

gossip

Technical description



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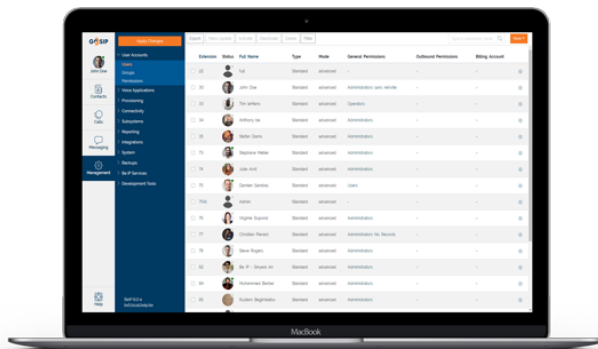
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1. Gosip

1. *Gosip Software Suite*

1.1.1. General

The Gosip **software suite** is a **unified communications and collaboration** solution entirely developed by **Be IP** that provides all the features offered by the most complete systems as standard, but at a competitive price and with a *simple* and *intuitive* configuration system.

The system offers you maximum *reliability* and *flexibility* in order to better meet your requirements.

It allows telephony integration with the IP network and thus supports:

- **teleworkers** ;
- **Remote offices**
- **employee mobility** ;
- **Virtually unlimited number of internal lines.**
- **traditional telephony to the national or international at a lower cost** via IP telephony providers or via traditional providers;

It also allows for extensive integration of telephony with the various IT tools:

- **integration with your CRMs and ERPs or SQL databases;**
- **integration with various Microsoft tools**

It is based on an intuitive web interface that will allow administrators to quickly and easily configure their system to get the most out of it, and users to have direct access to essential functions.

Being based on standards, it will allow you to **evolve** your phone system in line with the needs

of your business, **without constraints**.

Updates to both the phones and the exchange are planned and automated and automatically bring in the newly developed features.

In addition, the integration of **WebRTC** audio technology into our product allows any user of the system to make and answer their calls using a simple modern web browser. Whether the origin of the call or its destination is another **WebRTC** user or a landline or an external number!

It is no longer necessary to have a fixed computer or to physically install a softphone on the users' PCs since a simple browser pointing to the **Gossip user interface** is sufficient.

In addition, if you opt for the Cloud installation of our product, you will be able to subscribe to the **"Gossip Everywhere"** option which allows you to have access to the Gossip user interface and its **WebRTC component** from anywhere without requiring VPN access.



1.1.2. Installation Modes

Gossip can work in different environments, in order to adapt to any type of need. The features are the same in all cases.

The software can be installed:

- **In "On Premise" mode**
 - on a hardware platform provided by **Be IP**
 - ✓ or the **L50 hardware platform**, designed as a lightweight solution
 - ✓ the **L700 hardware platform**, for large deployments
 - on a customer-provided virtualization platform
- **En mode « Cloud »**
 - on a cloud virtualization platform located in our data center via our **Cloud offering** adapted to modern needs for flexibility

1.1.3. Features

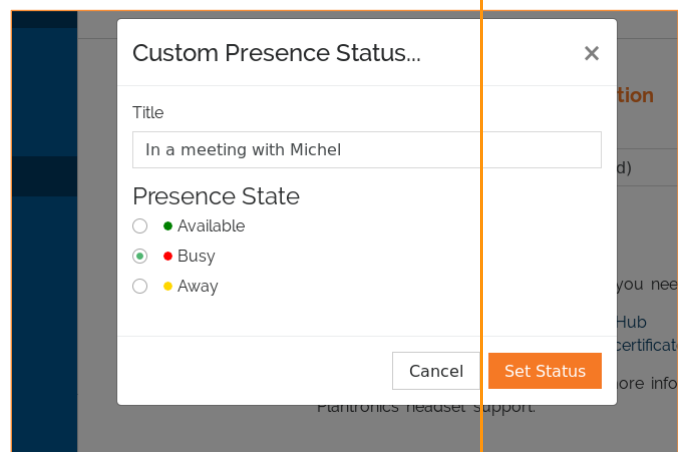
The main features are described in this section. All of the features described in this section are available with any system provided by **Be IP**.

Access to additional features is controlled by the system administrator based on user profiles.

Each user has access to their own portal allowing them to manage their communications, make and take calls, etc.

A dashboard is presented to him at the login in order to summarize the latest communications.

Each administrator has access to a simple and intuitive configuration system that does not require any special telephony or computer knowledge.



Please note that the entire software is **fully translated into French** but is also available in the following languages:

- *Dutch*
- *English*

The home screen:

New Call **New Message** **Addressbook**

“
Consider well the proportions of things. It is better to be a young June-bug than an old bird of paradise.
— Mark Twain, "Pudd'nhead Wilson's Calendar"”

Last Contacts Add Contact

	Walter White	+356608254400	wht.walter@kgd.lu	
	Elliot Manier	+3274589690	manier.elliott@dsu.com	
	Anne Burgeon	+3271258953	anne.b@mj.fr	
	Park wook	+3281511109	ms@lobor.be	
	Vincent Demeaker	+32474854614	vnt.stm@gears.be	

The address book:

Filter New Contact

Fav.	Status	Full Name	Ext.	Mobile	E-mail	
☆		Be IP - Sinyeol An	83	+32478830760	sa@beip.be	
☆		Damien Sandras	75	+32477771143	dsandras@beip.be	
☆		Muhammed Berber	84	00479332650	kikool@beip.be	
☆		Rustem Begimbetov	85		rb@beip.be	
☆		Steve Fréciniaux	78	00477771608	sfracinaux@beip.be	
☆		Virginie Delvaux	76	00479380010	vdelvaux@beip.be	

Group management for the administrator:

Apply Changes [Refresh] Save

Navigation: User Accounts, Users, Call Groups, Support Second-Line, Call Customer Display, Permissions, Voice Applications, Provisioning, Connectivity, Subsystems, Reporting, Integrations, System, Backups, Be IP Services, Management, Help (BeIP 6.0 α, bx5.local.beip.be)

Edit Group

Name:

Service
 Pickup Group

Group Members

Start typing to search users Add

- Damien Sandras (dsandras - 75)
- Christian Allardos (callardos - 77)
- Sinyeol 신열 An (san - 83)
- Steve Frexinaux (sfricinaux - 78)
- Steven Deferme (sdf - 74)

User management:

Apply Changes [Refresh] New +

Navigation: User Accounts, Users, Call Groups, Permissions, Voice Applications, Provisioning, Connectivity, Subsystems, Reporting, Integrations, System, Backups, Be IP Services, Management, Help (BeIP 6.0 α, bx5.local.beip.be)

Export	Mass Update	Activate	Deactivate	Remove	Filter	Search	
<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>		
73		Stéphane Clajot	Standard	Users	Standard Outbound	default	
74		Steven Deferme	Standard	Administrators	Standard Outbound	admin	
75		Damien Sandras	Standard	Administrators	Standard Outbound	default	
76		Virginie Delvaux	Standard	Users	Standard Outbound	default	
77		Christian Allardos	Standard	Administrators	Standard Outbound	Résidents	
78		Steve Frexinaux	Standard	Administrators	Standard Outbound	default	
79		Stijn Everaert	Standard	Administrators	Standard Outbound	-	
81		User 81	Standard	Users	Standard Outbound	-	
83		Sinyeol 신열 An	Standard	Administrators	Standard Outbound	default	
84		Muhammed Berber	Standard	Administrators	Standard Outbound	-	
85		Rustem Begimbetov	Standard	Administrators	Standard Outbound	-	

Our licensing policy is not to charge for system features but to charge for one license *per user* (a virtual fax is considered a user as well).

The same user can have as many terminals as they want (a mobile phone, a landline phone, a cordless phone, a softphone on a PC, tablet or smartphone, their WebRTC softphone) without requiring an additional license, whether the software is installed in "**On Premise**" or "**Cloud**" mode.

The user license gives you unrestricted access to all the telephony and unified communications features provided by the **Gossip software suite**.

