Kalliope Wiki

Release 0.1

NetResults

Nov 02, 2023

CONTENTS

The technical documentation is dedicated to Kalliope users and partners.

Warning: Starting from 05/06/2020 the IP address corresponding to the license (**license.kalliopepbx.it**), updates (**updates.kalliopepbx.it**) and KalliopeTribe registration (**ems.kalliopepbx.it**) servers has changed from 77.72.27.4 to **46.44.212.245**. All communication between the KalliopePBX and these servers is done in **HTTPS**, using the standard port **443**. If ACL filters are outgoing from Kalliope towards the old IP in the installation network of the central unit, it is necessary to update them with the new value.

We remind you that the VMs must periodically contact the license server to verify its validity. A prolonged period (30 days) of unreachability of the same can cause the block of the telephone functions.

We also recommend that the SMTP configuration of the central units be carried out in such a way as to receive, together with the others, the daily alert in case the server is unreachable.

CHAPTER

ONE

NEWS

- 10/08/2023 : released the new Release of KalliopePBX rev. 4.15.7, which introduces some security aspects. (*changelog*)
- 12/06/2023 : released the new Release of KalliopePBX rev. 4.15.6, which introduces support for the new Kalliope Phone mobile app. (*changelog*)
- 02/05/2023 : released the new Release of KalliopePBX rev. 4.15.5, which introduces the new WebCTI client. (*changelog*)
- 27/03/2023 : released the new Release of KalliopePBX rev. 4.15.4, which contains contains a number of bugfixes. (*changelog*)
- 03/01/2023 : released the new Release of KalliopePBX rev. 4.15.3, which contains the addition of the "protected" attribute to accounts. (*changelog*)

1.1 Changelog

The KalliopePBX firmware follows a 4.X.Y numbering system where:

- X is the Major Version. - releases with an even Major Version number (4.0.Y beginning from 4.0.8, 4.2.Y, etc.) are Maintenance Releases (MR), i.e. stable versions that mostly contain bugfixes; these releases add fewer features but guarantee greater stability; - releases with an odd Major Version number (4.1.Y, 4.3.Y, etc.) are Technology Releases (TR), i.e. versions which introduce new features;
- Y is the Minor Version, the sequential identifier of releases within the same Major Version.

1.1.1 Bootloader

Warning: The minimum RAM requirement for VMs is 2GB. If the RAM is below this requirement, the PBX will not start. The recommended RAM requirement for VMs with firmware version 4.9.8 or later is 4GB.

The bootloader is the base operating system of the PBX. It does not contain telephone features but is used to install and manage the firmware. The bootloader is updated similarly to the firmware, but it is updated in place and cannot be returned to return to a previous version. After a new bootloader is installed, the PBX will need to be rebooted (on the bootloader itself) to complete the procedure. After this, it will be possible to reboot the PBX normally on one of the available firmware versions.

Bootloader 1.1.0

This update adds to the bootloader software components usually distributed within the firmware, reducing the size of future updates. Bootloader version 1.1.0 is required for installing firmware updates starting from version 4.10 and 4.11

Note: There is a procedure to resolve the ERR_UNSAFE_PORT issue with Chrome and Firefox browsers. For more information on the procedure to follow (*click here*)

Changes

- System:
  - K-2630: Changed the firmware update download path to fix an issue where updates sometimes failed if the RAM assigned to the Kalliope VM was limited.

Bootloader 1.0.7

This update changes the file system of the storage partition to add a journaling feature. This change helps reduce the file corruption issues present in this file system (configurations, logs, databases) that may occur in case of a power failure when the system is writing the data.

Note: This bootloader is a prerequisite for installing firmware equal to or later than version 4.8.0.

Changes

- System:
  - 6694: Added journaling to file system/storage

Bugfixes

- General:
  - 6699: Fixed a bug that sometimes caused the system time not to sync via NTP if the system time was after the current time

1.1.2 Firmware series 4.15.x (TR)

Firmware updates from the 4.15.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.14, is the stable release recommended for generic use that does not require features introduced in the TR.

