the branch server to the PSTN go through the main server.
- Incoming calls from the PSTN to the branch server go through the main server.
- The prefix for outgoing calls to the PSTN is 0.
- External numbers from the SIP provider have the format 555 5xxx.



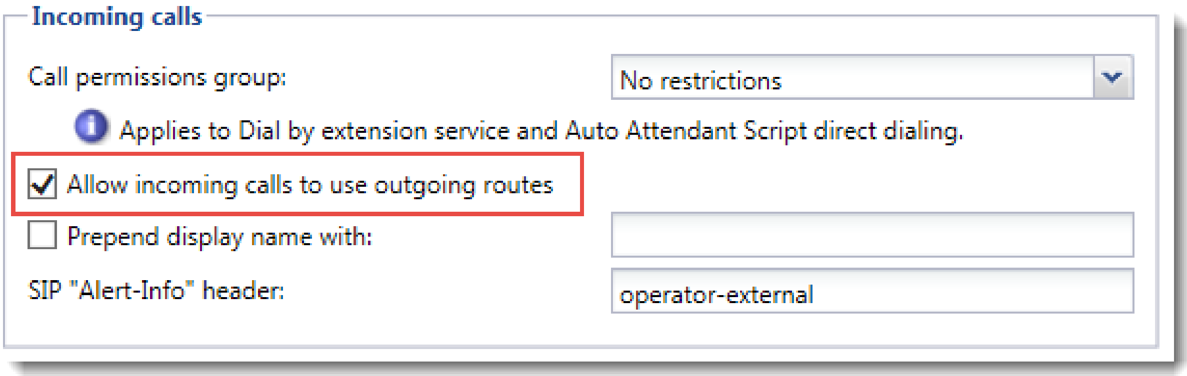
Calling to the PSTN through the main server

To call to the PSTN via the interface of the main server:

- Configure the interface on the main server.
- Create an outgoing route on the branch server.

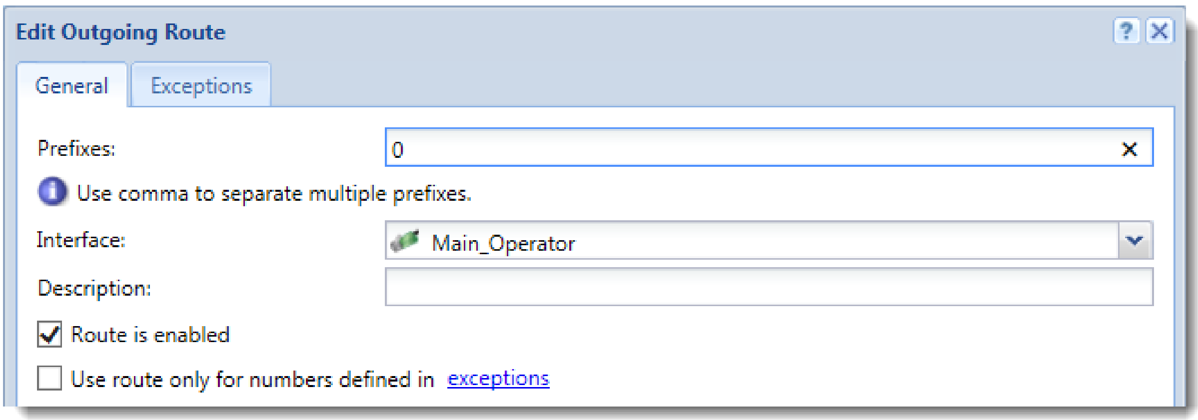
In the administration interface of the main server:

1. Go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**.
2. Double-click the interface for the branch server.
3. Go to the **Advanced** tab.
4. Select the **Allow incoming calls to use outgoing routes** option.
5. Click **OK**.



In the administration interface of the branch server:

1. Go to **Configuration** → **Call Routing** → **Routing of outgoing calls.**
2. Click **Add.**
3. Type the prefix for outgoing calls of the main server (0 in our example)
4. Select the interface of the main server.
5. Click **OK.**



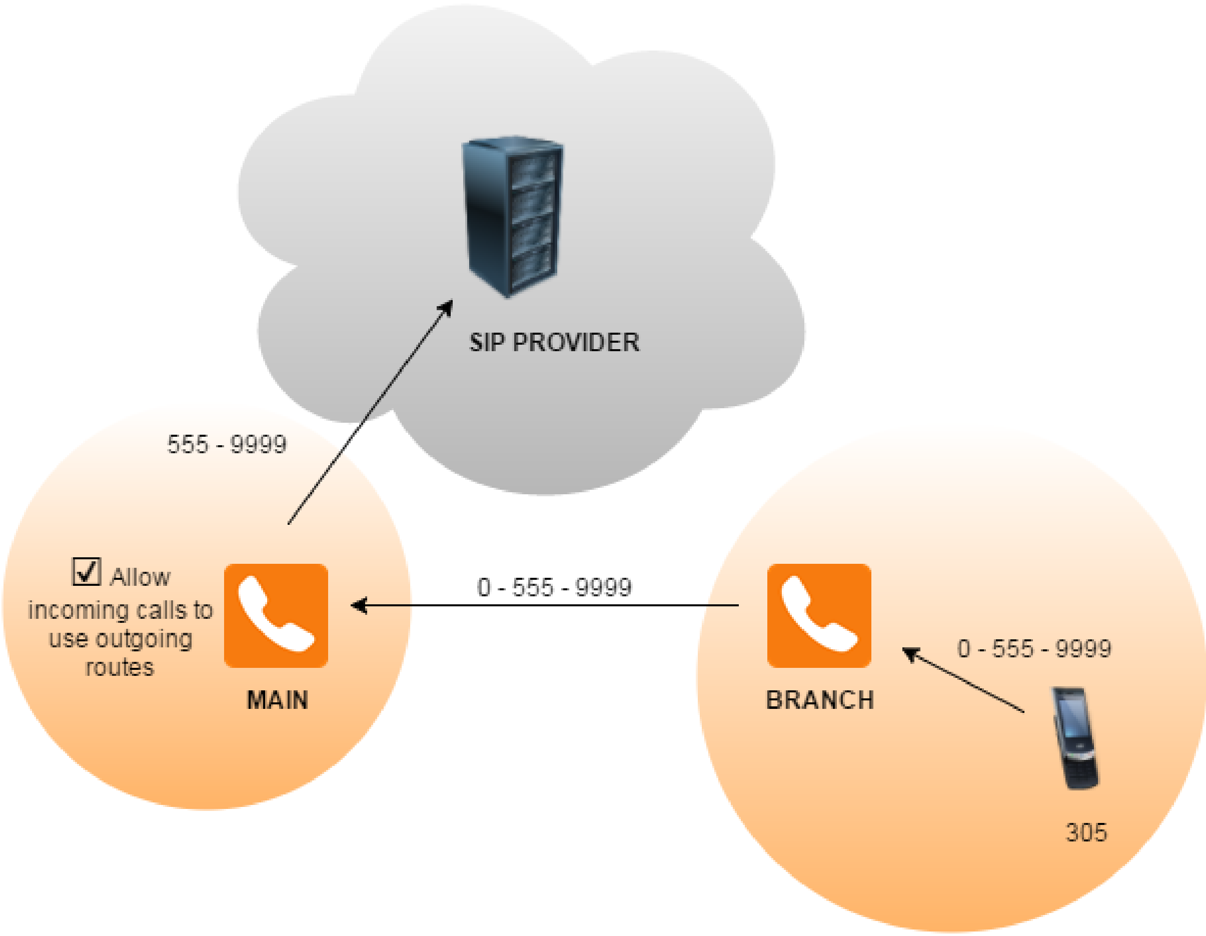
Make a test call to reach a number in the PSTN from the branch server.

Example of a test call

Call 555-9999 from extension 305:

1. The user with an extension 305 dials the number 0-555-9999.
2. Kerio Operator on the branch server recognizes the prefix 0 and routes the call to the main server.

3. The call arrives at the main server.
4. Kerio Operator on the main server recognizes the prefix 0 and strips the prefix off.
5. The main server routes the call to the SIP provider.



Routing incoming calls from the PSTN to the branch server

To route incoming calls to the branch server:

- If you have separate numbers, use speed dial extensions
- If you have a trunk of numbers, rewrite called numbers to match the internal extensions of the branch server

Using speed dial extensions

To use speed dial extensions:

1. Create a speed dial extension (9305 in the example) that dials an extension of the branch server (305 in the example).

For more details, see Creating and using speed dial.

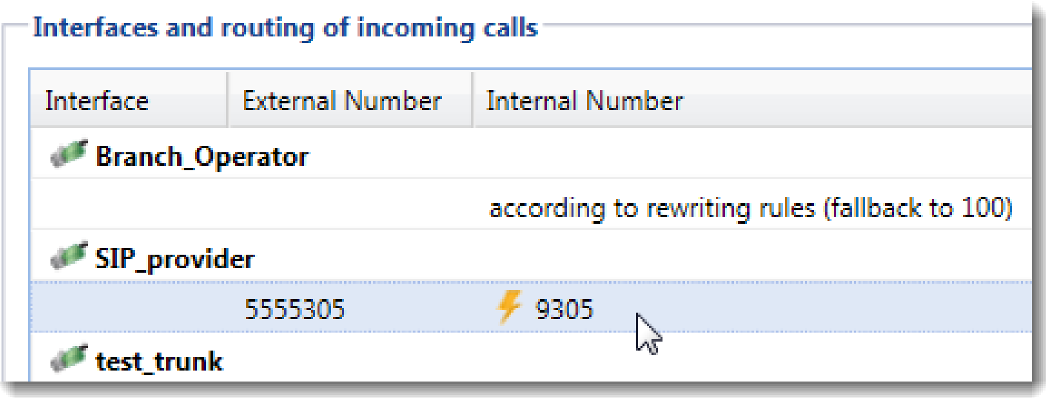
2. Go to **Call Routing** and double-click the interface for the branch server.

3. Go to the **Advanced** tab.

4. (Optional) To enable users to return calls, select **Do not substitute the calling number when forwarding calls** and click **OK**.

   This option also displays the caller ID of the caller instead of the number of the speed dial extension.

5. Double-click the number from your provider that you want to map.



6. In the **Route incoming calls to** field, select the speed dial extension and click **OK**.

7. Repeat steps 1—6 for all extensions you want to map to the branch server.

From now on, Kerio Operator uses the speed dial extension for all incoming calls that reach the external number and routes the call to extension 305 of the branch server.

Make a test call to reach an extension on the branch server.

Example of a test call

Call 555-5305 from 555-9999:

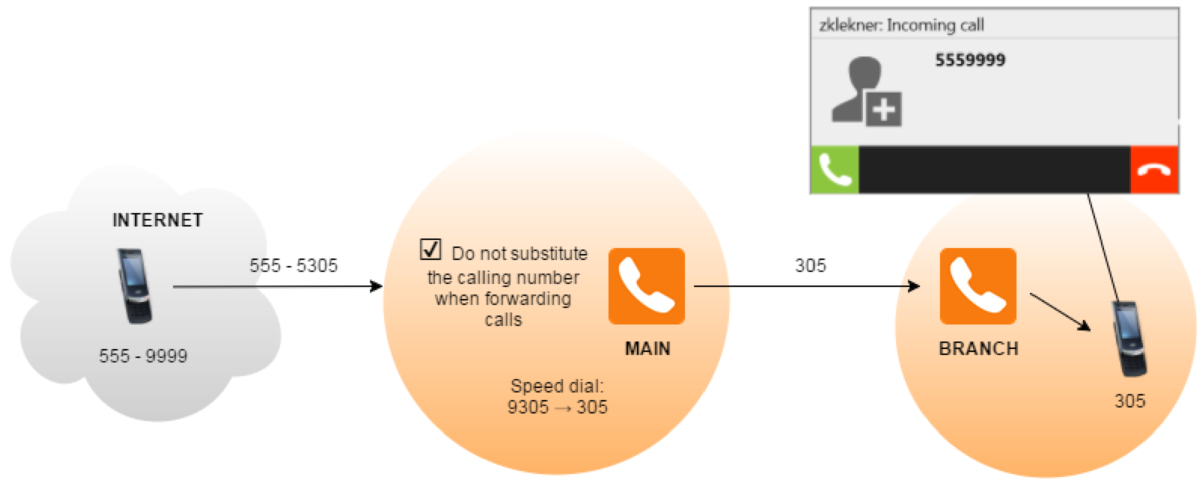
1. Caller dials the number 555-5305.

2. The call arrives at the main server.

3. Kerio Operator routes the call to the 9305 extension and then to 305.

4. The main server recognizes the prefix 3 and routes the call to the branch server.

5. The call arrives at the branch server and rings on the 305 extension.



Using number rewriting

To rewrite called numbers from your trunk and route them to the branch server:

1. In the administration interface of the main server, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls.**

2. Double-click the interface for the branch server.

3. Go to the **Advanced** tab.

4. (Optional) To enable users to return calls, select **Do not substitute the calling number when forwarding calls** and click **OK**.

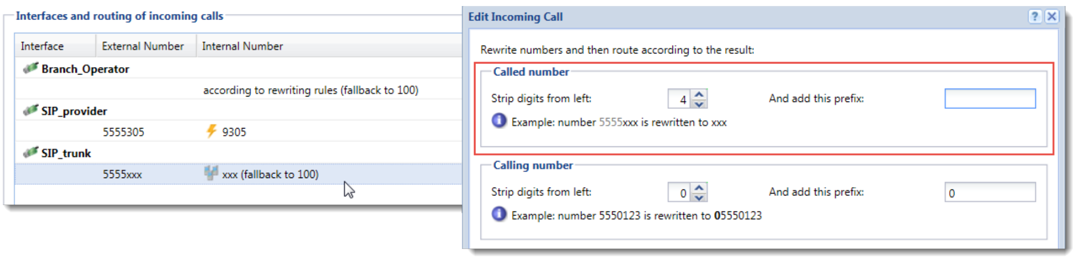
   This option also displays the caller ID of the caller instead of the number of the speed dial extension.

5. Double-click the interface for your provider.

6. Go to the **Advanced** tab.

7. Enable the **Allow incoming calls to use outgoing routes** option.

8. Double-click the trunk of numbers to verify that Kerio Operator rewrites the called number correctly.



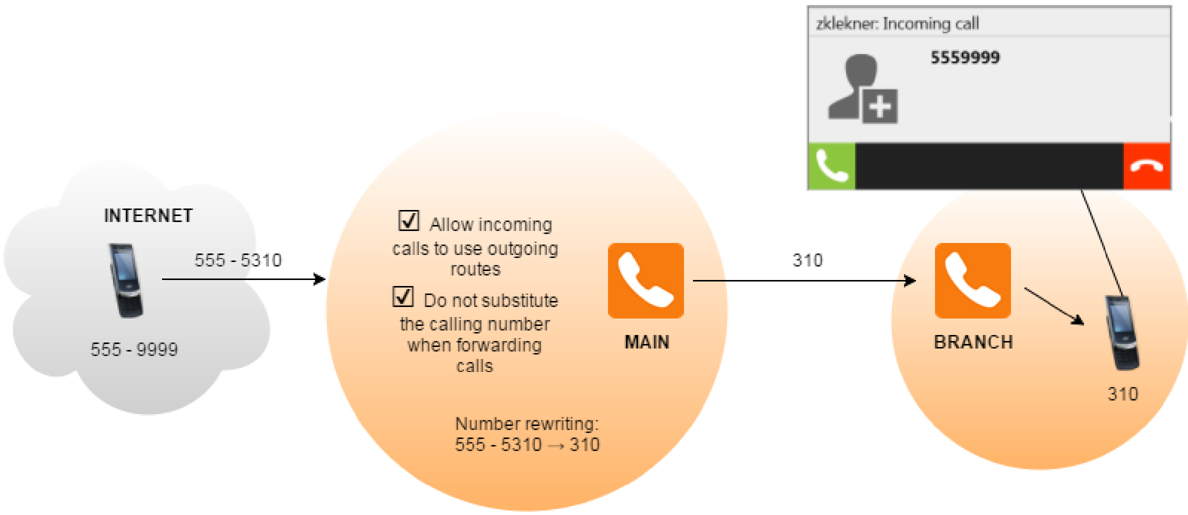
9. Click **OK**.

After configuring the interface, make a test call to reach an extension on the branch server from the PSTN.

Example of a test call

Call 555-5310 from 555-9999:

1. Caller dials the number `555-5310`.
2. The call arrives at the main server.
3. Kerio Operator matches the call to a SIP interface and strips off the first four digits of the number.
4. The call automatically uses the outgoing route with the prefix 3 and arrives to the branch server.
5. The call rings on the 310 extension.



Configuring standard phone interfaces

Overview

Redesigned in Kerio Operator 2.4!

You can connect Kerio Operator to your provider using hardware cards.

You can use the card distributed with Kerio Operator Box series 3000 or you can use your own card and connect it to your Kerio Operator server.

Supported cards

Kerio Operator supports the following cards:

- PRI card — The number of concurrent calls varies depending on whether you have a contract with an American or European provider:
    - T1 (in the USA) allows 23 concurrent calls.
    - E1 (in the EU) allows 30 concurrent calls.
- BRI card — Has four ports, each of which can operate two concurrent calls.
- FXO card — Has four ports each of which can operate only one call at a time.

For a specific list of supported cards, see the Supported Phone Cards section on the Kerio website.

Prerequisites

Before you configure an interface, you need to know:

- Telephone number (or numbers) from your telephone provider
- (PRI/BRI only) Which ISDN type to use for communication. This usually differs by your location: for example, EuroISDN for the EU, Nation ISDN Type 2 for the USA, and so on)
- Whether your provider requires overlap dialing

- Whether the provider sends or requires whole or abbreviated telephone numbers. See the Mapping external and internal numbers article for details.
- At least one internal extension defined in Kerio Operator (for example, the extension of an employee who redirects the calls).

Configuring interfaces

After connecting a card, configure the interface:

1. In the administration interface, go to the section **Configuration** → **Call Routing**.

   The **Interface and routing of incoming calls** table shows one of the following, depending on your card:

   - PRI card: one standard telephone interface
   - BRI or FXO card: four interfaces (one for each of the four ports)

2. Double-click an unconfigured interface.

   The configuration wizard opens.

3. Type a name for the interface (for example, your provider's name).

   The name must not contain spaces or special characters and must be unique.

One or multiple numbers

1. If you acquire one or multiple phone numbers from your provider, type the numbers in the **New provider** → **With external number** field. You can:

   - Separate individual numbers with commas (for example, 555450, 555451, 555452, and so on)
   - Type the whole range using – (for example, 555450-555459)

2. Click **Next**.
3. Select an extension to receive all calls from the provider.
4. (Optional) In the **Prefix to dial out** field, type a prefix for outgoing calls.

   Kerio Operator uses the prefix to route calls to your provider. This prefix can be same for other providers. See Working with prefixes for outgoing calls

5. Click **Next**.

6. (PRI and BRI only) Select the **Switch type** in the dialog box:
    - If you are in the EU, select the EuroISDN option
    - If you are in the USA, select the National ISDN Type 2 option
7. Click **Next**.
8. Verify the information in the **Summary** section.

   If you need to add more information from your provider select the **Edit details of created interface** option. For more information, see the Configuring additional details for an interface section.
9. Click **Finish**.
10. Create a rewriting rule to correctly map numbers to internal user extensions.

Interval of numbers

1. If you acquire a trunk with an interval of numbers from your provider, type the numbers in the **New provider** → **With external number** field.

   Use x in place of the numbers that vary (for example, 555xxx).
2. Click **Next**.
3. Select an extension to which you want Kerio Operator redirect all calls to unassigned (unused) extensions.
4. (Optional) In the **Prefix to dial out** field, type a prefix for outgoing calls.

   Kerio Operator uses the prefix to route calls to your provider. This prefix can be same for other providers. See Working with prefixes for outgoing calls
5. Click **Next**.
6. (PRI and BRI only) Select the **Switch type** in the dialog box:
    - If you are in the EU, select the EuroISDN option
    - If you are in the USA, select the National ISDN Type 2 option
7. Click **Next**.
8. Verify the information in the **Summary** section.

   If you need to add more information from your provider select the **Edit details of created interface** option. For more information, see the Configuring additional details for an interface section.

9. Click **Finish.**

10. Create a rewriting rule to correctly map numbers to internal user extensions.

Overlap dialing

Some telephone providers require telephone numbers as a whole, others require the telephone numbers one digit at a time. Ask your provider about their requirements. Follow these steps to configure the interface:

1. In the administration interface, go to the section **Configuration** → **Call Routing.**

2. Select an interface and click **Edit.**

   The **Edit External Interface** dialog opens.

3. Go to **Interface Card.**

4. Select the **Overlap dialing** option.

5. Click **OK.**

Configuring additional details for an interface

To set additional settings in your interface for incoming and outgoing calls:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls.**

2. Select an interface and click **Edit.**

3. On the **Interface Card** tab, change the interface settings.

   See the following chapters for details.

4. Click **OK.**

If you select the **Edit details of the created interface** option on the last page of the interface configuration wizard, this dialog box displays automatically.

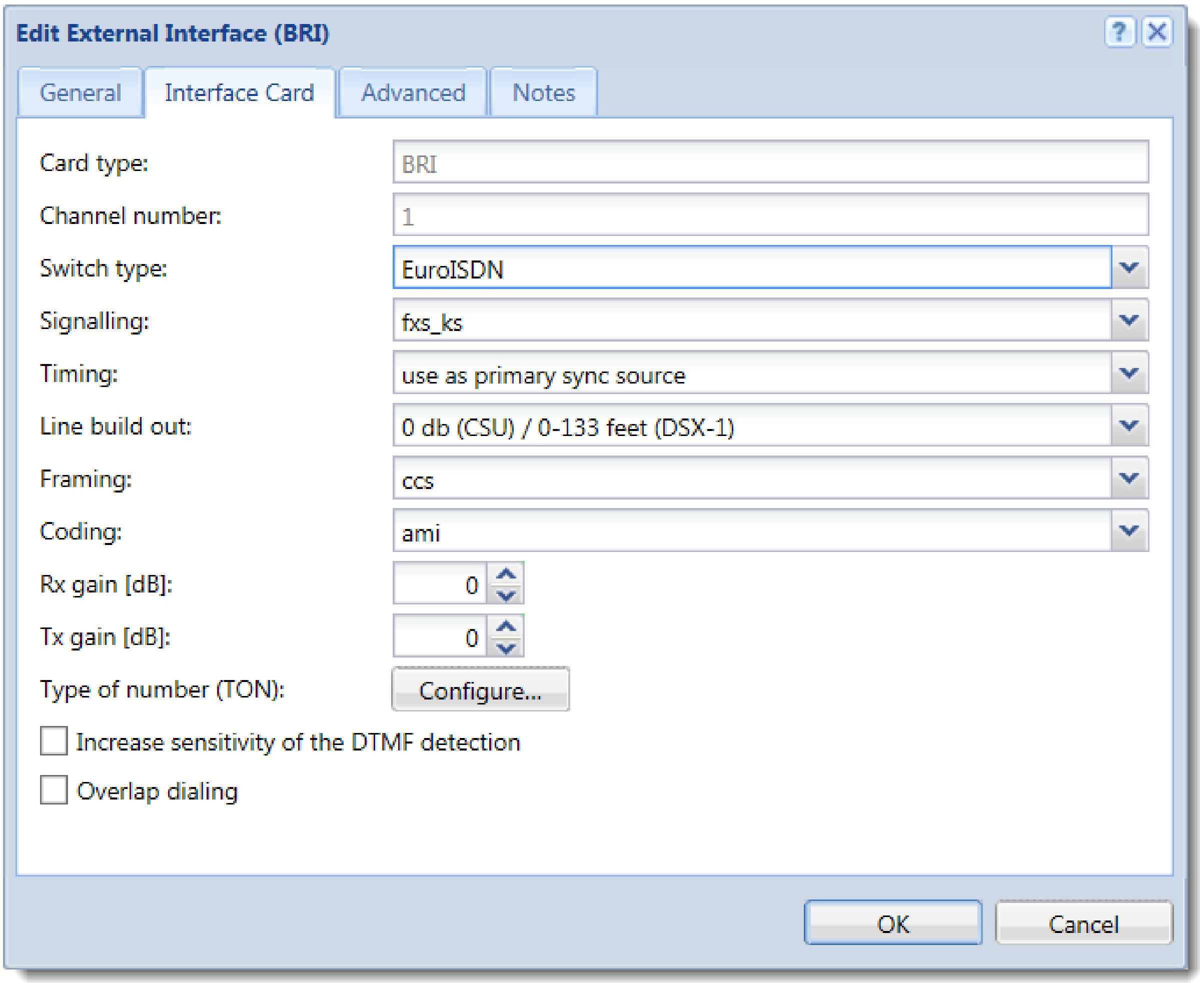


Figure 1

Adjusting audio gain for standard phone interfaces

New in Kerio Operator 2.4!

To adjust audio gain:

1. In **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**, select an interface and click **Edit**
2. Go to the **Interface Card** tab.
3. Set **Rx gain [db].**
4. Set **Tx gain [db].**
5. Click **OK.**

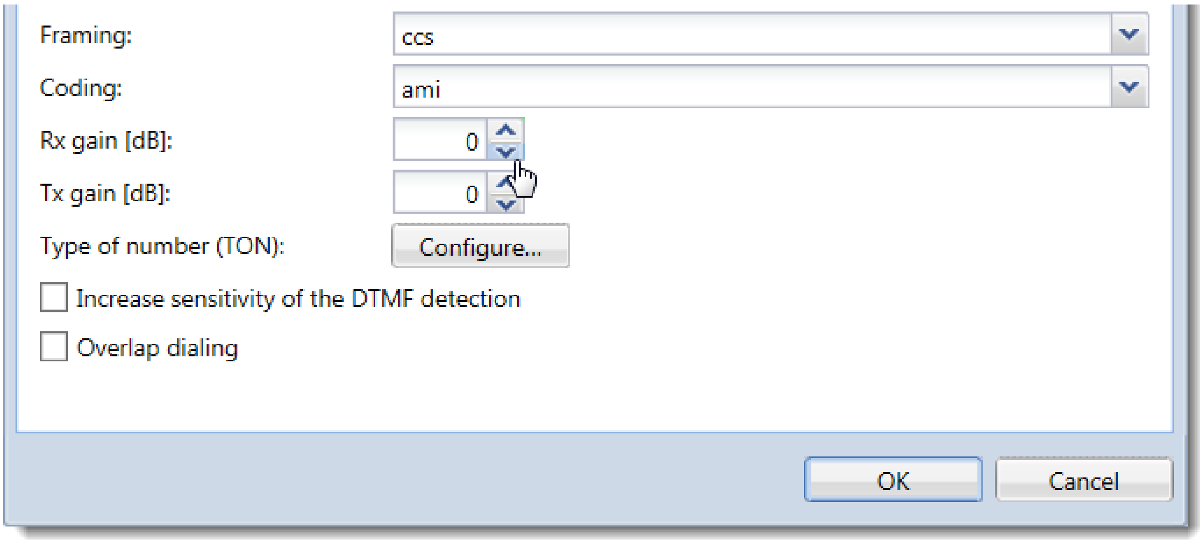


Figure 2

Configuring Type of number (TON)

New in Kerio Operator 2.4!

Some providers send a stripped number with additional information about the type of the number. Kerio Operator can read these types and assign a prefix to the stripped number.

To configure prefixes for **Type of number (TON)**:

1. In **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**, select an interface and click **Edit**.
2. Go to the **Interface Card** tab.
3. Click **Configure** next to **Type of number (TON).**
4. Type the prefixes you want to set.
5. Click **OK.**

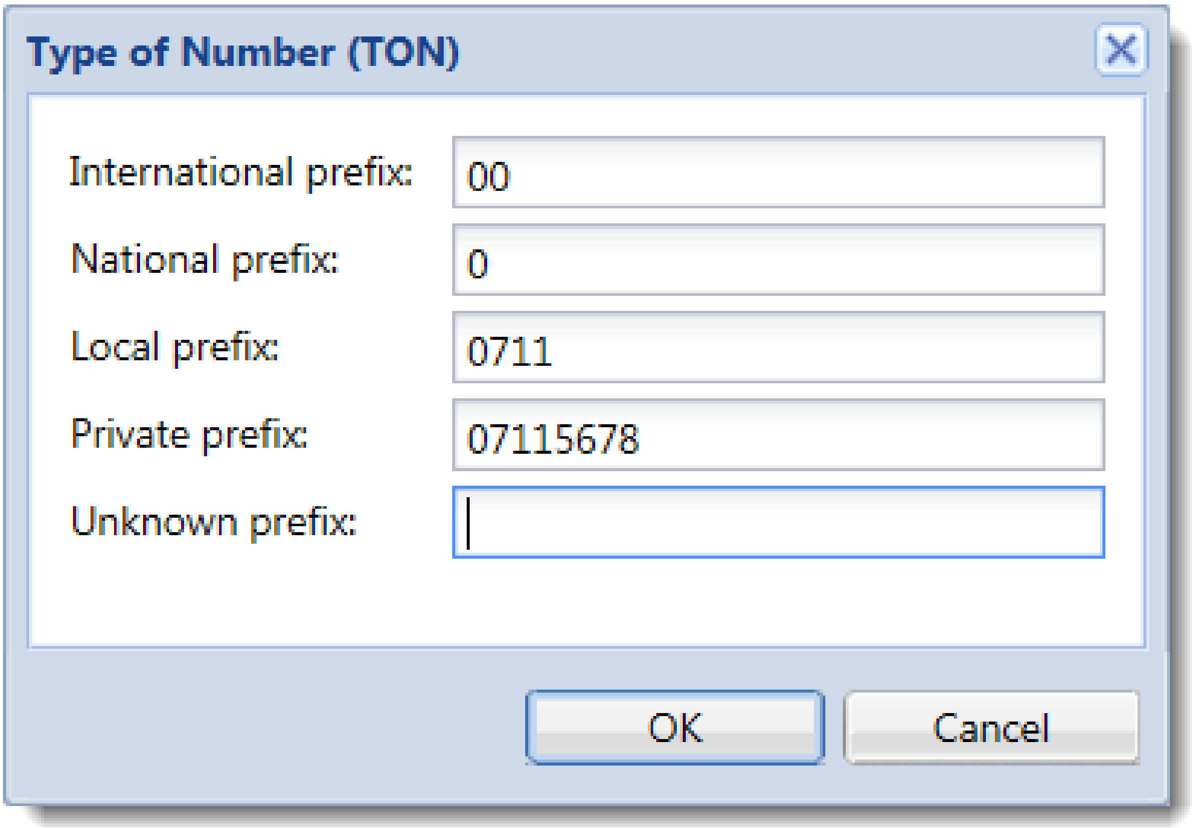


Figure 3

Increasing sensitivity of the DTMF detection

New in Kerio Operator 2.4!

To enable this option:

1. In **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls**, select an interface and click **Edit**.
2. Go to the **Interface Card** tab.
3. Select **Increase sensitivity of the DTMF detection**.
4. Click **OK**.

Mapping of numbers

See the Mapping external and internal numbers article for more information.

Configuring and using conferences

Telephone conferences overview

Telephone conference is one telephone call of three or more users.

Telephone conferences allow participation of Kerio Operator users and external participants. To join a conference, participants must dial the conference number and PIN.

You can use two different types of conferences — statically or dynamically configured.

Statically configured conference

Statically configured means that conferences are created in the administration interface and each new conference uses one extension.

If there is a lack of extensions, use dynamically configured conferences instead.

Configuring statically configured conferences

1. Go to section **Status** → **Dial Plan** and make sure that the extension you have selected for the conference is not used.
2. In **Configuration** → **Conferences**, click **Add**. The **Add conference** dialog is dispayed.
3. Enter the conference extension and its description.
4. In the menu **Conference type**, choose the **Statically configured** option.
5. Optional: Limit the number of participants.
6. Each conference can be protected by a PIN required from all participants upon attempting to enter the conference. If you wish to secure a conference, set a PIN and deliver it to the members.
7. To enable call recording, select **Record Calls.**

   Please note that call recording is a subject to special laws in many countries. It maybe illegal in your jurisdiction or require notice to the other party on the call. Accordingly, you assume all liability for using the call recording functions and are responsible for notifying all users of this system of this potential restriction, if applicable.

Connecting to a statically configured conference

1. Dial the conference telephone number / extension.
2. If the conference is protected, you will be asked to enter the PIN.

To leave the conference, simply terminate the call.

Dynamic conferences

A dynamic conference is created on one extension only. Users set the conference number and PIN after dialing the extension or the whole telephone number. On one extension, users can set unlimited number of conferences with different conference numbers.

The disadvantage of dynamic conference is that user has to enter three numbers when dialing the conference (the extension, the conference number and the PIN).

Configuring dynamic conferences

1. Go to **Status** → **Dial Plan** and make sure that the conference extension is not used by a user.
2. In **Configuration** → **Conferences**, click **Add**. The **Add conference** dialog is dispayed.
3. Enter the conference extension and its description.
4. In the **Conference type** menu, choose option **Dynamic, created on demand**.
5. To enable call recording, select **Record Calls**.

> Please note that call recording is a subject to special laws in many countries. It maybe illegal in your jurisdiction or require notice to the other party on the call. Accordingly, you assume all liability for using the call recording functions and are responsible for notifying all users of this system of this potential restriction, if applicable.

Connecting to a dynamic conference

To connect to an existing conference, enter the conference number and PIN (if required).

Creating a dynamic conference

1. Dial the conference telephone number / extension.
2. Enter any number for the conference.

3. Set PIN (if required).

4. Communicate these access numbers (extension, conference number and PIN) to other attendees.

To leave the conference, simply terminate the call.

Where to monitor conference activities

All current calls can be viewed under **Status** → **Conferences**.

Configuring call queues

Call queues overview

Call queues are used to distribute incoming calls between agents.

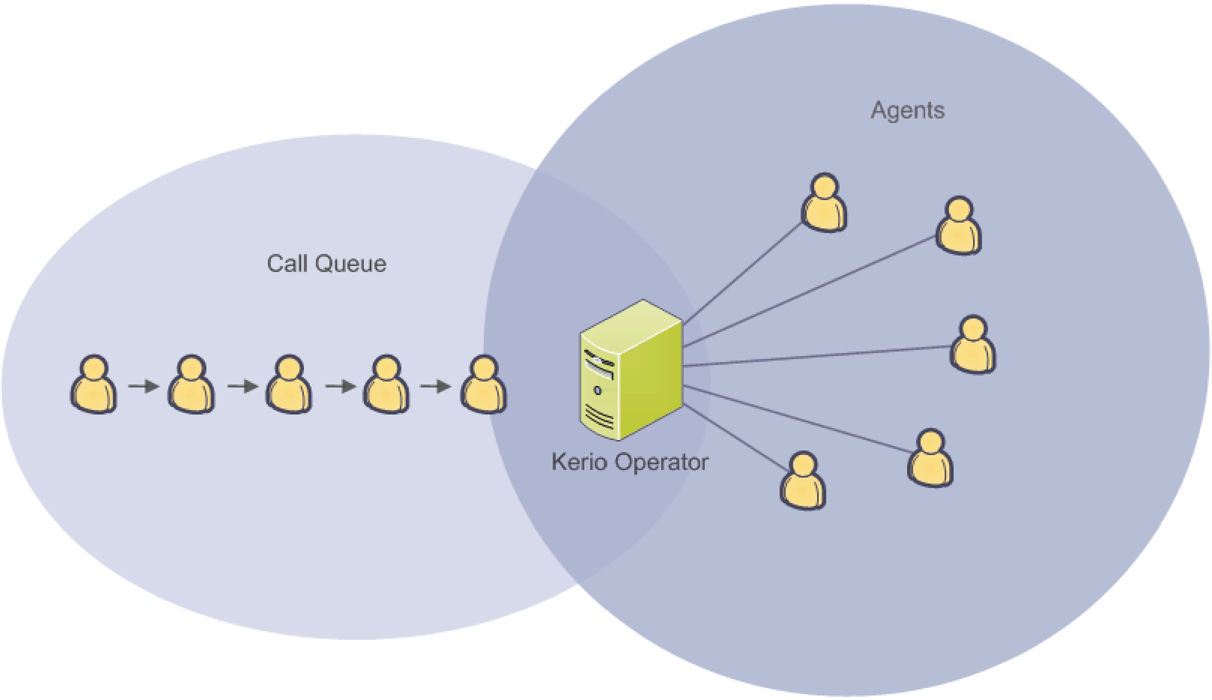


Figure 1 Call queue

Configuring call queues

1. In the administration interface, go to **Configuration** → **Call Queues**.
2. Click **Add** to open the **Add Call Queue** dialog. On the **General** tab, type the new queue extension number.
3. Select the queue strategy.
4. Click the **Agents** tab.
5. If you want your agents to log in dynamically, type login and logout code. For example, 12345 to login, and 54321 to logout. The calls will only go to agents logged into the queue.

   If you want to assign specific agents permanently to the queue, click **Add** to select their extensions.

