Release Notes

for Kerio Operator 2.1.4

Date: July 29th, 2013

1 Welcome to Kerio Operator 2.1.4

Welcome to version 2.1.4 of Kerio Operator. Please read below for an overview of new features in this version.

2 New features in Kerio Operator 2.1

Fax Support (since Beta 1)

Kerio Operator was without an official fax support so far. However the feedback of many Operator users persuaded us that fax is still a valid part of a modern Voice-over-IP PBX. Operator 2.1 comes with fax support in 3 flavors:

- T-38 support: Every extension you create in Operator can now handle T.38 traffic. T.38 is an ITU recommendation for the real-time transmission of faxes over IP networks. To send and receive faxes, you need to connect your fax machine to an Analog Telephone Adapter device (for example, Cisco SPA 112) and then assign one of Operator extensions to the ATA device. To send faxes reliably, your SIP service provider should support T.38 as well. (Most of them do, but it's always worth a check.)
- Fax-to-email: You can enable fax-to-email service for any of your users' extensions (in the tab "Advanced" of the extension edit dialog). All arriving faxes are then captured by Operator and sent to the user's e-mail address as PDF and TIFF attachments. In addition to this, you can define a PBX service at a separate extension number that will do fax-to-email as well. Remember you need to have SMTP relay defined so that your Operator server can send e-mails.
- PDF-to-fax: The MyPhone interface provides a simple method for sending faxes by uploading a PDF file to MyPhone. This is intended for occasional use. You first need to enable faxing in MyPhone's settings, then enter the recipient's telephone number in the dialing field and click "Send Fax". MyPhone will then open a file upload dialog.

Built-in DHCP server (since Beta 1)

There are deployment scenarios in which it is useful to have a separate DHCP server for VoIP devices. In larger networks, you may want a LAN segment dedicated to the voice traffic. In smaller networks, the router/firewall sometimes does not support the DHCP option 66 for automatic provisioning of the phones. For these situations, Kerio Operator 2.1 includes a DHCP server that is very easy to configure.

The integrated DHCP server is switched off by default so that it does not collide with your existing DHCP server. To enable it, edit a network interface, configure a static IP address on it and then check the option "Enable DHCP server". 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 (TFTP server address) set to Operator's own address with every address lease.

To manage DHCP leases, click the button "DHCP Leases" in the Network screen. You can reserve IP addresses for particular network nodes and it is also possible to disable option 66 for chosen addresses or set option 66 manually.

Dynamic voice conferences (since Beta 1)

Up to version 2.0, Kerio Operator lets you run one conference on each conference extension. Some of you need to run multiple conferences per extension (they do not want to waste DID numbers, for example). That is why Operator now has two conference types, called "statically configured" and "dynamic".

Callers who call the dynamic conference extension are asked to enter a conference ID and PIN. If the ID belongs to a conference that is currently in progress, the caller is connected to the conference, otherwise the conference is created and the caller becomes the first participant. The conference is ended when the last member leaves.

Other changes (since Beta 1)

There are many smaller changes throughout the Administration GUI and the Operator engine. This is the list:

- The activation wizard dialog (after first start) detects the time zone from the browser.
- The administration screens "Advanced Options" and "System" have been refactored. The screens "Security" and "Network" are now separate items in the menu tree. This change reflects their importance. There are several minor improvements in "Advanced Options".
- The import of provisioned phones from a CSV file was redesigned.
- All grids in the Administration GUI now react to keyboard commands (Ctrl-F sets focus to the search field, Enter opens the edit/view dialogs, Delete removes items).
- The Asterisk process can be restarted from the System Health screen ("Restart Telephony").
- Factory reset can be initiated from the System Health screen.
- The Operator Box edition plays a short tone sequence when the boot sequence completes (useful during upgrades).
- The Operator Box edition has a DHCP client enabled on the second network interface, in addition to the static address on the first interface.

Paging/Intercom (since Beta 2)

Paging is a special functionality supported by many SIP phones. It is also known as "Intercom" or "Public Address" function. If you call the phone and include a special header in the SIP signalling, the phone will answer the call automatically and will activate the loudspeaker.

Kerio Operator 2.1.0 Beta 2 supports paging in the form of paging groups. The administrator simply defines a paging group, assigns an extension to the group and adds some phones to the group. To page all members of the group, a user simply calls the group's extension.