Warning: To install version 4.15, bootloader 1.1.0 is required.

Firmware 4.15.7 (10/08/2023)

The main new features of this firmware concern security aspects such as:

- automatic logout for inactivity
- definition of custom password strength criteria
- warnings about the use of default passwords

New features

- General
  - K-16318 Added missed call event also in case of caller abandonment
  - K-16096 Added call answered event also for outgoing calls
  - K-15814 Added start and stop events for on-demand call recordings and readiness to download recordings
  - K-15756 Added ability to use placeholders identifying network interfaces as listening address in SNMP settings
  - K-15750 Added a password change invitation warning to login for administrator users using the default password
  - K-15747 Limited the validity of authenticated GUI sessions to 12 hours and added automatic logout after 2 hours of inactivity
  - K-15744 Added new panel for defining password robustness criteria
  - K-15607 Added configuration to show or not show the caller the real number of a called speed dial
  - K-14939 Added ability to assign provisioning profiles to applications developed with the KPE (Kalliope Phone Engine) library
- WebCTI
  - K-15894 Incoming call widget now also shows destination queue/group identifier
  - K-15891 Added the ability to transfer calls even to contacts not in the address book
  - K-13890 Added, in the audio settings panel, the audio device test

Reworking

- General
  - K-15720 Redesigned in tabular form the call recording settings panel
- WebCTI
  - K-15646 Simplified adding participants to the current call
  - K-15549 WebSocket connection between WebCTI and Kalliope for SIP signaling now uses standard port 443

Bugfixes

- General
  - K-16483 Fixed bug that prevented, in firmware installation from boot loader, correct initialization of data for application profiles
  - K-16209 Fixed bug that prevented FAX licenses from being assigned to FAX instances when restoring a backup
  - K-16073 Fixed bug that prevented, in firmware installation from boot loader, correct operation of log retention period configuration panel
  - K-14780 Fixed bug that in some conditions prevented proper activation of Multi-tenant license
  - K-14693 Fixed bug that in some conditions led to having extension address book with empty email and cell phone contacts
- WebCTI
  - General improvements and graphical fixes
  - K-16490 Fixed bug that prevented initiating calls to external numbers from the phonebook panel and CDR panel top bar
  - K-16434 Fixed bug that caused WebCTI to hang when there were contacts in phonebook with null fields
  - K-15349 Fixed bug that caused incorrect display of contact BLF status by switching between tabs in the CDR panel
- API
  - K-15528 Fixed bug that prevented editing an audio file associated with an IVR action

Firmware 4.15.6 (12/06/2023)

The main new feature of this firmware is the introduction of support for the new Kalliope Phone mobile app.

New features

- General
  - Kalliope-14505 Added panel, accessible by PBX administrator, to configure retention periods for logs
  - Kalliope-15579 Added new built-in user with permissions necessary for proper integration with Kalliope Nexus
  - Kalliope-15489 Added new panels for managing profiles of applications developed with the KPE (Kalliope Phone Engine) library
  - Kalliope-15924 Added new OID for exposing via SNMP the number of accounts in "Background" state
- API
  - Kalliope-15326 Added an API for filtering active calls by linkedId
  - Kalliope-15467 Added new API to transfer an incoming call, not yet answered, to another internal or external selection
  - Kalliope-14468 Added an API for uploading volatile audio files

Firmware 4.15.5 (02/05/2023)

The main new feature of this firmware is the introduction of the new WebCTI client.

New features

- K-15125 Released the new Web-based CTI client WebCTI

Bugfixes

- K-14803 Fixed bug that prevented calls originated by the advanced callback API from displaying in the list of active calls
- K-14958 Fixed bug that caused the line commitment prefix to be missing in the CDR for callback calls

Firmware 4.15.4 (27/03/2023)

This release contains a number of bugfixes.