> Both methods can be combined. One queue may have agents who are assigned permanently and agents who log in dynamically.

6. Click the **Announcements** tab.

   An announcement is a pre-recorded message that callers hear while waiting in a call queue. You can import pre-recorded announcements into Kerio Operator (see article Language settings in Kerio Operator) or record them by going to **Configuration** → **PBX Services** → **Record audio** (see article Using PBX services).

How to select a queue strategy

- Round robin with memory mode uses circular call distribution. It remembers the last agent who answered the phone, and new calls are directed to the next agent in the round queue.
- Ring all agents — calls always ring at all agents until one of them answers the particular call.
- Ring least recently called agent — the system selects the agents who have not answered the phone for the longest period.
- Ring agent with fewest calls — the system assigns the call to the agent with the lowest number of calls answered so far.
- Ring random agent — if you select this option, the system will choose an agent randomly.
- Ring in order — only for permanently assigned agents. You specify a fixed order in which they are always selected. This strategy is for companies where all calls are answered by a receptionist. In case the receptionist is not answering, the call is directed to the next agent in order (for example, an administration assistant).

What is the difference between permanently assigned and dynamic agents

- Permanent assignment — agent's extension is assigned permanently to the queue.
- Dynamic login — agents use special code for logging in and out of the queue.

Recording calls from call queues

Kerio Operator allows recording calls from call queues. No other module or equipment is necessary. Setting can be done as follows:

1. Open the **Configuration** → **Call Queues** section and select the queue in which you wish to record the calls.
2. On the **General** tab, select **Record calls**.

To play back recorded calls

Please note that call recording is subject to special laws in many countries. It may not be legal in your jurisdiction, or may require notice to the other party on the call. Accordingly, you assume all liability for using the call recording functions and are responsible for notifying all users of this system of this potential restriction, if applicable.

Section **Status** → **Recorded Calls** displays all calls recorded from call queues. Select a call to listen to it, download it to your computer or remove it.

Deleting Recorded Calls

Recorded calls can be periodically deleted once their total size reaches a certain limit. The limit can be set in section **Status** → **Recorded Calls**.

1. Click **Advanced** → **Periodically Remove Old Recorded Calls**.
2. In **Remove Old Recorded Calls** dialog box, enter the maximum size of recorded calls on a disk (in MB).

Configuring a call queue timeout

The call queue timeout period determines the maximum amount of time a caller can be placed in a call queue. Configuring the limit prevents from waiting in a queue infinitely.

The timeout limit is unlimited by default. For setting the limit, perform these steps:

1. In administration, go to **Configuration** → **Call Queues**.
2. Click **Add/Edit**.
3. On tab **General**, set **Queue timeout**.
4. (Optional) Go to tab **Announcements** and select **Timeout announcement**. Such announcement will play when the limit is reached and should include information about what happens next (tab Exceptions).

5. Go to tab **Exceptions**.
6. Choose an action for exceeded limit:
    - **Callers receive a busy signal** — if announcement was set, recording plays before call termination.
    - **Forward to** — type an extension. Kerio Operator forwards callers to the extension. If announcement was set, recording plays.
7. Save the settings.

Timeout is configured. If you want to check your settings, lower the limit to several seconds and dial the queue from several phones.

Configuring a music on hold and a while waiting period

A while waiting period is the period when users are waiting in a call queue for an agent. You can set what is playing during the period:

1. In administration, go to **Configuration** → **Call Queues**.
2. Click **Add/Edit**.
3. On tab **General**, select **While waiting**:
    - Music on hold — a music sounds during the while waiting period.
    - Ringtone — a ringtone sounds during the while waiting period.
4. If you selected the **Music on hold** option, select the particular recording in the **Music on hold** menu.

    If you want to add a new recording to Kerio Operator, go to the **Definitions** → **Music on Hold** section.
5. Save the settings.

Configuring a queue length

A queue length determines max. number of callers in the queue at the same time. Configuring the limit prevents from waiting too long in the queue.

The queue length is unlimited by default. For setting the limit, perform these steps:

1. In administration, go to **Configuration** → **Call Queues**.
2. Click **Add/Edit**.

3. On tab **General**, set **Queue length**.
4. (Optional) Go to tab **Announcements**, select **Full queue announcement**. Such announcement will play when the limit is reached and should include information about what happens next (tab Exceptions).
5. Click the **Exceptions** tab.
6. Select an action for exceeded limit:
    - **Callers receive a busy signal** — if an announcement was set, recording plays before a call is terminated.
    - **Forward to** — type an extension. Kerio Operator forwards callers to the extension. If an announcement is set, Kerio Operator plays the recording.
7. Save the settings.

The queue length is configured. If you want to check your settings, lower the limit to 1 and dial the queue from two phones.

Configuring exit keys

You can set exit keys for each call queue. Callers can use an exit key for transfer to an extension.

1. In administration, go to **Configuration** → **Call Queues**.
2. Click **Add/Edit**.
3. On tab **General**, click **Edit** next to **Exit keys**.
4. Edit **Exit Keys**, click **Add**.
5. In the **Add Exit Key** dialog, type an exit key (for example 1).
6. Type an existing extension to transfer calls.
7. Type a description.
8. Save the settings.

When users standing in the queue use the exit key, they are transferred to pre-configured extension.

Configuring call queues without agents

Follow these steps if no agents are logged into the queue:

1. In administration, go to **Configuration** → **Call Queues**.
2. Click **Add/Edit**.
3. (Optional) Go to the **Announcements** tab, select **No agents announcement**. Kerio Operator plays the announcement when there are no agents in the queue.
4. Go to tab **Exceptions**.
5. Select an action if the queue has no agents:
    - **Callers cannot join the queue. Callers already waiting are removed** — Kerio Operator disconnects all callers. If **No agents announcement** is selected, Kerio Operator plays the recording.
    - **Callers can join the queue** — new callers can connect to the queue. Current callers stay in the queue. If **No agents announcement** is selected, Kerio Operator plays the recording.
    - **Callers cannot join the queue** — new callers cannot connect to the queue. Current callers stay in the queue. If **No agents announcement** is selected, Kerio Operator plays the recording.
6. If you selected **Callers cannot join the queue** or **Callers cannot join the queue. Callers already waiting are removed**, select one of these actions:
    - **Callers receive a busy signal**
    - **Forward to** — type an extension or external phone number. Kerio Operator forwards callers to the number.
7. Save the settings.

Settings are complete now. If you want to check your configuration, test these cases:

1. No agent serves the queue. Try to join the queue as a caller.
2. One agent serves the queue. Join the queue as a caller. Agent logs out.

Prioritizing call queues

Agents can operate several call queues. In the following example, an agent is assigned to three queues.

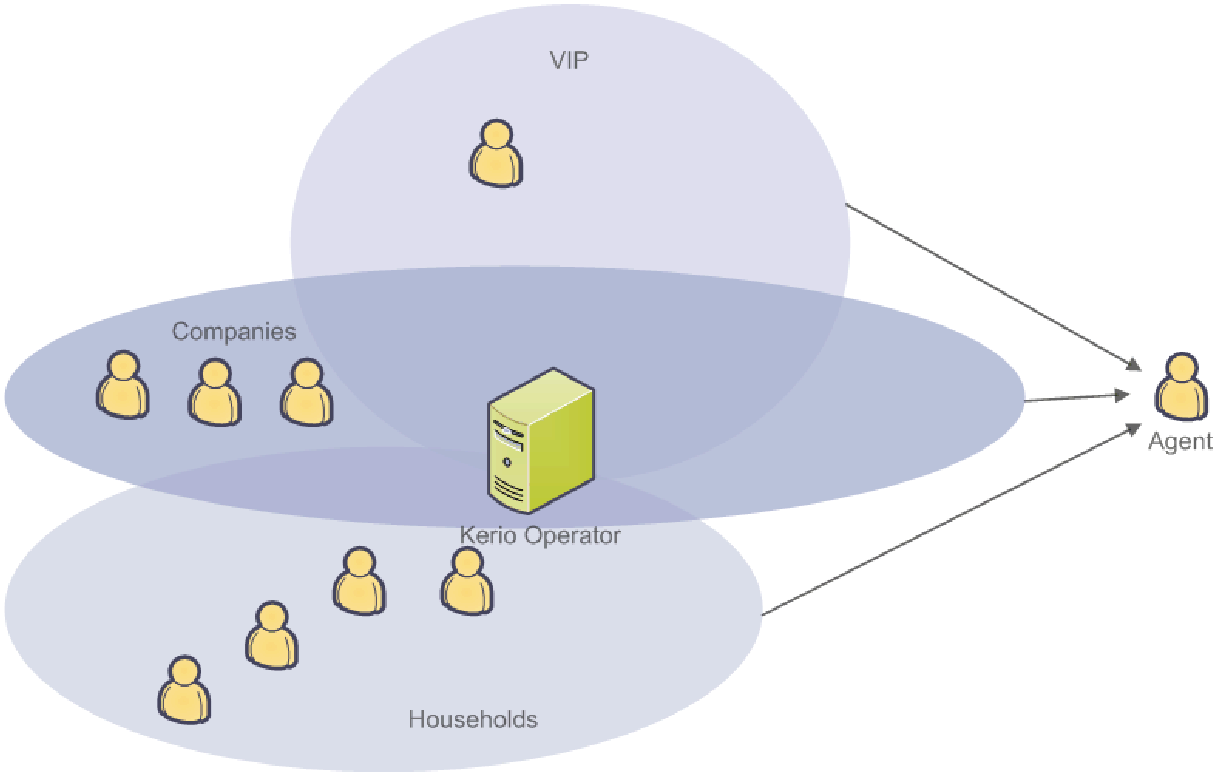


Figure 2 Operating multiple queues at once

To help agents identify the queues, you can upload various audio records for each queue. The record identifying the queue is played to the agent before a call from this queue is connected.

Upload new audio record as follows:

1. Select a call queue or create a new one in section **Configuration** → **Call Queues**.
2. In the displayed dialog, go to the **Announcements** tab.
3. Check the **Help agents identify the source queue by playing this announcement** and click on **Select**.
4. In the **Select Audio File** dialog box, double-click a record to select it, or upload your own record to Kerio Operator (it must be in WAV or GSM format). Use the **Upload** button.

It is also possible to set priorities for individual queues:

1. Open the **Configuration** → **Call Queues** section.
2. Select a queue or create a new one.

3. In the displayed dialog, go to the **General** tab and set the desired priority.
4. Repeat the configuration for other queues.

Queues with higher priority are processed first.

Displaying missed calls on phones in call queues

When an incoming calls rings in the call queue and an agent answers it, other devices in the queue register the call as missed anyway.

To not display missed calls on other devices:

1. In the Kerio Operator administration interface, go to **Configuration** → **Call Queues**.
2. Select an extension and click **Edit**.
3. Switch to **Advanced** tab.
4. Select **Do not display missed calls on the phones**.
5. Click **OK**.

Monitoring active call queues

1. In the administration interface, go to section **Status** → **Call Queues**.
2. The top table shows currently active queues.
3. The other tables display agents and callers in a queue. Just select a queue and the details in table **Agents** and **Callers** are updated.

You can also reset the call queue statistics to start from zero. Use the **Reset Statistics** button.

Configuring auto attendant scripts

What is auto attendant script

Auto attendant script is a simple collection of voice menus, submenus and announcements and actions defined for each of them according to the caller's behavior. It can:

- connect to an extension or voicemail,
- play an announcement,
- navigate through menus and submenus.
-

  New in Kerio Operator 2.5!

  send any faxes to a configured email.

Menus can be recorded in various formats. Kerio Operator supports the following formats:

Supported formats	Audio format
gsm	8KHz
wav	8KHz, 16 bits per sample, mono (Kerio Operator encodes all WAV files into this format automatically)

Table 1 Kerio Operator — supported audio formats

How to add new auto attendant script

See the following description of an auto attendant script as an example. Create a script which:

- starts after dialing extension 200,
- contains a voice menu with the following text: "LOL! You have just reached the Live And Let Laugh company's hotline (fiendish laugh)."
  - For Sales Department, press 1.
  - For Quality Assurance Department, press 2.

  - For Technical Support Department, press 3.
  - If you wish to speak to the receptionist, press 4.

The Sales Department manages two flagship products of the company. Therefore, two submenus (*Joke Lite*, *Laugh Home 2012*) are created.

- For *Joke Lite*, press 1.
- For *Laugh Home 2012*, press 2.
- If you wish to talk to the receptionist, press 3.

Create the same menu for technical support.

Before creating the script, it is necessary to create extensions (in the assigned range 123456XXX) which will be used in the script.

- *extension 100* — reception of Live And Let Laugh Inc. One of the receptionists Joan Giggle or Brian Snigger will connect the calls if the caller makes no selection from the menu.
- *extension 203* — Quality Assurance Department extension (David Jester).
- *extension 301* — common extension (you can create a call queue or a ringing group) for *Joke Lite* experts, such as Frederic Jovial, George Funpoker, Anne Kdotte.
- *extension 302* — common extension for Laugh Home 2012 experts (Tamara Bellylaugh, Otto Spass, Mary Merry).
- *extension 501* — call queue for Joke Lite technical support (Andrew Widegrin).
- *extension 502* — call queue for Technical Support of *Laugh Home 2012* (Alan Tickle).

Script settings

Configure the script in the administration interface in section **Configuration** → **Auto Attendant Scripts**:

1. Click **Add** and enter the **Script extension** (extension 200 in our example) and some description.

2.

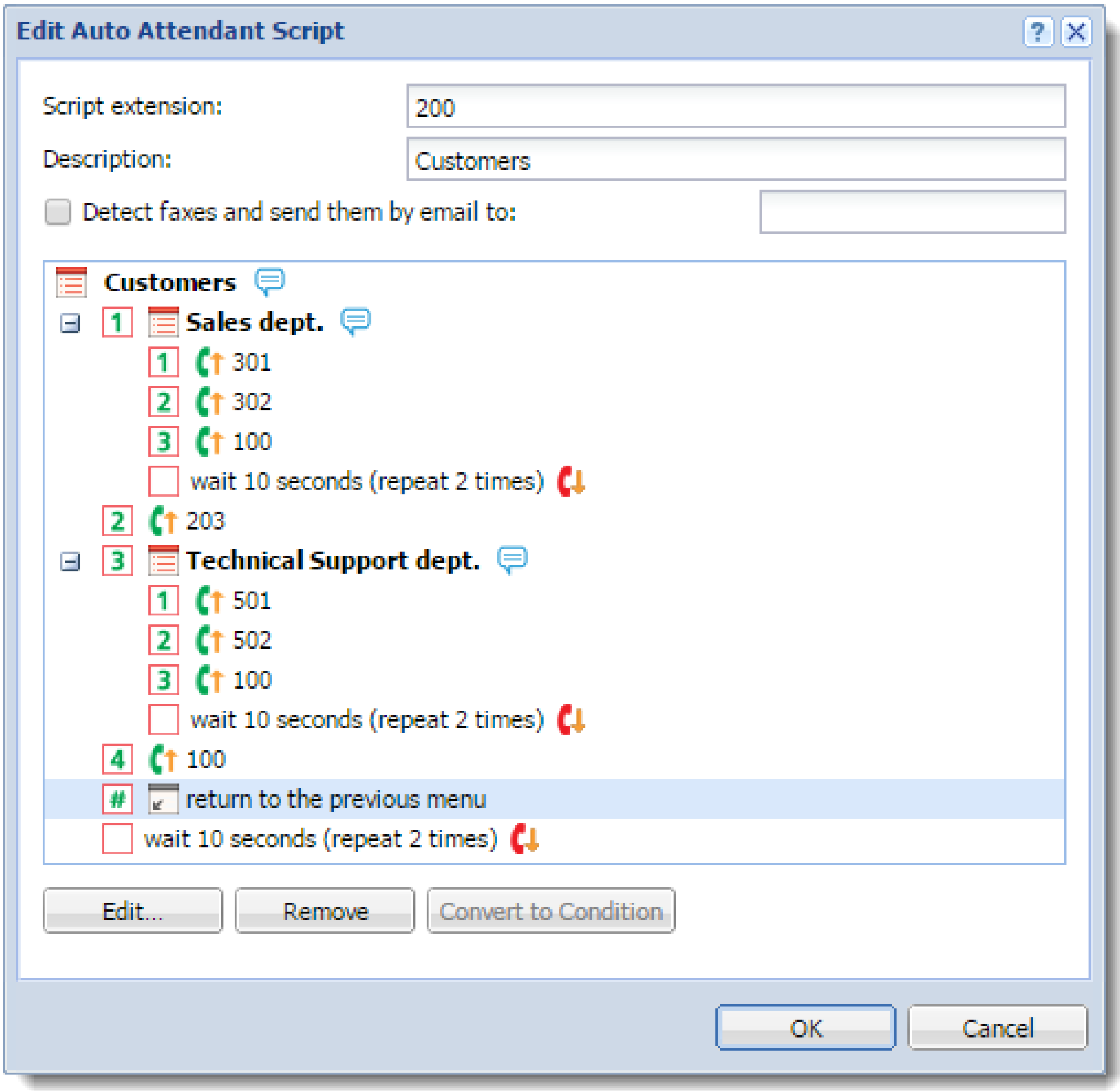


Figure 1 Auto Attendant

New in Kerio Operator 2.5!

(Optional) To receive faxes to configured email address, select **Detect faxes and send them by email to** and type an email address.

3. Click **Edit** and open the **Edit Menu** dialog.

4. In the **Announcement** field, select the recording for the main script. The **Select** button offers existing recordings or you can upload your own announcement to the PBX.

If you open the administration interface in Safari browser and you cannot play any recordings, read article Cannot play voicemails or audio files in Safari.

5. Set **Number of playbacks** to two which will ensure the menu is played to the caller twice.

6. Once the announcement is played, timeout is started with the default action taken upon its expiration. Set the timeout to 10 seconds. The default action is the preset hang up action. This means that if the announcement is played twice and the customer does not make any selection within 10 seconds, the call will be terminated.

7. Click **Add**. The **Key** column states the key which confirms the customer's choice. Enter number 1. Enter 1 in this column. Column **Action** defines what happens when the caller presses a key on their phone. Select **Go to submenu**. We need to direct calling customers to the extension of the product they are interested in (either Joke Lite or Laugh Home 2012). In the **Announcement** column, you can add a record which will be played upon pressing the particular key (for example: Stay tuned, now you will be redirected to the Live And Let Laugh Inc Sales Department). Finish the table according to figure.

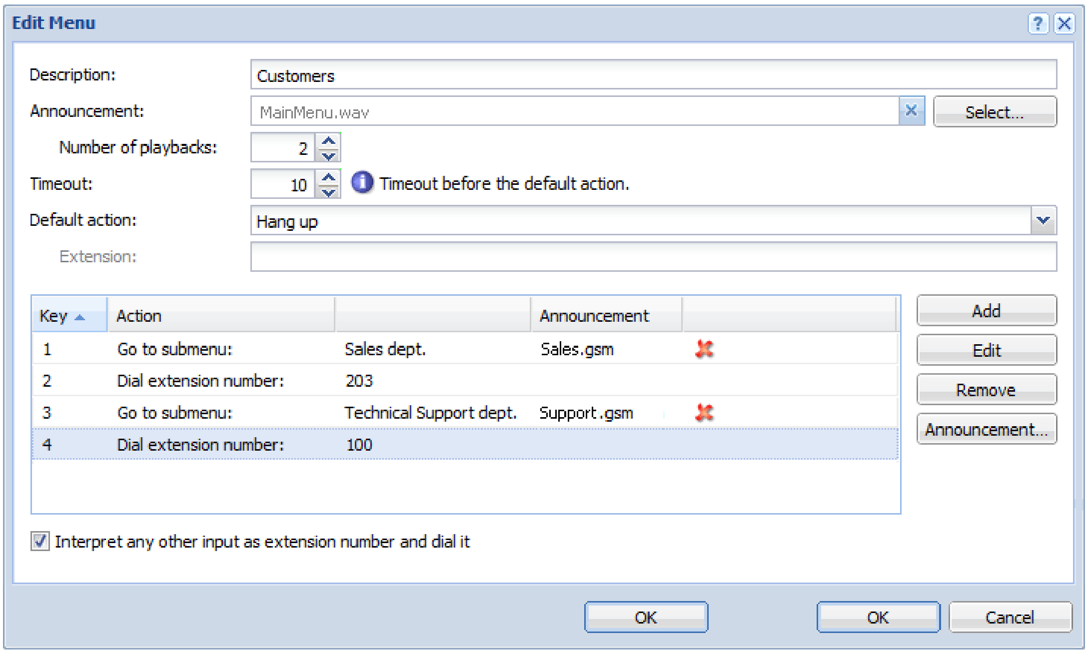


Figure 2 Editing main menu

8. Check **Interpret any other input as extension number and dial it**. This option allows to specify a direct extension while the auto attendant script is running.

9. Confirm the settings and return to the **Add Auto Attendant Script** dialog which is now similar to the one in picture above.

10. Click on menu **Sales dept.**. Again, the **Edit menu** dialog is opened but now the menu is for the Sales department. Follow the same procedure as with the main menu. The resultant menu will look as the one showed in the picture below.

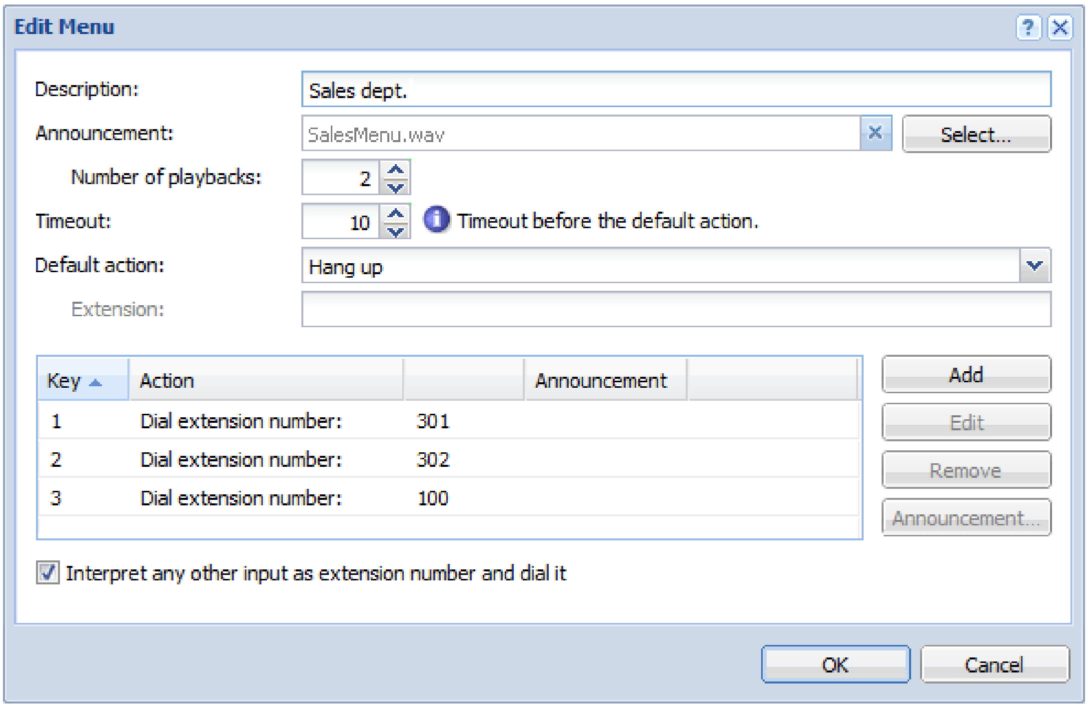


Figure 3 Submenu edit

11. Do the same for the **Technical Support dept.** menu.

Now the script is complete.

You can duplicate an existing script if you want to create a similar one — select a script and click **Duplicate**.

Time condition

The script can be limited to a specific time interval (office hours of your employees or night time when no call queue agents are available).

The time ranges (intervals) are configured in section **Configuration** → **Definitions** → **Time Ranges**. Once you have the time range configured, go back to the **Add Auto Attendant Script**, select the menu you wish to limit and click on the **Convert to Time Condition** button.

Instructions for time condition setting will be better understood through the following example focusing company's working hours. Sales department works from 9am to 5pm on weekdays. Configure the auto attendant script so that when customers call during office hours they will be connected to a sales department employee and when they call before or later they will hear a message announcing that the sales department is closed. To create the condition script, follow these instructions:

1. In the administration interface, go to **Configuration** → **Definitions** → **Time Ranges**.
2. Click **Add**. Dialog **Add Time Range** opens.
3. In section **Add to a group**, select the **Create new** option and enter a name for the new interval (for example, `Sales Department Office Hours`).
4. The **Description** is optional, for example **Weekdays from 9am to 5pm**.
5. Select `daily` in the **Type** menu and set the desired interval from 9 to 5 in the **From** and **To** fields.
6. In the **Valid on** menu, select **Weekdays**.
7. Click **OK** to confirm the changes.
8. Open the **Configuration** → **Auto Attendant Scripts** section.
9. Click on **Add**.
10. In the **Add Auto Attendant Script** dialog, create a corresponding menu (the script created in the previous section will be used in this example — see the picture below).
11. Select the **Sales Department** submenu and click **Convert to Time Condition**.
12. Divide the Sales Department submenu in two time conditions. The first one is played if the condition is met and the second if the condition is not met. Click on the red highlighted text **Set up the time condition**.
13. This opens dialog **Edit Time Condition**. In the **For time range** menu, select **Sales Department Office Hours**.
14. Click on the submenu representing the positive result of the condition. It is currently called **Unnamed**. In the dialog **Edit Menu** just opened, simply add a description (for example `Sales Department condition met`).
15. Click on the submenu representing the negative part of the condition (now it is empty and unnamed).
16. This opens dialog **Edit Time Condition** allowing to add a description (for example `Sales Department --- condition not met`).
17. Now you can modify the script. For example, in the **Announcement** field, add a message announcing that office hours of the Sales Department are from 9am to 5pm on weekdays.
18. Save the submenu. The resultant script is displayed in the next picture.

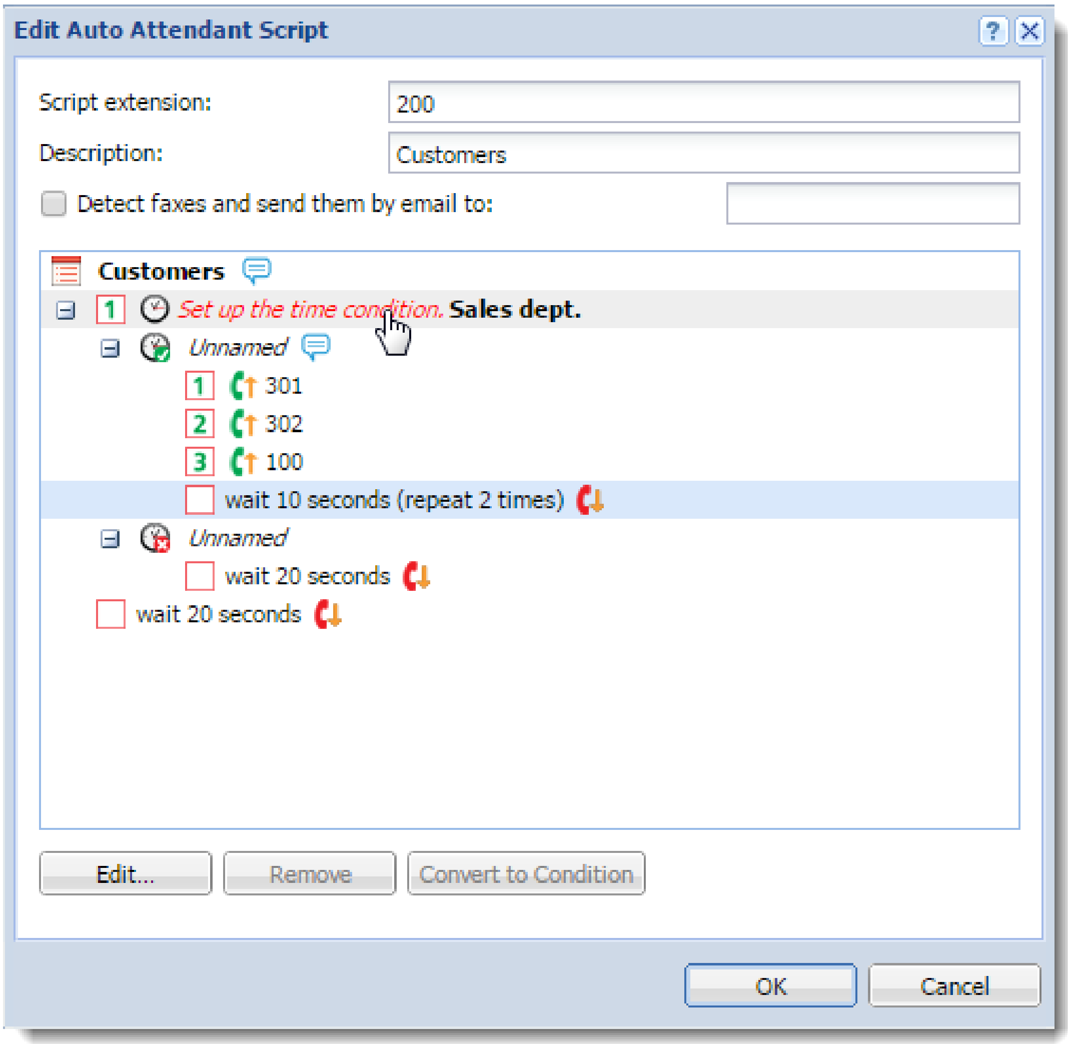


Figure 4 Setting the time condition

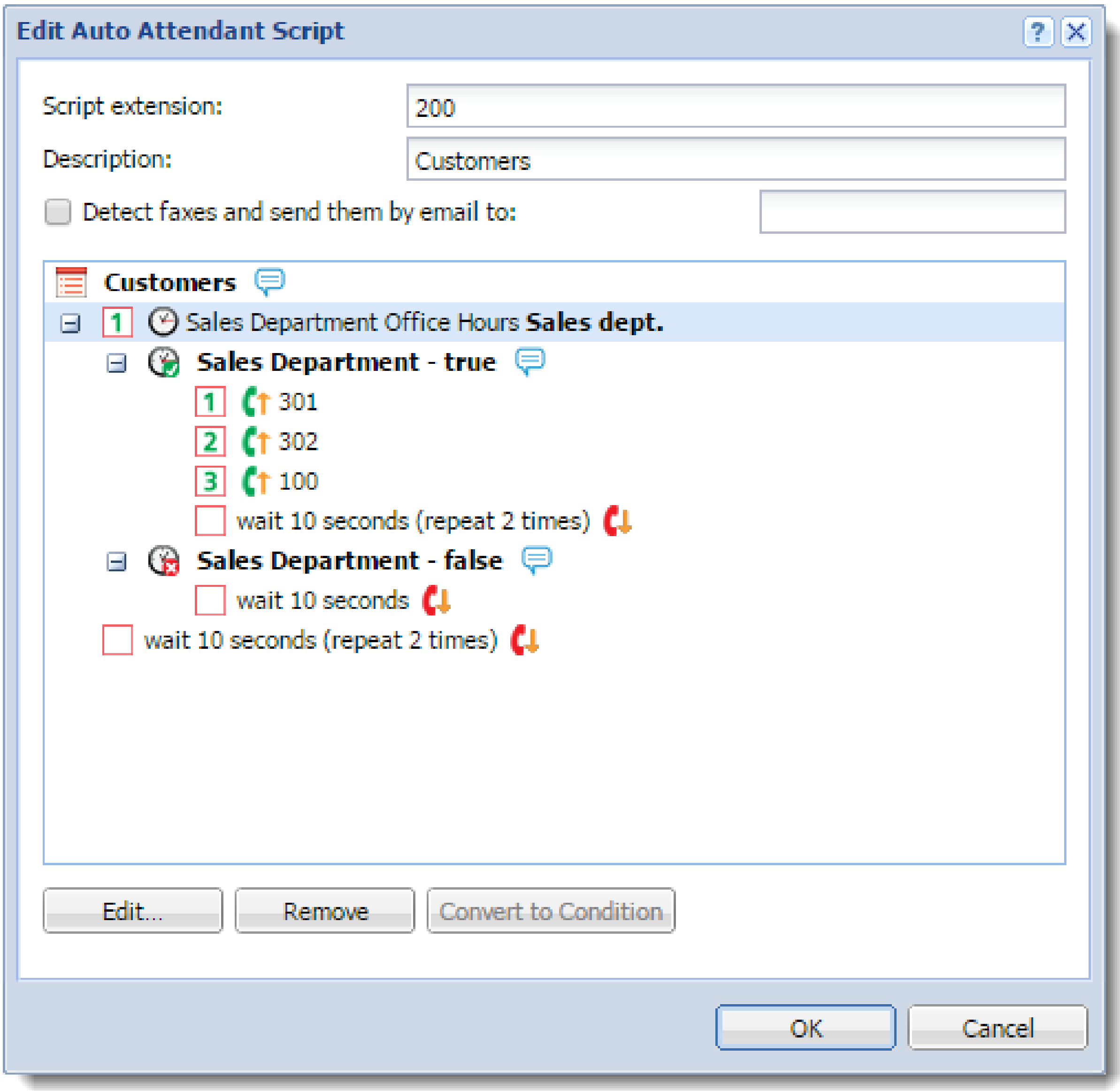


Figure 5 Time condition applied in the script

Setting time conditions in auto attendant scripts

Time conditions are best explained in an example

When configuring auto attendant scripts, Bob encountered the following problem. The company management created a new quality department. The responsible person is Alice. Bob created a new extension for this department. Alice came to Bob complaining that dissatisfied customers are calling constantly and she does not even have time for lunch.

Bob knew that Alice needs an auto attendant script which respects her working hours. And how to do it?

1. Bob created new time intervals for Alice's working hours, her lunch break and also for public holidays.

2. He created records for the following announcements:

   Hello. You are calling Live And Let Laugh Inc. We are having a delicious lunch at the moment. If you call after 1pm, we will gladly hear what you have to say. Talk to ya later!"

   "Hello. You are calling Live And Let Laugh Inc. We are off the clock at the moment. Please, call us on weekdays from 8am to 12pm or after lunch from 1pm to 6pm. We will gladly hear what you have to say. Talk to ya later!"

   "Hello. You are calling Live And Let Laugh Inc. Have a very merry holiday today. If you wish to make a complaint, call us on weekdays from 8am to 12pm or after lunch from 1pm to 6pm. We will gladly hear what you have to say. Talk to ya later!"

3. He created a new auto attendant script with time conditions.

You can also use the Day/night mode to create time conditions without a specific time set (see Using the Day/night mode in auto attendant scripts.

Setting time intervals for auto attendant scripts

1. In the administration interface, go to section **Definitions** → **Time ranges**.

2. Add three new time ranges. Two ranges will be of the daily time — `Lunch break` and `Working Hours`. Both ranges will be valid on weekdays.

3. The third range will be absolute. Add the first public holiday when creating the range. Add also other public holidays and do not forget to add them into the existing group `Holidays`.

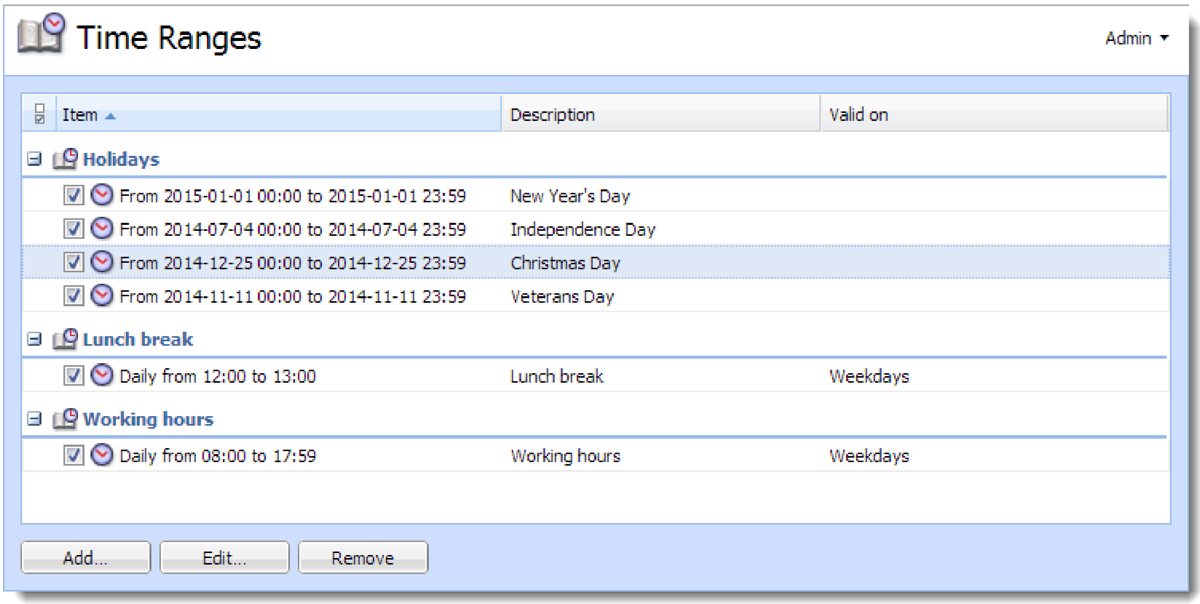


Figure 1 Setting time ranges for working hours, lunch break and holidays

Creating auto attendant scripts in Kerio Operator

The script will follow this scheme:

```
If Holidays
    publicholidays.wav
Else
    If Working hours
        If Lunch break
           lunchbreak.wav
        Else
           Action: Redirecting to Alice's extension.
    Else
        offtheclock.wav
```

1. In the administration interface, go to **Auto Attendant Scripts**.

2. Add a new script, assign it extension 300 and add a description (`Scripts for complaints desk`).

3. Create the first condition: Click **Convert to Time Condition**.

   Double-click on the red link **Set up the time condition** and in the **Edit Time Condition** dialog, select range `Holidays`. Save the settings.