In addition to the *standard* unified communications and collaboration features, *additional features* are also provided at no additional cost, as detailed below:

- *Multiple connectors:*
 - *Connectors for calendar integration such as Exchange/Office 365, Google Calendar, CalDAV*
 - *connectors allowing the authentication and/or import of users and/or the import of contacts via LDAP, Active Directory, MySQL or PostgreSQL, JSON or Excel*
- *Several modules:*
 - *an invoicing module*
 - *a call statistics module*
 - *a virtual FAX server module*
 - *an instant messaging module*
 - A **built-in** WebRTC softphone
 - An **additional smartphone softphone**
 - An **integrated** WebRTC operator console

In addition, it is interesting to note that a whole range of ancillary software can interface with the **Gossip software suite**.

1.1.3.1 Basic features

The items explained below are not an exhaustive list, but rather a summary.

Classic features

The Gossip **software suite** offers all the classic features offered by traditional exchanges, including, but not limited to:

- *call, and be called, with the calling number displayed,*
- *call redirection, conditional (online, no answer) and unconditional with a distinction between internal and external calls, and this also according to the state of presence (absent, busy, on leave),*
- *follow him,*
- *blind and advisory call transfer,*
- *suspension of appeal,*
- *support for multiple lines and simultaneous telephone communications,*
- *call groups, with the possibility of call interception (directed or group),*
- *hold and transfer music (via mp3 files),*
- *the possibility of intercom with or without pre-recorded announcement,*
- *Call parking, with recovery of a call in parking from a remote extension.*

Advanced features

It also offers advanced features (non-exhaustive list):

- support for *HD Voice*,
- advanced SIP **support**,
- support for **virtualization** and **cloud environments**,
- *queues*, with *accompanying music* and *configurable voice messages* ,
- *Automatic distribution of calls, among members of a group associated with a queue, according to a defined policy.*
- *(Conversation) channel monitoring and call intrusion*
- *interactive, multilingual voice menus with automated caller support, allowing precise actions to be associated with key presses,*
- *personalized voicemail*, with voicemail signaling on phones,
- possibility of organising *conference rooms*, with possible protection by *PIN code*,
- *intelligent routing of calls*, inbound or outbound, based on prefixes or tailor-made scripts and according to time periods,
- Possible definition of *call policies*, allowing restrictions to be modified according to the context of the call,
- Possible definition of *policies for access* to functionalities (software, exchange, redirections, etc.)
- **automatic or manual** call recording,
- *automatic phone configuration* ,
- advanced **redundancy support**,
- Cloud **monitoring** platform with advanced SIP protocol and system health statistics for **predictive maintenance** purposes (access to this platform may be subject to an additional cost),
 - *Shared address books* with categories and tags, accessible via LDAP,
 - *line supervision*,
 - *fixed/mobile convergence, integration with GSM* : single number concept and possibility of call transfer assisted by the exchange to internal extensions,
- support for *multi-sites* with automatic limitation of the number of inter-site calls and choice of the best codec according to the call path (**Call Admission Control**),
 - Support for *call tracking* within the system: recording of actions taken by the caller in interactive voice menus, transfer tracking, ...
 - support for *hot desking* : mobility of employees within the company via a login/logout on workstations,
 - video communications *support* (excluding WebRTC),
 - Visualization of the *presence status* of your colleagues in the address book (online, offline, in conversation, do not disturb mode, absent, busy, on leave, etc.)
 - **management of attendance statuses** (*absent, busy, on leave*), with automatic adaptation of the call flow accordingly (SIP support, not requiring the installation of specific software),
 - **Full WebRTC audio support** to take and make calls from the management interface without having to install a softphone,
 - ...

IP, CTI/Unified Messaging Features

These include, but are not limited to:

- *possibility of making calls in voice over IP, via a SIP provider, with the terminal best suited to the worker's needs: his WebRTC-compatible web browser, his fixed computer, his softphone, his mobile, etc.*
- facilities for *teleworking*, via the **WebRTC** component, or via any other SIP terminal, ...
- *Consolidation* of remote sites into a single virtual global site, with integrated telephony,
- Integration with IT (*CTI*): voicemails transmitted by e-mail, integration with traditional office tools, with databases, with CRMs, with ERPs, with the company's LDAP directory, Outlook integration, possibility of full integration of the solution with your business applications via a **complete REST API** to manage most aspects of the system, ...
- *Unified messaging* : Send and receive FAXs in PDF, via the user software
- *Extensibility and scalability*

The Gosip **solution** is validated and recommended (among others) for the following Belgian SIP providers:

- **Proximus**
- **Orange**
- **Nethys Flight / WIN**
- **Infrac**

Reliability and redundancy

L50 and L700 Servers

Whether the **Gosip** suite is installed on an **L50** or **L700 hardware platform**, the hardware server has been designed for reliability.

The **L700** server offers two hot-pluggable RAID-1 drives and a redundant, hot-swappable power supply as well. The **L50** model is based on an SSD disk. Both models have a backup network interface, thus allowing continuity of service despite a failure in the energy supply or in the network connections.

Installation in "On Premise" mode

All versions of the software suite can be placed in a redundant "active-passive" configuration. This means that two servers will be placed side by side (preferably connected to different power sources and on different switches), and that in the event of a failure of one of the servers, the other server will take over, in a completely transparent way for all users.

An "active-active" redundancy mode is also available for large deployments, which require many resources but do not want to compromise on security. In this mode of operation, all phones and applications are evenly distributed over two instances. In the event of a failure of one of the servers, the phones and applications managed by that server will be taken care of immediately by the second server hosting the **Gosip** suite, in a transparent manner, until the failed server is repaired.

In order to go further in terms of capacity, it is also possible to opt for an architecture based on several logical units, each of which may or may not be redundant. In such a mode, the number of users that can be managed is virtually unlimited. This also makes it possible to implement a survivability solution thanks to which one (or more) critical site(s) can continue to operate in the event of isolation from the rest of the infrastructure. This remains transparent both in terms of functionality for users and in terms of management for administrators.

Because the security of your data is priceless and all the technology in the world will never protect us from a fire, water damage or human error, the **Gosip** software suite also offers an automated backup system that will take care of putting your data in a safe place, allowing a very fast recovery of a system configured according to your needs and fully functional if by chance a

disaster occurs.

Redundancy in "active-active" mode or in "active-passive" mode can be enabled whether the software suite is installed on a dedicated hardware platform or in a virtualization infrastructure.

Cloud installation

When the choice is made to work in **the cloud** rather than **in on-premise** mode, rest assured that all modern redundancy mechanisms are implemented at the level of our data center infrastructure through, in particular, a high-availability virtualization platform distributed in two different buildings in the data center.

1.1.3.2 Additional features

This section describes some features that are ancillary to Unified Communications and that are nevertheless possible with any installation of our software at no additional cost.

LDAP / Active Directory Connector

The LDAP / Active Directory connector is a data and authentication connector that allows:

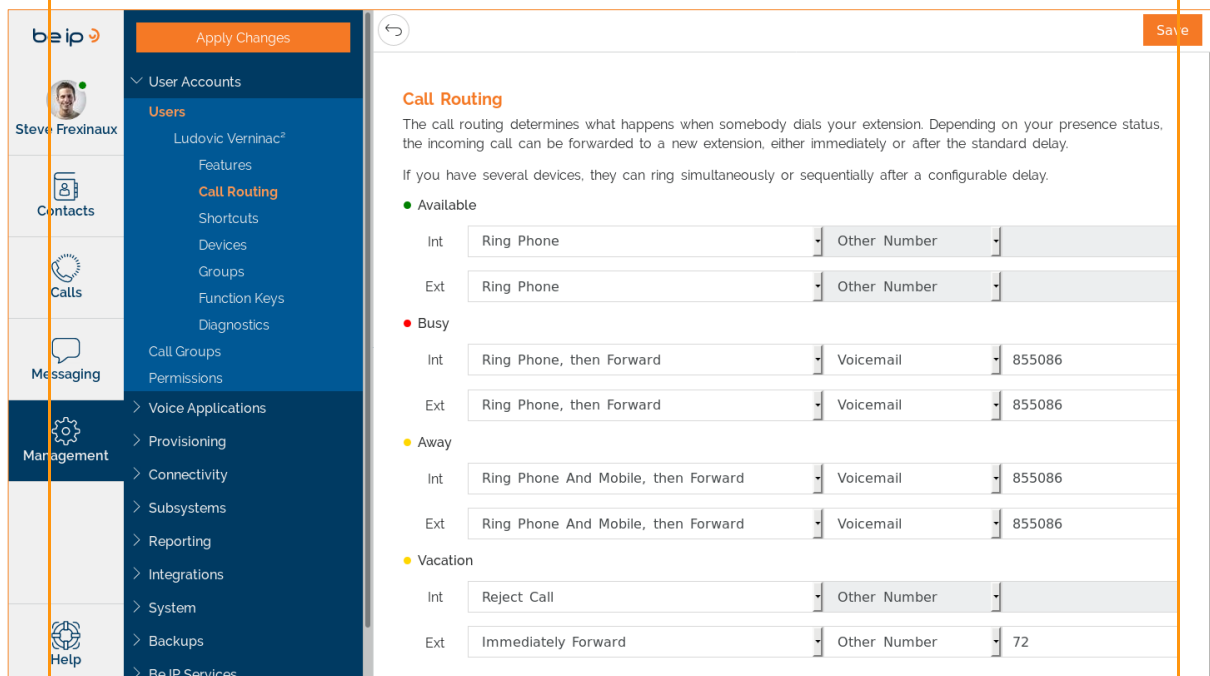
- *Users to log in to the web portal using their LDAP/Active Directory credentials.*
- *the automatic import of contacts and their phone numbers into the address book of our software;*
- *automatic import of system users from the LDAP / Active Directory server: this avoids the need to define the different system users for each service to which they have access.*

The synchronization of some of the data can be automatic or manual.

Google Calendar and Microsoft Exchange/Office 365 connectors

The Google Calendar and Microsoft Exchange / Office 365 calendar connectors make it possible to synchronize a user's presence status according to their appointments and holidays.

The call flow can be adapted depending on whether the person is in an appointment at the office, or on a trip, or on leave for a given period (note that not all calendar servers allow a distinction between several different presence states).



Call Routing

The call routing determines what happens when somebody dials your extension. Depending on your presence status, the incoming call can be forwarded to a new extension, either immediately or after the standard delay.

If you have several devices, they can ring simultaneously or sequentially after a configurable delay.

- **Available**

Int	Ring Phone	Other Number	
Ext	Ring Phone	Other Number	
- **Busy**

Int	Ring Phone, then Forward	Voicemail	855086
Ext	Ring Phone, then Forward	Voicemail	855086
- **Away**

Int	Ring Phone And Mobile, then Forward	Voicemail	855086
Ext	Ring Phone And Mobile, then Forward	Voicemail	855086
- **Vacation**

Int	Reject Call	Other Number	
Ext	Immediately Forward	Other Number	72

Compatibility is ensured with any system that supports the CalDAV standard, or with versions of Microsoft Exchange 2007 and newer, or with Office 365 or Google Calendar.

SQL, JSON, and Excel connectors

SQL (MySQL and PostgreSQL), JSON, and Excel data connectors enable:

- Automatic import of contacts and their phone numbers into our software's address book
- Automatic import of system users

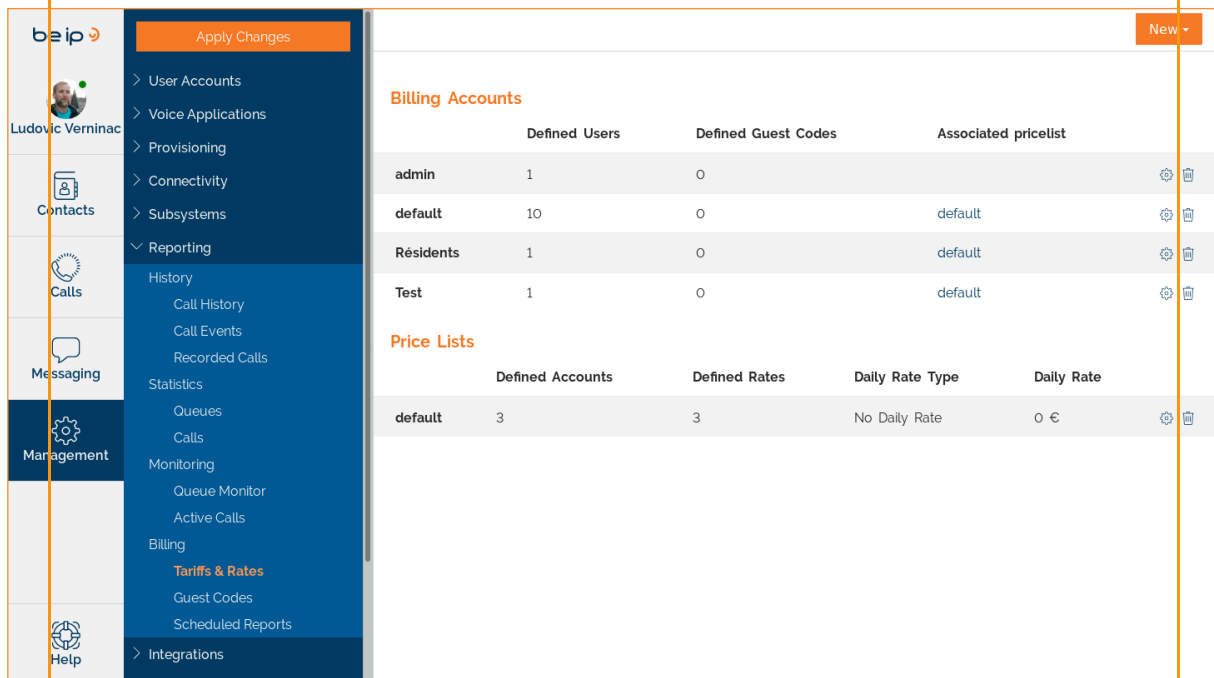
In the case of JSON and Excel connectors, files can be downloaded automatically from a web server.

The synchronization of some of the data can be automatic or manual.

Modules

Invoicing module

We also offer an invoicing tool.



Billing Accounts			
	Defined Users	Defined Guest Codes	Associated pricelist
admin	1	0	
default	10	0	default
Résidents	1	0	default
Test	1	0	default

Price Lists				
	Defined Accounts	Defined Rates	Daily Rate Type	Daily Rate
default	3	3	No Daily Rate	0 €

On the basis of call history and predefined cost tables, it will make it possible to produce a provisional bill based on specific search filters (calls abroad of more than 30 minutes, calls made to a given number, or from a given number, etc.).

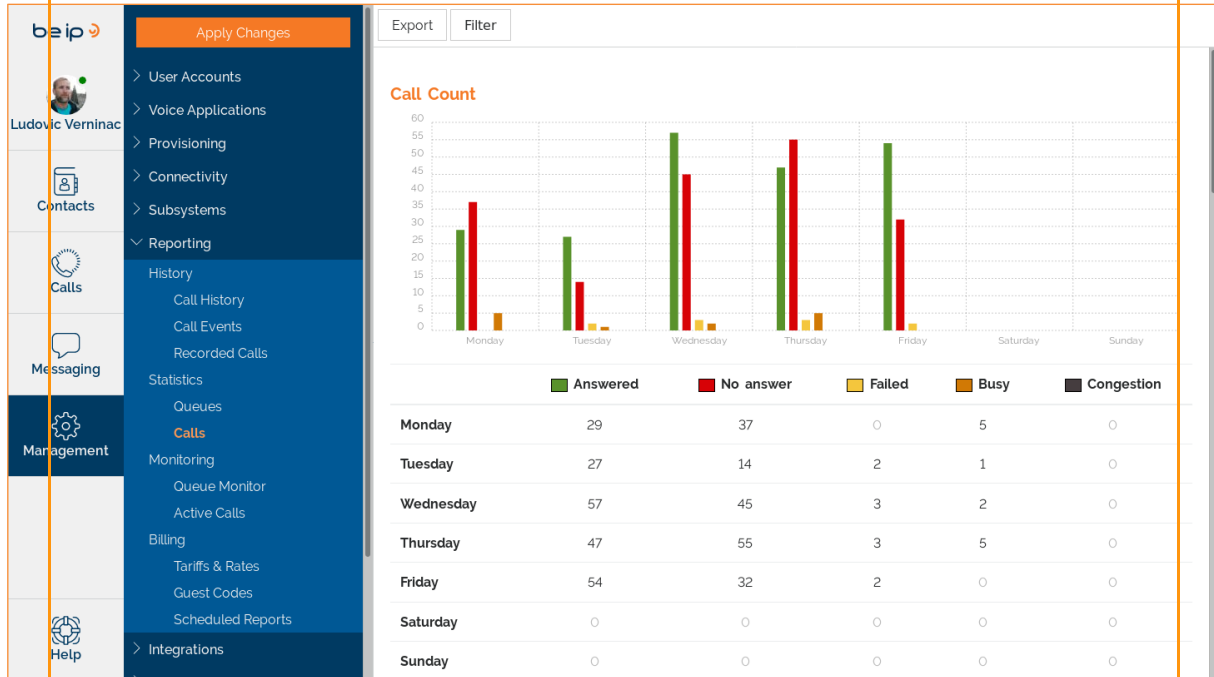
Cost tables can be introduced via a CSV file to be imported. The cost is calculated in real-time when the call ends.