Design improvements in the client interface (since Beta 2)

There are several design improvements in the Operator's client interface (also known as "My-Phone"). There's also an initial BLF support in the client interface that will be further improved in the next beta version.

Other improvements (since Beta 2)

Beta 2 brings these other changes:

- Voicemail can be disabled for individual user accounts.
- The Active Calls grid shows codecs used by individual calls.
- Call security constraints show limit utilization (how may calls have been counted for the given constraint, etc.)
- The login dialog for Administration and client interfaces has been redesigned.
- Voicemail messages can be up to 10 minutes long.
- The Call Queue edit dialog has been redesigned and improved.
- The audio player in the Administration and client interfaces now supports pause/play and search in both the variants (HTML5 and Flash).
- Asterisk engine upgraded to version 1.8.17.

Paging improvements (since Beta 3)

Beta 3 supports paging to individual extensions. Simply dial the prefix for individual paging (*910 by default) followed by the extension number. The prefix can be configured in the PBX Services screen in the Administration GUI.

The second improvement in both group paging and individual paging is that administrators can now control whether paging is allowed to interrupt calls in progress.

Other improvements (since Beta 3)

Beta 3 comes with these improvements and bug fixes:

- Support for distinctive ringing - Operator sends a SIP header that identifies whether the call is local (within the same PBX), external, or went via a ring group number.
- Added a watchdog utility for the Asterisk process.
- Added auto-provisioning support for Snom models 710, 720, and 760.
- Added auto-provisioning support for Cisco SPA 112, SPA 122, and Linksys 3102.

- Upgraded Asterisk to version 1.8.18 (fixes a possible deadlock in SIP channels).
- Fixed handling of repeated user import from a directory server that could break the user - extension assignment.
- Removed the maximum silence parameter that was too short and the silence detection was sometimes not reliable.
- Support for IE10 both in the administration GUI and the client interface.
- Show the number of digits used for call parking in the dial plan view (*5xx instead of *5...)
- Fallback on an interface can be disabled.
- Detect invalid extension numbers when importing users from a directory service and warn about them.
- The same telephone number can be used on multiple external interfaces (mainly BRI ports).
- Fixed access to Advanced options in case of upgrade from version 1.2.2 that had errors in configuration data.
- Fixed sending of fax by e-mail if the remote fax device ended the call too soon.
- The BLF functionality in the Kerio Operator client interface has been redesigned.
- Further look & feel improvements in the client interface.

Changes and improvements in Release Candidate 1

- Fixed warning about upcoming license expiration that was sometimes triggered too soon.
- Fixed the fax-to-email bridge, faxes could be received well but not forwarded by e-mail.
- Fixed update downloader, it might refuse downloading a new stable version if a previously downloaded beta/RC image was still present.
- Fixed import of users from a directory service, the Asterisk configuration was not regenerated in special situations.
- Fixed automatic assignment of extension numbers to newly provisioned phones, there was a race condition that could lead to duplicate extension numbers.
- Fixed Caller ID that could be sometimes incorrect for calls retrieved from call parking or for blindly transferred calls.
- Added fax detection options for fax-to-email bridge on standard extensions.
- Fixed page reloading that sometimes did not work correctly after having deleted some recorded calls in the Administration GUI.
- Client interface redesign finished.

Changes and improvements in Release Candidate 2

- Added a watchdog script for the LDAP synchronization binary to prevent frozen synchronization.
- Fixed race condition that could occur in voicemail IMAP integration when shutting down.
- The password generator now skips characters that are hard to read.
- Protect DNS resolving from errors if 127.0.0.1 was removed from the local address group.

- Fixed a possible incorrect agent order for call queues with the strategy "Ring in order".
- Fixed the configuration of DHCP option 66 in the DHCP server, the option could be sometimes disabled for a whole range.
- When downloading the original configuration of a provisioned phone, you can now select the network interface to be used in the configuration.
- Fixed a possible Javascript error when playing audio files.
- Fixed wrong timezone detection in Firefox 18 in Ubuntu 12.04.1 LTS
- Fixed scrolling in extension selection in IE10.
- Fixed sorting in Active calls grid.
- Fixed Javascript error that could occur when deleting call constraints in the Security screen.
- Fixed French translation of "Logout".
- Added drag & drop functionality for organizing favorite numbers in the client interface.

