BeIP LCS 4.0 Release Notes

Contents

1 Introduction

Based on successful Open Source software, the new *4.0* release of the **BeIP Software Suite** is a major milestone in the history of the project. The release introduces major modern features, as well as many user-visible changes for an always better telephony experience.

Figure 1: The brand new release.

Modernized and streamlined, this new release will enable IT managers to provide better user experiences with less time and effort.

The **BeIP Software Suite** consists in:

- The **BeIP Manager**
- The **BeIP FAX Module**
- The **BeIP Billing Module**
- The **BeIP Statistics Module**
- The *brand new* **BeIP Operator Console Module**

All software components have been reworked for *increased usability* and easier access to relevant information.

In total, more than **5 000 changes** have been introduced since the last major release.

Here is a summary of the most important changes.

2 What's new in the BeIP Manager?

2.1 Multi-Sites Handling

It is now possible to have one unique **LIBERTY Communication Server** handling several geographical sites with hard constraints on bandwidth requirements and network profiles.

It means the **LIBERTY Communication Server** will be able to dynamically:

- assign the appropriate network profile to any supported IP Phone requesting an IP address;

- switch between *HD Voice* and *Compressed* codecs to minimize bandwidth or maximize quality automatically;
- minimize the trafic going through the **LIBERTY Communication Server** to optimize bandwidth usage.

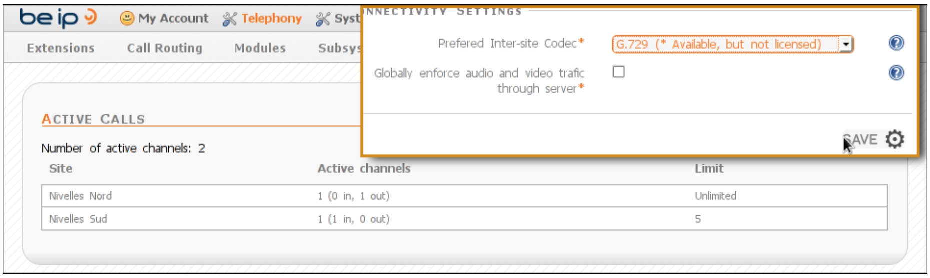

Figure 2: Inter-sites calling with bandwidth and quality management.

Moreover, IT Managers can keep an eye on the current traffic going in and out from each of the various sites thanks to the new dedicated page.

2.2 Load-Sharing and Redundancy

Customers with a high volume of calls and who want their telephony system to be up and running even in case of hardware failure can now rely on our brand new redundant active-active architecture.

In this mode, two redundant servers can share the load and allow inbound and outbound calls to keep working in the case of a failure of one of the two servers.

2.3 Cellphone Integration

This **One Number** approach consists in allowing incoming calls to be automatically redirected to the user cellphone.

If the call is redirected to the cellphone voice mail, it will be ignored and the **BeIP LIBERTY Communication Server** voice mail will be triggered instead.

Moreover, when a call is answered by a known cellphone, users can easily transfer the call back to any colleague or extension supported by the system.

This allows users to have only one unique number to reach their phone, their softphone or their cellphone

Moreover, users can now directly transfer or forward incoming calls to another user's cellphone using their internal extension followed by the `*3` *shortcut*. For instance, if you want to call the user `73` on their cellphone, you can dial the `73*3` number.

2.4 Granular Permissions Handling

Permissions handling has been completely reworked and now allows fine-grained permission definitions for all types of possible access restrictions like:

- Outbound routing;
- Call forwarding routing;
- Address book management;
- User interface possibilities;
- Telephony features.

All defined outbound routes can now be enabled or disabled individually in the permissions associated with the user profile. Dialing out can also be protected by a password for some users if configured.

2.5 Active Directory and LDAP Integration

The **BeIP Software Suite** now includes three levels of integration with Microsoft Active Directory and LDAP directories:

- LDAP-based user authentication;
- Automatic import of contacts into the shared address book;
- Automatic import of the **LIBERTY Communication Server** configuration.

2.6 Enhanced Call Detail Records

It is now possible to track call events and caller actions thanks to an enhanced CDR module. Administrators can see who called in, when, for how long, but also **monitor** events like *call transfers*, *call pickups*, *actions* in *interactive voice response* systems, who answered calls in the various defined *call queues*, ...

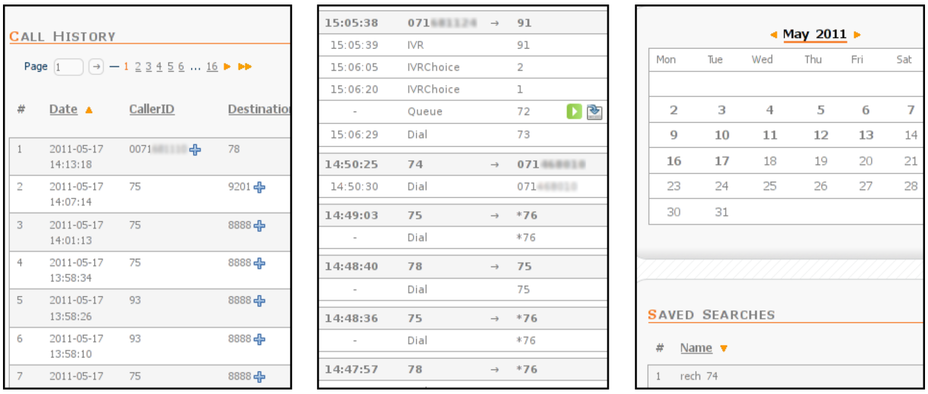

Figure 3: The enhanced CDR now comes with events monitoring and date-based filtering.