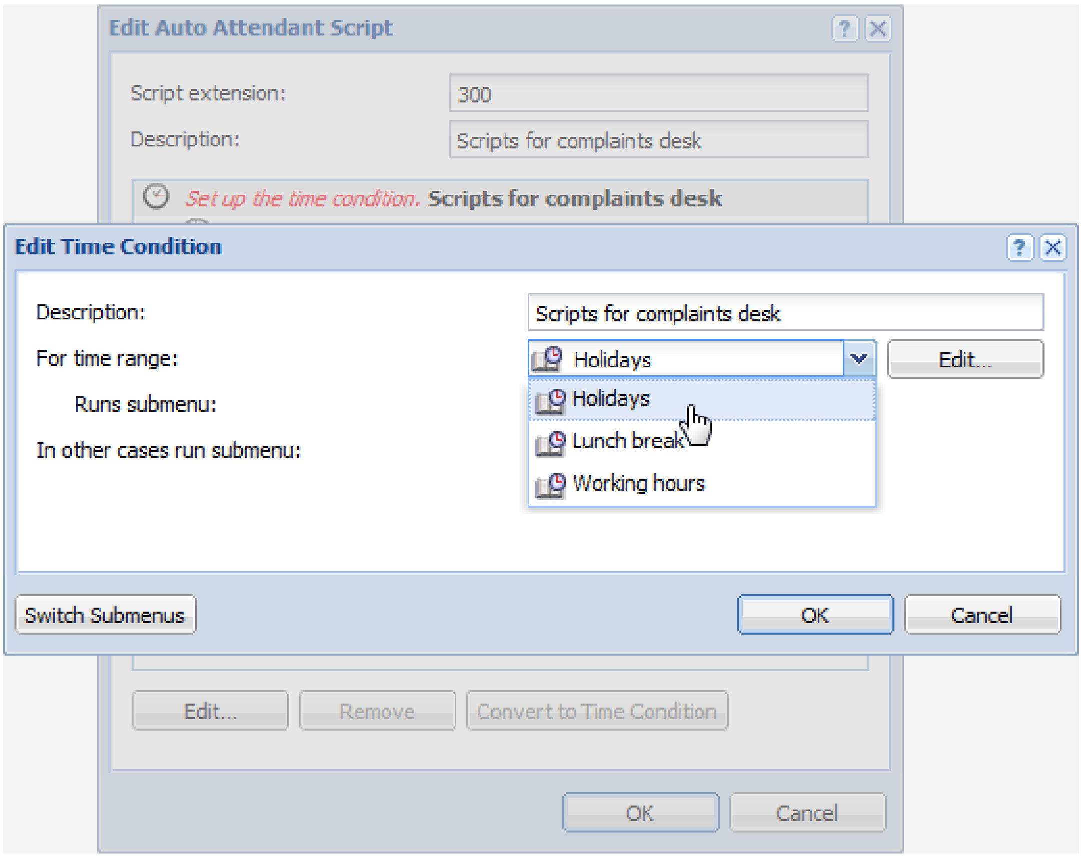


Figure 2 Auto Attendant Scripts — creating first condition

4. Now, edit the first part of the condition in dialog **Add Auto Attendant Script**. Double-click *Unnamed*).

5. In the **Edit Menu** dialog, type description `Holidays` and add a file with the announcement about a holiday. Set timeout to 5 second (this will suffice) and save the settings.

6. Create the second condition: Select the *Unnamed*) icon and click **Convert to Time Condition** (thus the “Working hours” condition will be nested into condition “Holidays”).

   In the **Description** field, enter **Working hours**; in the **For time range**, select **Working hours**. Save the settings.

7. In the **Edit Auto Attendant Script** dialog under the **Working hours** line, two new conditions appear.

8. Create the third condition: Click *Unnamed* and click **Convert to Time Condition**. In the **For time range** menu, select **Lunch break**. Save the settings.

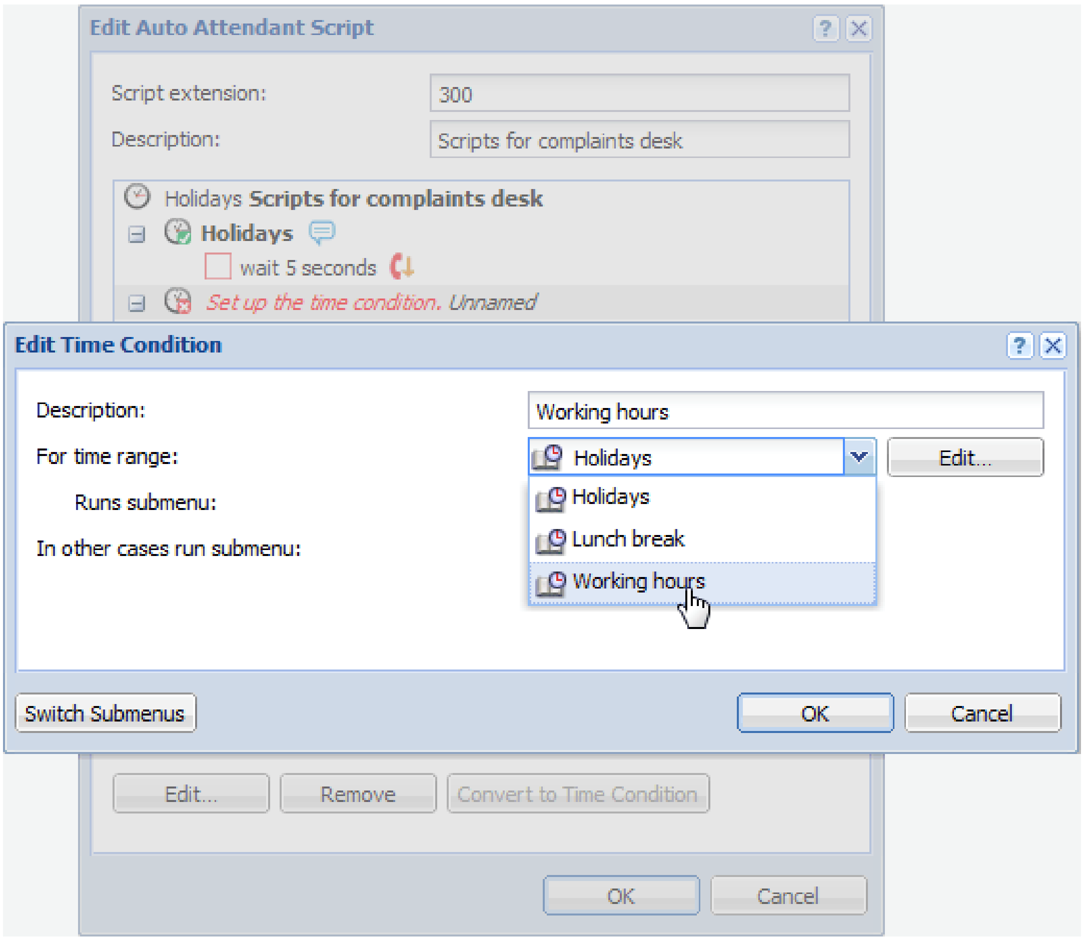


Figure 3 Auto Attendant Scripts — creating second condition

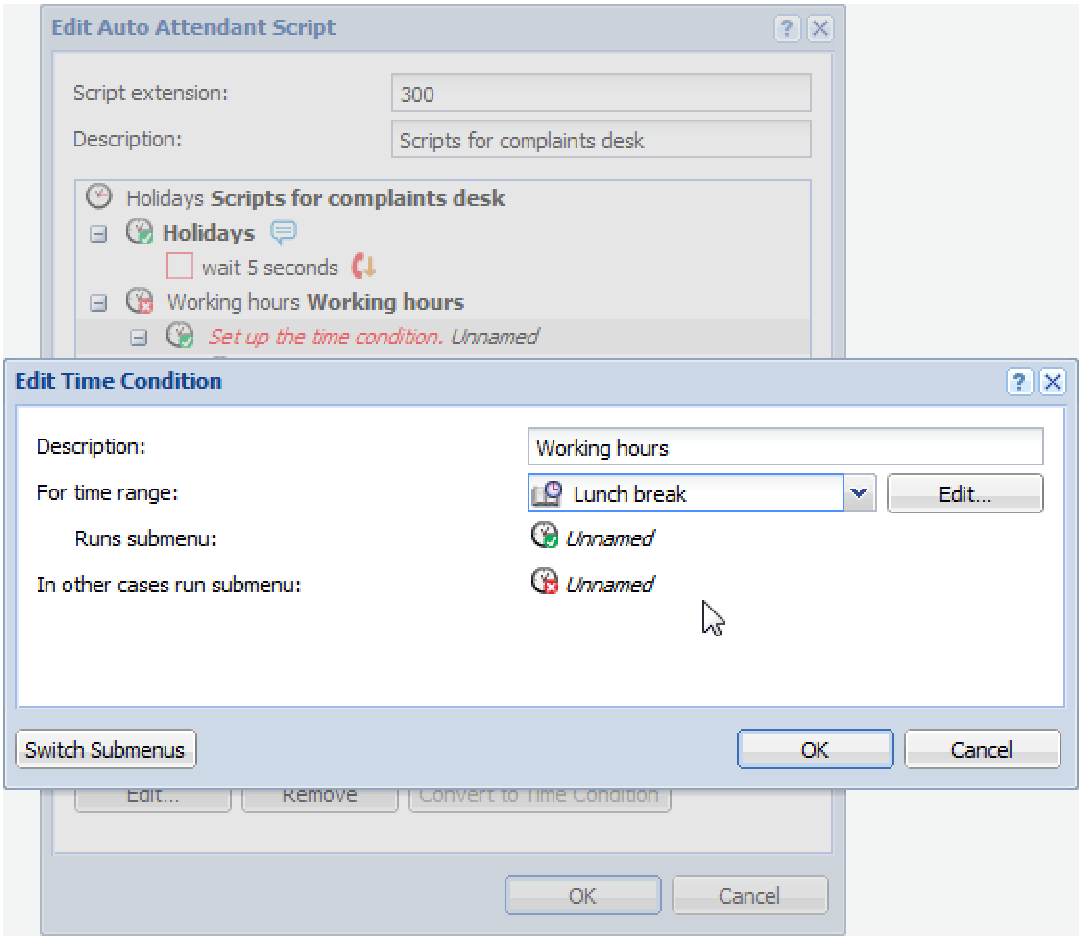


Figure 4 Auto Attendant Scripts — creating third condition

9. Double-click the last *Unnamed* icon. In the **Edit Menu** dialog, type description **Lunch break** and add a file with the announcement about a lunch break. Set timeout to 5 second and save the settings.

10. Double-click the *Unnamed* icon (last but one in the scheme). In the **Edit Menu** dialog, type description **Working hours (dial Alice)**.

    You can add an **Announcement** with information about redirecting to the Complaints department. Set **Timeout** to 1 second. In the **Default action** menu, select **Dial extension number**. Type Alice's extension (211) in the **Extension** field and save the settings.

11. Double-click the last condition (icon *Unnamed*). In the **Edit Menu** dialog, type description **After Hours** and add a file with the announcement that the Complaints department is close at the moment. Set timeout to 5 second and save the settings.

> ⚠ If you open the administration interface in Safari browser and you cannot play any recordings, read article Cannot play voicemails or audio files in Safari.

The resultant script is displayed in figure Auto attendant script for the Complaints department.

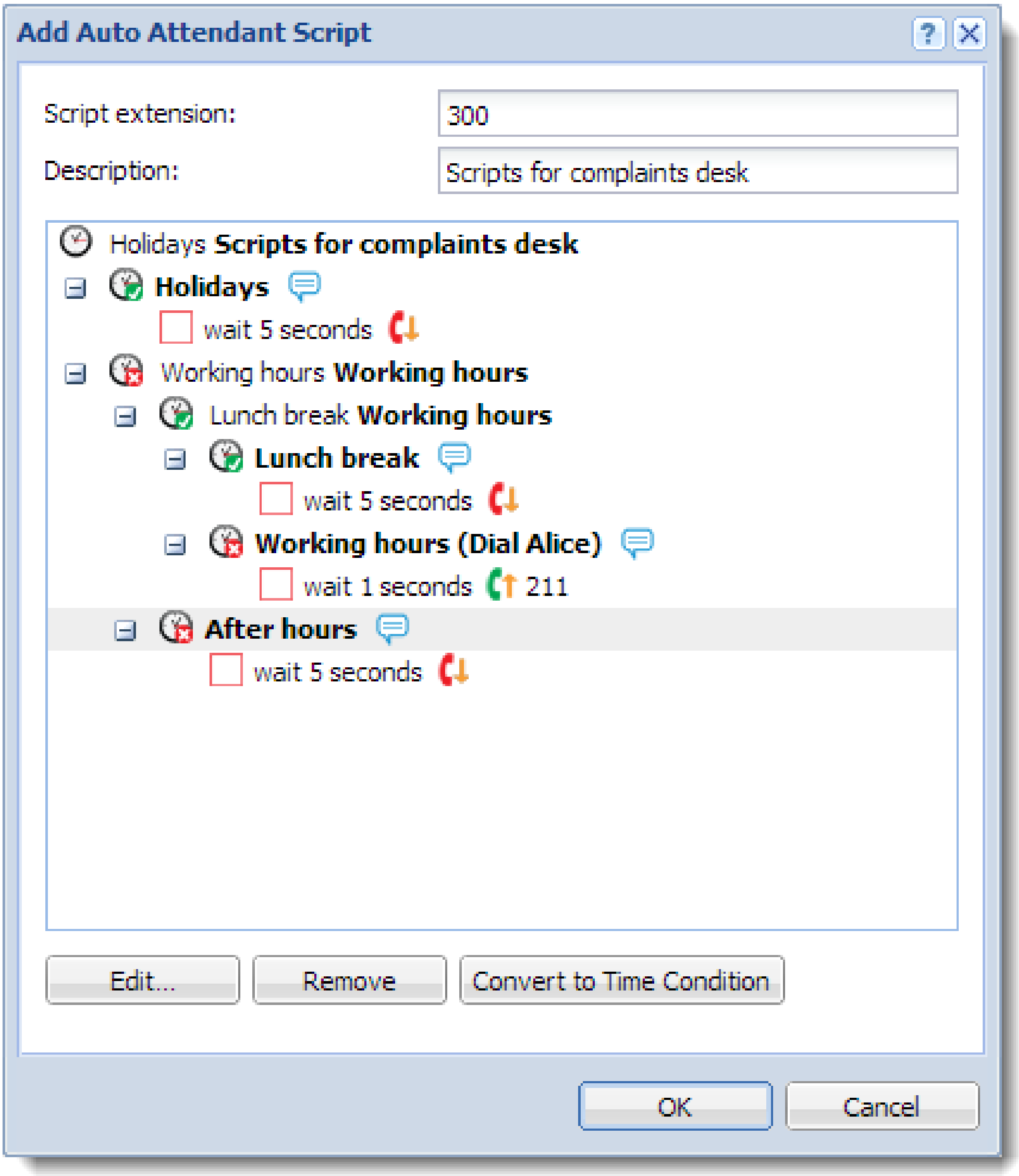


Figure 5 Auto attendant script for the Complaints department

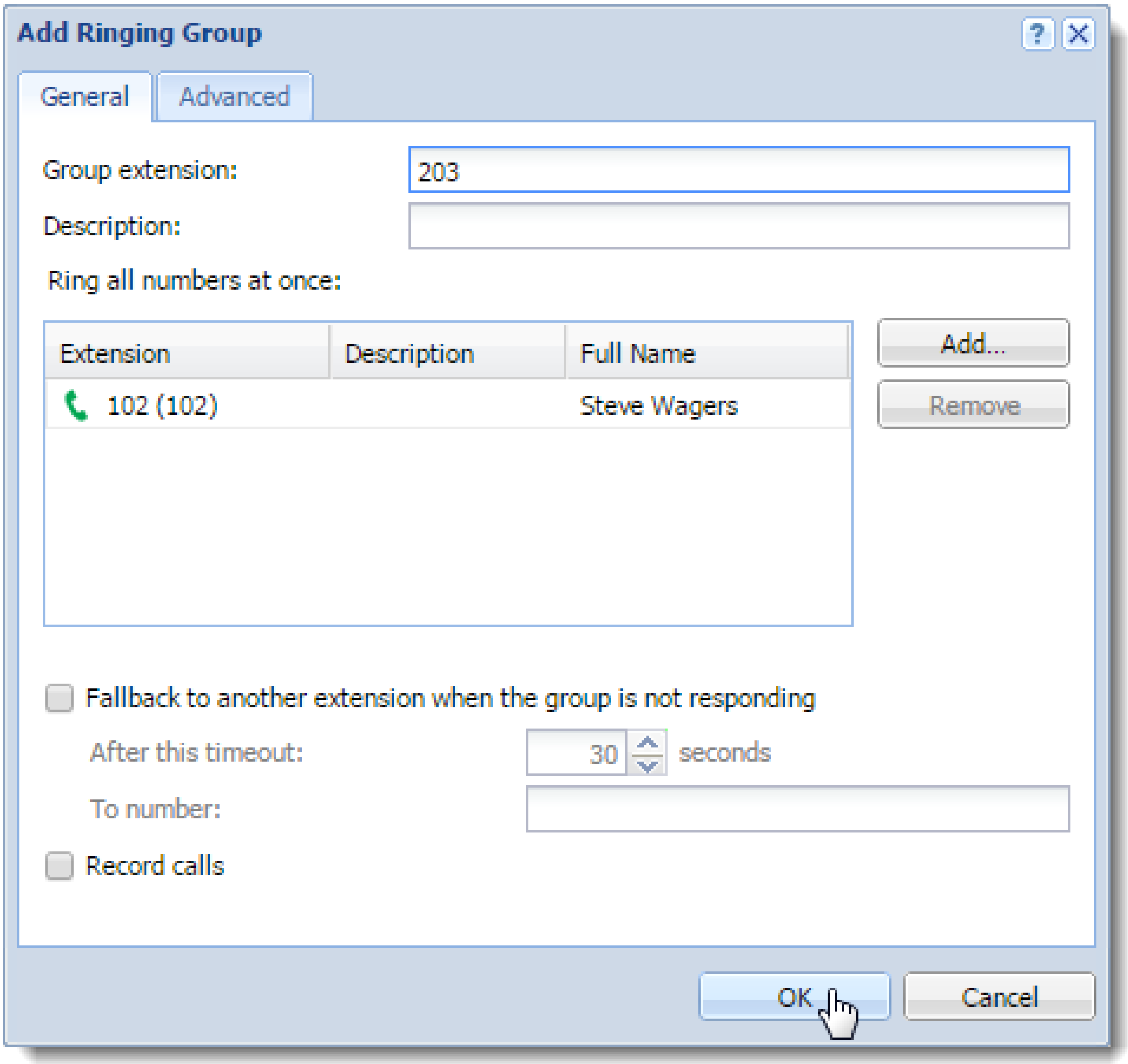
Creating ringing groups

Overview

You can use ringing groups to make calls ring simultaneously on multiple extensions.

Adding new ringing groups

1. In the Kerio Operator administration interface, go to **Ringing Groups** and click **Add**.
2. In the **Group extension** field, type the extension number for the group.
3. Add extensions you want to ring simultaneously to the table.
4. (Optional) To redirect the call to another extension when no one answers the phone, select **Fall back to another extension when the group is not responding** and set a timeout and destination extension.
5. (Optional) If you don't want to display the answered call as missed on other phones in the group, go to **Advanced** tab and select **Do not display missed calls on the phones**.
6. Click **OK**.


Add Ringing Group
General
Advanced
Group extension:
203
Description:
Ring all numbers at once:
Extension
Description
Full Name
102 (102)
Steve Wagers
Add...
Remove
Fallback to another extension when the group is not responding
After this timeout:
30
seconds
To number:
Record calls
OK
Cancel

Using PBX services

PBX services overview

Kerio Operator has special phone extensions which run the following services:

- Directed call pickup
- Call parking
- Call monitoring
- Call pickup
- Voicemail — a service extension to access voicemail. Kerio Operator recognizes which extension is used and you can set if PIN is required or not.

  This service is set automatically for provisioned phones.
- **Voicemail with login prompt** —a service extension to access voicemail. Kerio Operator is not able to recognize which extension is used. Users must authenticate with typing their extension and PIN.
- **Echo** — this option helps you monitor whether phones are correctly connected and what is the sound delay. Speak to the phone after hearing the automated message. If done correctly, your message is recorded and played back.
- **Music** — music plays upon dialing the extension (used for checking the connection).
- **Current time** — auto attendant tells the current date and time.
- **Dial by extension** — auto attendant invites the user to enter the extension which the operator will dial.
- **Dial by name** — user enters first several letters of the callee's surname and system searches among the users created in Kerio Operator and dials the extension.
- **Record audio** — Kerio Operator starts recording. Thus you can easily create records for auto attendant scripts in excellent quality.
- **Receive fax messages** — the service enables you to receive fax to defined email address. Necessary condition for enabeling the service is entering email address for receiving faxes in PDF format.

To configure PBX services, go to the administration interface → PBX Services.

If you wish to use any service, tick the box next to this service. Extensions offering the services are disabled by default.

Creating voice files

This chapter shows how to create a records for an auto attendant script easily, fast and in sufficient quality.

1. Prepare texts.
2. In the administration interface, go to **PBX Services**, enable **Record audio** and save the settings.
3. Pick up the handset of your phone which is connected to Kerio Operator.
4. Dial the *Record audio* service.
5. Say individual voice recordings into the headset.

The record is stored in the audio file library in Kerio Operator. You can listen and manage the recordings in the **Definitions** → **Audio File Library**.

If you open the administration interface in Safari browser and you cannot play any recordings, read article Cannot play voicemails or audio files in Safari.

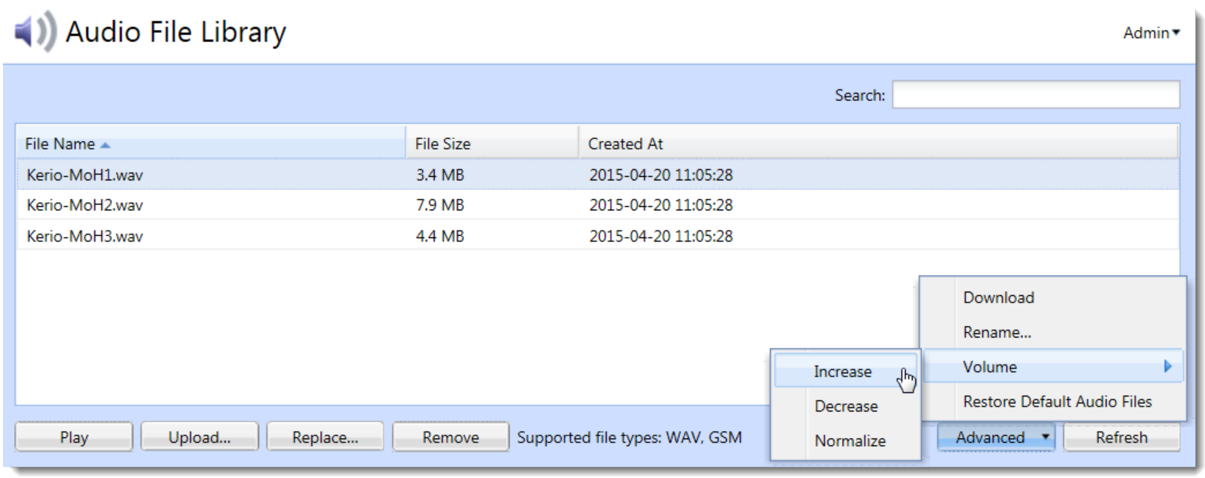


Figure 1

Configuring and using call parking

Overview

Call parking is a special type of call transfers. Parked calls wait for the callee on a special number.

Configuring call parking

You can park calls on numbers which consist of:

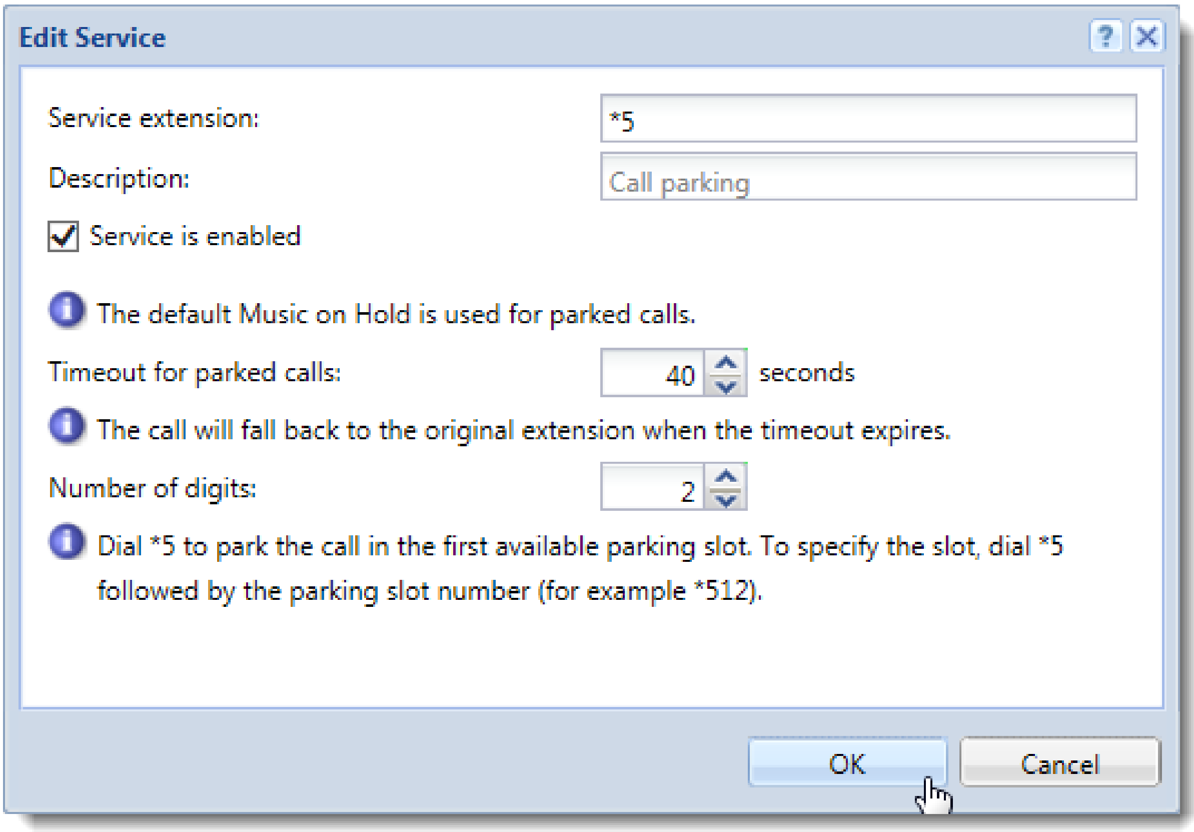
- PBX service prefix
- Parking position number

1. In the administration interface, go to the **PBX Services** section.
2. Double-click **Call parking** to open the **Edit Service** dialog box.
3. Select the **Service is enabled** option.
4. In the **Service extension** field, type the call parking prefix.

   You can leave the default prefix setting *5.
5. Set the timeout (40 seconds by default).

   When the timeout expires, the call falls back to the original extension.
6. Set the number of digits for parking positions.

   > se the same number of digits as for extensions (your dial plan). Users can park calls on positions which match their extension numbers.
7. Save your settings.



Using call parking

1. Initiate or answer the call.
2. Select the call transfer function on your phone.

   See Hardware telephone basic usage.
3. Dial the call parking number. You can:
   - Dial the whole parking slot number (for example, *512) to park the call to the specific slot.
   -

     New in Kerio Operator 2.4.2!

     Dial the **Call parking** extension only (for example, *5) to park the call in the first available parking slot.

     The voice-prompt message tells you the number of the first available parking slot.

4. Select the call transfer function on your phone.

5. Terminate the call.

To answer a parked call:

1. Pick up the phone.

2. Dial the call parking number (for example, `*512`).

If nobody answers the parked call before the timeout expires, the call falls back to the original extension.

Monitoring active calls

Call monitoring overview

New in Kerio Operator 2.3!

Call monitoring allows you to participate in any active call by dialing a special prefix, followed by an extension.

You can use call monitoring in call centers where supervisors need to monitor trainees during coversations with customers.

When you join an active call, the active callers have no indication that you have joined the call.
Call monitoring is protected by a PIN number. Whoever knows the PIN can listen to any extension in your telephony subsystem. Therefore, we recommend to set special call permissions for people who can use the call monitoring prefix.

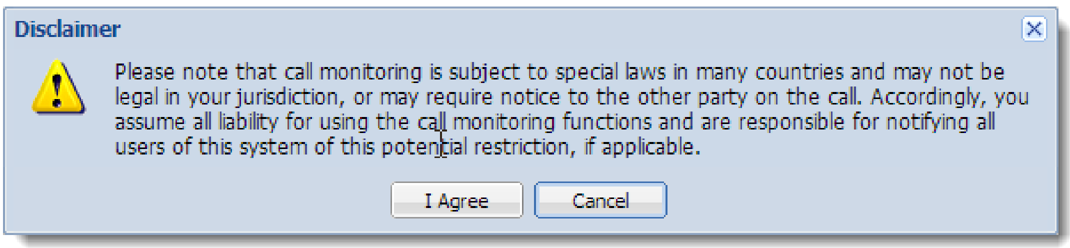
The default prefix for call monitoring is *6, and it is configured in the

PBX services. The prefix is disabled by default and you have to enable it manually.

Configuring call monitoring

To configure the call monitoring service, follow these steps:

1. In the administration interface, go to **PBX Services**.
2. Double-click **Call monitoring**.
3. In the **Edit Service** dialog, you can change the service extension.
4. Check the **Service is enabled** option.
5. Read the disclaimer carefully and click **I Agree**.



6. Click the keys icon and remember the PIN number. You can also change the PIN number.

   The PIN protects the call monitoring from misuse.

7. Select the **Audio mode**:

   - Listen only — muted

     When joining an active call in listen only mode, there is no indication to the active callers that you have joined the call.

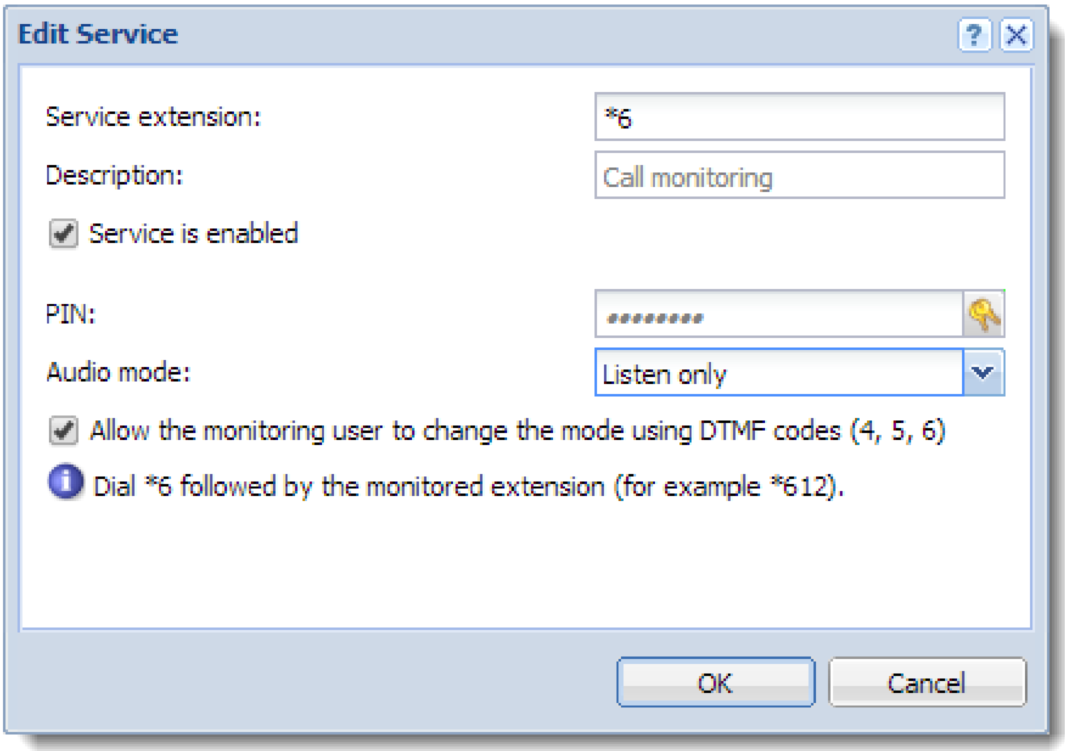
   - Whisper to the extension only — muted only to remote party

   - Talk to both — unmuted

8. To allow users to change the audio mode with DTMF codes, check the **Allow the monitoring user to change the mode using DTMF codes (4, 5, 6)** option.

   Users can change the audio mode with a key on their phone devices (**4** is for **Listen only**, **5** is for **Whisper to the extension only** and **6** is for **Talk to both**).

9. Click **OK**.



The call monitoring service is configured.

Setting call permissions

Set a call permission group for users who can use the call monitoring feature (people who knows the PIN number):

- call monitoring is allowed on extensions, which can be monitored (rules 1, 2 and 3 in the figure)
- other calls with *6 are forbidden (rule 4 in the figure)

Example:

The first three rules allow call monitoring on extensions 111, 112, 113:

1. In the administration interface, go to **Definitions** → **Call Permission Groups**.
2. Click **Add**.
3. In the **Add Call Permission Group** dialog, add the name of the group.
4. In the **Description** field, type `Group restricts call monitoring to listed extensions`.
5. Click **Add**.
6. In the **Add Prefix** dialog, type *6111.
7. Switch the rule to **Allowed** and click **OK**.
8. Repeat the steps 5, 6 and 7 for extensions 112 and 113.

The fourth rule disables general usage of *6 prefix:

1. Click **Add** in the **Add Call Permission Group** dialog (it is still opened).
2. In the **Add Prefix** dialog, type *6.
3. Switch the rule to **Denied** and click **OK**.

   Now, you can compare your result with figure. They should be the same.

   > The denial rule must be placed below the allowing rules.