The invoicing module also offers a **prepaid system** with support for **guest accounts**. When configured, it allows users with a guest account who have loaded their account with sufficient credit balance to make calls from any extension that is configured with the feature. A billing system then makes it possible to issue an invoice showing the activity of the account (charging, calls, etc.) at any time. This can be particularly interesting in hospitals, nursing homes, etc.

Statistics Module

Our statistics tool allows you to establish call statistics relating to all calls in the different queues. It makes it possible to determine, for example:


- *call volumes,*
- *the most active agents,*
- *the busiest times of the day,*
- ...





The list of statistics measured is quite complete.


The **graphs** are displayed intuitively and give a clear and accurate representation of the situation for a given period of time.


Real-time visualization or access to more complete statistics require **Gossip Advanced** licenses for the profiles concerned (see "Call center features").






Ludovic Verninac


Contacts


Calls


Messaging


Management


Help

Apply Changes


- > User Accounts
- > Voice Applications
- > Provisioning
- > Connectivity
- > Subsystems
- > Reporting
 - History
 - Call History
 - Call Events
 - Recorded Calls
 - Statistics
 - Queues**
 - Calls
 - Monitoring
 - Queue Monitor
 - Active Calls
 - Billing
 - Tariffs & Rates
 - Guest Codes
 - Scheduled Reports
- > Integrations
- > System
- > Backups

Export


Filter

■ Given up by Caller	2
■ Forwarded after timeout	2
■ No agent logged in	0
■ Forwarded because queue was full	0
■ No agent available	0


Answered Calls



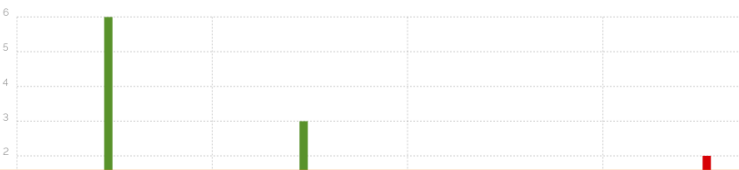
Unanswered Calls



Total Inbound Calls

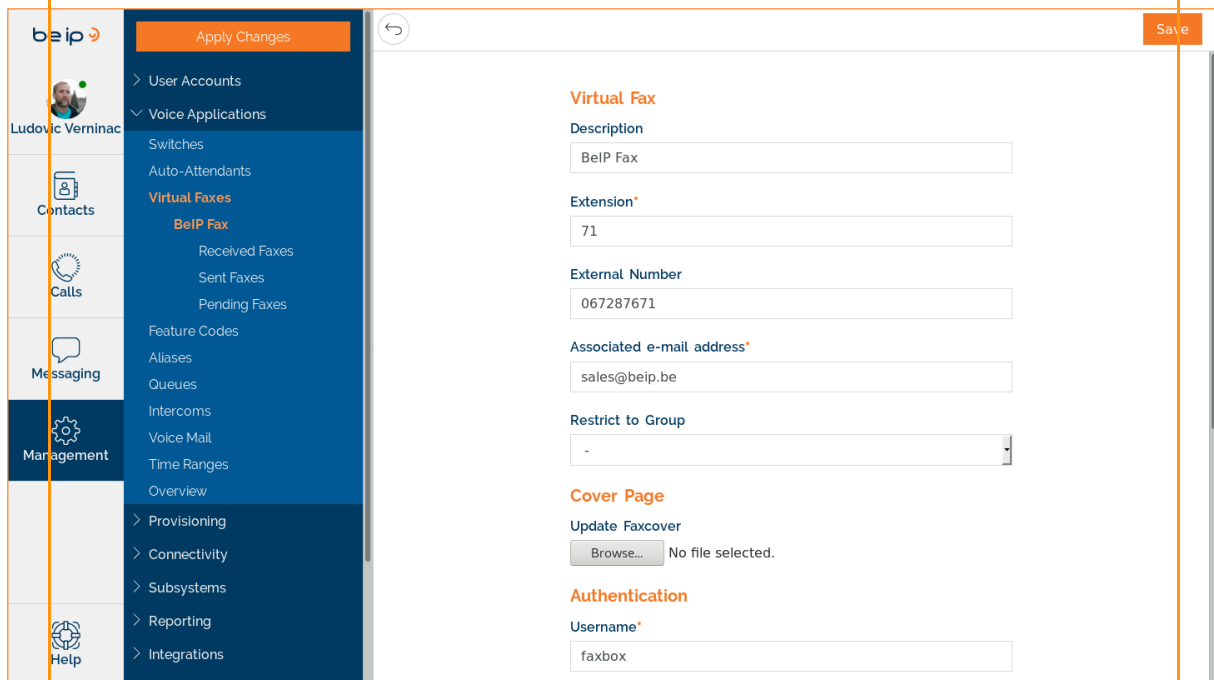


Quality of Service



Virtual FAX Module

We also provide a virtual FAX server, both for sending and receiving.



As input, any incoming FAX is converted to PDF and emailed as an attachment.

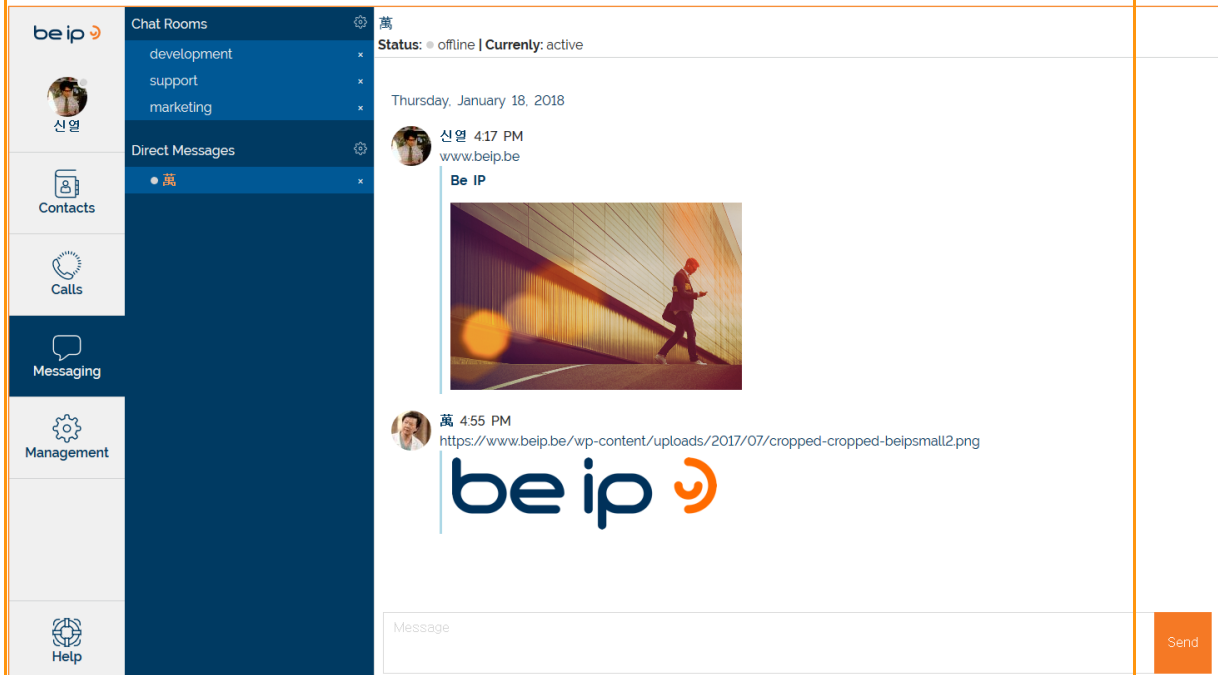
As an output, any email sent to a particular email address is converted into a FAX and sent to the recipient. This mode allows you to receive acknowledgments of receipt by return e-mail.