The module also comes with several improvements regarding *search filters*, *automatic reports*, and *reports export*.

The *search filters* and *searchable information* now contain advanced information about sites and trunks used for outbound calls.

2.7 New Address Book with Categories

This new release comes with a brand new address book allowing the system administrator to define categories in which users can put shared and private contacts following their personnal *permissions*.

Incoming calls will automatically present the appropriate Caller ID if the contact information is available to the callee.

BeIP Manager users will also be able to:

- Monitor their colleague status in realtime;
- Click to call address book contacts,
- Click to transfer their active calls to address book contacts.

The module also comes with the ability to import VCards.

Figure 4: The new address book interface with transfer and extension monitoring capabilities.

2.8 Enhanced Call Recording

The **BeIP LIBERTY Communication Server** now has the ability to automatically record calls placed and received by predefined users. Call recording is dynamically started or stopped as required when the call is transferred from one peer to another.

The *Enhanced Call Detail Records* interface provides a convenient way to access records. *FTP* access for fast downloading is also possible.

2.9 Enhanced Voice Mail Module

The module allows more fine-tuning with respect to user specific needs. Callers can be automatically transferred to an operator if they wish to.

Support for different *busy* and *unavailable* voice messages has been added, as well as the ability to mark voice mails as important. Moreover, former global settings can now be user-specific in order to improve frequent requirements.

Moreover, users can now directly transfer or forward incoming calls to another user's voice mail using their internal extension followed by the *2 *shortcut*.

2.10 Enhanced Call Forwarding

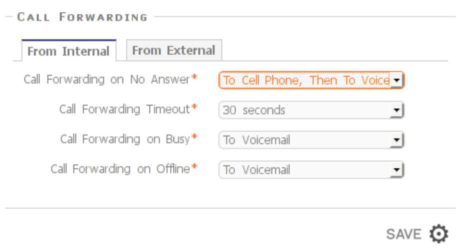

Figure 5: Better call forwarding.

Several *Call Forwarding* conditions have been added in user profiles allowing to have distinct behaviours when the user is *away*, *busy*, or if their phone is *offline*.

Distinct behaviors for *internal* and *external* calls are now possible.

A nice addition is the new ability to redirect calls to your cellphone if you are away or busy, and back into your voice mail on the **LIBERTY Communication Server** if you don't answer your cellphone. And if you take the call, you will be able to forward it back to another user using your cellphone's keyboard!

2.11 Miscellaneous Changes

- The **LIBERTY Communication Server** now supports *HD Voice* when available, as well as T.38 for *reliable FAX over IP* transmission and reception.
- It is now possible to create *Paging* groups for which calls are automatically recorded. In addition to this, it is also possible to broadcast prerecorded announces thanks to paging groups.
- Server-side *Do Not Disturb*, *Temporary Call Forward* and *Call Waiting* features.
- New *Switch* appplication allowing to change the routing of a call following the switch is *enabled or not* or following *time-based routing*.
- More *monitorable* features (Follow-Me, Do Not Disturb, Switches, ...)
- New user interface to send *SMS messages*.
- New possibility to *export* the users and devices lists.
- New *pickup notifications* so that users can monitor who is calling their colleagues and decide to pickup the call or not. Notifications are delivered with a *distinctive ring*.
- Improved *Call Queuing* with new *distribution strategies*, new *configurable timeouts* and the addition of *queue helpers* (dynamic agents).

- Improved default *Feature Codes*.
- Allow users to authenticate with the **BeIP Manager** using their IMAP account.
- Allow new action associations in the *Interactive Voice Response* systems.
- Easier integration with third-party software depending on the *Asterisk Manager Interface*.
- Improved support and integration with various *device types*: Kirk DECT Systems, Patton Gateways, new Polycom devices, ...
- If equiped with appropriate hardware, the **LIBERTY Communication Server** is able to generate and handle alerts in case of accident thanks to the new *Man-Down* support.
- Improved *service monitoring* and alerting with live monitoring of the various network services provided with the **BeIP LIBERTY Communication Server**.
- Enhanced configuration generation *performance*.

The software suite also comes with various other *enhancements* and several *bugfixes*.

3 What's the new in the optional modules?

3.1 The new Operator Console module

The **BeIP Software Suite** now offers a brand new operator console module. This module allows having a global visualization of all ongoing calls handled by the system or waiting in a queue to be answered in all geographical sites.