4. Click **OK** in the **Add Call Permission Group** dialog.

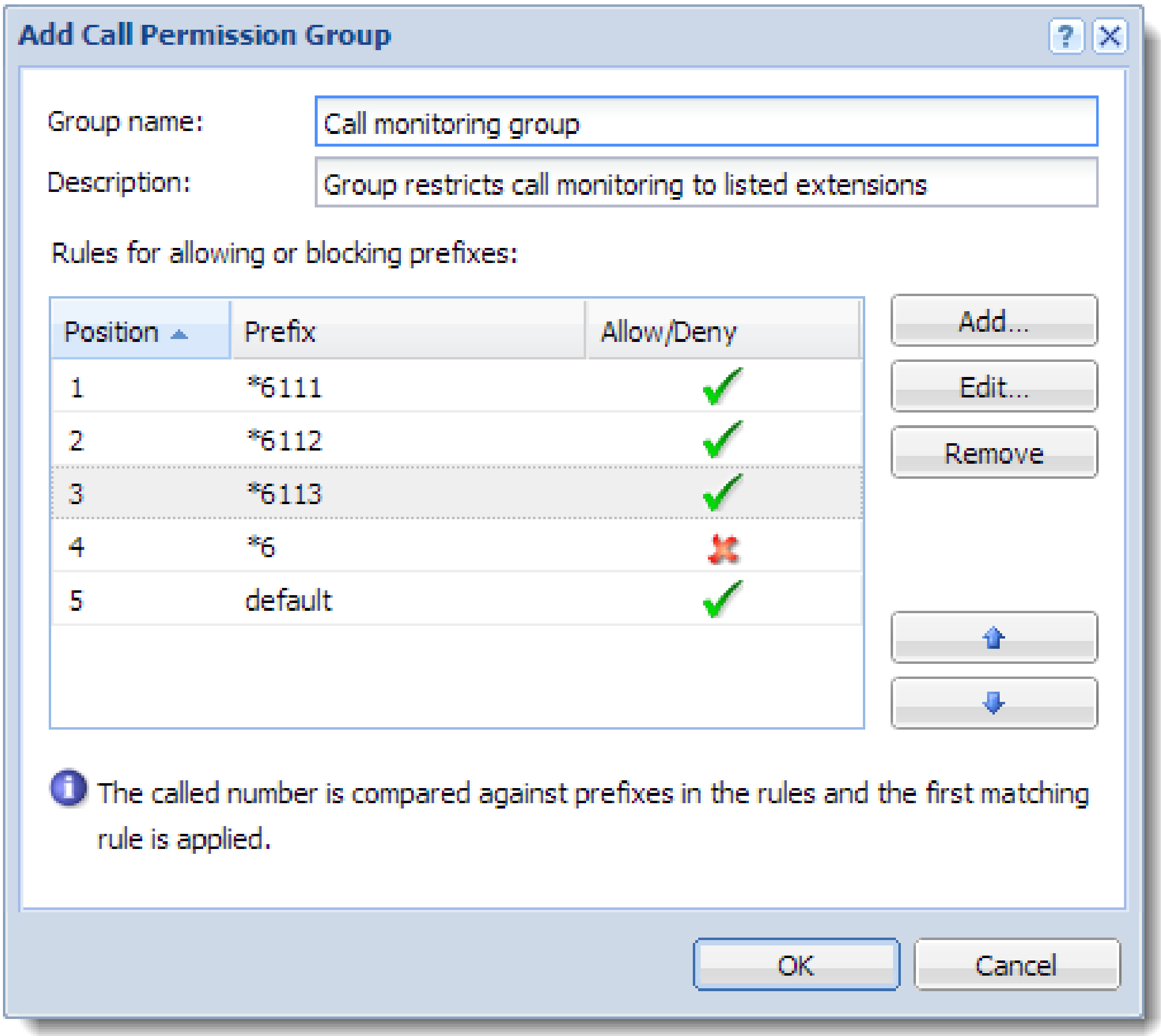


Figure 1 Call permission group for call monitoring

The group for call monitoring is established.

Now, you must assign the group to users eligible to use the call monitoring prefix and know the PIN number:

1. In the administration interface, go to **Configuration** → **Extensions**.
2. Select an extension assigned to John Smith (in figure it is extension 201) and click the **Edit** button.
3. In the **Edit Extension** dialog, change **Call permissions group** to **Call monitoring group** (see screenshot).
4. Click OK.
5. If the user has assigned more extensions, you must set **Call monitoring group** for all of them to avoid a risk of misuse of the call monitoring.

The cal monitoring group is assigned the user who is eligible to use the call monitoring prefix.

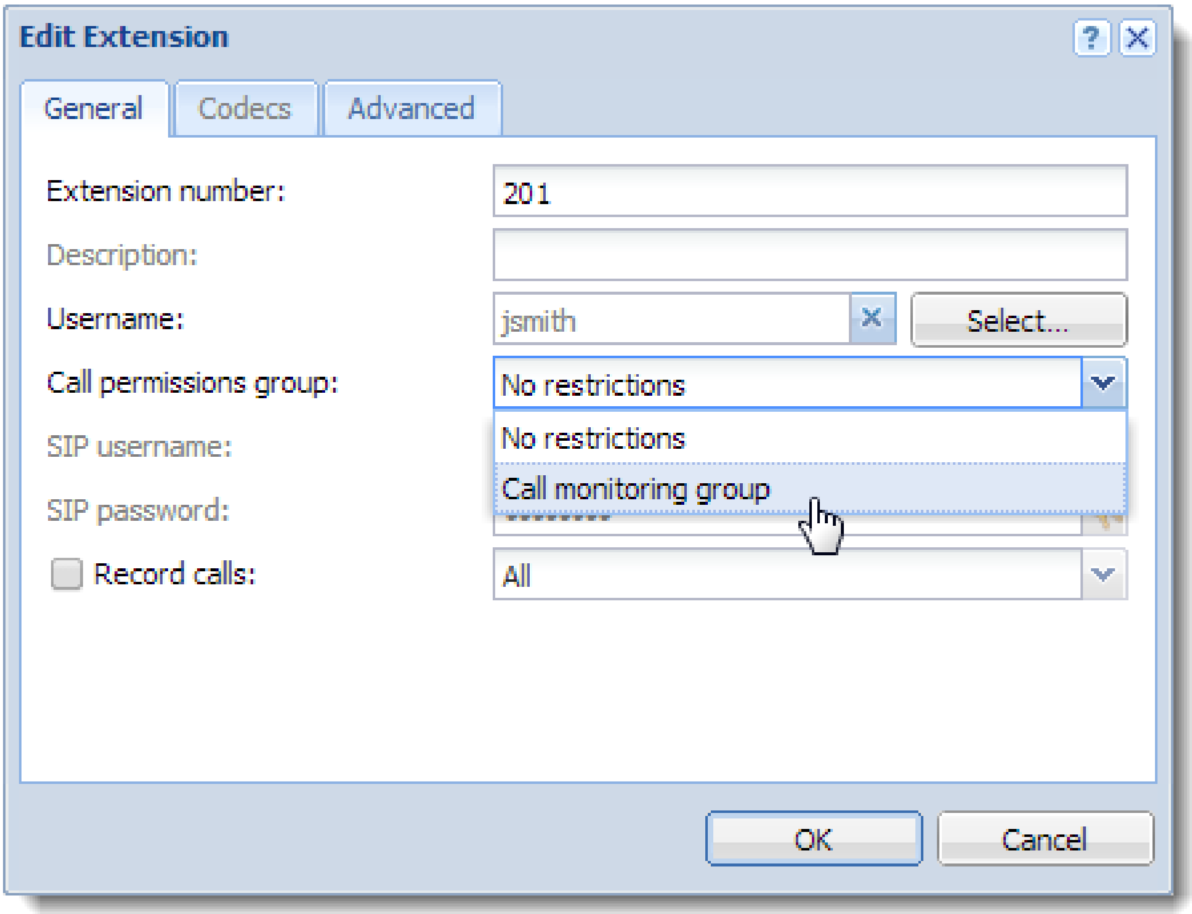


Figure 2 Call permission group for call monitoring

Using call monitoring

To use the call monitoring service you must know:

- the service extension (*6 by default),
- the PIN,
- the monitored extension (for example **111**).

For extension **111**, dial ***6111** to listen to the conversation. Then, you will be asked for the PIN number. Now, you are silently connected to the call on extension **111**.

If you are connected to the **111** extension, you can change a mode during the call (if allowed by the call monitoring service):

- press **4** for listen only mode

- press **5** for whisper to the extension only
- press **6** for talk to both

You can also monitor all employees in your office:

- extensions in your office start with 11
- five of them are assigned to employees (111, 112, 113, 114, 115)

If you dial ***611**, you can connect to the first ongoing call from all extensions starting with 11

If you dial ***61**, you can connect to the first ongoing call from all extensions starting with 1

If you dial ***6**, you can connect to the first ongoing call from all extensions of your telephony subsystem.

Pressing * key will look for another call to monitor.

As you can see, the user can monitor all calls in your telephony subsystem. Therefore, it is important to set call permissions for all users, who are eligible to use the call monitoring prefix.

Configuring call pickup

What to use call pickup for

This function enables users to answer a call ringing on an extension on a device at another extension. The PBX distinguishes between two types of call pickup:

- Call pickup within defined groups (so called rooms) by using specific code (by default, this code is *8),
- Call pickup by using a special code (by default, this code is **) with the called extension appended at the end.

How to configure call pickup rooms

1. In the administration interface, go to **Configuration** → **PBX Services**, enable **Call Pickup** and save the settings. Keep the default pickup code (*8) unless you do have a reason to change it.
2. Go to **Definitions** → **Call Pickup Rooms** and click **Add** to open dialog **Add Call Pickup Room**.
3. Type the name of the department or the office in the **Name** field.
4. In the table, add all users and extensions that will be able to pick up calls for one another.
5. Make sure the **Room is enabled** option is checked.

Example:
The Live And Let Laugh company network administrator uses the **Add Call Pickup Room** dialog to add a group with room name Local Sales for HPR (Happy people Republic). He adds all sales assistants for local market and their extensions: Frederic Jovial, Mary Merry, George Funpoker.
Frederic Jovial has a day off today. His phone is ringing. Thanks to the call pickup rooms feature, Mary Merry does not need to dash for the Frederic's desk every time a customer calls his extension. She simply dials the magic code *8 and serves the customer at her desk.

How to configure directed call pickup

Directed call pickup is a service allowing to pickup calls directed to any extension at the PBX. Imagine the following situation:

- the managing director Peter Prank uses extension 101
- the financial director Oscar Jape uses extension 102
- they share an assistant, Ms Alessandra G. Uffaw.

If Alessandra's phone shows information that someone is calling the managing director (Peter Prank) during his meeting with the financial director (Oscar Jape), she can accept the call by dialing **101. Once she picks up the call, she learns that the caller is the International laughter Association manager and arranges a meeting for him and her company's executive manager. A few minutes later, the phone at the desk of the financial director Oscar Jape starts ringing. Again, the assistant can accept this call at her desk phone. now she enters the code **102 and recommends the caller (the Cirque de Rire ringmaster) to call Mr Jape back later.

As you can see, by dialing the call pickup code, you can answer a call for any extension of the PBX.

For directed call pickup, apply settings as described below:

1. In the administration interface, go to **Configuration** → **PBX Services.**
2. Enable **Directed Call Pickup**.
3. Directed call pickup is now fully functional.

> You can use directed call pickup in Kerio Phone.

Video calling in Kerio Operator

About video calls

New in Kerio Operator 2.4!

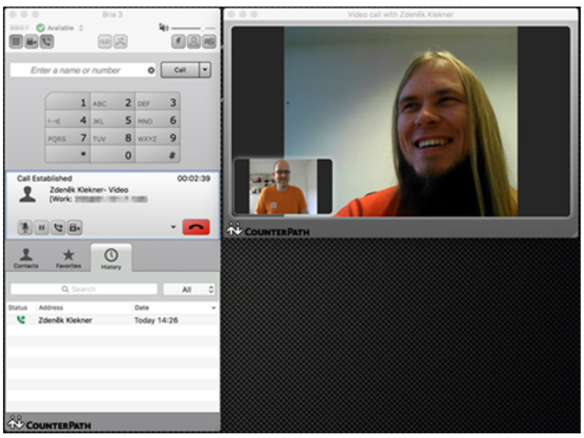
Kerio Operator now supports video calls with video enabled devices or software.

Prerequisites:

- Devices or software that use the same supported video codecs
- Configured extensions and interfaces to use the same video codec as your devices

Kerio Operator supports these video codecs (all are pass-through only):

- H.261 Video
- H.263 Video
- H.263+ Video
- H.264
- MPEG4 Video



Adding video codecs to extensions

To enable video codecs for any extension:

1. In the administration interface, go to **Configuration** → **Extensions**.
2. Select an extension and click **Edit**.

   The **Edit Extension** dialog box opens.
3. Go to the **Codecs** tab.
4. Select a codec and click **Add** to insert the codec in the **Selected codecs** list.

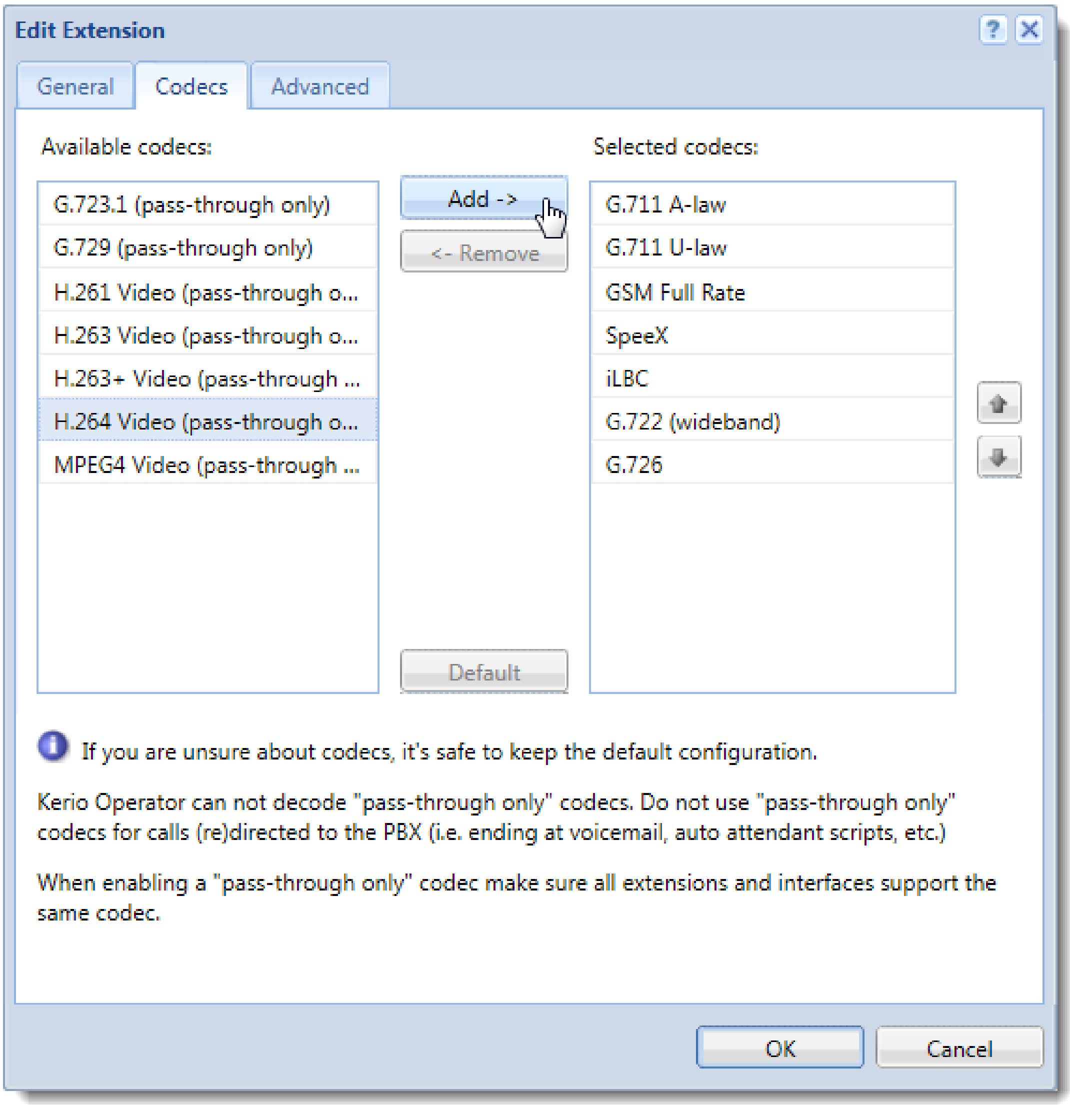


Figure 1

5. Click **OK** to save your settings.

All extensions participating in a video call must have the same codec. You can select a single codec and assign it to all your extensions.

Adding video codecs to interfaces

To enable video codecs for any interface:

1. In the administration interface, go to **Configuration** → **Call Routing** → **Interfaces and routing of incoming calls.**
2. Select an interface and click **Edit.**

   The **Edit External Interface** dialog box opens.
3. Go to the **Codecs** tab.
4. Select a codec and click **Add** to insert the codec in the **Selected codecs** list.
5. Click **OK** to save your settings.

Interfaces must have the same codecs as all extensions participating in a video call.

Troubleshooting

Video codecs in Kerio Operator are pass-through only and Kerio Operator cannot transcode them. For a proper connection, all devices must use the same codec. See the examples below:

Example of improper configuration

Device A tries to manage a video call with **Device B**:

- **Device A** works with the **H.261 Video** codec.
- **Device B** works with the **H.263+ Video** codec.

This configuration does not work, because the devices have different codecs and Kerio Operator cannot transcode them.

Example of proper configuration

Device A tries to manage a video call with **Device B**:

- **Device A** works with the **H.264 Video** codec.
- **Device B** also works with the **H.264 Video** codec.

This configuration works, because both devices work with the same codec, so Kerio Operator does not need to transcode any codecs.

Phones do not display any video

If your phone does not display any video during the call:

- Set the same codecs for each device.

  To verify which codecs devices use, see the call history.

- Lower the resolution on the caller's phone.

For example, **Grandstream GXV3272** sends video call with 720p resolution to **Grandstream GXV3140**, but **Grandstream GXV3140** cannot decode the video. User decreases the resolution on **Grandstream GXV3272** and both phones start to display the video.

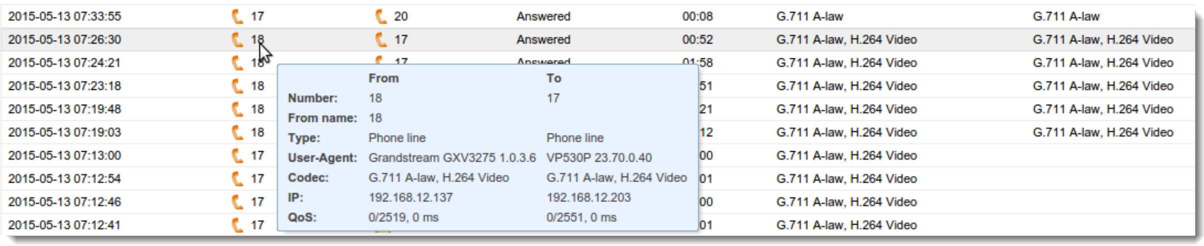


Figure 2

Phones do not transmit video

If your phone does not transmit video call, configure the device to make a video call.

For example, before you make the call, configure **Yealink VP-530** to prefer video calls.

Video is unstable

Devices with slow CPU or without a hardware acceleration can have problems with decoding the video:

- Decrease the resolution on the caller's phone.
- Verify that the network is not jammed.

  For example, transmitting a VGA signal using a **H.264 codec** takes 400 kbps in each direction.

Customizing the Kerio Phone login page

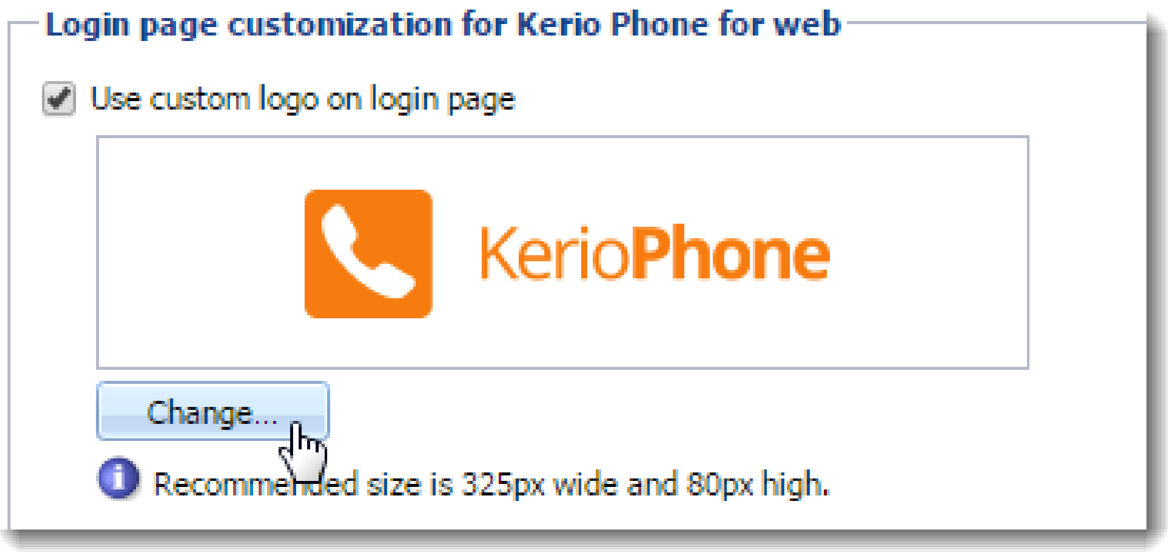
The login page of the administration login page does not change.

Adding your custom logo

To change a logo of your login page:

1. In the administration interface, go to **Configuration** → **Advanced Options** → **Login Page**.
2. Select the **Use custom logo on login page** option.
3. Click **Change** and locate the new logo file.

   The logo must be in the PNG format. The recommended maximum size is 325 x 80 pixels.
4. Click **Apply** to save your settings.



Configuring your custom button style

To change a style of a button:

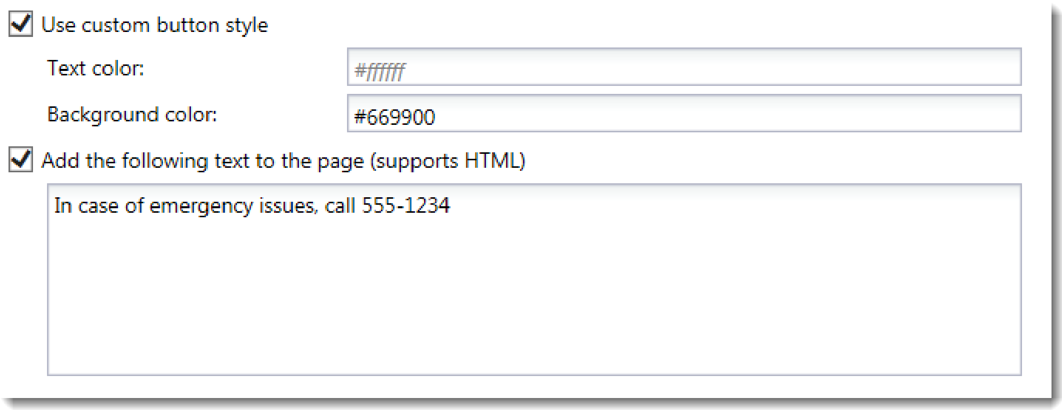
1. In the administration interface, go to **Configuration** → **Advanced Options** → **Login Page**.
2. Select the **Use custom button style** option.

3. Type a color's hex value for **Text color** (for example, #ffffff).
4. Type a color's hex value for **Background color** (for example, #669900).
5. Click **Apply** to save your settings.

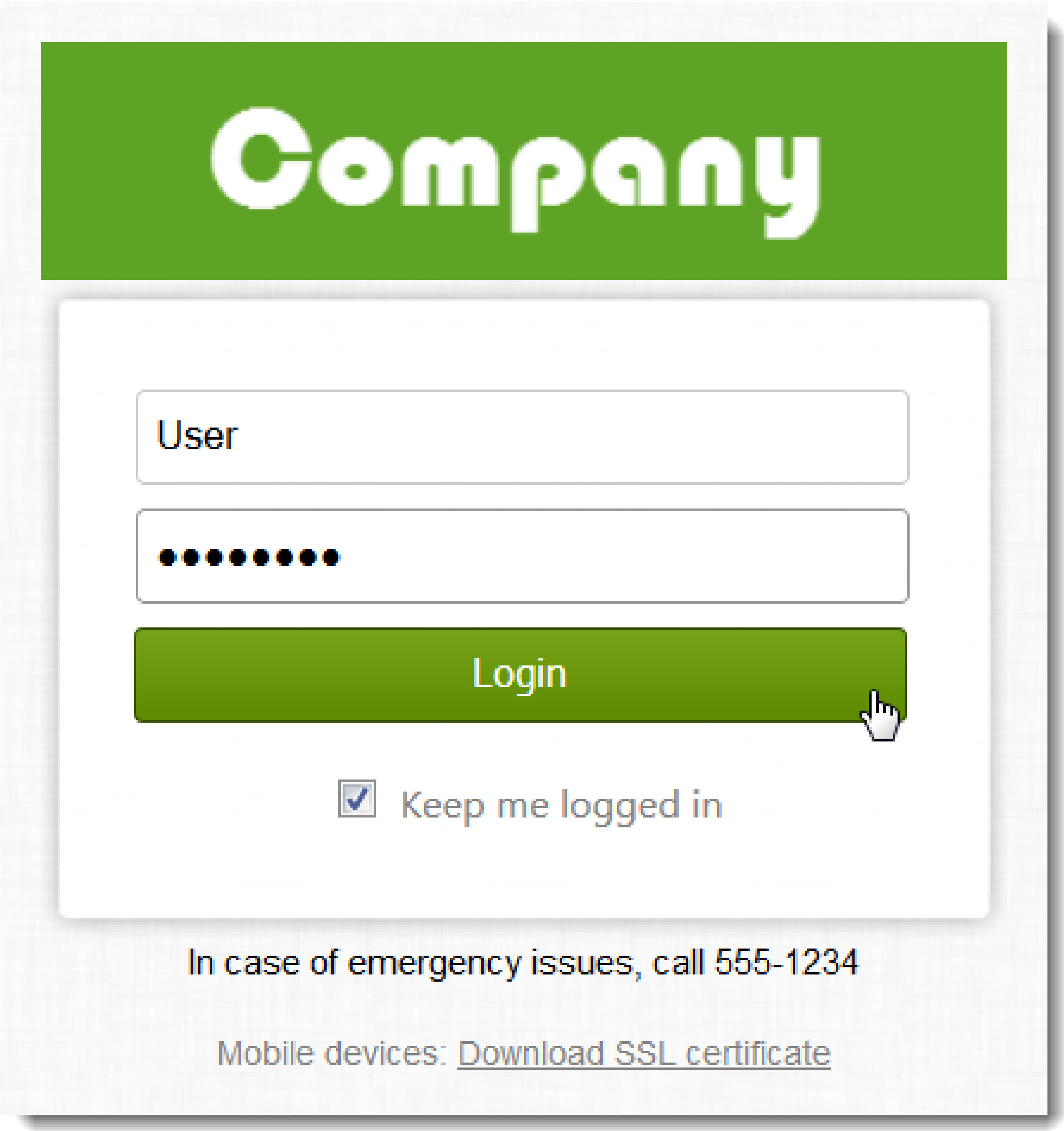
Adding your custom text

To add a text to your login page:

1. In the administration interface, go to **Configuration** → **Advanced Options** → **Login Page**.
2. Select the **Add the following text to the page (supports HTML)** option.
3. Type your text (for example, **In case of emergency issues, call 555-1234**).



4. Click **Apply** to save your settings.


Company
User
Login
Keep me logged in
In case of emergency issues, call 555-1234
Mobile devices: Download SSL certificate

Creating and using speed dial

Speed dial overview

Speed dial is a shortcut for phone numbers (for both the internal extensions and external phone numbers).

Adding speed dial

Before you begin creating speed dial, select a numerical range you will use. Speed dial must be different from current extensions. Generally, it is convenient to create speed dial so that they will not coincide with your dial plan in future.

1. Open **Speed Dial**.
2. Click **Add**.
3. In the **Add Speed Dial** dialog box, type a speed dial in the **Speed dial extension** field.
4. In **Dial number**, type the callee's phone number including the prefix for outbound calls.
5. Click **OK**.

Configuring speed dial with DTMF

The speed dial with DTMF (Dual-tone multi-frequency signaling) is intended for calling special services likelong distance phone service providers. If you need to place a call via such a service, you usually need:

- provider's number (usually it is a toll-free number that starts with 800: 800555333)
- user ID (78901234)
- PIN (8808)
- a number you want to call (011420111222333)
- # character — denotes the end of the number and starts the call.

When you set the speed dial with DTMF, the number of steps is shortened to dialing the speed dial extension followed by the number you want to call: 89011420111222333

Configuring speed dial with DTMF

1. In the administration interface, go to **Speed Dial**.
2. Click **Add**.
3. In the **Add Speed Dial** dialog box, type a speed dial in the **Speed dial extension** field (

   In our example it is 89.
4. In **Dial number**, type the provider's access number including the prefix for outbound calls.
5. Select **DTMF tones are enabled**.

   Once you enable DTMF, the speed dial behaves as a dial-out prefix.
6. In the **DTMF prefix** field, type the access code and PIN.

   Your provider's IVR system may require a pause between typing the access code and PIN. Therefore use the w character for a half second pause.

   In our example it is 78901234w8808.
7. In the **DTMF suffix** field, type #
8. Click **OK**.

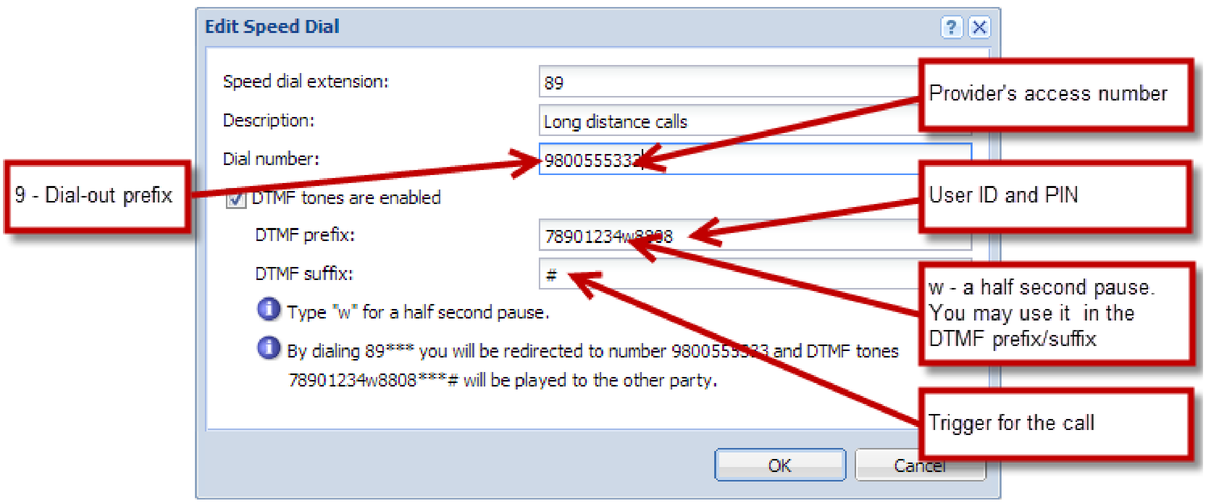


Figure 1 Add Speed Dial dialog

What is happen if you use the speed dial 89?

You want to call the number 011420111222333.

To place the call, you dial: 89011420111222333. The service will dial the access number 800555333 and once the call is connected, the following DTMF digits are sent:

789012348808 011420111222333 #

Using paging groups and services

Paging overview

Paging, also known as "intercom" or "public address", enables Kerio Operator users to broadcast a message to a user or a group using a phone's speakers. Phones included in the paging group or service answer the call automatically, and activate the loud speaker.

Paging works with phones that support auto-answer functionality.

The paging group is a group of users to whom you can make a call with using loud speaker.

The paging service is a prefix for paging. You dial the prefix + an extension to page a particular user.

Configuring paging groups

1. In the Kerio Operator administration interface, click **Paging**.
2. Click **Add Group**.
3. Type the paging group extension.
4. To add members to the group, click **Add.**
5. (Optional) Check **Page only idle extensions.**

   Paging does not interrupt active calls.
6. (Optional) Check **Beep when the call is established**.

   Your phone beeps when all phones from the paging group are connected.
7. Select audio transfer strategy:

   Select **only to the receiving party** to broadcast the message without giving paging group members ability to answer.

   Select **in both directions** to enable two-way communication.
8. (Optional) To enable call recording, select **Record Calls**.
9. Click **OK**.

If you want to check your configuration, dial the group extension and do a test call.

Configuring a paging service

1. Go to the administration interface, and click **Paging**.
2. Click **Add Service**.
3. Type **Paging service prefix**.
4. (Optional) Check **Page only idle extensions**.

   Paging do not interrupt active calls.
5. (Optional) Check **Beep when the call is established**.

   Your phone beeps when all phones from the paging group are connected.
6. Decide, if you want to transfer audio **only to the receiving party** (telephones play the message and users cannot answer) or **in both directions** (telephones play the message and users can answer).
7. (Optional) To enable call recording, select **Record Calls**.
8. Click **OK**.

If you want to check your configuration, dial the service prefix and an extension and do a test call.

Securing paging

Anyone who knows the extension or whole telephone number of the paging group can use this feature. You can secure your paging groups and service with **Call Permissions**. You can create a new call permission group, where paging an extension or a prefix is denied and add people without permission for using paging:

1. In the administration interface, go to **Definitions** → **Call Permission Groups**.
2. Click **Add**.
3. In the **Edit Call Permission Group** dialog, type a group name (for example `Paging`).
4. Click **Add**.
5. In the **Add Prefix** dialog, type a paging extension or service.
6. Click **OK**.

7. Go to **Configuration** → **Extensions**.
8. Select the user who will have paging disabled and click **Edit**.
9. In the **Call permissions group** menu, select the paging rule (in our example it is `Paging`).
10. Repeat step 9 to disable paging for additional users.

For testing purposes you can add yourself to restricted group called `Paging`. Try to call the paging group or service.

List of supported and tested phones

Paging was tested by Kerio Technologies with the following telephones:

- Cisco SPA508G, SPA525G
- Linksys SPA942, SPA922
- Polycom IP335, IP650
- Well SIP-T38G
- Snom 360, 820 and MeetingPoint

Configuring voicemail

Configure voicemail

Voicemail does not need any configuration. It works automatically once Kerio Operator starts. All users have forwarding to voicemail inbox enabled by default:

- when unavailable
- when busy

You can change the settings in section **Users** (Ringing rules). Users can also modify the settings in their Kerio Phone.

You can find the advanced voicemail configuration in the administration interface in section **Voicemail**

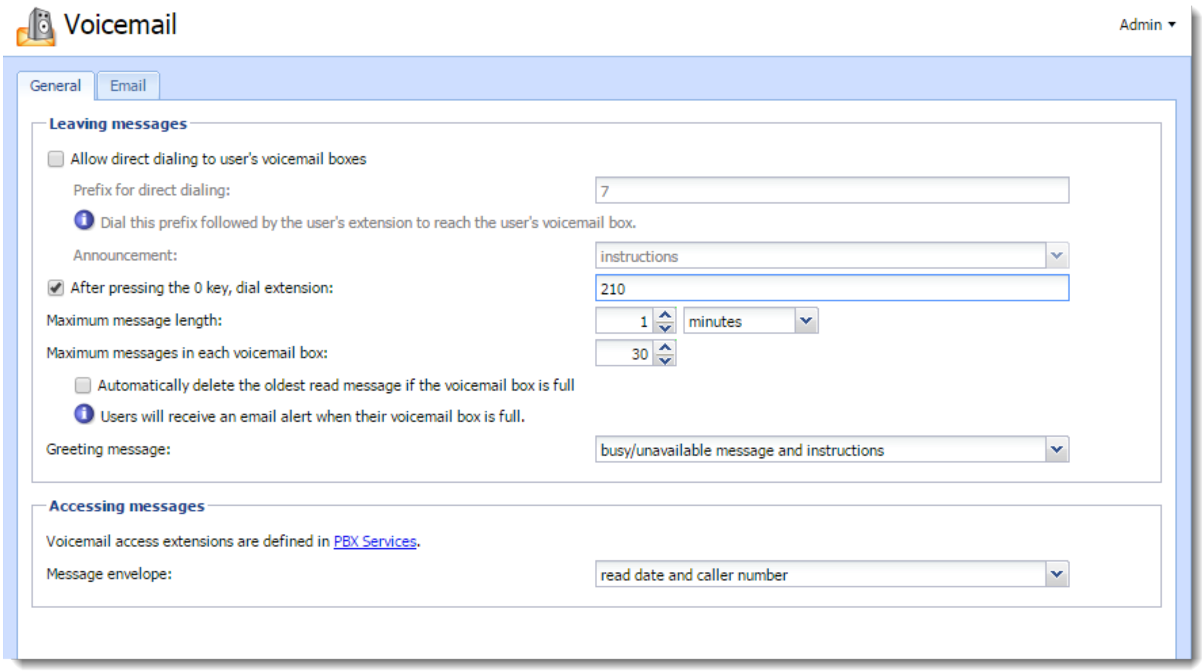


Figure 1 Configuration → Voicemail

What is direct access to voicemail inbox and how to configure it

Direct access to users' voicemail enables the receptionist to connect calls directly to callee's voicemail.

1. In the administration interface, go to **Voicemail** → **General**.
2. Check **Allow direct dialing to user's voicemail boxes**.

3. Type a prefix in **Prefix for direct dialing.**

4. (Optional) Set an announcement (greeting message). If a call is redirected to voicemail, the caller hears a recorded message. This message can consist of two parts:

    - **Instructions** inform callers what they should do next: "Leave a message after the beep".

    - **Message** informs callers that the callee is unavailable (the phone is switched off) or busy (the callee speaks with someone else).

5. (Optional) To change the size of users' voicemail boxes, adjust the value in **Maximum messages in each voicemail box.**

6. (Optional) To automatically delete read messages in full voicemail boxes, select **Automatically delete the oldest read message if the voicemail box is full.**

7. Click **Apply**.

Now the receptionist can dial the extension for direct access followed by the user's extension. The caller will be directed to the voicemail box of the person they are calling.