The FAX module supports the **T.38 protocol** for reliable transmission of FAXs over IP.

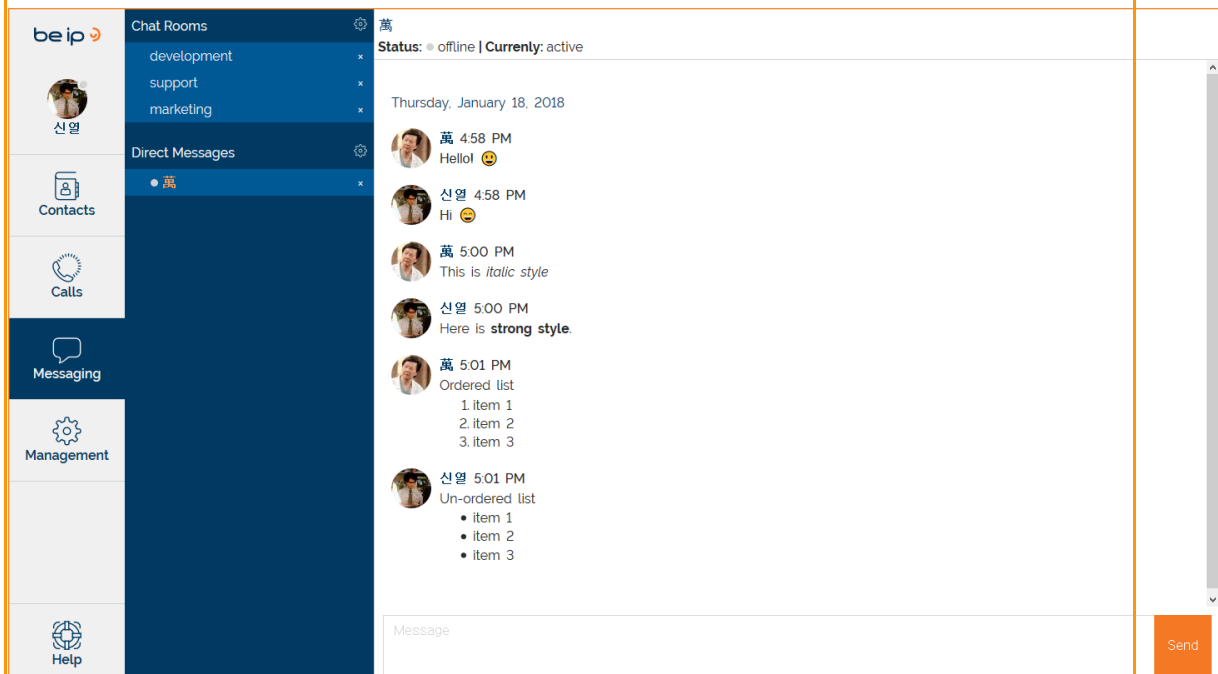
Instant messaging module

The **Gosip** system also comes with an instant messaging module that allows different employees to communicate with each other through text messages.

There are two ways to exchange messages: direct messages between two employees and messages in discussion groups between employees interested in the same issue.



2. Peripherals



1.2.1. General Description

While compatible with any SIP device, our offering offers a selection of IP phones and various

peripherals that combine *flexibility* and *robustness* to communicate with the **Gosip**, while maximizing *integration* with the latter.

We offer, among other things:

- *IP phones;*
- *WIFI IP phones;*
- *DECT IP phones and gateways;*
- *analogue (PSTN) and digital (ISDN BRA, ISDN PRA, etc.) gateways;*
- *conference stations;*
- *des SIM-Boxes ;*

Each of our IP phones is connected to the LAN, allowing you to use the DATA network to make calls. They have standard classic features such as call forwarding, conditional and unconditional call redirection, voicemail signaling, call suspension, multi-line support, LCD screen, hands-free mode, etc. They also include a built-in switch, allowing you to connect your desktop computer to the phone in order to gain access to the LAN, which reduces the number of network sockets required. In addition, most of the models offered are equipped with *Power over Ethernet*, allowing each substation to take its power via the network cable. All models offered work in a *plug-and-play* way with our system.

In addition to wired IP phones, we also offer IP phones that work in WIFI or DECT.

In the case of a DECT installation, we work with IP DECT technology, acting as a SIP – DECT gateway that allows a seamless interconnection with the **Gosip** software. Each employee can have an IP phone and a DECT handset in their possession.

The following hardware is required for DECT deployment:

- *SIP - DECT server*
- *antennas (and repeaters)*
- *DECT phones*

In the case of a DECT deployment, a prior audit is required.

Finally, we also offer gateways to the analogue and digital ISDN world.

Analog gateways allow one or more traditional analog device(s) such as a fax machine or analog/DECT telephone to be connected and signaling translated to appear to the system as "voice over IP" devices. In the case of FAXs, the T.38 protocol is supported to ensure flawless transmission reliability.

Digital gateways allow translation to be carried out between the traditional digital world (ISDN BRA or PRA) and the SIP world.

Finally, you can connect GSM/SIP gateways to the system that allow you to make calls to GSM at the GSM rate in a transparent way for your users.

We invite you to discover the technical sheets of our different models in the appendix of this document.

1.2.2. Gateways

Gosip software can connect to networks running with traditional technologies via dedicated gateways. The use of a gateway allows for greater granularity at the overall system level and therefore higher redundancy.

1.2.2.1 ISDN and analog gateways

We work with the Patton brand for analog and ISDN gateways.



Patton has been manufacturing voice and data communications equipment for carrier, enterprise and industrial networks around the world since 1984.

Patton VoIP gateways can interface with all types of telephone interfaces on the market, including ISDN PRI & BRI, T1, E1, FXS and FXO and more.

Combined with **Be IP** technology, SIP compatibility is ensured, as well as compatibility with the T.38 protocol, a major prerequisite for reliable FAX transmission.

Patton's VoIP solutions are the preferred choice of many companies or operators around the world.

In terms of analog gateways, the Poly brand also offers interesting connectivity solutions to connect fewer analog ports. This solution is ideal for connecting a fax machine in a corridor or a remote analog station.



1.2.3. SIP Terminals

Our offer is compatible with phones of various brands. The use of standard protocols in the **Gosip** software suite ensures compatibility with many different brand models.

For your convenience, we have grouped the phones available into various categories suitable for any type of need. The advice given in this section is based on our opinion, following a long experience of several years.

We mainly offer a range of landline phones:

- *The Poly Edge E range, ideal for standard use*

All Poly phones are PoE and come without power but with a network cable.



Poly (formerly Polycom) phones are high-quality phones, offering a lot of features, as well as listening comfort and ease of handling that are hard to match.

The new **Poly Edge E** range is particularly suited to the latest technological developments in IP telephony and unified communications. The Poly range also offers **HD Voice** technology, which is fully compatible with our solution and offers **superior audio quality**.

1.2.3.1 Poly Edge E Series



The **Poly Edge E100** supports two lines (4 via pagination).

It has a 2.8" color screen and 4 context-sensitive soft keys. It offers two shortcut keys to monitor up to 7 contacts thanks to a scrolling system.

It has an internal 10/100/1000 Mbps switch, 1 USB-C port and is HD Voice compatible.

The support for NFC standards.



The **Poly Edge E220** supports four lines (16 via pagination).

