Release Notes

for Kerio Operator 1.2.2

Date: May 2nd, 2012

1 Welcome to the Kerio Operator 1.2.2

Kerio Operator is a PBX software for small and medium business customers. The main focus of Kerio Operator is ease of use, both for the end users and the administrator. Kerio Operator is based on the VoIP technology internally but it also supports the standard telecom interfaces like PRI and EuroISDN.

For a description of what is new in version 1.2, see the section "New features in Kerio Operator 1.2" below.

2 Changes since Kerio Operator 1.2.1

Issues fixed in Operator's engine

- Fixed measuring of trial period - the trial could be shortened if the algorithm happened to run at one particular time.
- Corrected time zone settings for auto-provisioned Cisco 7961 phones.
- Fixed downgrade from version 1.2 to 1.1 that could loose call forwarding configuration in some situations.
- The voicemail access in MyPhone sometimes did not observe the IMAP configuration (port 143 vs. 993), and showed no voice messages as a result.
- Updated the Operator/Connect handshake for voicemail/e-mail integration to be compatible with Kerio Connect 7.4.
- Fixed a crash in the Asterisk process that could occur when stopping Operator.

3 Changes since Kerio Operator 1.2.0

Issues fixed in the Administration GUI

- Upgrade by uploading the upgrade image failed if you reloaded the GUI in the browser after having finished the file upload
- The mapping of external phone numbers to local extensions could shift after inserting a new number at the beginning of the list

Issues fixed in Operator's engine

- Fixed IMAP warning about unknown data that could sometimes occur when using the voicemail/e-mail integration
- The SIP user ID that differs from telephone number can be now used for registration only (phone number used in calls)
- Corrected downgrade from 1.2.x to 1.1.x that could fail if a new phone model unsupported in 1.1.x was auto-provisioned before the downgrade [Workaround: delete the phone entries before downgrading]
- Fixed BRI error messages that could occur during boot
- Fixed phone firmware upgrade that could fail for auto-provisioned SPA504G, SPA942 and SPA525G
- You can now override display name for outbound calls on a SIP interface
- The syslog service could freeze in a situation with an extremely high amount of data being written to the debug log
- One of the web server's processes could crash when attempting to test LDAP connection with missing configuration data
- Caller ID override could still display the original number in some situations

4 Changes since Kerio Operator 1.2.0 Release Candidate 2

Issues fixed in the Administration GUI

- The field that holds external numbers has been extended to allow up to 100 individual phone numbers on a SIP interface
- Corrected translations for languages in the Administration GUI
- Fixed Javascript error when accessing the provisioned phones screen as a read-only administrator (Auditor)

Issues fixed in Operator's engine

- Corrected BRI module reload procedure that could log warnings
- Incorrect called number was reported in Status->Calls for calls that went through a Ring group
- Fixed a memory leak and/or crash in the phone provisioning TFTP process that could sometimes occur if the TFTP traffic was filtered in one direction by a firewall
- Removing voicemail integration with Kerio Connect could sometimes fail
- SIP registration and SIP proxy setup could sometimes work with different IP addresses for the same SIP server when the SIP carrier uses a DNS round robin setup
- Fixed several resource leaks and potential deadlocks in Asterisk's voicemail IMAP module
- Did a change in Operator's web server that should prevent Microsoft's KB2585542 update from influencing the Admin GUI when used from Internet Explorer

5 Changes since Kerio Operator 1.2.0 Release Candidate 1

Issues fixed in the Administration GUI

- Completed Admin GUI translations - English, Czech, German, Russian, Italian, and French are now available
- Fixed a bug when creating a new IP address group (caused by recent code refactoring)
- The filter for displaying new phones in Auto provisioning could activate itself automatically when new phones had been detected shortly before the user refreshed the grid
- The administrator can now control what the caller hears when his/her call is being forwarded (play an announcement, then ring; ring immediately; ...)
- Removing a dynamic agent from a call queue could fail for some configurations

Issues fixed in the MyPhone interface

- Completed translation updates for all supported languages (English, Czech, German, Spanish, French, Hungarian, Croatian, Italian, Japanese, Dutch, Polish, Brazilian Portuguese, Russian, Slovakian, Swedish, Chinese)

Issues fixed in Operator's engine

- Added missing voice prompts that are required by Asterisk 1.8 (British English, American English, Czech, German)
- Fixed the configuration problem reported by the DAHDI driver on some machines (timing issue)
- Fixed a bug when automatically assigning extension numbers after an extension containing asterisk was created manually
- The default maximum call duration is now 2 hours (the former security limit was too strict)
- Fixed an issue that could cause a crash dump in the process that generates configuration files for Asterisk (related to along cards support)
- Operator incorrectly generated some user-level settings when automatically provisioning Polycom phones. Operator will reset these settings after the upgrade to version 1.2.0 RC2 and will not touch them after that.
- Automatic provisioning now supports a number of newer Cisco/Linksys models: SPA301-G2, SPA303-G2, SPA501G, SPA502G, SPA504G, SPA508G, SPA509G, SPA520G, SPA525-G2
- Dialing patterns for auto-provisioned phones could be generated incorrectly for special combinations of extension numbers due to an error in pattern optimization.
- Call history and voicemail envelope could be incorrect for calls that passed through a ring group

6 Changes since Kerio Operator 1.2.0 Beta 2

Issues fixed in the Administration GUI

- Network upgrade is now supported, downloads upgrade image directly into Operator
- Corrected text overflows in Russian translation

Issues fixed in the MyPhone interface

- Dialed phone numbers are normalized by removing spaces, dots, commas, dashes.
- Improved visual feedback when moving mouse over the tabs in MyPhone.