Enabling caller to escape voicemail by dialing 0

If you want to enable escaping voicemail by dialing 0, you must set an extension where the call is redirected:

1. In the administration interface, go to **Voicemail** → **General**.

2. Select **After pressing the 0 key, dial extension**.

3. Type an extension.

4. Click **Apply**.

Configuring forwarding of voicemail messages to user's email inbox

To send voicemail messages to email inboxes of the users, you need to set their email addresses in the administration interface in **Users.**

> If the users' INBOXes are unavailable (the mailserver is down), the user accounts are disconnected from voicemail and try to reconnect every 5 minutes. Each attempt to connect is recorded in logs.

My mailserver is Kerio Connect

You can find more information in article Integrating Kerio Connect and Kerio Operator.

My mailserver is MyKerio

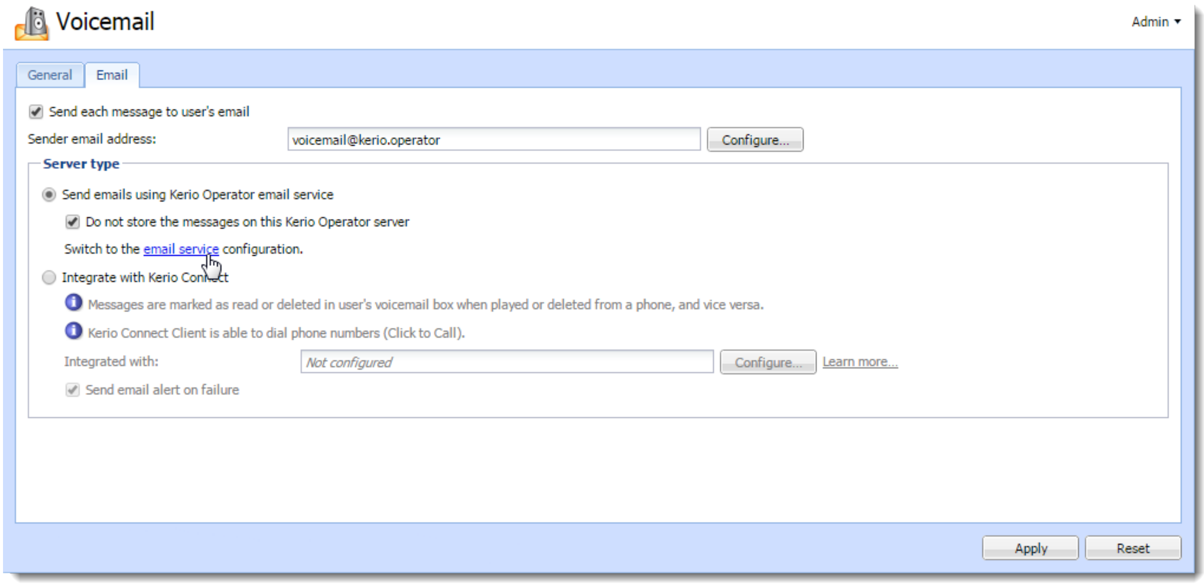
To use MyKerio as an email service, you must enable the communication between Kerio Operator and MyKerio first.
For detailed information, see Adding Kerio Operator to MyKerio.

1. In the Kerio Operator administration interface, go to **Configuration** → **Voicemail.**
2. Switch to the **Email** tab.
3. Select **Send each message to user's email.**
4. (Optional) Modify the format of the sender's email address.
5. In the **Server type** section, select **Send emails using Kerio Operator email service.**
6. (Optional) To save disk space on your Kerio Operator server, select **Do not store the messages on this Kerio Operator server**.
7. Click **Apply.**
8. Click the **Email service** link.

   Kerio Operator redirects you to **Advanced Options** → **General**.
9. In the **Email service** section, select **MyKerio.**
10. Click **Apply.**

You can now test the email service:

1. In the **Email service** section, click **Test**.
2. Type your email address.
3. Click **Send**.



My mailserver is a different SMTP server

1. On your mail server, create a special user which will be used for sending the voicemail messages. You can name them for example `operator`.
2. Go to administration interface to **Voicemail** → **tab Email** and check **Send each message to user's email**.
3. In the **Server type** section, select **Send emails using Kerio Operator email service**.
4. (Optional) To save disk space on your Kerio Operator server, select **Do not store the messages on this Kerio Operator server**.
5. Click **Apply**.
6. Click the **Email service** link.

   Kerio Operator redirects you to **Advanced Options** → **General**.
7. In the **Email service** section, select **SMTP server**.
8. Click **Configure**.
9. In **Hostname**, type the SMTP server hostname.
10. Set the port number of the port used by your SMTP server.

    Usually 25 for SMTP and 465 for SMTPS
11. Decide, whether to communicate through secured connection. If the configuration of your mail server allows it, we recommend the encrypted connection to establish more secure communication.

12. If your SMTP server requires authentication, check **Server requires authentication**. Use the username and password for the account you created on your mail server in step 1.

13. Type a valid email address in **Sender email address** (so that your antispam rules accept it). The address should also represent the origin of the message. Example: `operator@live-and-let-laugh-inc.com`

14. Click **Save**.

Configuring the welcome message for callers

If a call is redirected to voicemail, the caller hears a recorded message. This message can consist of two parts:

Instructions inform callers what they should do next: “Leave a message after the beep”.

Message informs callers that the callee is unavailable.

How to set the greeting message?

1. Open section **Voicemail**.

2. In the **Greeting message** menu, select whether the caller will hear the instruction, the message or both.

Greeting message for the direct dialing is described in the What is direct access to voicemail inbox and how to configure it section.

Changing the extension and voicemail PIN

Users use a special extension number to access their voicemail (by default: 50 or 51) and PIN.

To change the extension or enable/disable PIN, go to section **PBX Services** and read article Using PBX services.

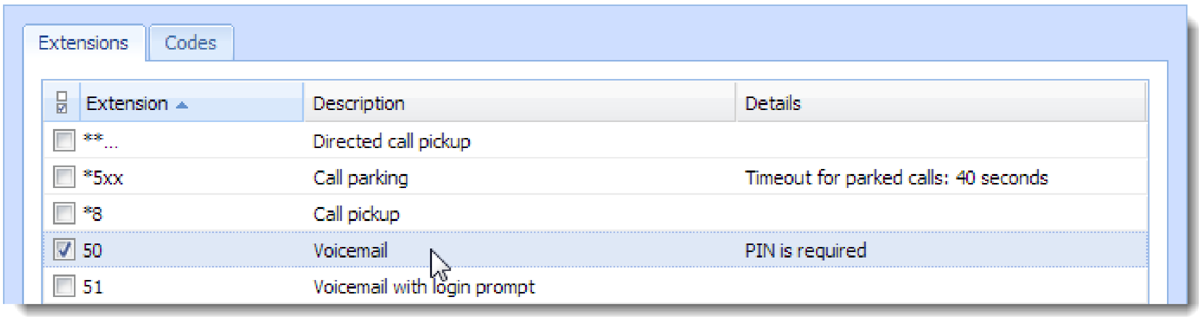


Figure 2 PBX Services

To set the user's PIN, go to account configuration in section **Users** to tab **Extensions**.

Accessing voicemail

- On your phone, press voicemail button or dial voicemail number and play the message.
- Through Kerio Phone.

> For users of Apple iPhone, iPad or Apple Mac OS X: If you cannot play your voicemail messages in Kerio Phone, contact the Kerio Operator administrator. An invalid certificate may be the reason.

- By forwarding voicemail to your mailbox (to get more information on this option, contact your network administrator).

Removing voicemail data for selected user

You can remove all local data connected with the particular user.

Local data is:

- voicemail
- custom voicemail greeting message

> Local data means that you cannot use this feature when you use the Kerio Connect integration — voice messages are stored in Kerio Connect in this case.

1. In the administration interface, go to **Users**.
2. Right-click the table heading.
3. In the context menu, select **Columns** → **Voicemail** and **Columns** → **Local Voicemail Size**.
4. Right-click the selected user and click **Erase Local Voicemail Data**.

If you succeed, there is value `0B` in the **Local Voicemail Size** column.

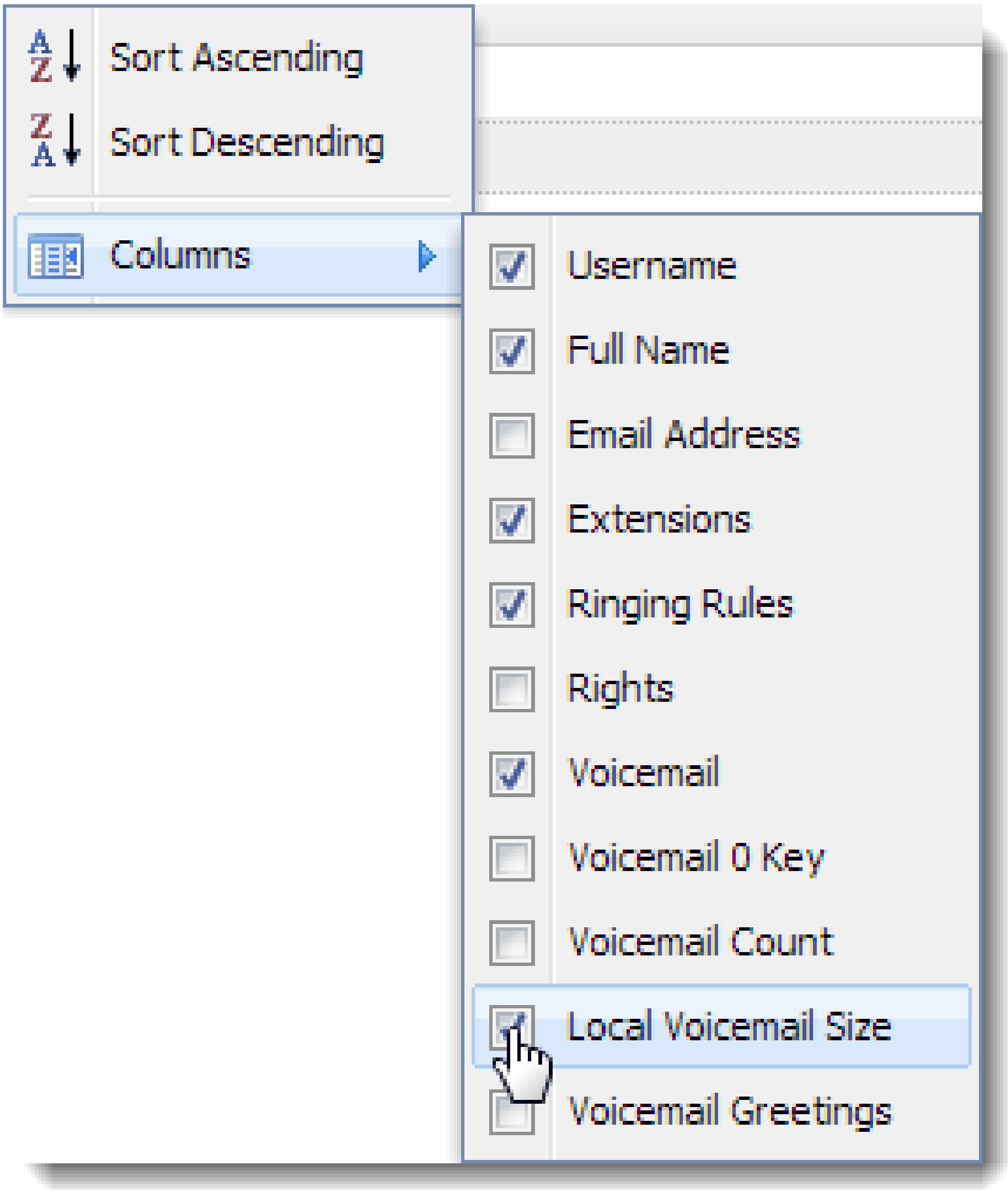


Figure 3 Table context menu

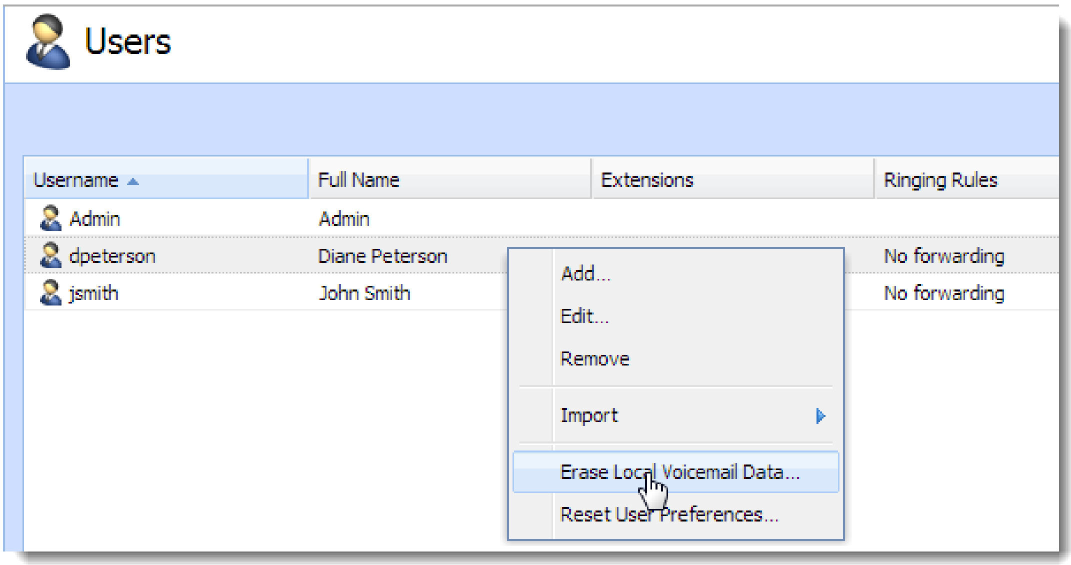


Figure 4 User's context menu

Managing voicemail via Kerio Phone

You can find more information in the Using voicemail section.

Voicemail Menu

For more information visit the Voicemail Menu section.

Integrating Kerio Connect and Kerio Operator

What are the possibilities of Kerio Operator and Kerio Connect integration

There are several posibilities how to integrate Kerio Operator and Kerio Connect:

Integrating voicemail

The integration synchronizes flags which marks whether a voicemail message has been read/played. If you mark a message as read in Kerio Phone or if the message is marked as read after you hear it on your phone, the message will also be flagged as read in your mailbox (and vice versa).

If integration with Kerio Connect is set, voicemail messages are not stored in Kerio Operator but in user's **Inbox** on the mailserver.

Limitation: You can integrate Kerio Connect with a single Kerio Operator only.

Searching the address book on Kerio Connect on provisioned phones

For more details, refer to the Accessing company contacts through LDAP on provisioned phones article.

Calling directly from Kerio Connect Client

New in Kerio Operator 2.3!

Users of Kerio Connect Client can click a contact's phone number to initiate a call from Kerio Operator. By clicking a number, you can select the registered phone/device to dial from. The selected phone/device will ring. Answer the call and Kerio Operator will place the outbound call to the dialed number.

To set up and use the Click to Call feature in Kerio Operator, go to the Using number transformation article.

To set up and use the Click to Call feature in Kerio Connect, go to the Integrating Kerio Connect with Kerio Operator article.

If you want to Click to Call for Kerio Operator plugin for Chrome and Firefox, go to the Using Click to Call for Kerio Operator plugin for Chrome and Firefox article.

Configuring voicemail integration

If you want to set up voicemail integration, follow these steps:

1. Go to **Configuration** → **Users**.
2. In the users' settings, type their email addresses.

> Use the primary email address (not an alias) — otherwise sending of messages to Inbox will not work.

3. Go to **Configuration** → **Voicemail** → **Email**.
4. Change the SMTP server settings to **Integrate with Kerio Connect**.
5. Click **Configure** and type the DNS name of Kerio Connect.

> If the IMAP service runs on a nonstandard port in Kerio Connect, enter the server name including the port number (`hostname:12345`)

6. Specify the name and password of a user with admin rights for Kerio Connect.

Authentication details are used for the first connection to Kerio Connect and creation of a special account using JSON-RPC2 API for authentication. Once this special account is created, the PBX drops the administrator's name and password.

> To synchronize flags between the two servers, Kerio Operator uses protocol IMAP with TLS or IMAPS. If Kerio Connect is behind firewall, enable at least one service on standard port. The IMAP or IMAPS services need to be allowed on Kerio Connect server.

Troubleshooting

If Kerio Connect is protected by firewall, open the ports for the IMAP/IMAPS protocols.

The IMAP/IMAPS services must be running in Kerio Connect.

If you cannot connect Kerio Operator with Kerio Connect, consult the following logs:

- In Kerio Operator, consult the Warning log for any problems with the IMAP service.
- In Kerio Operator, consult the Error log for problems with connection to Kerio Connect's IMAP server.
- In Kerio Connect, consult the Mail log for information about delivered voicemails.

Setting emergency numbers

Emergency numbers overview

When configuring emergency numbers, you can:

- add emergency numbers to the system,
- enable direct dialing (without the prefix for calling external networks).

Call permitions and security restrictions are not applied to emergency numbers.

Configuring emergency numbers

1. In the administration interface, go to **Configuration** → **Call Routing**.
2. Click the **Emergency Numbers** button placed in the lower left corner.
3. Click **Overwrite** and select the country.
4. If the lists of emergency numbers do not suit your needs, click **Add** to create your own emergency numbers.

Enabling direct dialing

All outgoing calls to external networks use a prefix. You can configure an exception for emergency numbers:

1. In the administration interface, go to **Configuration** → **Call Routing**.
2. Click the **Emergency Numbers** button placed in the lower left corner.
3. Check **Enable direct dialing**.
4. Select **Used outgoing route**.

   This route will be used for all calls to the emergency numbers.

If the direct dialing is enabled, you cannot create extensions which equal the emergency numbers.

Configuring SSL certificates

SSL certificates overview

To secure the PBX by SSL/TLS encryption, you need a SSL certificate. SSL certificates authenticate an identity on a server.

Kerio Operator creates the first self-signed certificate during the installation. The server can use this certificate but users will have to confirm they want to go to an untrustworthy page. To avoid this, generate a new certificate request in Kerio Operator and send it to a certification authority for authentication.

- If you use the Safari browser in your environment (on Apple iPhone, Apple iPad, Mac OS X systems and on Microsoft Windows), you will not be able to play voice messages in Kerio Phone on their devices with a self-signed certificate. You must have a trustworthy certificate available.
- If you use a self-signed certificate, users with Apple mobile devices will not be able to play voice messages in Kerio Phone on their devices. They must have a trustworthy certificate available.
- To encrypt the communication between Kerio Operator and hardware phones (and only a self-signed certificate available), you have to import or configure information in the phones that the invalid certificate is to be ignored.

Creating self-signed certificates

To create a self-signed certificate, follow these instructions:

1. In the Kerio Operator administration interface, open section **Definitions** → **SSL Certificates**.
2. Click **New** → **New Certificate**.
3. In the **New Certificate** dialog box, type the hostname of Kerio Operator, the official name of your company, city and country where your company resides and the period of validity.

   The **Hostname** and **Country** entries are required fields.

4. Click **OK**.

5. To enable the server to use this certificate, select the certificate and click **Set as Active**.

Creating certificates signed by certification authority

If you wish to create and use a certificate signed by a trustworthy certification authority, follow these instructions:

1. In the Kerio Operator administration interface, open section **Definitions** → **SSL Certificates**.

2. Click **New** → **New Certificate Request**.

3. In the **New Certificate Request** dialog box, type the hostname of Kerio Operator, the official name of your company, city and country where your company resides and the period of validity.

   The **Hostname** and **Country** entries are required fields.

4. Click **OK**.

5. Select the certificate and click **Export**.

6. Save the certificate to your disk and email it to a certification organization.

7. Once you obtain your certificate signed by a certification authority, go to **Definitions** → **SSL Certificates**.

8. Click **Import**.

9. To enable the server to use this certificate, select the certificate and click **Set as Active**.

Intermediate certificates

Kerio Operator supports **intermediate** certificates.

To add an intermediate certificate to Kerio Operator, follow these steps:

1. In a text editor, open the server certificate and the intermediate certificate.

2. Copy the intermediate certificate into the server certificate file and save.

   The file may look like this:

```
MIIDOjCCAqOgAwIBAgIDPmR/MA0GCSqGSIb3DQEBBAUAMFMxCzAJBgNVBAYTAl
MSUwIwYDVQQKExxUaGF3dGUgQ29uc3VsdGluZyAoUHR5KSBMdGQuMR0wGwYDVQ
     ..... this is a server SSL certificate ...
ukrkDt4cgQxE6JSEprDiP+nShuh9uk4aUCKMg/g3VgEMulkROzFl6zinDg5grz
```

```
QspOQTEYoqrc3H4Bwt8=
-----END CERTIFICATE-----
-----BEGIN CERTIFICATE-----
MIIDMzCCApygAwIBAgIEMAAAATANBgkqhkiG9w0BAQUFADCBxDELMAkGA1UEBh
WkExFTATBgNVBAgTDFdlc3Rlcm4gQ2FwZTESMBAGA1UEBxMJQ2FwZSBUb3duMR
     ..... this is an intermediate SSL certificate which
           signed the server certificate...
5BjLqgQRk82bFi1uoG9bNm+E6o3tiUEDywrgrVX60CjbW1+y0CdMaq7dlpszRB
t14EmBxKYw==
```

3. In the administration interface, go to section **Definitions** → **SSL Certificates.**

4. Import the modified server certificate by clicking on **Import** → **Import a New Certificate.**

If you have multiple intermediate certificates, add them one by one to the server certificate file.

Securing Kerio Phone with SSL certificates

For more information about securing Kerio Phone, see the Securing Kerio Phone with SSL certificates section in the **Provisioning for Kerio Phone** article.

Language settings in Kerio Operator

Languages in Kerio Operator are:

- Application language — language for the administration interface and for Kerio Phone.
- PBX language — “the voice of the PBX”. Voice records which are used for communication with users (internal and external).

You can also change the type of indication tones according to individual countries (read section Changing indication tones according to countries).

Changing the application language

The language for the administration and softphone interfaces can be set in the **Admin** menu in the right top corner of the of the application window.

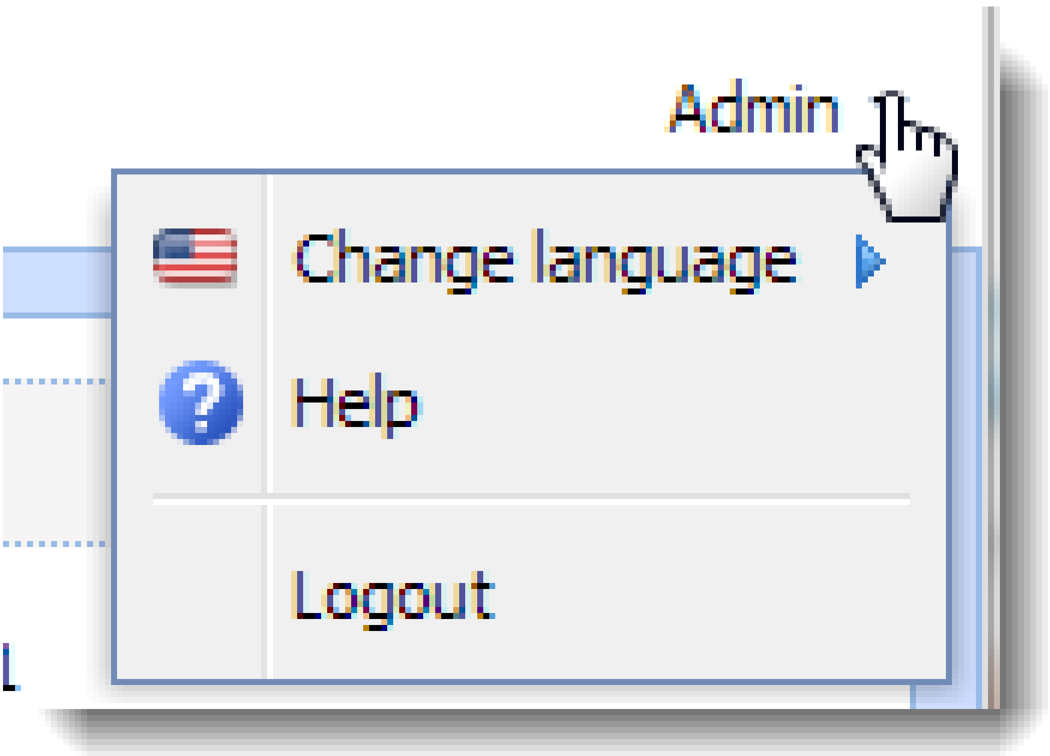


Figure 1 Changing the application language

Changing the language of the PBX

You can change the default language of the PBX in the administration interface in section **Configuration** → **Advanced Options** → **Telephony**.

There, you can also upload new language version or different voice records of the same language (for example, less formal records).

When setting language, bear in mind the following rules:

- Default language set in section **Advanced Options** → **Telephony** has lower priority than settings of individual users in section **Users**. If users do not have any language set, the default one is used.
- Default language set in section **Advanced Options** → **Telephony** has lower priority than settings for interfaces for incoming calls (section **Call Routing**). The language set for the interface of incoming calls has lower priority than files uploaded to call queues (see screenshot below). If no language is set, the default one is used. The same goes for call queues.

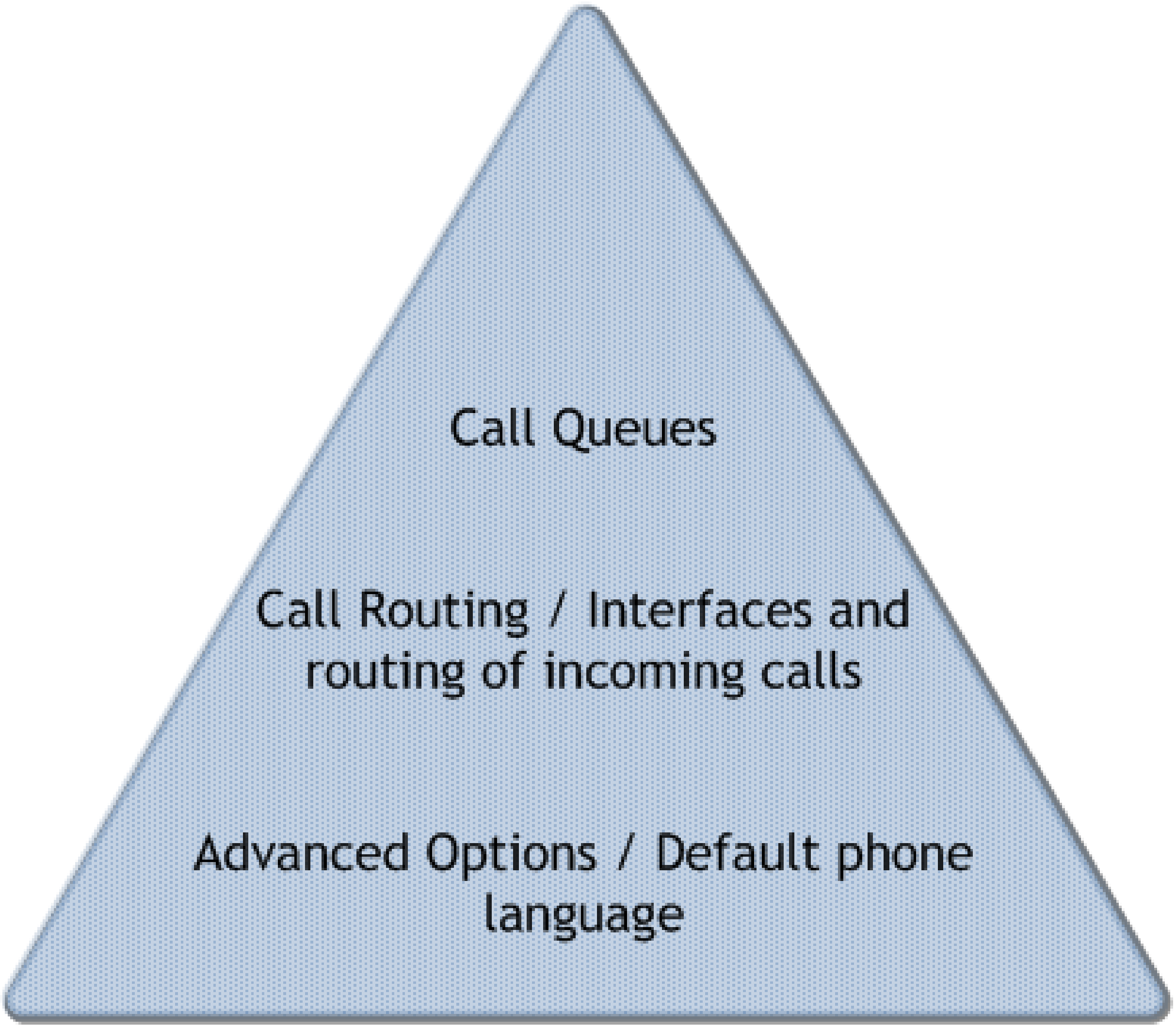


Figure 2 Language settings priority

How to change the language for individual users

Thomas Punchline, the network administrator at Live And Let Laugh Inc, faces the following problem: New employee has arrived in the company. Alessandra G. Uffaw has moved from the Bliss Seekers Land to the Happy People Republic and cannot speak the Happish language. She complains she can't understand her voicemail. Thomas has to switch the PBX language to the Cravish language for her. Do you need to solve a similar problem? Check the following example:

1. In the administration interface, go to **Configuration** → **Users**.
2. In the user's settings, go to tab **General** and change the **Phone language**.

How to change the PBX language for a group of users

Thomas was instructed to create a new interface in Kerio Operator and change its language to the Cravish. He has to create a new interface for incoming calls and set a language for this interface. He called his VoIP service provider and purchased new phone numbers for the employees who will communicate with foreign customers. And how he configured Kerio Operator?

1. In the administration interface in section **Configuration** → **Call Routing** , add a new route for incoming calls.

2. Connect it to the provider, open the edit dialog by clicking on the route in table **Interfaces and routing of incoming calls**.

3. Select a language on tab Advanced.

4. Select a country on tab Advanced.

   Each country has different standards for indication tones during calls (e.g. beeps, ringing tones, etc.).

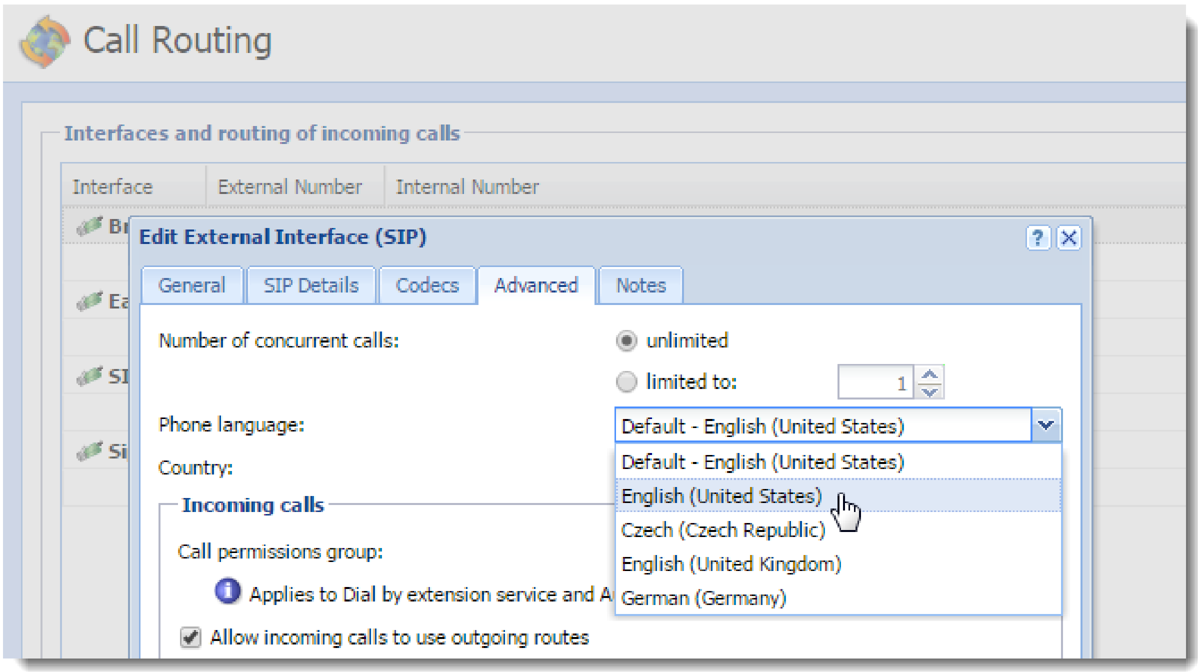


Figure 3 Changing a language for an entire route

Setting a different language for a call queue

If you wish to change the language for call queues, not for the entire route, go to section **Configuration → Call Queues.**

> Language files used in call queues has automatically higher priority than language set for incoming calls.

How to add a new language to the PBX

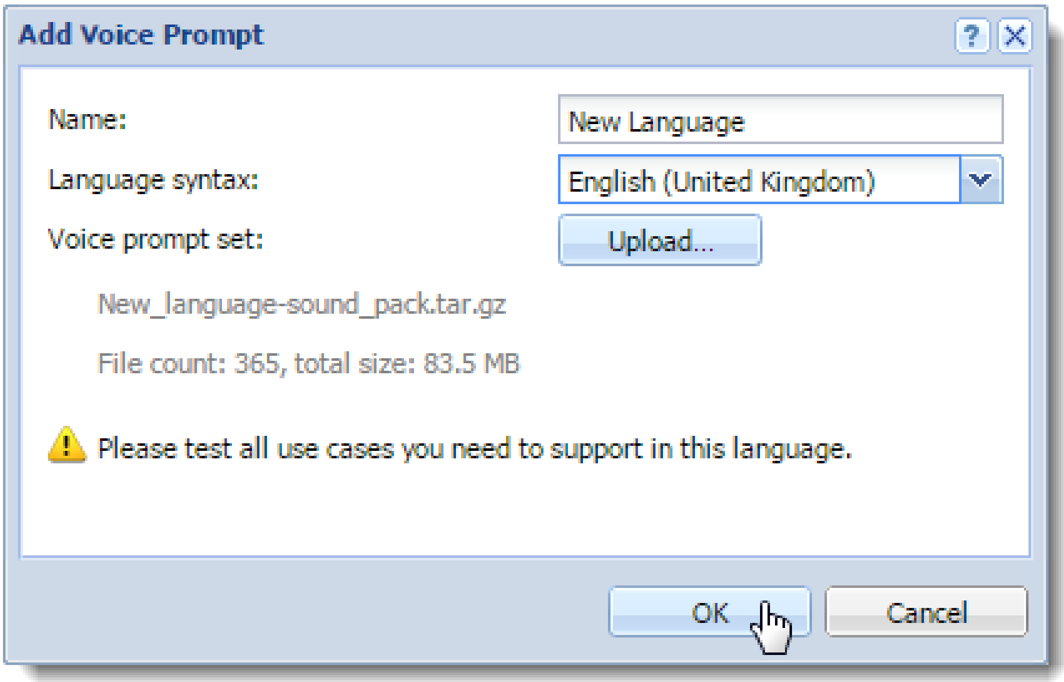
If the language sets (voice records) provided in Kerio Operator do not satisfy your needs, you can download or buy different language sets and import them to the PBX. You can download the language sets (free or paid), for example, in the following sites:

- http://www.voip-info.org/
- http://downloads.asterisk.org/

You can extract any language set archive and create your own voice records (provided you keep the file structure).

To add a new language:

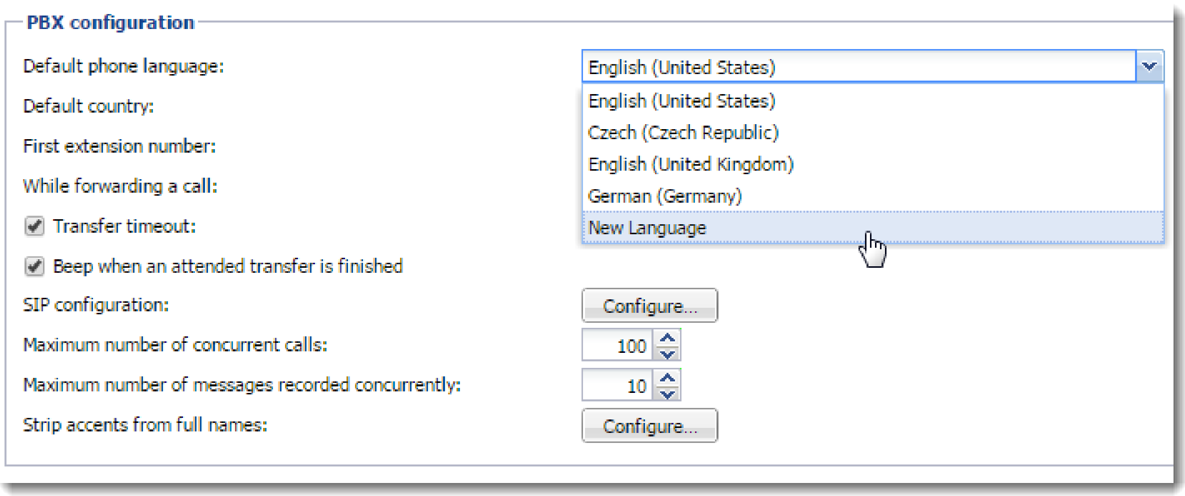
1. In the Kerio Operator administration interface, go to **Configuration** → **Advanced Options** → **Telephony**.
2. Next to the **Default phone language** field, click **Configure**.
3. In the **Voice Prompts** dialog box, click **Add**.
4. Type a name of the voice prompt.
5. Select a language syntax.
6. Click **Upload** and select your sound package.