It has a 2.8" color screen and 4 context-sensitive soft keys. It offers four shortcut keys to monitor up to 15 contacts thanks to a scrolling system.

It has an internal 10/100/1000 Mbps switch, 1 USB-C port and is HD Voice compatible.

It also supports the NFC standard and is Bluetooth-enabled.



The **Poly Edge E300 / E320 / E350** support eight lines (32 via pagination).

They have a 3.5" color screen and 4 context-sensitive soft keys. They offer eight shortcut keys to monitor up to 31 contacts thanks to a scrolling system.

They have an internal 10/100/1000 Mbps switch, 1 USB-C port and are HD Voice compatible.

They support the NFC standard.

The E320 and E350 are also Bluetooth-enabled.

The E350 adds Wi-Fi support.



The **Poly Edge E400 / E450** support eight lines (32 via pagination) + six lines (12 via pagination).
 They allow the addition of an expansion module.
 They have a 3.5" main color screen, a 2.4" secondary screen and 4 context-sensitive soft keys. They offer fourteen shortcut keys to monitor up to 43 contacts thanks to a scrolling system.
 They have an internal 10/100/1000 Mbps switch, 1 USB-C port and are HD Voice compatible.
 They support the NFC standard.
 The E450 is Bluetooth enabled and adds Wi-Fi support.



The **Poly Edge E500 / E550** supports twelve lines (48 via paging).
 They allow the addition of two expansion modules.
 They have a 5" color screen and 4 context-sensitive soft keys. They offer twelve shortcut keys to monitor up to 47 contacts thanks to a scrolling system.
 They have an internal 10/100/1000 Mbps switch, 2 USB-C ports and are HD Voice compatible.
 They support the NFC standard.
 The E350 is additionally Bluetooth-enabled and adds Wi-Fi support.



The **Poly Edge E EM** supports twenty-two lines (66 per module via pagination).
 They have a 5" color screen.

1.2.3.2 Poly CCX Series



The **Poly CCX 400** supports twenty-four simultaneous communications.

It has a 5-inch color LCD screen for displaying video.

It has an internal 10/100/1000 Mbps switch, and is HD Voice compatible. It supports PoE.



The **Poly CCX 500** supports twenty-four simultaneous communications.

It has a 5-inch color LCD screen for displaying video.

It is available with and without a handset.

It has an internal 10/100/1000 Mbps switch, and is HD Voice compatible. It supports PoE.




The **Poly CCX 600** supports fifty-four simultaneous communications.

It has a 7-inch color LCD screen for displaying video.

It is available with and without a handset.

A built-in camera can be added separately.

It has an internal 10/100/1000 Mbps switch, and is HD Voice compatible. It supports PoE.



The **Poly CCX 700** supports fifty-four simultaneous communications.

It has a 7-inch color LCD screen for displaying video.

It has a built-in camera.

It is available with and without a handset.

It has an internal 10/100/1000 Mbps switch, and is HD Voice compatible. It supports PoE.

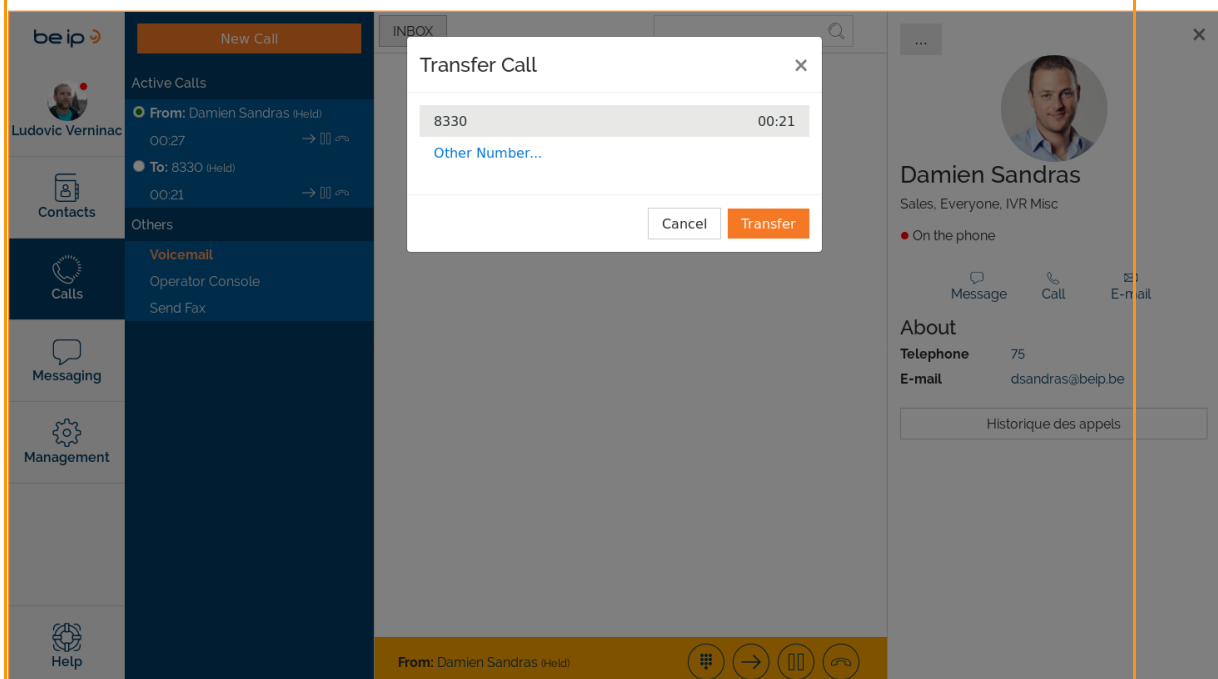
1.2.3.3 Softphones WebRTC

These softphones are an integral part of the Be IP product and its **Gossip** software, at no extra cost.

Standard users

Each user of the system can take and make calls from their standard-compliant web browser.

When the solution is installed in **"Cloud mode"**, it can even take advantage of the **"Gossip Everywhere"** option which allows, without VPN access, to make and receive calls from



anywhere via a network connection.

The **"Gossip Everywhere"** option is subject to a usage cost per desired communication channel.

The softphone allows all the classic operations of a terminal such as *waiting*, *blind* or *consultative transfer* to be carried out using the solution's address book.

We recommend the use of a Poly headset or a Poly Sync speaker for better communication quality.

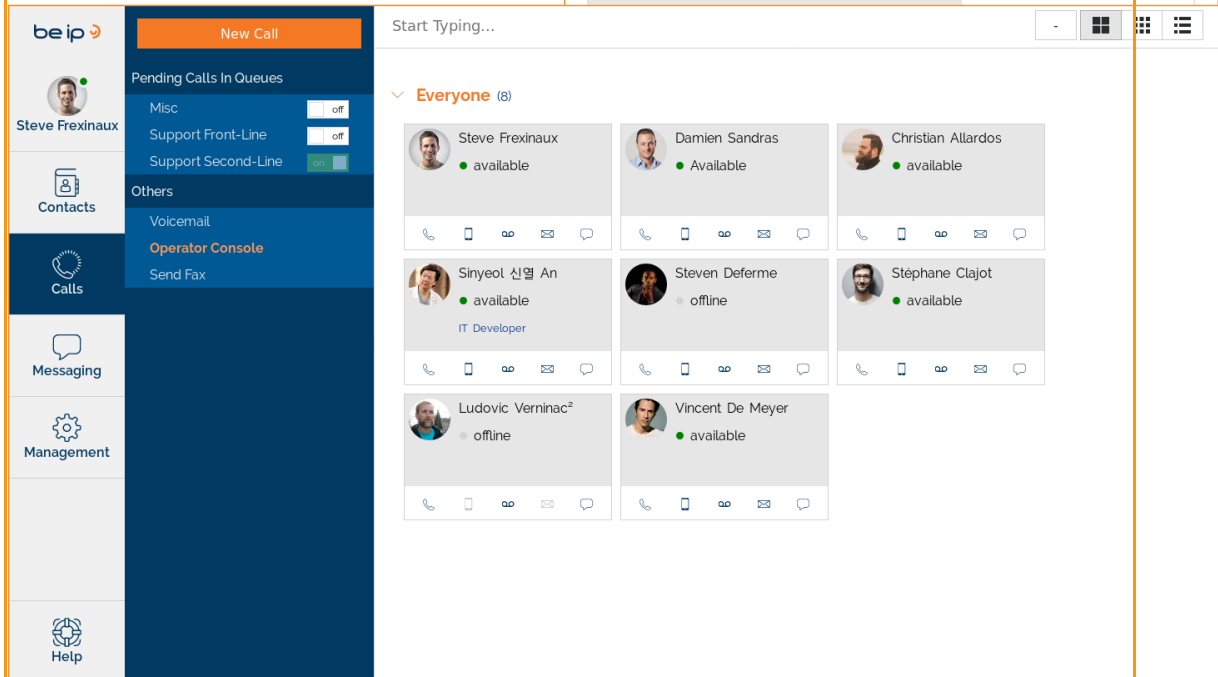
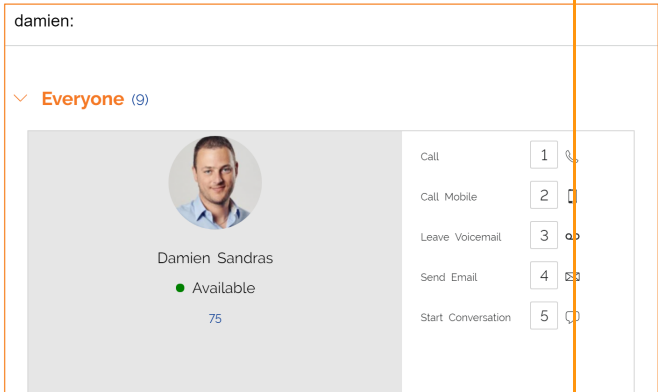
Operators