Issues fixed in Operator's engine

- When multiple SIP interfaces are assigned to a dial-out prefix, the interface to be used for the call is now selected using the rules for incoming calls to ensure consistent mapping of users to external numbers.
- Firmware upgrade via Operator is now supported for Snom models 300, 320, 360, 370, 820, 821, 870.
- Fixed an issue when having two or more call queues - Asterisk sometimes parsed the configuration incorrectly and callers could not join the queue. Because of the same bug, a call queue with static agents might not work.
- Fixed importing of a signed certificate from a certificate authority.
- Fixed call queue behavior for agents using extensions with multiple registrations
- Simplified the text of the e-mail message when sending voicemail over SMTP.
- Corrected an issue where the Auto attendant menu could play music on hold instead of silence when waiting for users' input.
- Single-digit extensions could not be dialed on auto-provisioned Linksys phones
- Transliteration of national letters to plain ASCII (needed to display caller's name correctly on phones that do not respect UTF-8) was not used when dialing from MyPhone.
- Transliteration of national letters is now used in the Dial-by-name function (so that ASCII characters match for their accented counterparts).
- Transferring an external call to another external number could fail on an interface with an empty dial-out prefix.
- Corrected dial patterns for Snom desk phones.

7 Changes since Kerio Operator 1.2.0 Beta 1

Issues fixed in the Administration GUI

- The list of music on hold could be displayed as empty.
- Auto-refresh in Status->Call Queues reset the selection in the grid.

- Fixed usability issues in Status->Call Queues.
- Auto-attendant editor allowed dialed numbers to be 10 digits at most. Corrected to 16 digits.
- It was sometimes not possible to delete call constraints.
- Import of provisioned phones from a CVS file could sometimes fail.

Issues fixed in Operator's engine

- User activated from a directory server did not have default empty ringing rules.
- Testing directory server connection with an empty password could end with a process segmentation fault.
- Server restart is enforced after time zone change to ensure that all system processes have the right zone.
- Call declination was sometimes not correctly signaled on a PRI line (fixed with new Dahdi driver version)
- Fallback was sometimes not working for incoming calls on EuroISDN
- Firmware upgrade via automatic provisioning now works correctly for Snom desktop phones (firmware upgrade not yet supported for M3, M9, and MeetingPoint models)

Minor new features

- Added server-side speed dials.
- Auto-provisioning now supports Linksys SPA 921/941.
- Auto-provisioning supports Snom M9.

8 New features in Kerio Operator 1.2

Upgrade to Asterisk 1.8

Operator 1.2 is based on Asterisk 1.8. This is an important change as it will allow us to build on some new Asterisk features in the future - for example the IPV6 support.

New Linux kernel

Operator 1.2.0 ships with Linux kernel 2.6.38. This should significantly improve support for new HW like new motherboards with integrated network interfaces.

Support for analog telephony lines

Operator can be now connected to analog telephone lines using the Digium's TDM410 card with FXO modules. This will help users in countries where SIP connectivity is not yet available or in situations where users prefer the plain old telephony. TDM410 supports up to 4 analog lines.

Call Pickup

One of the most demanded functions is finally there! Call pickup allows you to accept a call that is ringing on someone else's phone by dialing a special code. Call pickup is possible within defined rooms or even across rooms when using the directed pickup option.

Support for phones with BLF

BLF stands for "Busy Lamp Field". It is a function of some office phones that are able to indicate the status of other extensions (available, busy, call ringing). This is a feature that is useful for receptionists. It is similar to status indications in instant messaging clients and in fact it uses IM extensions of the SIP protocol.

Improved call forwarding configuration

The call forwarding now works better in situations where a user has several extensions. It is possible to have a different forwarding setup for each extension. In connection to this, voice mail can be enabled individually on a per-extension basis. Call forwarding is more user-friendly on the receiving end as well because we replaced Asterisk's "find-me" module (that has been a source of confusion for some users) with our own implementation.

9 Using Call Pickup in Kerio Operator 1.2

Call pickup is a feature that allows you to take a call that is ringing on someone else's phone.

Configuring Call Pickup

To configure Call pickup, you need to visit two configuration screens in the Admin GUI:

1. The PBX Services screen where you can configure the special numbers you need to dial to pick up a call. By default, the number for room call pickup is "*8" and "**" is a prefix for directed call pickup.
2. Definitions > Call Pickup Rooms where you can group extensions by rooms. The room call pickup is then limited to the same room. The size of rooms is entirely up to you - for example two adjacent offices where people can hear a phone ringing in the other office can form a single room in Operator.

Using Call Pickup in rooms

If a call rings in your room and you want to take that call using your phone, simply dial "*8" (or whatever code you have configured in PBX Services). If two or more calls ring in your office, "*8" will take the one that started ringing first.

Using Directed Call Pickup

Directed Call pickup allows you to take a call that rings on any extension on the same Operator server. You need to know the extension number. For example, if you want to take a call from extension 11, you dial "**11".

Anyone is allowed to do directed pickup by default. If you want to change that, simply restrict dialing of the prefix "**" in Call Permissions.

10 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/.

11 Legal Notice

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