7. Click **OK** and click **Close**.

8. In the **Default phone language** field, select the new language.

9. Click **Apply**.



Changing indication tones according to countries

Each country has different standards for indication tones during calls (e.g. beeps, ringing tones, etc.).

You can change the settings in the administration interface.

To select a default country for your PBX, go to **Configuration** → **Advanced Options** → **Telephony**.

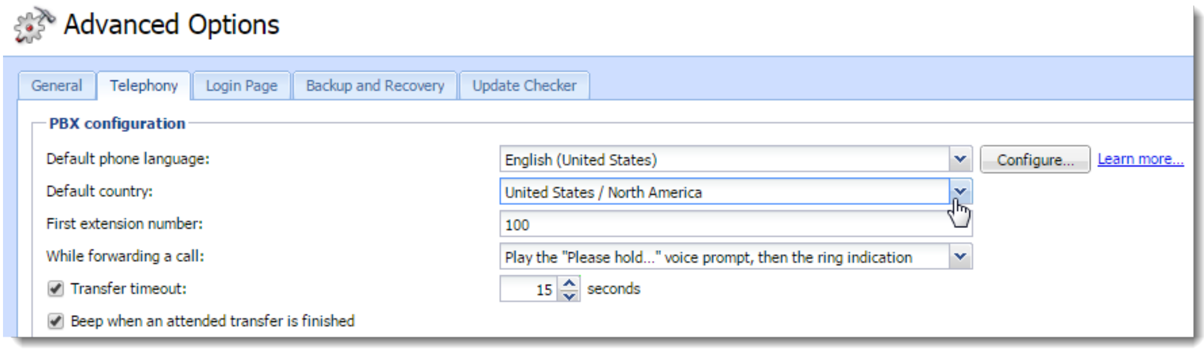


Figure 4 Changing the default country

Example

Live And Let Laugh Inc has the following configuration:

- Joan Giggle, receptionist and operator, uses extension 100 and wishes the phone to communicate with her in the Happish language.
- Brian Snigger, receptionist and operator, uses extension 200 and is satisfied with the default language, which is English.
- Phoney VoIP, an interface for incoming calls, is configured in Kerio Operator with the default language — English. This interface is operated by Brian Snigger.
- Telephium VoIP, an interface for incoming calls, is configured in Kerio Operator for communication with customers from the Bliss Seekers Land (in Cravish). This interface is operated by Joan Giggle.
- The default language in Kerio Operator is English.
- Voicemail is enabled and the extension for accessing the voicemail is 50.

Scenario 1:

When Brian Snigger calls Joan Giggle (200 → 100) or when Brian Snigger calls the voicemail (200 → 50), the automatic announcements are in English.

Scenario 2:

When Joan Giggle calls Brian Snigger (200 → 100) or when Joan Giggle calls the voicemail (200 → 50), the automatic announcements are in Happish.

Scenario 3:

Customers calling via the Phoney VOIP interfaces will hear announcements in the default language (English).

Scenario 4:

Customers calling via the Telephun VOIP interfaces will hear announcements in Cravish.

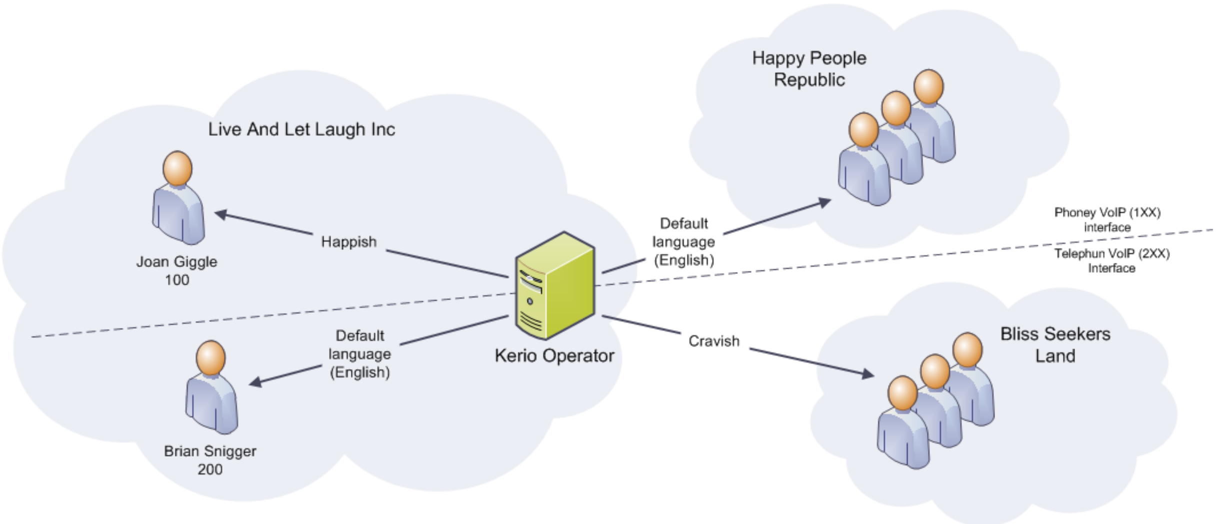


Figure 5 Figure showing the language settings in Live And Let Laugh Inc

Customization of voice sets

Summary

This summary provides information on how to customize/change voice sets in Kerio Operator.

Details

The Internet provides many sources of localized and customized basic sounds and voice prompts. Voice sets for various languages can be found at http://www.voip-info.org.

If you wish to customize a voice set (for example, substitute numerals), begin with the basic sounds:

1. Unpack them.
2. Substitute relevant files.
3. Renew the archive or zip the folder tree (Kerio Operator supports many formats for archiving).
4. Login to Kerio Operator administration.
5. Open **Advanced Options** → **General**.
6. Click **Configure** which is located next to option **Default phone language**.

Once you upload a voice and sound set, you can use it for Kerio Operator, individual interfaces or individual users. For detailed information on this setting, refer to article Configuring languages in Kerio Operator.

Configuring server date, time and time zone in Kerio Operator

Time Settings

Correct time and time zone settings of your PBX are necessary for correct configuration of telephone communication, time ranges and logs. If the time zone is not set properly, log messages or call history may contain confusing information. Therefore Kerio Operator is automatically synchronized with an NTP server.

Do not change the settings unless you have a good reason.

NTP (Network Time Protocol) is a protocol for synchronizing time in your computer with time of the NTP server.

Time and time zone settings on this tab refer to the administration interface time. It is the server time. Kerio Phone will display the time zone using the computer settings. If users are in a different zone to Kerio Operator, logs in call history will be displayed in users' time zone.

Configuring synchronization with NTP

1. In the administration interface, go to section **Advanced Options** → **General**.
2. Select the **Keep synchronized with NTP servers.**

   Date and time can be set manually but it is better to use an NTP server which provides information about the current time and allows automatic management of the firewall's system time.
3. Kerio Technologies offers the following free NTP servers for this purpose: 0.kerio.pool.ntp.org, 1.kerio.pool.ntp.org, 2.kerio.pool.ntp.org and 3.kerio.pool.ntp.org.
4. Click **Apply**.

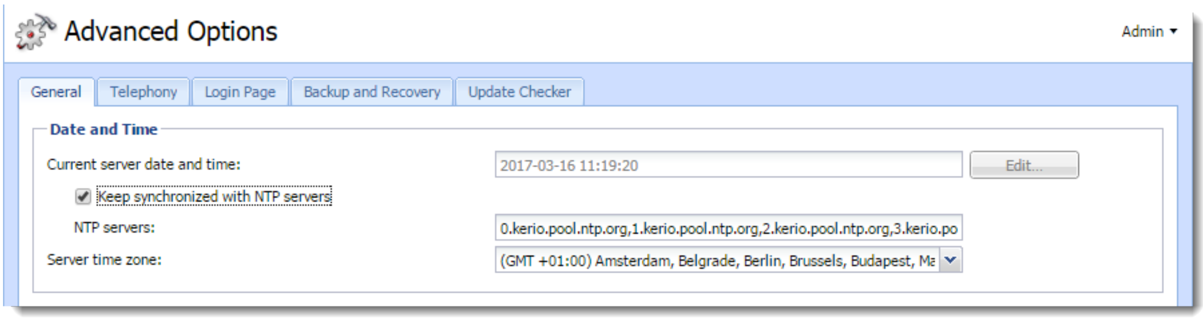


Figure 1 Advanced Options — date and time settings

Configuring time zone

1. In the administration interface, go to section **Advanced Options** → **General**.
2. Select a time zone from the **Server time zone** list.
3. Click **Apply**.

The current date and time will be changed according to the new time zone.

CRM integration using AMI

Asterisk Manager Interface (AMI) overview

Asterisk Manager Interface (AMI) is an interface which enables other applications to connect to Kerio Operator (which includes Asterisk) and to communicate via the AMI commands. You can use it to make phone calls. It enables you to:

- dial calls from your CRM system,
- monitor call statuses in your CRM system (e.g., create logs),
- direct calls to another extension or terminate calls in your CRM system.

Connecting Kerio Operator with other applications

You can connect an application with Kerio Operator very easily. The settings are different for connections with a client (the “server-to-client” connection) and with a server (the “server-to-server” connection).

How to connect a client application (desktop application for dialing numbers) with Kerio Operator

To connect the applications, you need the username and password of the client application user:

1. In the administration interface, go to **Configuration** → **Users**.
2. Select a user and open the **Edit User** dialog.
3. Go to tab **Advanced** and check option **Password for dialer (AMI)**.
4. Click on the  icon and note down the displayed password.
5. Enter the username and password in the client application to authenticate.

How to connect a server (CRM system) with Kerio Operator

You need the authentication data which you enter to your CRM system:

1. In the administration interface, go to **Configuration** → **Integration** → **General**.
2. Click **Configure** at **Third party CTI integration (AMI)**.

3. Check **Third party CTI integration is enabled.**
4. Click **Add.**
5. Enter **Account name** (usually the name of the CRM system).
6. The password is generated automatically. Click on the icon and note down the password.
7. To test the communication, set the permissions to full control. If the communication is successful, you may limit the permissions.

> Some applications allows you only to originate calls but they use asterisk commands which require a higher level of permission (usually full control).

8. Login to your CRM system and enter the password for the AMI integration.
9. Test the communication by dialing an extension.

Application we have tried and prepared a configuration guide

OutCALL configuration for dialing from the Microsoft Outlook contacts

What to do when communication fails

Consult the logs in Kerio Operator:

1. In the administration interface, go to section **Logs** → **Debug**.
2. Right-click on the log screen and select option **Messages** in the context menu.
3. This opens the **Logging Messages** dialog box. Check the **AMI (CRM Integration, Desktop Dialer Applications).**

Configure the internal firewall of Kerio Operator

1. In the administration interface, go to section **Configuration** → **Network** → **Firewall** and check the settings.
2. If your CRM system is located outside your local network, add its IP address in section **Configuration** → **Definitions** → **IP Address Groups**,
3. Go back to section **Configuration** → **Network** → **Firewall** and select a new IP address group for the integration with the CRM system.

Salesforce integration with Kerio Operator

Salesforce integration overview

Lightning Experience from Salesforce doesn't allow the settings described below. Switch to Salesforce Classic to complete the configuration.

Kerio Operator App for Salesforce is based on Call Center. The Call Center is an application embedded in Salesforce and integrates Salesforce with Kerio Operator. For more information about Call Centers, go to https://help.salesforce.com/.

Kerio Operator App for Salesforce enables:

- click-to-dial
- displaying contacts, accounts and leads during the call
- logging calls into Salesforce

To use Kerio Operator App for Salesforce, install the application. You can download it from Kerio Operator administration interface.

Kerio Operator supports:

- Salesforce Enterprise Edition
- Salesforce Performance Edition
- Salesforce Unlimited Edition

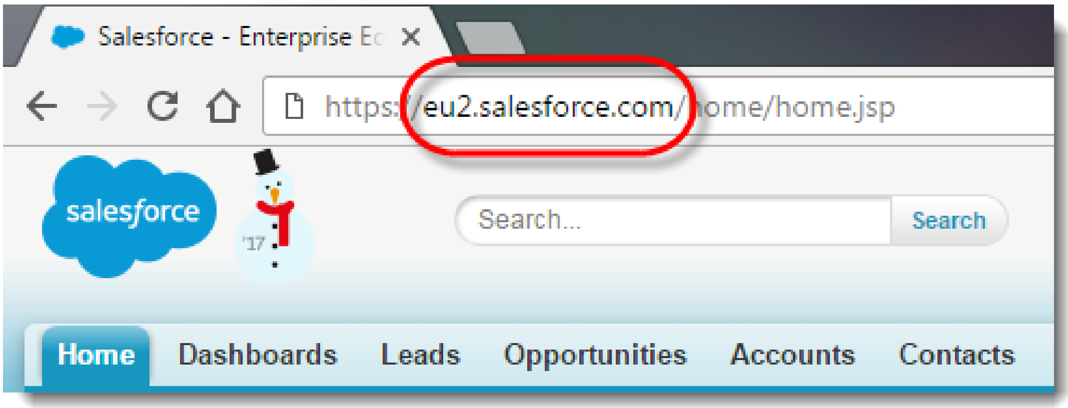
This article helps you to install and configure Kerio Operator App for Salesforce. If you need to use Kerio Operator App for Salesforce, go to Using Kerio Operator App for Salesforce.

Configuring Kerio Operator

Add the Salesforce hostname to Kerio Operator and download **Call Center Definition** for Salesforce.

1. In the administration interface, go to **Integration**.
2. In the **Salesforce integration** section, click **Configure**.
3. Login to your Salesforce and copy the Salesforce hostname.

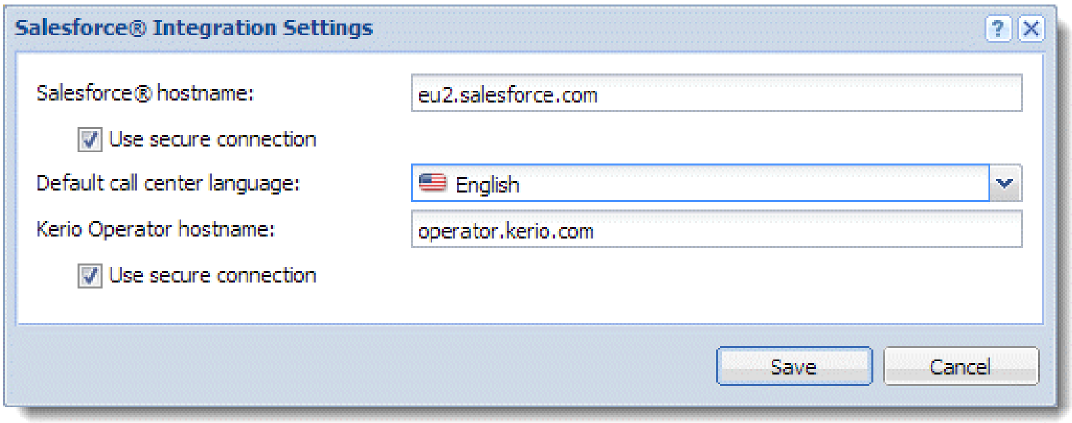
   Paste the hostname to **Salesforce hostname** in Kerio Operator.



4. Check if the Kerio Operator's hostname is complete.

   If the field is empty, type a correct Kerio Operator's hostname.

5. Save the settings.



6. Click **Download Call Center Definition**.

The communication is based on HTTPS by default. Verify that port 443 is open in both directions and make sure that the hostname of the SSL certificate matches the Kerio Operator hostname (read more in the Configuring SSL certificates article).

Configuring salesforce.com

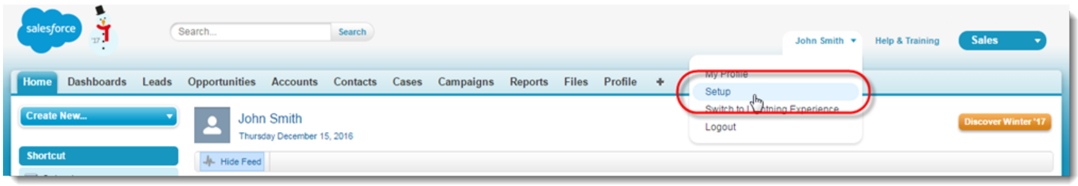
Configuration is divided into three steps:

- Adding Call Center
- Adding users to the Kerio Operator Call Center
- Installing the Kerio Operator Open CTI Package

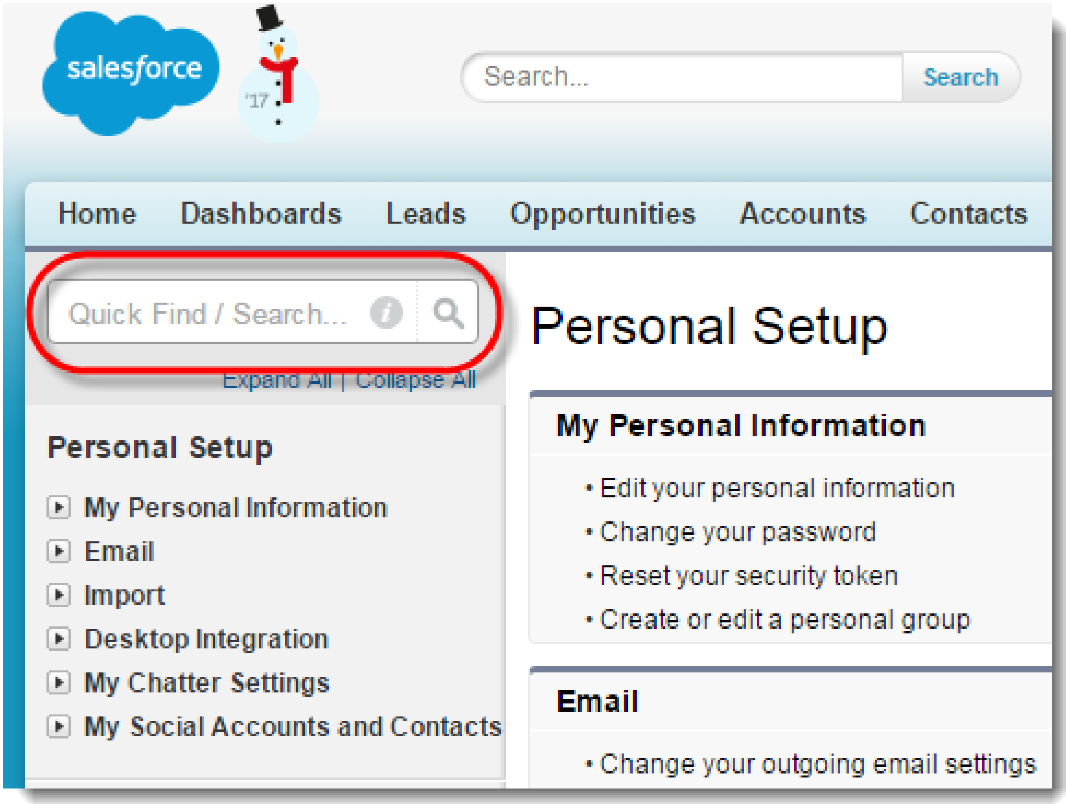
Adding Kerio Operator Call Center

To add Kerio Operator Call Center to Salesforce, follow these steps:

1. In Salesforce, click your name and go to **Setup**.



2. In the **Quick Find**, type **Call Center** and click **Call Centers** in the results.



3. Skip the help page if it appears.
4. Click the **Import** button in the **All Call Centers** page.
5. Click the **Choose File** button and select the call center definition file you downloaded earlier.
6. Click **Import**.

Kerio Operator Call Center (Kerio Operator App for Salesforce) is installed in Salesforce. Now add users to the call center.

Adding users to the call center

To add users (your colleagues) from Salesforce to Kerio Operator Call Center, follow these steps:

1. In **Kerio Operator Call Center**, click **Manage Call Center Users.**
2. Click **Add More Users.**
3. Leave the form as it is and click **Find.**
4. Select users and click **Add to Call Center**.

The users appear in the **Kerio Operator Call Center: Manage Users** table.

Go to **Home** in the main menu. You can see the Kerio Operator Call Center application if your user account is added in the Kerio Operator Call Center.

Installing the Kerio Operator Open CTI Package

Kerio Operator Open CTI Package enables searching salesforce contacts, accounts and leads in the Kerio Operator Call Center application.

1. Go to Salesforce.
2. In the address bar of your browser, add this string after your Salesforce hostname (in our case it is `https://eu2.salesforce.com/`):

   `packaging/installPackage.apexp?p0=04tb0000000QG2n`

   The final result is similar to:

   `https://eu2.salesforce.com/packaging/installPackage.apexp?p0=04tb0000000QG2n`

   A **Package Upgrade Details** page is opened.
3. On page **Package Upgrade Details**, click **Continue**.
4. On page **KerioOperatorOpenCti**, click **Next.**
5. Select **Grant access to all users** and click **Next.**
6. Click **Install.**
7. If you are successful, the application answers that the installation is complete.

You can test all features of Kerio Operator App for Salesforce. For details, go to article Using Kerio Operator App for Salesforce.

Configuring number transformation for calls from Salesforce

To make calling via Kerio Operator App for Salesforce easy, add number transformations which ensure that numbers are dialed correctly from Salesforce.

Read the article Using number transformation for detailed information.

Configuring outgoing prefixes

You can also configure prefixes in Kerio Operator Call Center. However, number transformation is recommended.

1. Go to Kerio Operator Call Center.
2. Click **Edit**.
3. Change prefixes in the **Dialing Options** section.
4. Click **Save**.

Prefixes are the same for Kerio Operator and Salesforce now.

Saving Kerio Operator configuration to FTP or local storage

Backup overview

Kerio Operator can back up system settings and data:

- to an FTP server
- to your local storage (the file can be downloaded from Kerio Operator)
- to MyKerio

Kerio Operator can backup the following items:

- System configuration — system settings, IVR (auto attendant scripts), users, logos, firmwares etc.
- Local voicemail data — if you use integration with Kerio Connect, Kerio Operator sends voicemails via IMAP to Kerio Connect. These voicemails are not backed up.
- SSL certificate — only an active SSL certificate is backed up.
- System logs — all logs from the **Logs** section.
- Call history log — all logs from the **Status** → **Call History** section.
- License — a `.key` file with your licence.
- Recorded calls — locally saved recorded calls.

  You can also back up recorded calls to a FTP server.
- Custom provisioning files /var/tftp

Saving backups to an FTP server

1. In the administration interface, go to section **Integration** → **Remote Storage**.
2. Type a hostname of your FTP server
3. Type a username and password if it is necessary.
4. Click **Apply**.

5. Go to **Advanced Options** → **Backup and Recovery**.
6. Change **Type** to **FTP**.
7. Test the settings by clicking on the **Backup on Remote Storage** button.
8. Select **Enable automatic backup to remote storage**.
9. In the **Start at** field, specify the time at which backups should be performed.
10. In the **Period** field, specify how often backups should be performed.
11. Next to **Content**, click **Edit** and select content types for backup.

    By default, Kerio Operator backs up only a system configuration. Full backup (all items are selected) increases size of the backup.
12. Save the settings.

Saving a single backup file

1. Go to **Advanced Options** → **Backup and Recovery**.
2. In the **Backup** section, click **Download Backup File.**
3. Select a backup content.

   By default, Kerio Operator creates a full backup.
4. Click **Create Backup for Download.**
5. Click **Download** and save the file.

Recovering data from a backup

1. Download a backup file from an FTP server or locally saved backup from your computer.
2. In the administration interface, go to **Advanced Options** → **Backup and Recovery**.
3. Click **Upload Backup File.**
4. Select the file and upload the backup to Kerio Operator.
5. When the **Recovery** dialog box appears, select the configuration and data for recovery.
6. Click **Recovery**.
7. A warning about restart appears, click OK.

After the restart, the backup recovery is complete.

After restoring from a backup, restart your browser in order to log back into the administration interface.

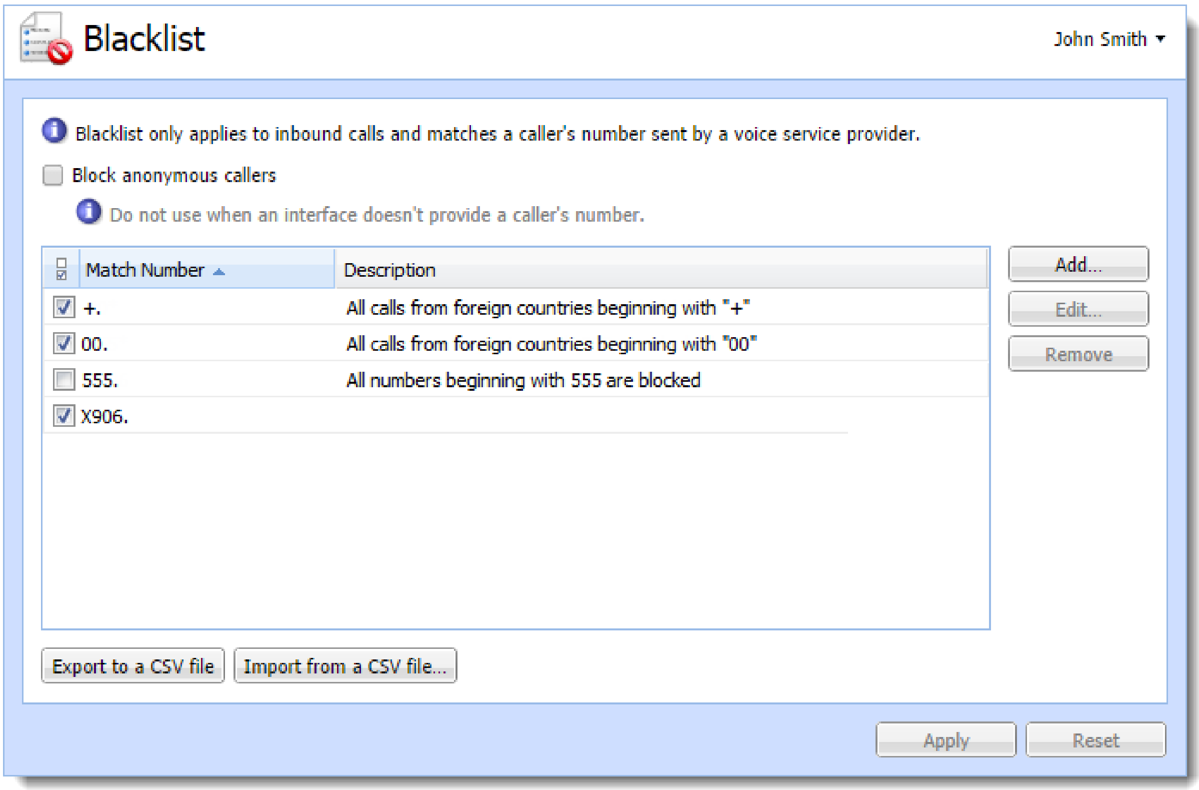
Blocking incoming calls in Kerio Operator

About the blacklist

New in Kerio Operator 2.3.3!

If you want to block incoming calls from certain numbers, you can add the numbers to Kerio Operator's **Blacklist** .

Kerio Operator then blocks all numbers in the blacklist. No incoming calls from these numbers are connected.



Adding numbers to the blacklist

1. In the administration interface, go to **Configuration** → **Blacklist.**
2. Click **Add.**

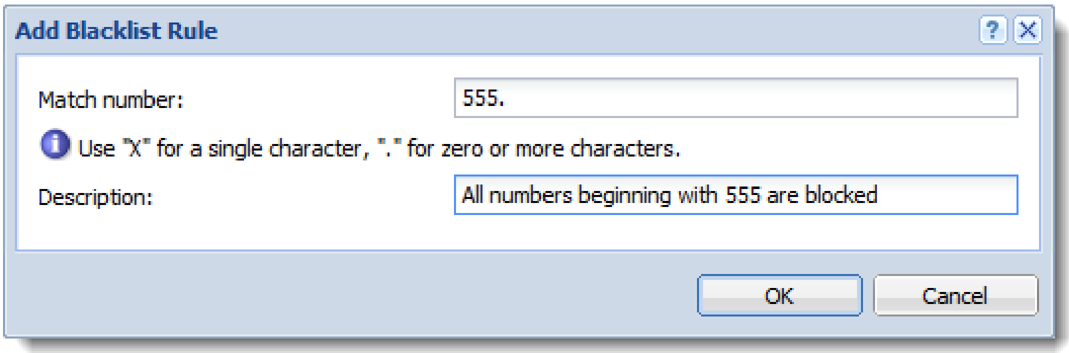
3. Type the number you want to block (**Match number**).

   You can match an entire number, or you can use X for single characters and . (dot) for multiple characters.

4. Add a description to document the reason for blacklisting the number.

5. Click **OK**.

6. Add as many rules as you need.



7. (Optional) You can also block anonymous callers.

   Do not use this option if your provider does not show the caller's number. Otherwise, all incoming calls are blocked.

8. Click **Apply**.

When you receive a call from any of the numbers in the blacklist, your extension appears to be busy and the call is not connected.

Adding numbers from Call History

In **Call History**, you can select any incoming call number to add to the blacklist.

Right-click the number (a line) and select **Blacklist**.

When a call is blocked by blacklisting, you see **Blacklisted** in the **Status** column in **Call History**.

Adding/removing numbers with a PBX service

You can also use your phone to add numbers to the blacklist.

Kerio Operator has three pre-defined PBX services:

- *30 for adding numbers to the blacklist
- *31 for removing numbers from the blacklist
- *32 for adding the last caller to the blacklist

To add a number to the blacklist:

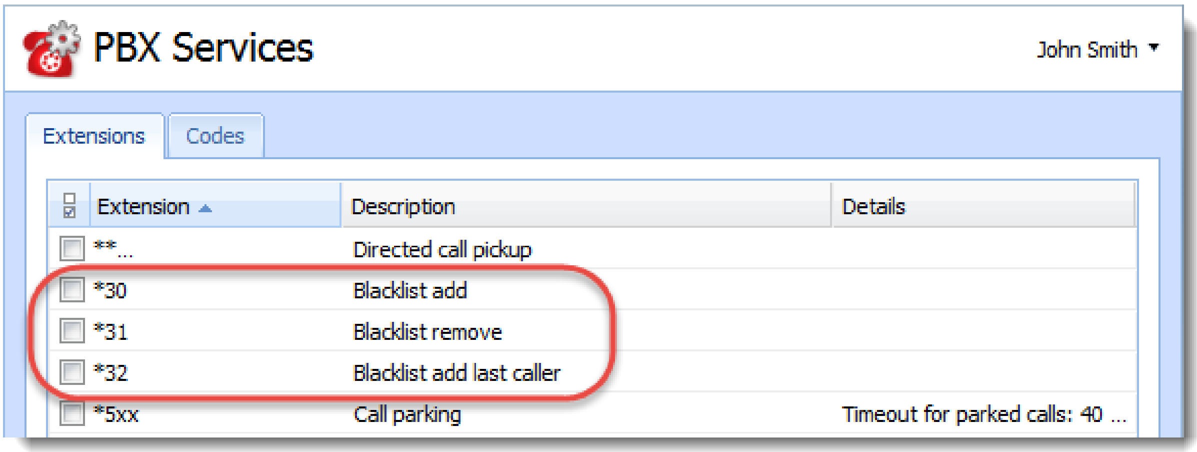
1. Dial the service number for adding numbers: *30.
2. After the beep, enter the phone number.
3. Hang up.

To add the last caller to the blacklist:

1. Dial the service number for adding last number: *32.
2. Confirm the number.
3. Hang up.

To remove a number form the blacklist:

1. Dial the service number for removing numbers: *31.
2. After the beep, enter the phone number.
3. Hang up.



Importing blacklists

You can prepare a CSV file of numbers to be blocked and import init to Kerio Operator.

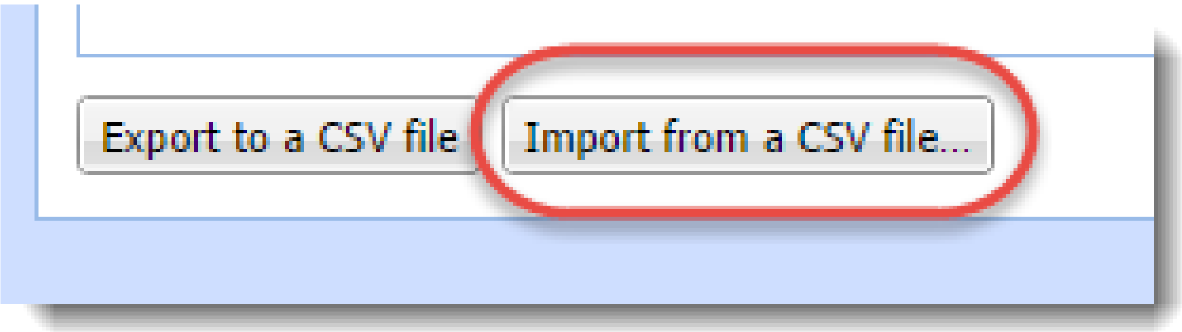
Each line in the file defines one entry. Entries must have the following format:

```
0,"555.","All numbers beginning with 555 are blocked"
1,"+.","All calls from foreign countries beginning with +"
1,"00.","All calls from foreign countries beginning with 00"
1,"X906.",""
```

Notes:

- All items are separated by commas
- Number definitions and descriptions must be inside quotation marks
- If any item is empty, keep the quotation marks

To import the file, go to the **Blacklist** section and click **Import from a CSV file.**



Exporting blacklists

You can export the list of blacklisted numbers to a `*.csv` file.

1. Click **Export to a CSV file.**
2. Go to the correct folder, assign a file name, and save.

Setting outgoing calls constraints in Kerio Operator

Overview

You may want to limit some or all outgoing calls for a variety of reasons. For example, should an outside party obtain the username and password of one of your employees, they could use your PBX for international calls—possibly involving fraud and costing you money. It is therefore critical to have calls to external networks well configured.

You can set outgoing call constraints to prevent these types of attacks.

Restricting outgoing calls

Restricting the length of individual outgoing calls

To set the maximum call duration:

1. In the administration interface, go to **Configuration** → **Security**.
2. Set **Maximum duration of each outgoing call**. The recommended value is 2 hours.

Restricting the number and length of outgoing calls

You can limit all outgoing calls by creating special rules in the section **Configuration** → **Security** in table **Outgoing calls constraints.**

The default rule limits the number of outgoing calls to 50 per hour and total call duration to 2 hours per day.

Example

A manufacturer in the United States sells and primarily has contacts just in the U.S. and Canada, but has a factory in Mexico. Management wants to limit calls to other countries.

1. In the administration interface, open **Configuration** → **Security** and click **Add**.
2. Type a rule name, such as `Constraints for Mexico`).
3. In the **Apply to these outgoing calls** section, select **All except listed** and click **Add**.
4. Add the calling prefixes as a single string:

- For local calls: 9 (outside line)
- For U.S. and Canada: 91 (outside line + 1 preceding the area code)
- For Mexico: 901152 (outside line + 011 for international call + 52 for Mexico's country code)

5. Define the conditions: Set **Maximum calls count** to 10 per hour and **Maximum total calls duration** to 1 hour a day.
6. When the conditions are met, Kerio Operator can send a warning email or block all outgoing calls.

We recommend creating:

- One soft rule with lower limits that sends warning messages via email.
- Another rule with higher limits that blocks the PBX.

If the limits are reached and the PBX is blocked, no one will be able to make calls to the restricted prefixes. However, an administrator can unlock the PBX in section **Configuration** → **Security**. We recommend making a thorough analysis of your calls before setting restrictions so that the PBX is not blocked by standard operations.

In addition to these settings, you can also configure similar rules for specific users or groups of users. See Disabling outgoing calls to certain prefixes.

Configuring Built-in DHCP server in Kerio Operator

Why to use built-in DHCP server

Kerio Operator includes a built-in DHCP server. There are deployment scenarios in which it is useful to have a separate DHCP server for VoIP devices:

- In larger networks, you may need a LAN segment dedicated to voice traffic.
- In smaller networks, the router/firewall sometimes does not support the DHCP option 66 for automatic provisioning of phones.

DHCP server is disabled in the default mode so that it does not collide with your existing DHCP server.

Configuring DHCP server

The built-in DHCP server must have a static IP address:

1. In Kerio Operator administration interface, go to the **Network** section.
2. Select a network interface and click **Edit**.
3. In the **Interface Properties** dialog, switch configuration to **Use the following configuration** and type a new static IP address, mask and gateway.
4. Check **Enable DHCP server**.
5. Click **OK** to save the settings.