Each user of the system can also take and make calls from their standard-enabled web browser and use the operator console provided for this purpose.

Unlike the WebRTC softphone described above, this one has a view adapted for operators and allows for quick transfer of calls to the right party, in particular via keypad code sequences.

It also presents a synthetic view adapted to female operators.

When the solution is installed in "Cloud



mode", the operator can even take advantage of the "**Gossip Everywhere**" option which allows, without VPN access, to make and receive calls from anywhere via a network connection.

The "**Gossip Everywhere**" option is subject to a usage cost per desired communication channel.

1.2.3.4 Mobile Softphones

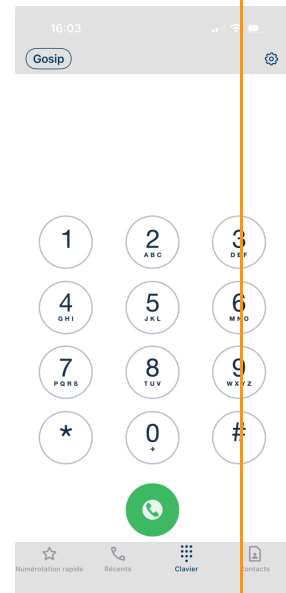
For smartphone users, we have developed the **Gossip Mobile softphone**.

It has the following features:

- *Call forwarding*
- *Pausing*
- *Numeric keypad*
- *Contact List*
- *Voicemail*
- *Voice mail integration*

Note that it requires the presence of an SBC in the case of an "On Premises" installation as well as an additional license per user.

It is download on Google Play and Apple Store.



1.2.3.5 Conference Stations & Speakers

Our offer can be enhanced with conference stations or speakers compatible with our WebRTC solution.

These devices allow you to conduct meetings involving multiple people while providing optimal audio quality. We highly recommend the Poly brand, leader in the field.



The **Trio C60** has the following features:

- *Microphone range: 6.1 meters*
- *Supports Ethernet, Wi-Fi and Bluetooth*
- *Supports the addition of additional external microphones*



The **Sync 20, 40 and 60** are USB and Bluetooth smart speakers with the following features:

- *Microphone range: 2 to 3 meters*
- *Sync 40 and 60 can be chained*

1.2.3.6 Gamme Audility

Our offer also includes phones specially designed for the elderly. They have a simplified format where all unnecessary features have been removed.



The **TOPIC Senior IP** is an excellent product with the essential features for an elderly person:

- *3 shortcut keys with photos*
- *10 memorable keys*

It has an internal switch.

1.2.4. Helmets



We also work with the Poly range to cover all your needs in headphones and speakers, whether they are wireless or with Bluetooth, with 1 earpiece or two, but also regardless of how the headphones are worn: head circumference, neck circumference, etc.

Note that some headset models allow the use of GSM and our WebRTC softphone together.

We recommend the use of a headset when using our softphone.

2. Integration with your business software

Our software suite can be integrated with several software programs available either free of charge or in paid form. **Gossip Advanced** licenses also open up additional features.

Here is a non-exhaustive list. Do not hesitate to discuss your needs with our teams so that we can offer you the most suitable solution.

1. *Native features*

2.1.1. Bringing up files

Profiles with a **Gossip Advanced license** have the possibility to take advantage of the file escalation.

This requires one or more licenses in addition to the **Gossip user license** depending on the number of integrations to be made.

This makes it possible to display web pages or components during incoming calls and therefore allows integration with any type of business application that is sufficiently open.

For example, it is possible to view a customer or patient file and interact with it. This sheet can

present a summary related to the calling number, and interact with the customer's business software according to the desired programming.

The development of web pages or components as well as any "middleware" allowing you to interact with your business application is not part of our offer. Please discuss this with your sales representative.

2.1.2. Dynamic fields in contact cards

A **Gossip Advanced** license also adds the ability to have "tailor-made" fields in the app's contacts.

The values for these fields can be:

- a text field,
- a web link,
- a telephone number,
- an email,
- a list of key/value fields corresponding to the previous types,
- A table with values corresponding to the types defined above.

The whole can be organized into sections.

2.1.3. Event reporting via API

A **Gossip Advanced** license also adds the possibility of escalating events via calls to REST APIs (Webhooks) to your business applications. These events can be interpreted by your business software to be recorded or display a customer file based on the calling number directly within the latter.

In addition, integration with Zapier (<https://zapier.com/>) is also possible, opening up almost unlimited possibilities for integration with more than 6,000 applications.

This requires one or more licenses depending on the number of integrations to be made.

The development of any "middleware" to interact with your business application is not part of our offer. Please discuss this with your sales representative.

2.1.4. Conditional call routing via API

A **Gossip Advanced** license adds the possibility of routing calls according to the result returned by REST APIs (Webhooks) to your business applications to which our system can provide the calling number, sequences of DTMFs entered by the user, etc. The result returned via the APIs can be used to route the call differently.

In addition, integration with Zapier (<https://zapier.com/>) is also possible, opening up almost unlimited possibilities for integration with more than 6,000 applications.

This requires one or more licenses depending on the number of integrations to be made.

The development of any "middleware" to interact with your business application is not part of our offer. Please discuss this with your sales representative.

2.1.5. Click-to-call and other operations

The **Gossip software suite** provides a REST API that allows, among other things, to add click-to-call functionality to your own software by simply calling URLs.

It also allows you to configure almost all the elements of the system without going through our WEB interface. This is in order to better integrate with your business software.

Support for "tel://" URLs allows native support to be added to all compatible applications,

resulting in the ability to click-to-call from anywhere.

This feature is free and is an integral part of our core product.

3. Call Center Features

1. General Description

Gosip Advanced **licenses** allow associated users to go beyond **Gosip's** standard call distribution module and the standard statistics described above. These licenses can be activated through the purchase of the advanced licenses in addition to the **standard Gosip** user license .

A **Gosip Advanced** license is required per agent, supervisor, or integration (see above). However, it requires exclusive use of the WebRTC softphone if you want to have the full range of features.

2. Features

3.2.1. Automatic call distribution

Automatic call distribution allows incoming calls to be organized into queues and distributed according to different call strategies to different call center agents.

Many features can be configured:

- *ad serving,*
- *prioritization of calls from one queue to another,*
- *the priorities of the agents in a queue,*
- *redirection in the event of overflow or too long a waiting time,*
- *integration with REST APIs,*
- *Setting up timeout*
- *pausing agents,*
- *the possibility for callers to submit a request for a callback;*
- ...