Changes and improvements in version 2.1.0 (final)

- Added dialog for managing Kerio Operator Softphones. The Kerio Operator Softphone is a SIP phone application for iPhone and Android. The softphones can be auto-provisioned securely from the Kerio Operator server.
- Fixed the Auto attendant that could play music on hold instead of silence when waiting for user's input.
- Improved displaying of the Privacy Policy and Legal Notices (a dialog window instead of a new browser window).
- Fixed status reporting when restoring a backup file on the box edition - the "server not responding" message could be incorrectly shown.
- Fixed Javascript error that could occur in IE9 after logging in.
- Fixed displaying of Release Notes during upgrade that could sometimes fail (occurred between 2.1.0 RC1 and RC2)
- Fixed downloading of very large packet dump files.
- Fixed access to UserVoice via the "Suggest Idea" button - it could fail for timezones west of GMT-2.

Changes and improvements in version 2.1.1

- Added support for certificate chains.
- Asterisk's sip.conf now explicitly contains the option "alwaysauthreject=yes". (This is the default setting but let us be sure.)
- Improved detection of telephony cards when cards are swapped or removed.
- SIP usernames containing a slash character are now supported by the SIP interfaces.
- The NTP server is restarted once a week to prevent problems when the remote NTP server assigned to Operator by the DNS round robin goes offline during the week.
- Log no errors when the computer Operator runs on has no serial port.
- The SSL certificate is now parsed correctly even if it does not end with a new line character.

- The distinctive ring flags for calls from a queue or group now override the external call flag.
- Fixed compatibility issues with Cisco 7940 that could occur after upgrading to Operator 2.1.0.
- Fixed configuration restore that might not set the correct timezone.
- Fixed fallback for inbound calls that might not work if the beginning of an internal extension matched the dial-out prefix.
- The web server restart from the command line could shorten the trial period.
- Fixed crashes that might occur when shutting down the Asterisk process.
- Fixed the detection of fax tones on PRI/BRI cards.
- It is now possible to enter several NTP servers.
- Improved handling of telephony cards when they are swapped or removed.
- Fixed update checker notifications, it might report the new version even after an upgrade in some situations.
- Fixed the Javascript errors that might occur in Firefox 19
- Fixed the Javascript errors that could be reported by the "Report problem" function in special circumstances (the Administration GUI displays multiple notifications and it runs in the second browser tab)

Changes and improvements in version 2.1.1 patch 1

- Fixed a Javascript error in the external interface edit dialog. The error influenced customers with the phone language set to British English.

Changes and improvements in version 2.1.2

- Improved support for IE10 in the administration GUI.
- Added support for several SATA drivers and RAID controllers.
- Improved busy tone detection on analog lines in several countries.
- Fixed the handling of number rewriting rules for auto-provisioned Kerio Operator Softphones. The rules might not be removed from the softphone when deleted in Operator administration.
- Fixed syslog server configuration - it could be lost after reboot.
- Fixed upgrade process that did not enable paging on Snom phones.
- Fixed dial patterns for Snom phones - newer Snom firmwares require that asterisk (*) is escaped.
- Fixed Javascript error that could appear when removing button rows in the client interface.

Changes and improvements in version 2.1.3

- Packet sniffer can be started on all interfaces at once.
- Disable the "Remove" button when no extension is selected.
- Fixed editing of IP address groups with more than 50 items.
- Fixed NTP daemon restart every Sunday.

- Fixed recording of calls to numbers that contain "+" or "*" signs.
- Fixed ring indications for Malaysia.
- Fixed the missing Date header in e-mail messages inserted by the voicemail/e-mail integration.

Changes and improvements in version 2.1.4

- Fixed reassigning of an extension from one user to another that propagated the original user's advanced forwarding options to the new user.
- Fixed the direct dialing function in the auto-attendant that did not work for single-digit extensions.
- Fixed the dialing patterns generated for auto-provisioned Snom phones - the pattern did not work well when using an outbound route with empty prefix.

3 Open Source Software Notice

Kerio Operator includes open source software. The complete open source code packages of these components are available in Kerio Software Archive at http://download.kerio.com/archive/.

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