Kerio Operator will derive the configuration of the DHCP server from the values you set for the interface's IP address, network mask, and gateway. The DHCP server sends option 66 to Kerio Operator's own address with every address lease.

Assigning IP addresses

Kerio Operator generates the range of IP addresses from a configured mask of a network interface and assigns these addresses automatically.

Example:

- The configured IP address for a network interface is `192.168.62.1`
- The configured mask is `255.255.255.0`
- The gateway has the address `192.168.62.254`

In this example, the range of IP addresses is **192.168.62.2 — 192.168.62.253**.

Example — LAN segment is dedicated to voice traffic

In our example, you have LAN and you need to add an other network interface as a special telephony segment (see scheme DHCP is running on the particular segment).

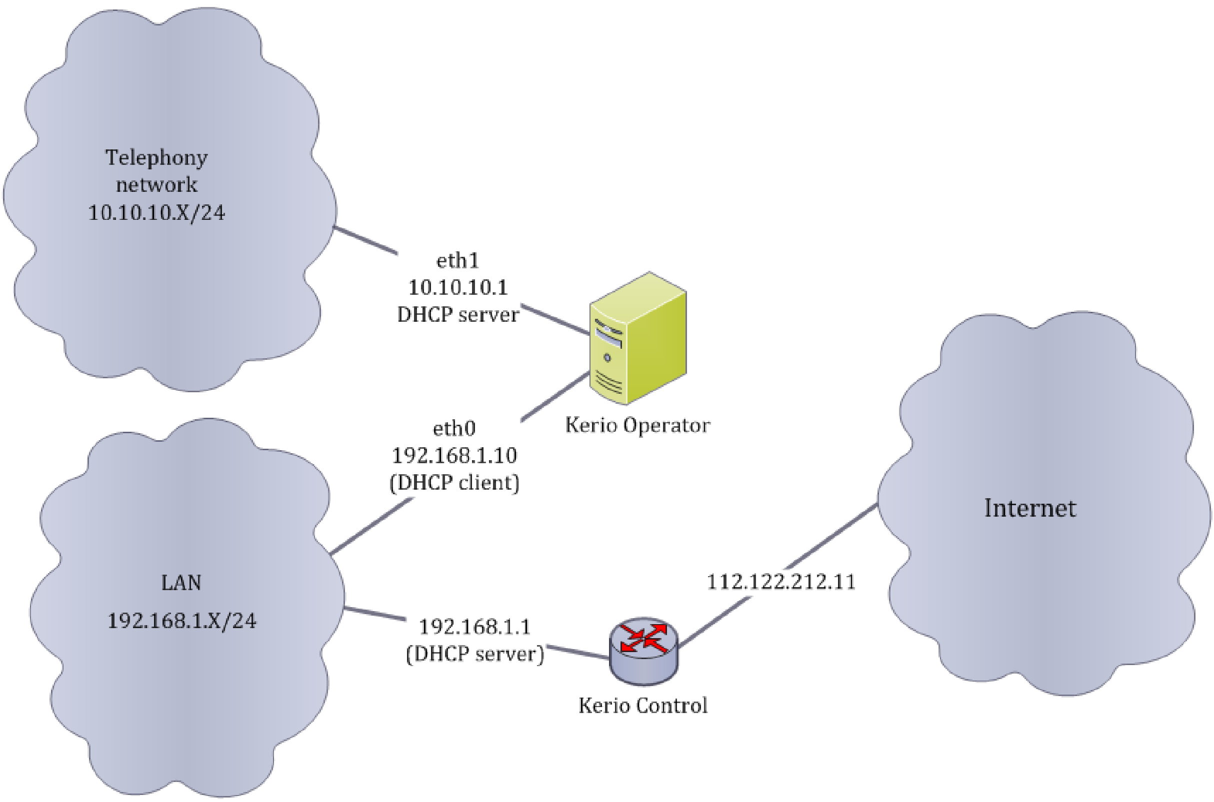


Figure 1 DHCP is running in the particular segment —scheme

You need to configure two interfaces in Kerio Operator administration interface:

1. Go to section **Configuration** → **Network** → **General**.
2. Configure interfaces as displayed in the screenshot DHCP is running on the particular segment.

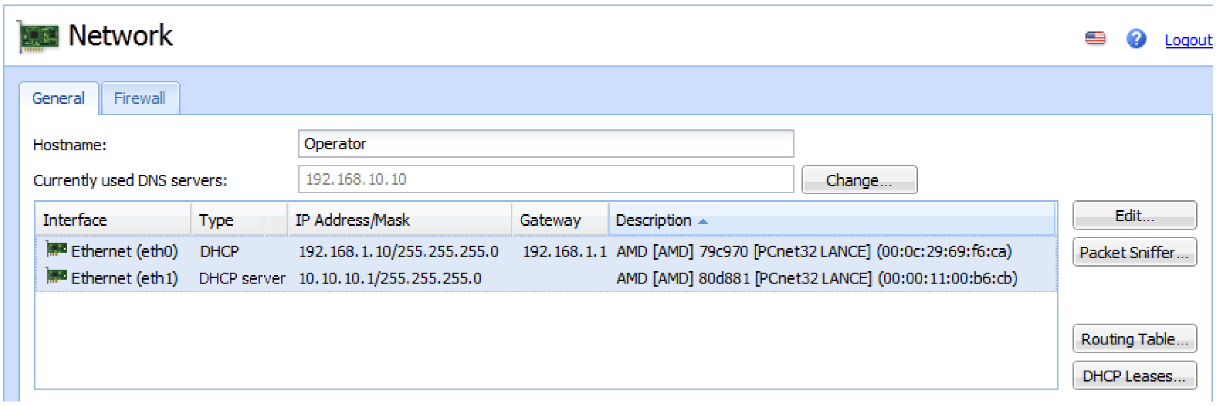


Figure 2 DHCP is running in the particular segment

Configuring NAT

Kerio Operator is behind NAT and phones are in the Internet

1. In the administration interface, open section **Network** → **General**.
2. In the NAT support section, enable NAT by checking the option.
3. Enter the public address which should be used in SIP protocol messages.
4. For phones in the same private network as Kerio Operator, create an appropriate IP address group in section **Configuration** → **Definitions** → **IP Address Groups** with all addresses on which phones communicate in your private network. Thus, the PBX will communicate with phones within the network directly.
5. (Optional) You can also limit the RTP port range.

   Each call requires 4 ports for communication.
6. Also, map the following ports from firewall to Kerio Operator:

   - TCP+UDP/5060
   - TCP/5061
   - UDP/443
   - TCP+UDP/3478
   - TCP+UDP/3479

   It is usually necessary to map a port range for RTP (according to the specified interval).

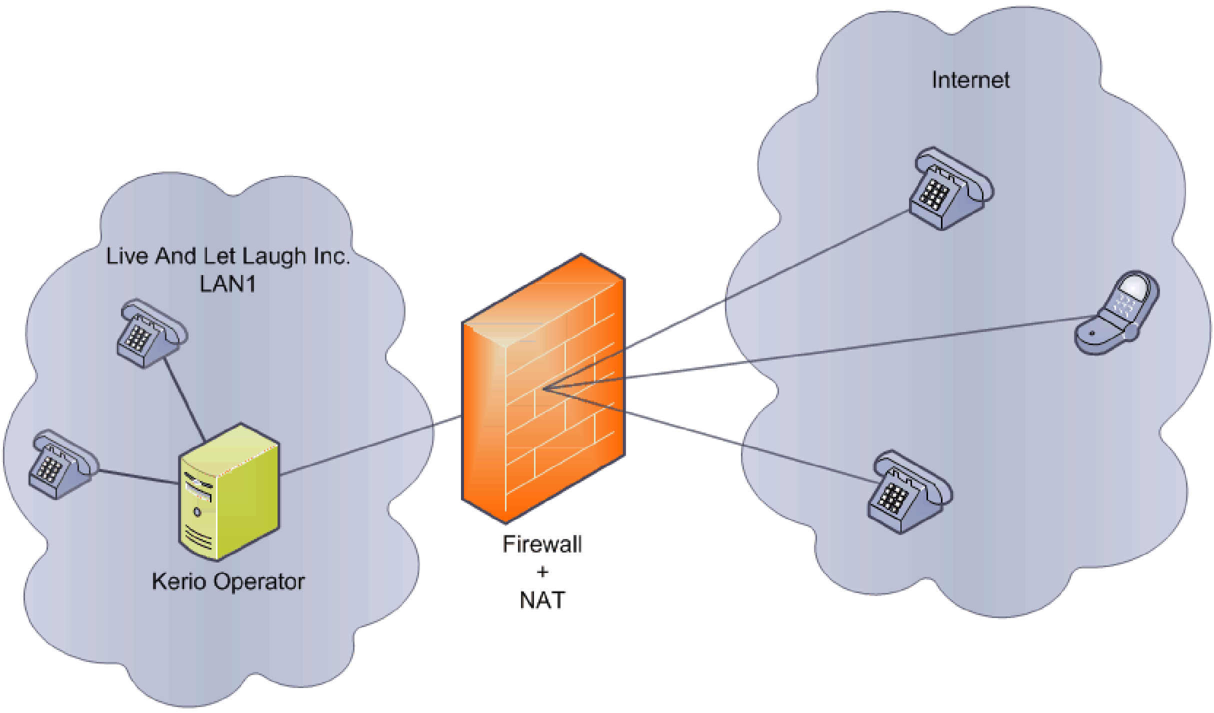


Figure 1 Kerio Operator is behind NAT and hardware phones are in the Internet

Kerio Operator is in the company network and hardware phones are behind NAT

Firstly, configure NAT for Kerio Operator.

The scenario in figure requires only one minor configuration in the PBX settings:

1. In the administration interface, open the **Extensions** section.
2. Select the extension of the user whose phone is in a private network.
3. In the **Edit extension** dialog, go to tab **Advanced**.
4. Check the **Extension is behind NAT** option.

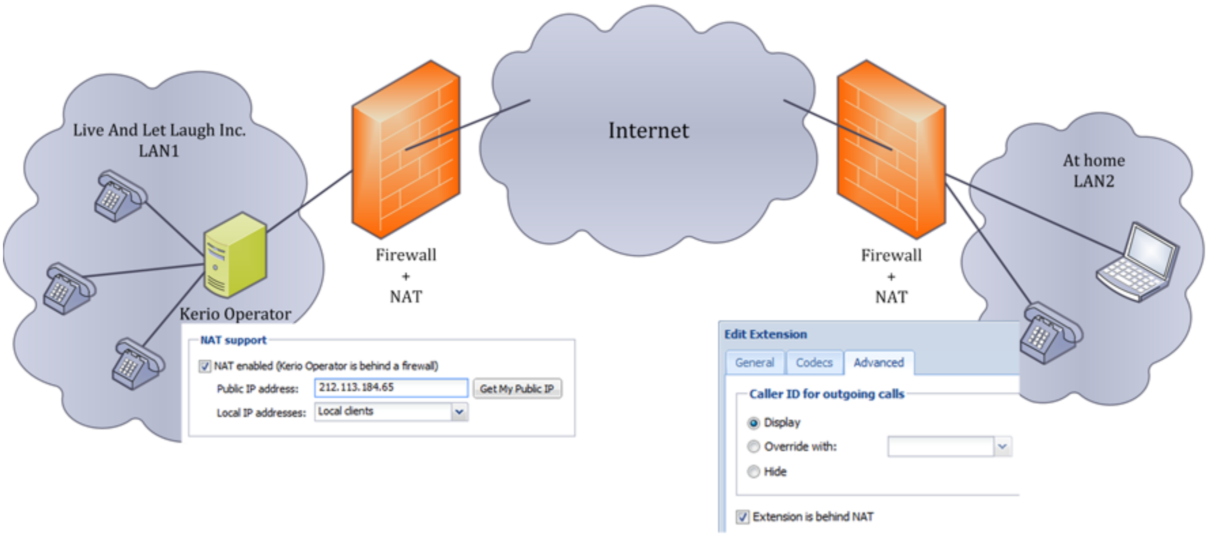


Figure 2 Kerio Operator is in the company network and hardware phones are behind NAT

Kerio Operator is behind NAT and hardware phones are in the Internet

Firstly, configure NAT for Kerio Operator.

If the telephone is in the Internet (not behind NAT), Kerio Operator does not require special configuration.

Phones which are in the Internet cannot be managed in section Provisioned Phones.

Configuring music on hold

Overview

While a caller is waiting for connection or in a call queue (see the Configuring call queues article), they can hear recorded music. Kerio Operator has a default music collection. You can add and configure other audio files. You can upload any file in GSM and WAV format in section **Definitions** → **Music On Hold.**

Adding new collections

To add a new music collection (with one or more file), follow these instructions:

1. Go to **Definitions** → **Music On Hold** and click the **Add** button.
2. In the **Add Music on Hold Collection**, enter a name for the collection and a description.
3. Click the **Add** button situated on the right side of the table with added audio files.
4. In the **Select Audio File** dialog, add file one by one by clicking **Upload.**
5. Select a file in the list and double-click it. Repeat this step until all your uploaded files are listed in table **Audio files in the collection.**

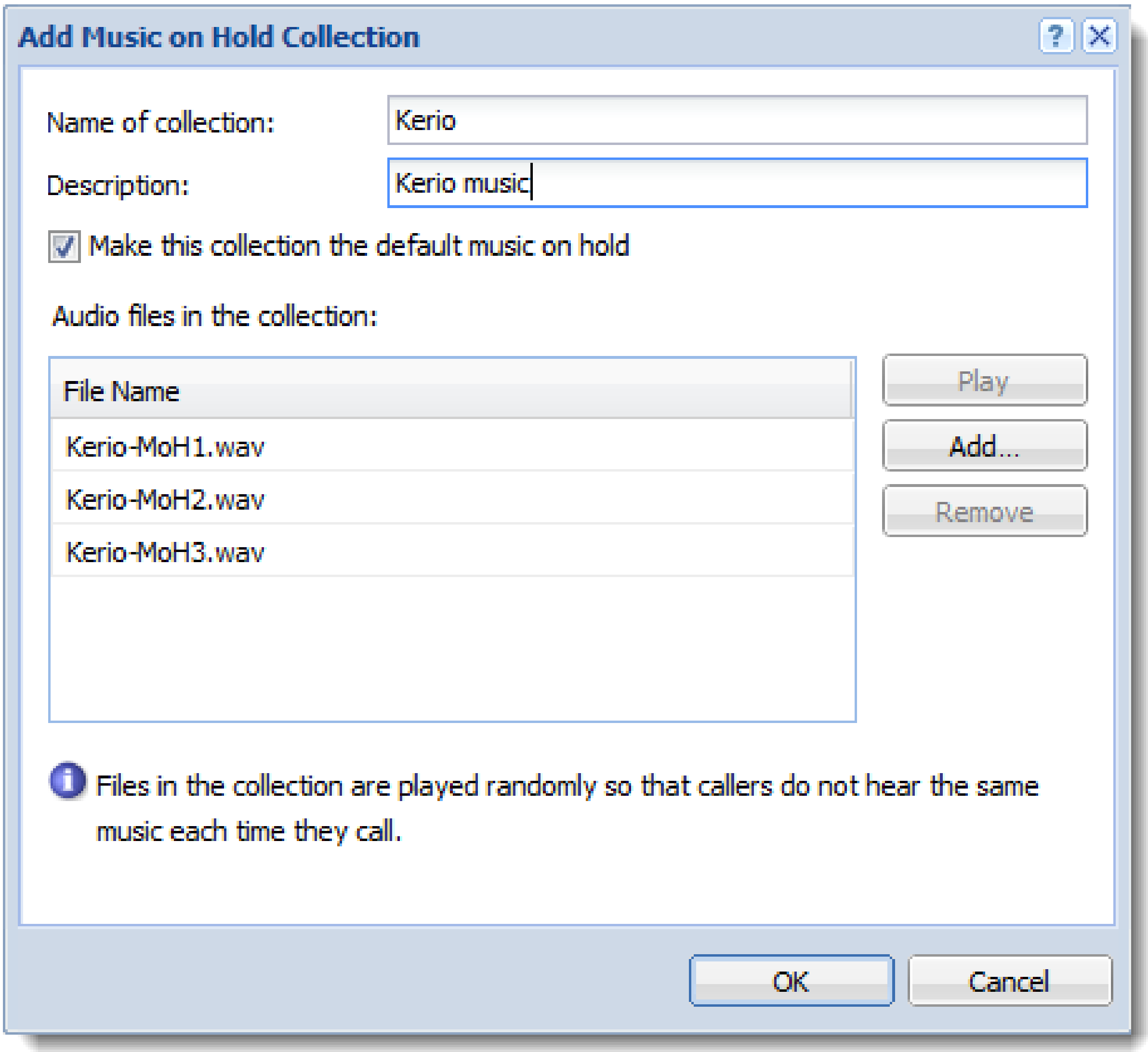


Figure 1 Adding New Collection

Setting Default Collection

In the **Add Music on Hold Collection** dialog, check the **Make this collection the default music on hold** to ensure this collection is used as default in all other Kerio Operator Administration settings.

The default collection is used while holding the line (usually the **Hold** button on most phones). The other collections can be used, for example, in call queues.

Disabling outgoing calls to certain countries or regions

Overview

For security reasons, disable calls to countries users never call, create call permission groups and assign them to extensions.

Call permission groups can:

- Allow everything and disable certain prefixes, or
- Disable everything and allow certain prefixes

Disabling outgoing calls

1. In the administration interface, go to **Definitions** → **Call Permission Groups**.
2. Click **Add** or select an existing group and click **Duplicate**.
3. In the **Add Call Permission Group** dialog box, type the name and a description for the group and click **Add.**
4. Type a specific string of numbers, and choose the option to allow or deny access.

   To limit outgoing calls, include the prefix for outbound calls (usually 9).

5. (Optional) Repeat steps 2 and 3 to add additional numbers.
6. Click **OK** to save the settings.

   Kerio Operator applies the calls permissions in order, one by one.

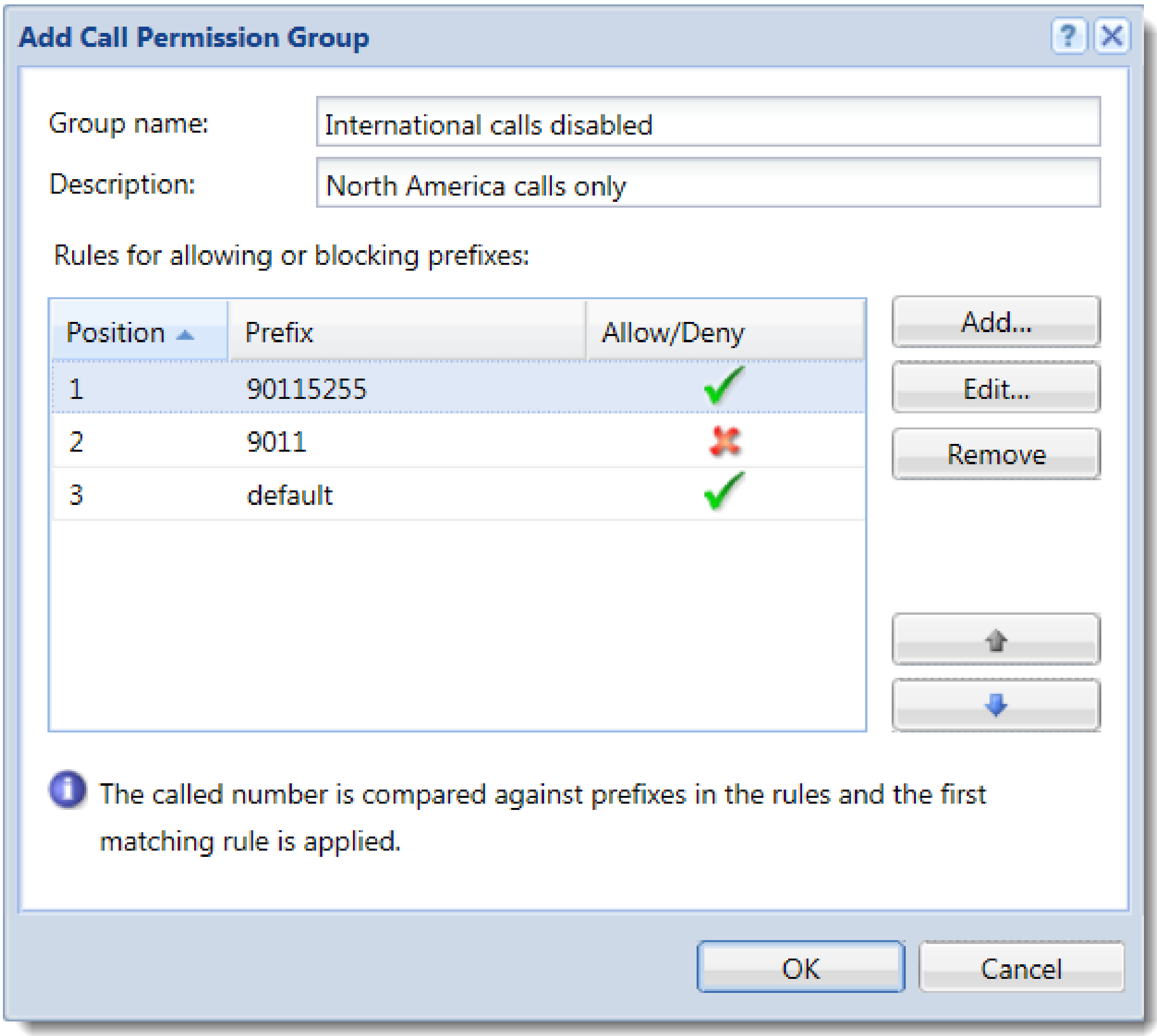


Figure 1 Call permission groups settings

Example

Live and Let Laugh Inc.'s headquarters is located in Mountain View, California. Employees in this office can make calls within the United States and Canada, and to the branch office in Mexico City. Calls to other countries are expensive and there is no reason to make such calls. Thomas Punchline (the company network administrator) can change the settings to block such calls.

- The prefix for calling to an external network (outside line) must be a part of the string (number 9 in our example).
- All international calls (011) are forbidden.
- To enable the prefix for Mexico City, allow prefix (country code = 52, city code = 55). Including the prefix 9 for outside dialing, the prefix is 90115255.

Apply the following settings:

1. In the **Configuration** → **Definitions** → **Call Permission Groups** section, click **Add**.
2. Type a name for the group (for example, **International calls disabled**) and a description.
3. In the **Add Call Permission Group** dialog box, click **Add** and enter the prefix for calls to external networks (9), followed by the prefix for international calls `011`. The result is `9011`.
4. Set the rule to **Deny**.
5. Repeat step 3 to add another rule for calls to Mexico City: 90115255.
6. Set the rule to **Allow**.

Here is an example of the final configuration:

Assigning call permission groups to extensions

1. In the administration interface, go to **Configuration** → **Extensions** and assign the created call permission groups to individual extensions.
2. Select an extension and click **Edit**.

   The **Edit Extension** dialog box opens.
3. Select a **Call permissions group**.
4. Click **OK**.

To assign a call permission group for multiple extensions, select multiple extensions and click **Edit**.

Adding area codes to called numbers

For adding area codes to your called numbers, see this article.

Adding area codes to called numbers

How to add a prefix for outgoing calls

In some situations you need to add an area code to your dialed numbers. In Kerio Operator, you can set the area code automatically.

Example:

- You use only 7-digit schema for your phone numbers (for example 555-5555)
- Your provider accepts only 10-digit numbers

To change your schema from 7-digit to 10-digit numbers:

1. Go to **Configuration** → **Call Routing**.
2. Click **Add...** under the **Routing of outgoing calls** section.
3. On the **General** tab, add prefix number (for example 9).
4. Select your interface.
5. For **Called Numbers** set **Strip digits from left** to 0 and type a 3-digit number prefix (for example 450).
6. Click **OK**.

After that, all outgoing calls dialed with the 9 prefix have 10-digit format (in our example 450555-5555 instead of 555-5555).

Disabling outgoing calls to certain countries or regions

For disabling calls to certain countries or regions, see this article.

Fax support in Kerio Operator

Using fax in Kerio Operator

Kerio Operator supports:

- T.38 protocol
- Fax-to-email
- PDF-to-fax

T.38 support

T.38 is a protocol for realtime transmission of fax over IP.

Kerio Operator uses T.38 by default. Ask your provider whether they support this protocol. If not, read section My provider does not support T.38.

Connecting a fax machine to Kerio Operator

1. Connect your fax machine to an Analog Telephone Adapter device (ATA — for example, Cisco SPA 112).
2. Assign one of Kerio Operator extensions to the ATA device.

Fax machine is connected to the network. You can send and receive faxes.

Configuring an ATA device

You can use various ATA devices. Each device has different settings. The following must be configured:

1. enable T.38
2. set **fax passthru** to **ReInvite**

Phone provisioning in Kerio Operator sets these variables automatically.

Receiving faxes to a user’s email address

You can enable fax-to-email service for any extension. Kerio Operator then sends all incoming faxes to the user’s email address as PDF attachments.

In the administration interface, define SMTP relay in section **Advanced Options → General** so that your Kerio Operator can send emails.

In the administration interface:

- go to **Users** and enter an email address for each user.
- go to **Extensions** and enable option **Forward incoming faxes to user’s address** for the particular user’s extension.

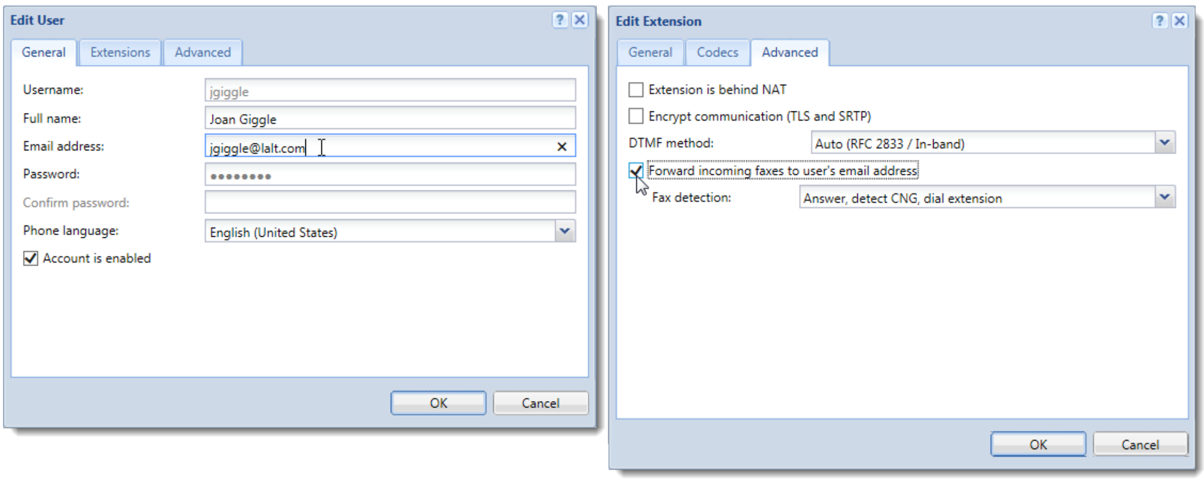


Figure 1 Forward incoming faxes to user’s address

Configuring fax detection (CNG signal)

A CNG signal is the fax machine sound you may hear when there is a fax machine connected to the other end of line. Kerio Operator can detect the signal and start receiving faxes automatically.

1. In the administration interface, go to **Extensions**.
2. Double-click a selected extension.
3. On tab **Advanced**, select:
    - **Dial extension, wait for answer, detect CNG** — PBX dials an extension, waits for an answer and then starts detecting the CNG signal. User has to answer a call first

in order to receive faxes. When a fax tone is detected, the call will be taken over by Kerio Operator.

- **Answer, detect CNG, dial extension** — PBX answers a call first, then detects the CNG signal and immediately dials an extension. If users don't answer the phone, a fax mail is received and users have a missed call on their phone display. This option is good for occasional fax transmissions.
- **Answer, detect CNG, wait 3.5 seconds, dial extension** — Extension is dialed after a 3.5 seconds delay which is used to detect faxes. There will not be any missed calls shown on the phone's display. Regular calls will be automatically answered and will be followed by a 3.5 second delay of silence. This option is good for more frequent fax usage.
- **Answer, detect CNG, wait 3.5 seconds (ringing tone), dial extension** — the PBX will generate a ringing tone instead of waiting in silence. This option is also good for more frequent fax usage and may be less confusing to human callers.

4. Save the settings.

Receiving all faxes to a specific email address

Kerio Operator can send all incoming faxes to a single email address.

1. Go to **PBX Services.**
2. Open **Receive fax messages**.
3. Type email address in the **Send received faxes by email to** field.

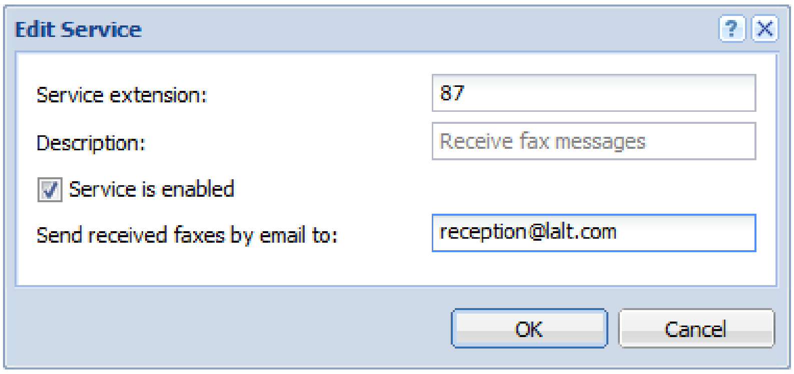


Figure 2 Setting a service for sending incoming fax to email address

Kerio Operator will send all incoming fax messages to the specified email address.

My provider does not support T.38

Fax support without T.38 is not reliable. Using codecs G.711 A-law/U-law instead of T.38 is a workaround.

If your SIP provider does not support T.38, you have to solve these issues:

- Enable codecs G.711 A-law/U-law for the transmission. High compression codecs would distort signal.
- Reduce the speed on your fax machines (if supported).

Enabling G.711 A-law/U-law codecs for the interface

1. Login to the administration interface.
2. Go to **Configuration** → **Call Routing**.
3. Click the provider's interface.

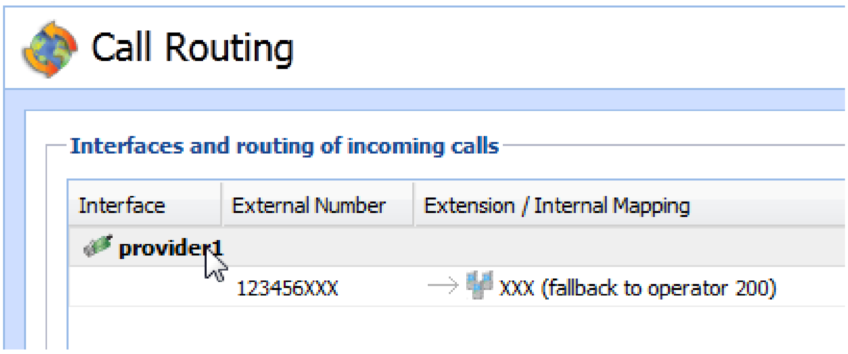


Figure 3 Choosing the provider's interface

4. Click the **Codecs** tab.
5. Move **G.711 A-law** and **G.711 U-law** to the **Selected codecs** table.
6. Move **G.711 A-law** and **G.711 U-law** codecs up in the table.

Moving G.711 A-law/U-law codecs up in the table can cause bandwidth consumption.

7. Click **OK**.

Fax messaging now uses codecs G.711 A-law/U-law.

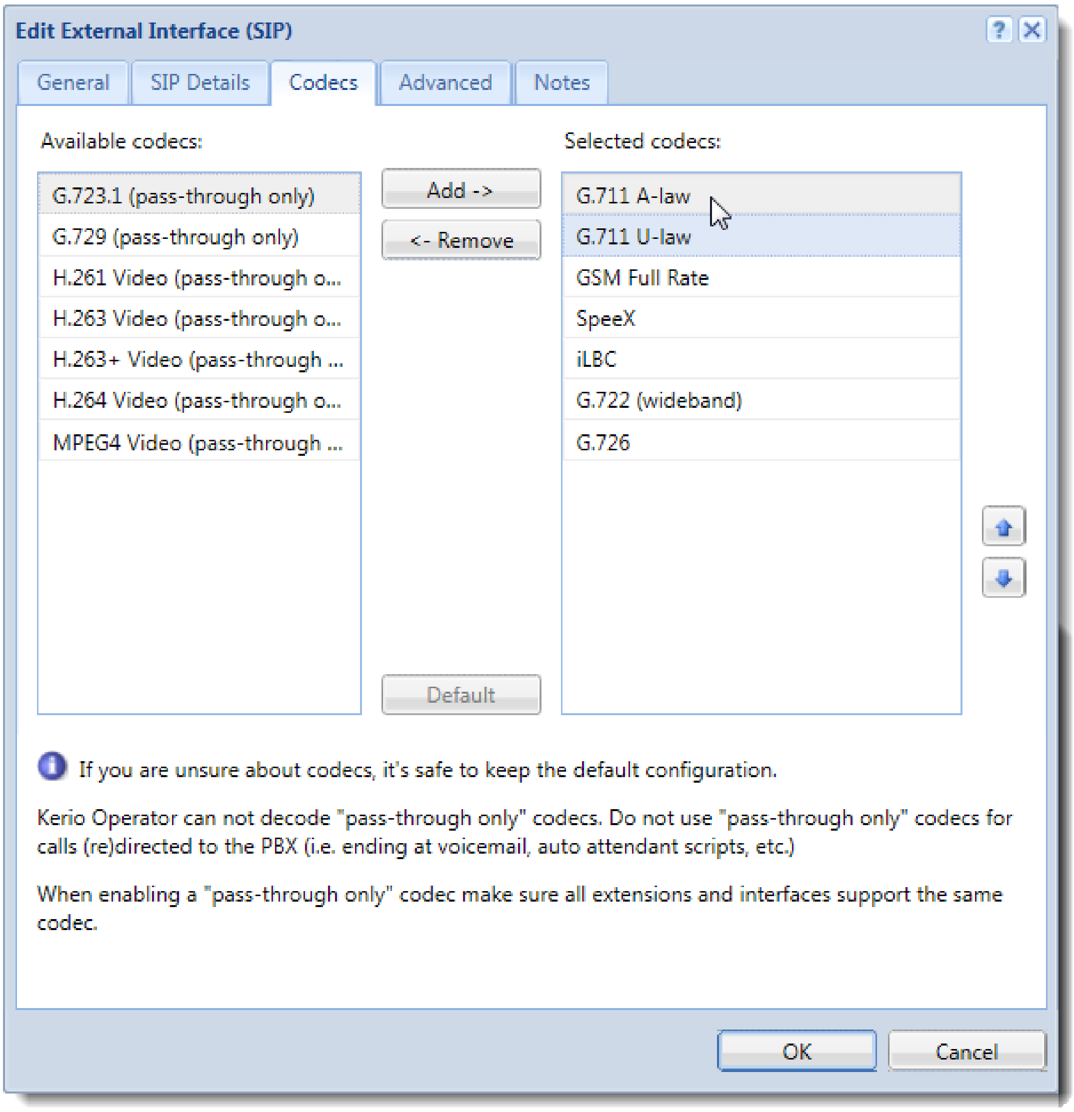


Figure 4 Moving codecs up

Disabling the T.38 support

Although your SIP provider supports T.38 protocol, you may experience some difficulties in communication. Conclusion is disabling a support of the T.38 protocol:

1. In the administration interface, go to **Configuration** → **Advanced Options**.
2. On the **General** tab, click **Configure...** next to the **SIP Configuration**.
3. Unselect **Use T.38 standard for faxing**.

If you still have problems with a fax communication, visit Enabling G.711 A-law/U-law codecs for the interface section.

Sending PDF to fax

This feature is described in a special article: Sending PDF to fax in Kerio Phone.

Distinctive ringing support

Distinctive ringing overview

Kerio Operator supports setting different ring tones for different types of calls (external calls, internal calls or ringing groups).

Configuring strings Kerio Operator

New in Kerio Operator 2.5!