3.2.2. Recall campaigns

Each queue can also be associated with reminder campaigns.

These can be fed automatically by the automatic call distribution system or provisioned.

This allows the module to be used in a model for carrying out outbound call campaigns.

gosip									
yyyy-mm-dd to yyyy-mm-dd		Queue - Support Front-Line		Search		Next Call			
Number	Agent	Date	Attempts	State	Outcome	Queue	Source	Action	
7458	vdelpierre	10/5/20, 8:54 AM	1	Complete	Complet	Support Front-Line	-	-	attempts
458	ngodfraind	10/5/20, 10:00 AM	2	Complete	Faux numéro	Support Front-Line	-	-	attempts
52	amizero	10/28/20, 8:56 AM	1	Complete	Faux numéro	Support Front-Line	-	-	attempts
53	amizero	10/28/20, 8:56 AM	1	Complete	Faux numéro	Support Front-Line	-	-	attempts
761'3	vdelpierre	1/29/21, 9:25 AM	1	Complete	Complet	Support Front-Line	-	-	attempts
761'3	amizero	1/29/21, 9:32 AM	3	Complete	Abandon	Support Front-Line	-	-	attempts
023490223	amizero	2/5/21, 2:48 PM	1	Complete	Mauvais numéro	Support Front-Line	-	-	attempts
023490223	amizero	2/9/21, 10:16 AM	1	Expired	Non décroché	Support Front-Line	-	-	attempts
+32472121621	amizero	2/24/21, 8:46 AM	1	Expired	Non décroché	Support Front-Line	-	-	attempts
		2/24/21,				Support			

3.2.3. The Agent Dashboard

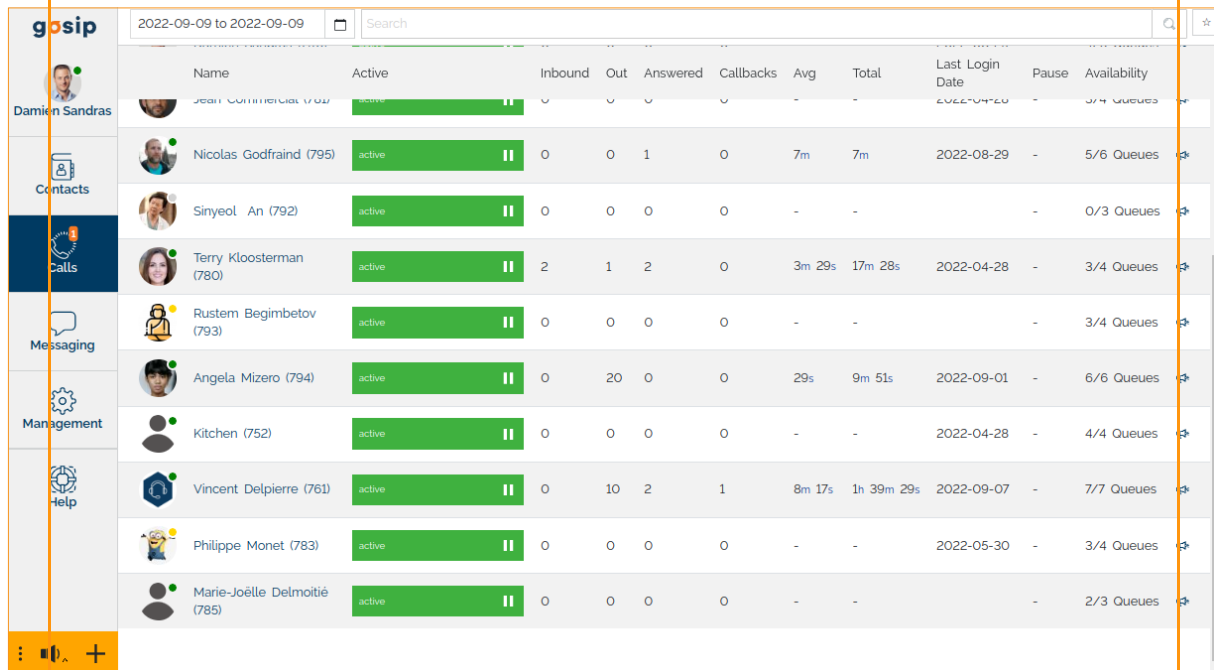
The Agent Dashboard presents, for a given period of time (by default, the current day) and agent by agent, many statistics that allow conclusions to be drawn about the activity and efficiency of a call center from the agent's point of view.

The Dashboard is refreshed in real time to allow for quick action.

For *each of the agents*, all lanes combined or file by queue, he summarizes the following statistics:

- **Direct calls:** the number of incoming and outgoing calls;
- **Calls associated with queues:** the number of incoming calls answered and callbacks made;
- **Talk time:** average and total;
- **Several useful information about the agent's availability :** the time and date of the last connection to a queue, the break time, the number of queues to which the agent is currently connected, the session time, the duration of availability, etc.;

The view also allows the *supervisor* to act on the *pause status* of the agents, but also to *connect or disconnect* the agent from each of the presented queues as well as to indicate the current connection status.



Name	Active	Inbound	Out	Answered	Callbacks	Avg	Total	Last Login Date	Pause	Availability
Jean Commercial (792)	active	0	0	0	0	-	-	2022-09-09	-	3/4 Queues
Nicolas Godfraind (795)	active	0	0	1	0	7m	7m	2022-08-29	-	5/6 Queues
Sinyeol An (792)	active	0	0	0	0	-	-	-	-	0/3 Queues
Terry Kloosterman (780)	active	2	1	2	0	3m 29s	17m 28s	2022-04-28	-	3/4 Queues
Rustem Begimbetov (793)	active	0	0	0	0	-	-	-	-	3/4 Queues
Angela Mizero (794)	active	0	20	0	0	29s	9m 51s	2022-09-01	-	6/6 Queues
Kitchen (752)	active	0	0	0	0	-	-	2022-04-28	-	4/4 Queues
Vincent Delpierre (761)	active	0	10	2	1	8m 17s	1h 39m 29s	2022-09-07	-	7/7 Queues
Philippe Monet (783)	active	0	0	0	0	-	-	2022-05-30	-	3/4 Queues
Marie-Joëlle Delmoitié (785)	active	0	0	0	0	-	-	-	-	2/3 Queues

3.2.4. The Queue Dashboard

The Queue Dashboard presents, for a given period of time (by default, the current day) and queue by queue, numerous statistics that allow conclusions to be drawn about the activity and efficiency of a call center from the point of view of queues, i.e. from the point of view of automatic call distribution.

For each of the queues, the following statistics are present:

- **answered and unanswered incoming calls: their number and distribution by type of call closure, average waiting time, number of pending and ongoing calls, percentage of calls respecting the SLA;**
- **Talk time: average and total;**
- several useful information about the **capacity** of the queue: the number of available agents and the number of connected agents, *thenumber of calls on hold*, the number of *calls in progress*, the *waiting time* of the oldest call;
- the *waiting time* for the oldest call
 - **Callback or outbound call campaign:** The number of pending callbacks and completed callbacks.

gosip	Concise	Detailed	Charts	Legend	2022-09-09 to 2022-09-09	Search	Q	*						
Queue	Answered				Unanswered				Callbacks		Call duration		Current calls	
	Total	SLA	Avg waiti.	Max waiti.	Abandon.	Callbacks	Avg waiti.	Max waiti.	Avg	Max	Waiting	Longest	Establi.	
Sales Fr.	1	1 / 1 100 %	8s	8s	0 0 %	0	-	-	0	4m 45s	4m 45s	0	-	0
Misc Fro.	3	3 / 3 100 %	29s	48s	0 0 %	0	-	-	0	4m 29s	5m 25s	0	-	0
Support	1	1 / 3 33.33 %	13s	13s	0 0 %	1	32s	1m 1s	1	1m 41s	1m 41s	0	-	0
Support	1	1 / 1 100 %	14s	14s	0 0 %	0	-	-	0	6m 59s	6m 59s	0	-	0

3. Available statistics

Statistics are available through the various dashboards, reports in CSV format but also our APIs. The complete list of implemented statistics and graphs is available upon request.