The operator can easily filter system extensions by site and decide to transfer their own calls in one click to a colleague extension or cellphone, or directly to their voice mail.

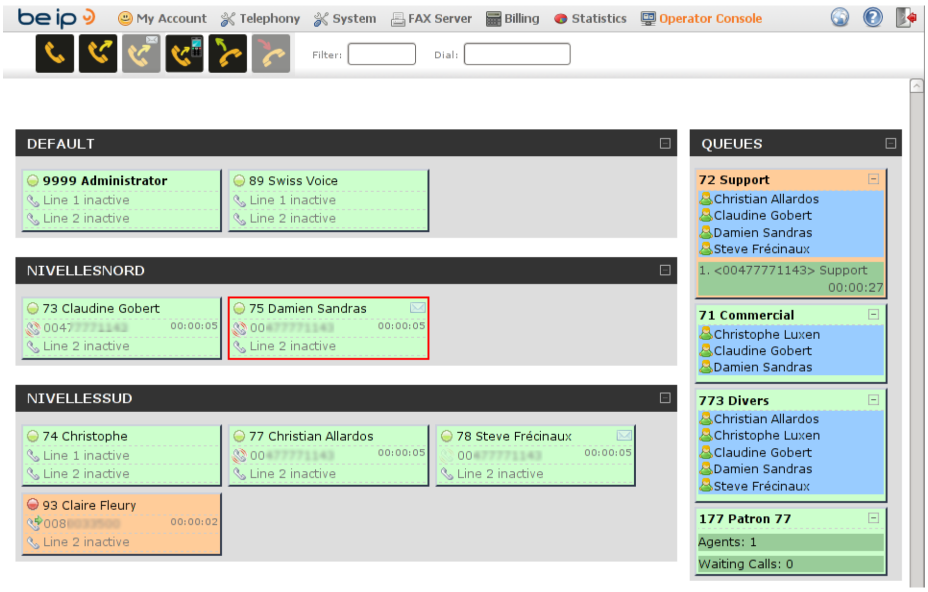

Figure 6: The new operator console module allowing full call control.

3.2 BeIP Statistics

The **BeIP Statistics** module now comes with improved statistics display. Accessing call statistics and analizing helpdesk service level agreements compliance has never been so easy.

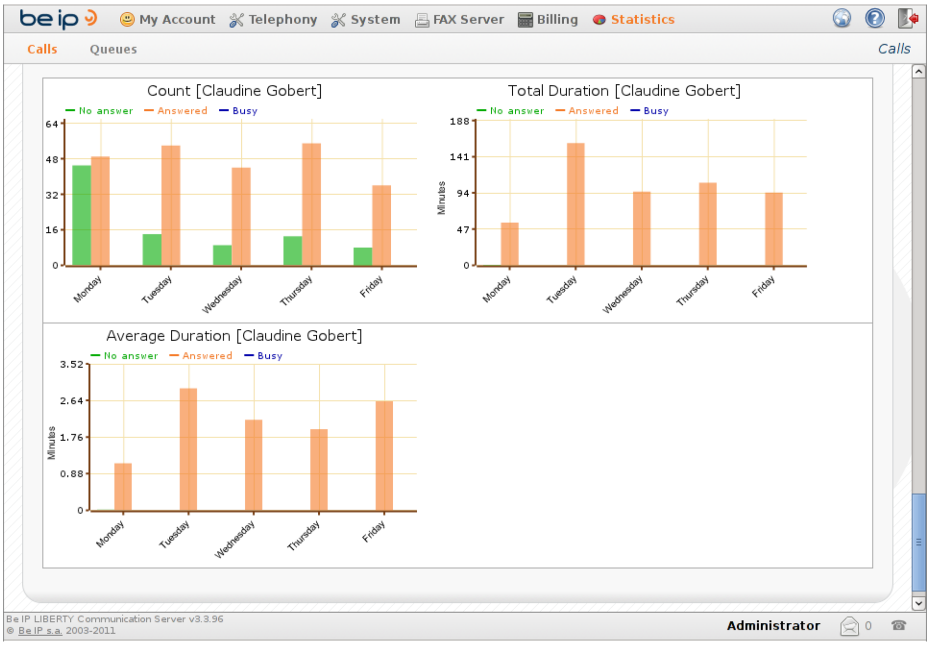

Figure 7: The improved statistics module.

3.3 BeIP FAX Server

The **BeIP FAX Server** module now adds the ability for users with the appropriate access rights to upload pages to be sent by FAX.

3.4 BeIP Billing

The **BeIP Billing** module is now able to handle several geographical sites. It also takes into account the billing related to SMS delivery.

4 About Be IP

Be IP is focused on bringing IP Telephony solutions based on open standards and technologies.

The company was founded by Voice over IP pioneers and is part of the European *TELKEA* group whose key members are Téléphonie Luxembourg, Quantum ICT, Riviera Telecom and Netline.

With major customers both in the public and private sectors, **Be IP** is positioned as a key player in the conception and the deployment of friendly *IP telephony* and *unified communications* solutions.

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