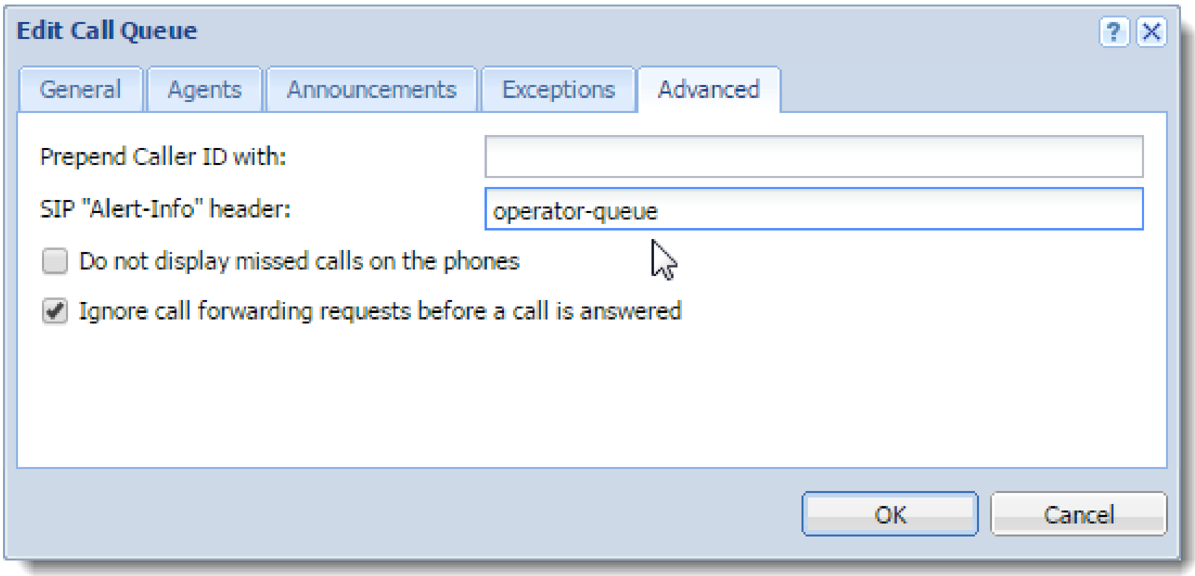
By default, Kerio Operator uses the following strings for the Alert-Info header:

- `operator-external` (calls from an interface)
- `operator-queue` (calls from a call queue)
- `operator-group` (calls to a ringing group)

To configure different ring tones for your SIP interfaces, call queues and ringing groups, change the default string in **SIP "Alert-Info" header**:

1. Go to **Call Routing**, or **Call Queues**, or **Ringing Groups**.
2. In **Call Routing**, double-click a SIP interface.

   In **Call Queues** or **Ringing Groups**, double-click an extension.
3. Switch to the **Advanced** tab.
4. In **SIP "Alert-Info" header**, change the default string.
5. Click **OK**.



Configuring telephones (example: snom 360)

1. Go to web administration of your telephone.
2. Go to **Setup** → **Preferences**
3. Find the alert-info settings.
4. Set different ringers for different alert-info strings (see screenshot).
5. Save the settings.

For testing purposes: Try to make a call from an external telephone number, from an internal extension and to ringing group.

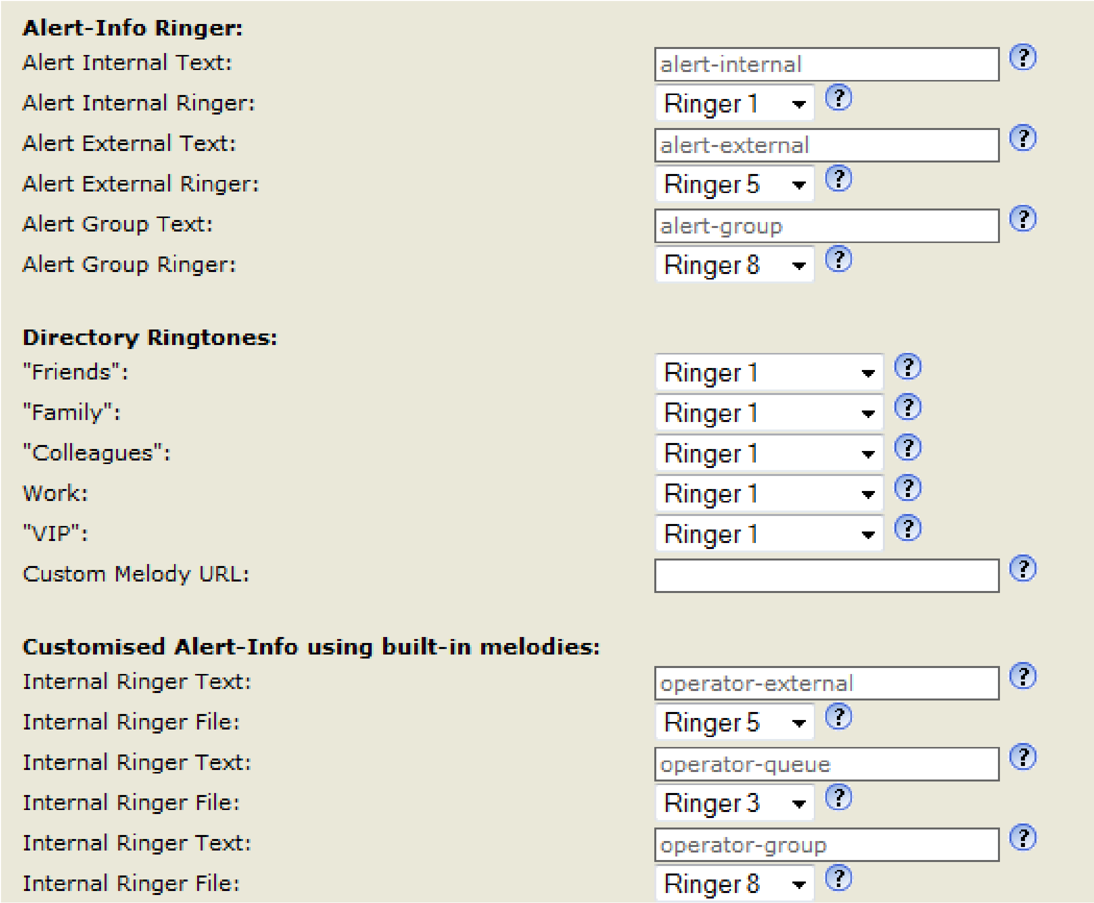


Figure 1 Customising Alert-Info strings

Using Click to Call for Kerio Operator plugin for Chrome and Firefox

Click to Call for Kerio Operator plugin overview

New in Kerio Operator 2.3!

Click to Call for Kerio Operator allows you to dial any phone number in Chrome and Firefox browsers.

Click to Call for Kerio Operator provides the following features for users whose phone is connected to the Kerio Operator PBX:

- The plugin detects phone numbers in the web page and makes them clickable.
- The detection of phone numbers is enabled/disabled by clicking on the extension's icon.
- The detection of phone numbers is repeated when the web page changes.
- You can select the phone number manually and then dial it from the context menu (right-click on the selected text).

If you want to use Click to Call in Kerio Connect Client, go to the Integrating Kerio Connect and Kerio Operator article.

Installing and configuring the Click to Call for Kerio Operator plugin for Chrome

Kerio Operator does not need any configuration, however, users must configure the Click to Call for Kerio Operator plugin.

Follow these steps to install the Chrome version of the plugin:

1. Open the following link in the Chrome browser:

   Click to Call for Kerio Operator

2. Install the plugin.

   A configuration dialog appears after the installation.

3. In the configuration dialog, type the Kerio Operator URL.

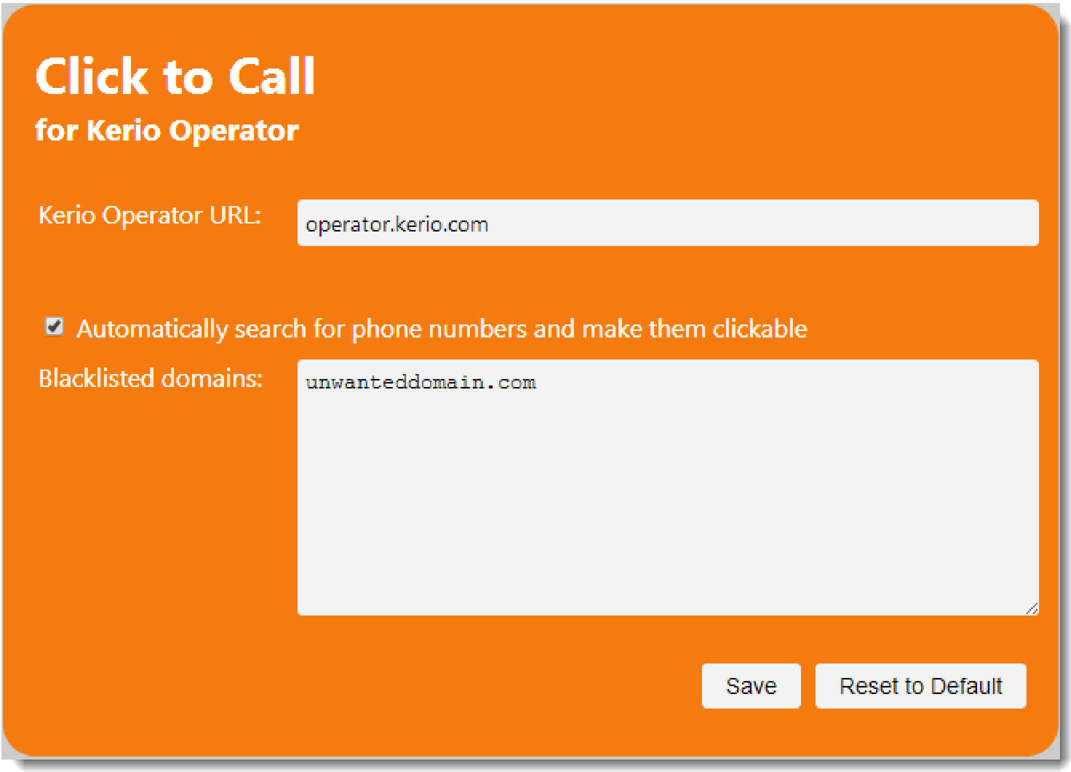


Figure 1 Click to Call for Kerio Operator

If you want to display the configuration dialog later, right-click the **Click to Call for Kerio Operator** icon placed next to the address bar and select **Options**.

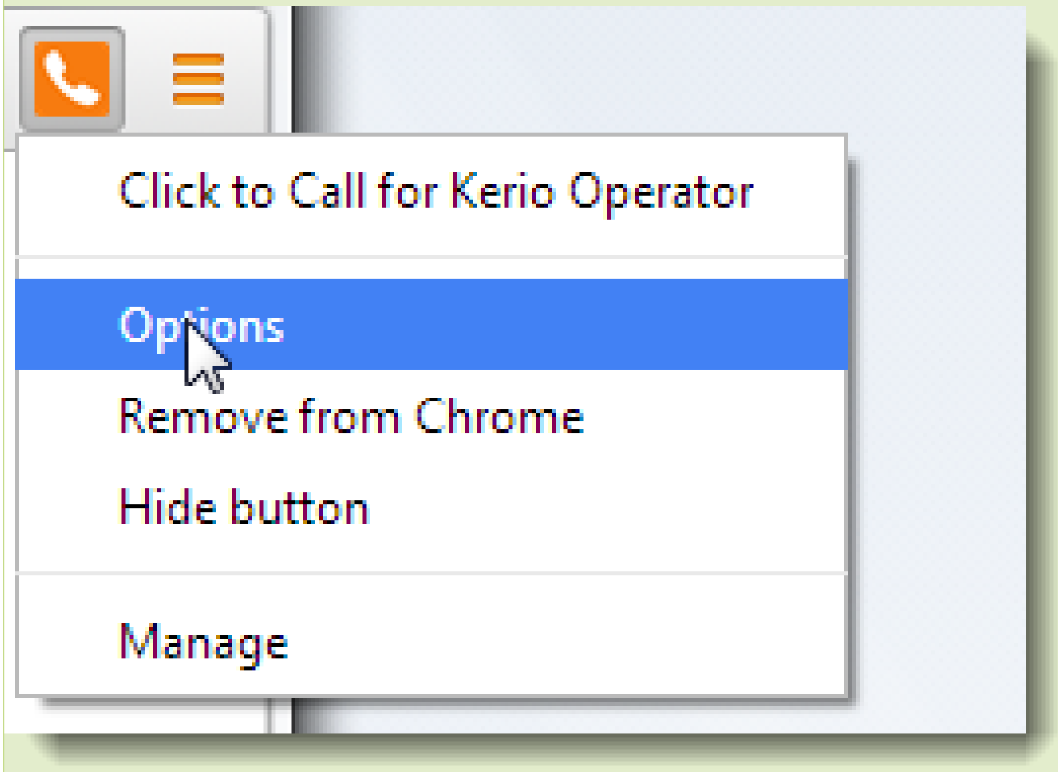


Figure 2 Click to Call for Kerio Operator context menu

4. Check the **Automatically search for phone numbers and make them clickable** option.

   This option enables/disables the Click to Call for Kerio Operator plugin.

5. If you know some websites, which should not use the Click to Call for Kerio Operator plugin, type the URLs into the **Blacklisted domains** field.

6. Click **Save**.

Click to Call for Kerio Operator is configured.

Configuring the Click to Call for Kerio Operator plugin for Firefox

Kerio Operator does not need any configuration, however, you have to configure the Click to Call for Kerio Operator plugin.

Follow these steps to install the plugin:

1. Open the following link in the Firefox browser:

Click to Call for Kerio Operator

2. Install the plugin.

   The plugin appears on the Firefox **Extensions** after successful installation (type `about:addons` in your browser).

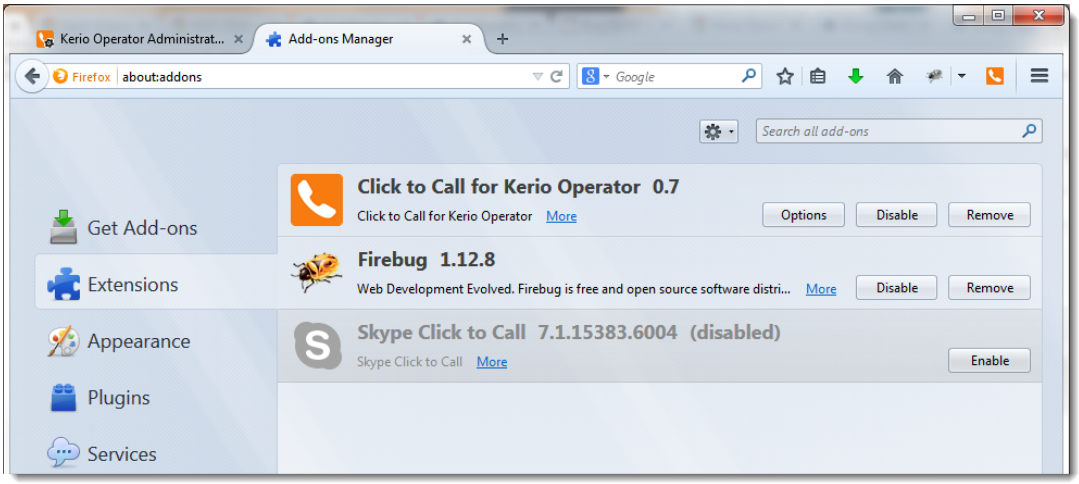


Figure 3 Firefox Extensions

3. Click **Options**.

4. In the configuration dialog, type the Kerio Operator URL.

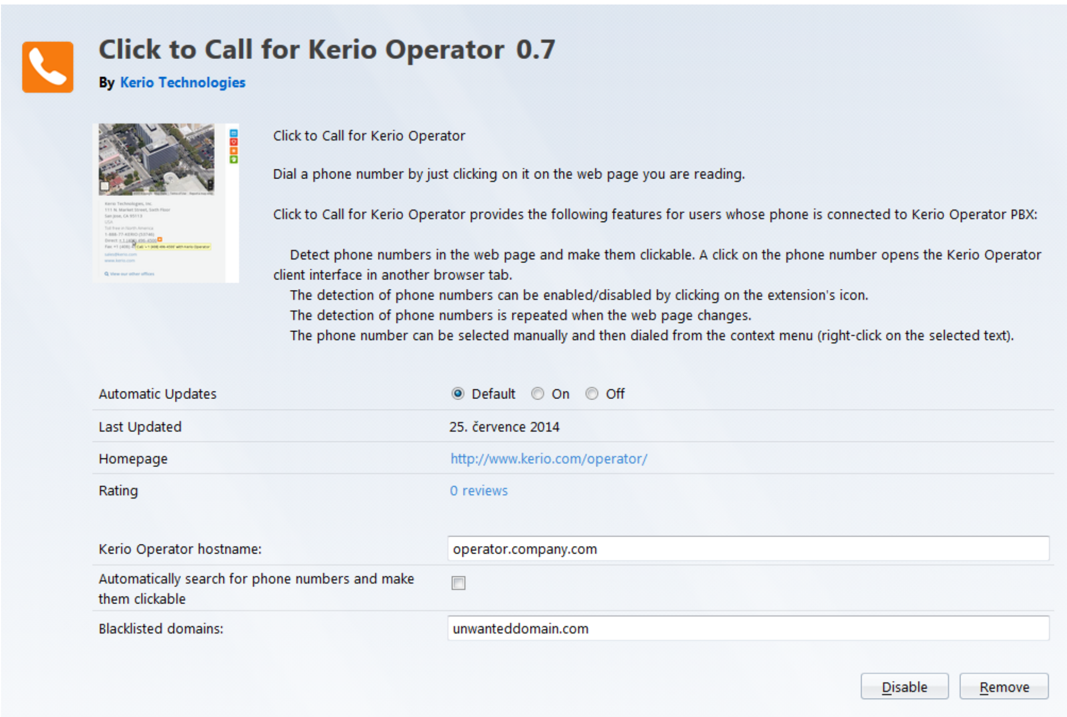


Figure 4 Click to Call for Kerio Operator — Options

5. Check the **Automatically search for phone numbers and make them clickable** option.

   This option enables/disables the Click to Call for Kerio Operator plugin.

6. If you know some websites, which should not use the Click to Call for Kerio Operator plugin, type the URLs into the **Blacklisted domains** field.

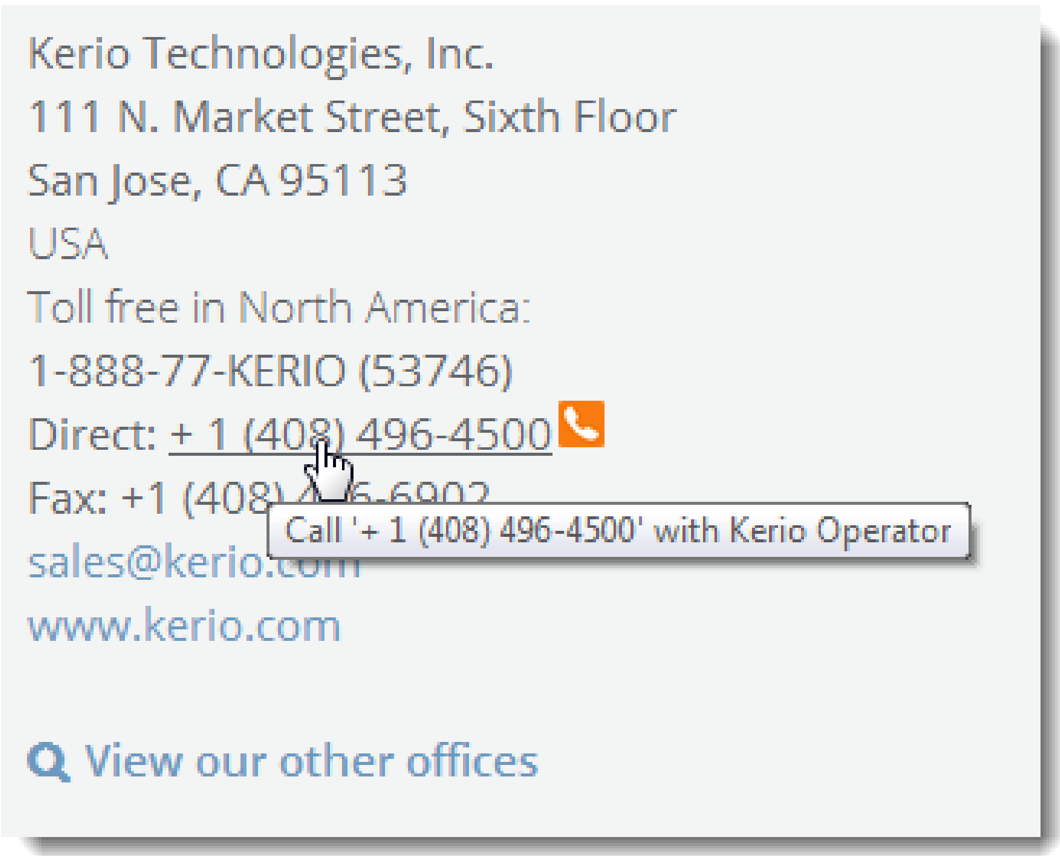
7. Click **Save**.

Click to Call for Kerio Operator is configured.

Using the Click to Call for Kerio Operator plugin

Click to Call for Kerio Operator plugin allows you to initiate a call from Kerio Operator using Chrome and Firefox browsers:

1. Double-click the phone number marked with Kerio Operator logo.



2. The browser opens the Kerio Phone interface in another browser tab. The phone number is predefined.

3. Click **Dial.**

   Dialing in the Kerio Phone works on a callback basis. Kerio Phone connects directly with the PBX and the PBX contacts back your phone. Your phone starts ringing as well as the called person's one. Pick it up and wait for the called person to answer.

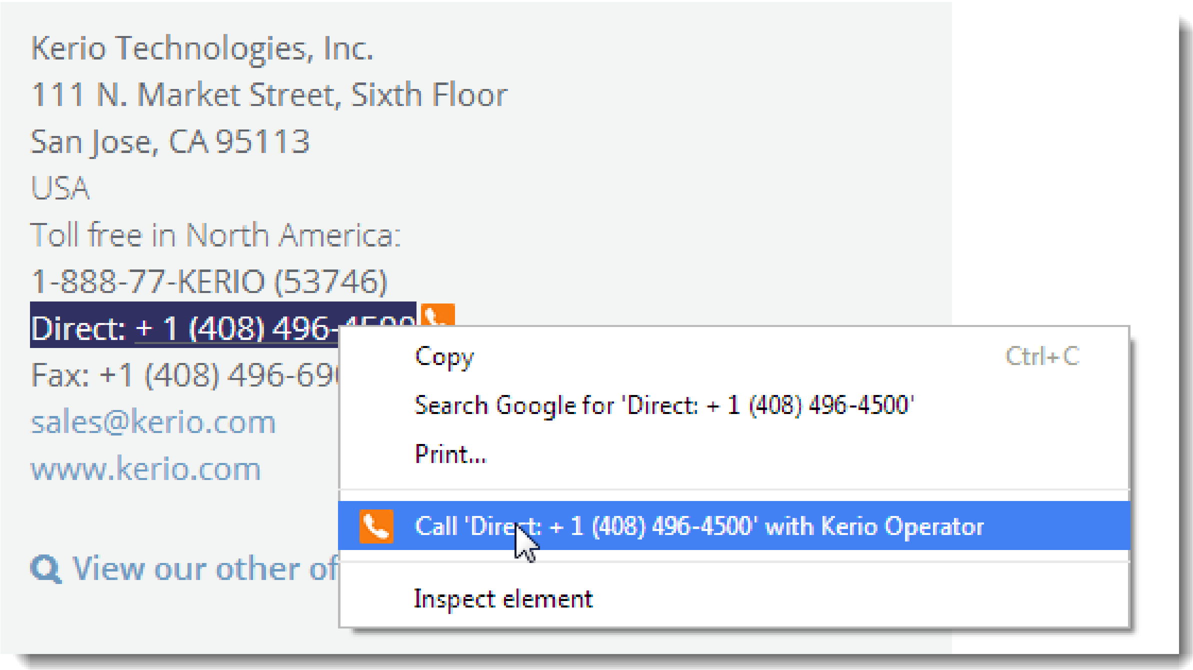
Enabling/disabling Click to Call for Kerio Operator

To enable/disable Click to Call for Kerio Operator, click Kerio Operator icon in the browser (see figure below).

Selecting the phone number manually

You can select the phone number manually and then dial it by right-click on the selected text (see figure below).

> The context menu option is available even if the plugin is disabled.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Print...
Call 'Direct: + 1 (408) 496-4500' with Kerio Operator
Inspect element

Monitoring Kerio Operator

Monitoring overview

When you are experiencing problems with your connection, we recommend to use tools for monitoring the status of your PBX. The tools are available in section **Status**:

- Calls
- Call History
- Recorded Calls
- Dial Plan
- Conferences
- Call Queues
- System Health

Monitoring active calls

All current calls can be viewed under **Status** → **Calls**.

You can see a table where each call occupies one line and a graph displays a number of calls in time in the **Calls** section.

Go to the **Calls** section, especially in case that you plan to restart the PBX which may result in an undesired termination of a call in progress.

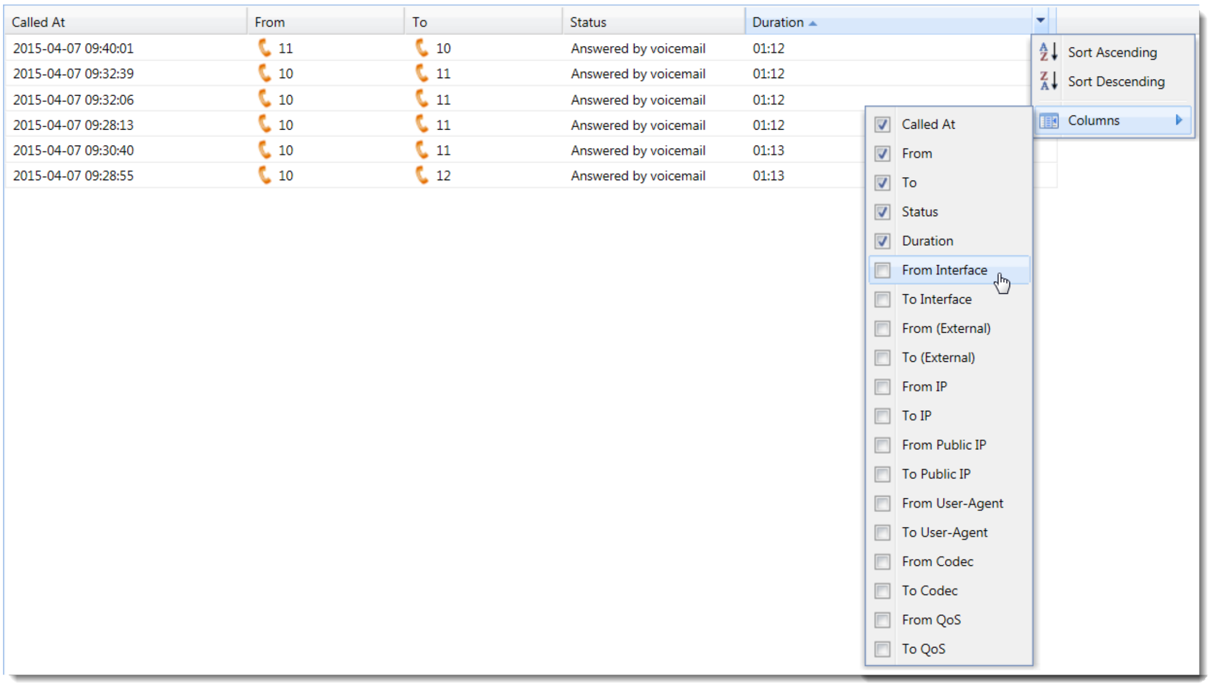
Call History

The Call History section keeps a list of all internal and outbound calls of the PBX.

Call History can be viewed under **Status** → **Call History**.

To add or remove columns in the call history:

1. In the administration interface, go to **Status** → **Call History**.
2. Mouse-over a name of a column and click the arrow on the right side.
3. In **Columns**, you can:
    - select new columns to add them to the **Call History**,
    - deselect columns to remove them.



Each line contains information about one call. The following actions can be applied to the call history:

Export to a CSV file
: You can click on **Advanced** → **Export to a CSV file** to save the file on your local drive.

Clear
: Click on **Advanced** → **Clear** and confirm your decision in the corresponding dialog.

> Individual users can delete their history in the *Kerio Phone*. However, this operation only hides the data. They are not removed from the PBX and logs.

Monitoring Recorded Calls

Section **Status** → **Recorded Calls** displays all calls recorded from call queues. This section displays a table where each recorded call occupies one row. Select a call to listen to it, download it to your computer or remove it.

Click **Settings** to record calls locally or to a remote storage. For more details, refer to the Setting optional call recording article.

Monitoring a Kerio Operator dial plan

A dial plan contains a list of all the used extensions and their users. You can export this list to a CSV file or print it.

Go to section **Status** → **Dial Plan** to see the list:

Export to CSV — the button exports the data in the format described in table.

Extension Number	Type ID	Description
111	1	Winston Smith
112	1	Ada Monroe
50	7	Voicemail

Table 1 CSV file content

Changing the Dial Plan

If you use automatic phone provisioning and the change in your dial plan may affect automatically provisioned phones, update of the phones configuration is needed. Kerio Operator detects such changes automatically and displays a warning. If you confirm this warning, phones will be restarted at the time you selected in the dialog. You can restart the phones later manually in section **Provisioned Phones**. To restart the phones, click on the **Advanced** → **Restart all phones** button.

Monitoring active conferences

All current conferences can be viewed under **Status** → **Conferences**. The window displays two tables. Each line in the first table displays one conference. The second table displays information about individual conferences. Just select a conference and the details in the bottom table are updated.

Monitoring call queues

All active call queues and their parameters can be observed in section **Status** → **Call Queues**. The window displays three tables. Each line in the first table displays one call queue.

The other tables display agents and callers in a queue. Just select a queue and the details in table **Agents** and **Callers** are updated.

You can also reset the call queue statistics to start from zero. Use the **Reset** button.

System Health

The administration interface allows you to view the status of CPU, memory and disk space of your computer with Kerio Operator.

System status can be viewed under **Status** → **System Health**.

In this section, click **Tasks** to:

- restart telephony subsystem
- reboot Kerio Operator
- power off Kerio Operator
- do factory reset of Kerio Operator

The **Support information** link generates an asterisk configuration file and last 100 lines of all logs. This information may be helpful especially when solving issues in cooperation with the Kerio Technologies technical support.

See detailed information about disk space usage by clicking on **Details**. This opens a dialog with information about disk usage of audio files, voicemail and configuration file of Kerio Operator.

Managing logs in Kerio Operator

What are Kerio Operator logs for

Logs are files where information about certain events (e.g. error and warning reports, debugging information) is recorded. Each item is represented by one row starting with a timestamp (date and time of the event). Messages in logs are displayed in English for every language version of Kerio Operator.

Configuring logs

Logs are available in the Kerio Operator administration interface in section **Logs**.

When you right-click in a log, you can configure the following settings (available in all logs):

Save log
: You can save whole logs or a selected part in a `txt` or `HTML` format. See also **Log Settings** option.

Highlighting
: You can save any part of text in logs for better reference. Specify a substring or regular expression and all rows containing such text will be highlighted.

Log Settings
: Apart from immediate savings, you can configure regular saves of individual logs, specifying the size and number of saved files.
You can also enable external logging to a Syslog server.

Clear Log
: Use this option for deleting a log.

Types of logs

Auth

The **Auth** log includes information about all successful attempts to login to Kerio Operator (to the administration or client interfaces).

Failed login attempts are logged into the **Security** log.

Config

The **Config** log stores the complete history of communication between Kerio Operator Administration and the server. It is possible to determine what administration tasks were performed by a specific user.

Debug

Debug log is a special log which can be used to monitor specific information. This is especially useful for problem-solving.

To enable the **Debug** log, right-click in the log window and select the **Messages** option in the context menu. In the opened dialog window, select specific information you wish to monitor.

In addition, displaying too much information slows Kerio Operator's performance. We recommend that you only display information that you are interested in and only when necessary.

Error

The **Error** log displays serious errors that affect the functionality of the entire PBX. The Kerio Operator administrator should check this log regularly and try to eliminate problems found here. Otherwise, users might have problems with some services or/and serious security problems might arise.

Event

New in Kerio Operator 2.3.3!

The **Event** log gives information about phone and interface registrations, phone provisioning, new versions of Kerio Operator, etc.

Kernel

The **Kernel** log contains records generated by the operating system. It includes information about starting and stopping of the server, logs generated by individual processes, etc.

Security

The **Security** log contains the failed login attempts to Kerio Operator.

Warning

The **Warning** log shows error warnings which are not severe. Typical examples of such warnings are messages stating that a user with administrator rights has a blank password or that a user account of a given name does not exist.

Events recalling warning messages in this log do not seriously affect the PBX functionality. However, they can point at current or possible problems. The **Warning** log can help if for example a user is complaining that services are not working.

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Wireshark® is registered trademark of Wireshark Foundation.

X-Lite is a software phone developed by CounterPath Corporation with registered trademark of CounterPath®.

Used open source software

This product contains the following open-source libraries:

adapter.js
: Shim to insulate apps from spec changes and prefix differences.
Copyright (c) 2014, The WebRTC project authors. All rights reserved.

Appliance OS Sources
: Kerio Operator devices are based on open software from various resources. For detailed information on conditions of each particular software used in the product, refer to acknowledgments.
To download the source package, go to http://download.kerio.com/archive/.

asterisk

Asterisk - An open source telephony toolkit.

AudioContext-Polyfill

Polyfill for AudioContext and its parties on Web Audio API.

coturn

coturn TURN server project

Heimdal Kerberos

Heimdal is an implementation of Kerberos 5, largely written in Sweden. It is freely available under a three clause BSD style license (but note that the tar balls include parts of Eric Young's libdes, which has a different license). Other free implementations include the one from MIT, and Shishi. Also Microsoft Windows and Sun's Java come with implementations of Kerberos.

jsonrpccpp

C++ framework for json-rpc (json remote procedure call)

Kerio Asterisk Module

The Kerio Asterisk Module extends the functionality of the Asterisk PBX to match Kerio Operator needs. It is distributed and licensed under GNU General Public License version 2. The complete source code is available at:

http://download.kerio.com/archive/

libcurl

Libcurl is a free and easy-to-use client-side URL transfer library. This library supports the following protocols: FTP, FTPS, HTTP, HTTPS, GOPHER, TELNET, DICT, FILE and LDAP.

libiconv

Libiconv converts from one character encoding to another through Unicode conversion.

Author: Bruno Haible

Homepage: http://www.gnu.org/software/libiconv/
The *libiconv* library is distributed and licensed under GNU Lesser General Public License version 3.
Kerio Operator includes a customized version of this library. Complete source codes of the customized version of *libiconv* library are available at:

http://download.kerio.com/archive/

libmbfl

libmbfl is a streamable multibyte character code filter and converter library. The *libmbfl* library is distributed under LGPL license version 2.

The library is available for download at:

http://download.kerio.com/archive/

libopus

Opus is a high-quality audio codec developed in cooperation among Xiph.org, Broadcom, and Microsoft (Skype). The codec is standardized in RFC 6716. The reference implementation of the codec is licensed under a 3-clause BSD-style license. The copyright and patent licenses for the Opus algorithm are automatically granted to everyone and do not require application or approval. The patent licenses are included below together with the BSD-style license.

libxml2

XML parser and toolkit.

nginx-nchan

Fast, horizontally scalable, multiprocess pub/sub queuing server and proxy for HTTP, long-polling, Websockets and EventSource (SSE), powered by Nginx. https://nchan.slact.net/
Written by Leo Ponomarev (slact) 2009-2015.

nginx-upload-module

A module for nginx web server for handling file uploads using multipart/form-data encoding (RFC 1867). http://www.grid.net.ru/nginx/upload.en.html

OpenLDAP

Freely distributable *LDAP* (*Lightweight Directory Access Protocol*) implementation.

OpenSSL

An implementation of *Secure Sockets Layer* (SSL v2/v3) and *Transport Layer Security* (TLS v1) protocol.
This product includes software developed by the *OpenSSL Project* for use in the *OpenSSL Toolkit* (http://www.openssl.org/).
This product includes cryptographic software written by Eric Young.
This product includes cryptographic software written by Tim Hudson.

PHP

PHP is a widely-used scripting language that is especially suited for Web development and can be embedded into HTML.

This product includes PHP software, freely available from http://www.php.net/software/

php-ev

ev provides interface to libev library - high performance full-featured event loop written in C.

PHP-JWT

A simple library to encode and decode JSON Web Tokens (JWT) in PHP, conforming to RFC 7519.

pjproject

Asterisk fork of PJSIP

ScoopyNG

This product includes software developed by Tobias Klein.

SIP.js

A simple, intuitive, and powerful JavaScript signaling library http://sipjs.com

Copyright (c) 2014 Junction Networks, Inc. http://www.onsip.com

tftpd

TFTP daemon. TFTP is a simple protocol used for file transmission.

Copyright ©1983 Regents of the University of California. All rights reserved.

uwsgi

uWSGI application server container http://projects.unbit.it/uwsgi

Copyright (C) 2009-2014 Unbit S.a.s. <info@unbit.it>

WAVPlayerProject

WAV player.

Denis Kolyako May 28, 2007, see http://etcs.ru/copyright/

zlib

General-purpose library for data compressing and decompressing.

Copyright ©1995-2005 Jean-Loup Gailly and Mark Adler.