New features

- General
  - K-14637 Added Snom D862 and D865 devices to the list of built-in devices
- API
  - K-14107 Added an enhanced callback API
  - K-12645 Added new set of API V2 for address book management

Reworking

- K-14055 Improved on-call event generation mechanism to reduce Call Setup Time
- K-12088 Disabled insecure HTTP access to the interface and API. If enabled in provisioning configuration it remains active ONLY for terminal provisioning
- K-10246 Fixed potential vulnerability to clickjacking.

Bugfixes

- General
  - K-14951 Fixed bug that, in some scenarios, on incoming call rejection caused the call to reoccur instead of performing the set overflow action
  - K-14682 Fixed bug that, in some scenarios, reported TIMEOUT as the cause of exit in CDR Call-Center operator events even though the operator had answered the call
  - K-14213 Fixed filtering on the “protected” column in the account list
  - K-14149 Fixed bug that caused massive account import to fail when at least one account had empty label within the file to be imported

- K-13915 Fixed bug that, in some scenarios, prevented Mail2FAX from working properly
- K-13838 Fixed bug that prevented the export of the audit log
- K-13832 Fixed bug that prevented disabling ACL rules of switches
- K-13816 Fixed bug that, in some scenarios, led to incorrect caller identity if two click2calls were executed at the same time

- REST API
  - K-14348 Fixed bug that caused a 500 error on operation API invoked as a non-administrator user
  - K-13938 Fixed bug where user roles created via API did not contain all available actions but only those included in the body of the request
  - K-13652 Fixed bug that caused a 500 error on the invocation of some APIs when the lock was acquired by another user
- Backup
  - K-13066 Fixed bug that, in some scenarios, prevented correct restoration of user email addresses when restoring a single tenant machine backup
  - K-12971 Fixed bug that, in some scenarios, prevented correct restoration of dynamic operators and pause states in queues when restoring a backup

Firmware 4.15.3 (03/01/2023)

The main new feature of this firmware is the addition of the "protected" attribute to accounts. Through the new permission "Manage Protected Accounts" it is possible to define the level of access of various roles to accounts with this attribute.

New features

- K-12840 Added the protected attribute to accounts.

Reworking

- K-11536 Added restoring user license assignments when restoring a backup to Single-tenant machine
- K-13510 Added the ability to use placeholders that identify network interfaces as bind addresses in SIP settings

Bugfixes

- General
  - K-13423 Fixed bug that prevented message-on-demand functionality in queues from working properly
  - K-13452 Fixed bug that caused, during playback of custom messages in queues, playback to operators of hold music that was not the default music set for the tenant
  - K-13587 Fixed bug that prevented decryption of calls recorded with firmware <4.13.7
  - K-13626 Fixed bug that prevented access to FAX instance template page
  - K-13686 Fixed bug that, in some scenarios, allowed deletion of entities without showing confirmation pop-up

- CDR
  - K-13206 Fixed bug that, in some scenarios, prevented populating the destination name in the CDR details
  - K-13210 Fixed bug that caused NOANSWER status instead of CANCELED to be entered in the CDR for calls abandoned by the caller during audio message playback in IVRs
  - K-13214 Fixed bug that caused the empty IVR exit reason to be entered into the CDR in the case of direct dial dialing
  - K-13690 Fixed bug that caused NOANSWER status to be entered into the CDR for calls originating from outside and directed to closed queues

Firmware 4.15.2 (21/10/2022)

The main change in this firmware is the upgrade of the Asterisk phone engine to version 18 LTS

New features

- General
  - K-2597 Upgraded Asterisk to version 18 LTS
  - K-12580 Added new role to enable downloading of provisioning files
- API
  - K-12332 Added mobile number among parameters allowed in extension creation and modification
  - K-12485 Added an API for viewing the audit log

Reworking

- K-10240 Increased security of communications between KCTI and Kalliope clients by disabling TLS versions less than 1.2

Bugfixes

- K-6520 Fixed bug that, in some Multi-tenant scenarios, prevented creation of all accounts assigned to the tenant
- K-9684 Fixed bug that, in some scenarios, caused massive import of extensions to fail
- K-12361 Fixed bug that left AMI access enabled even when disabled from settings
- K-12495 Fixed bug that caused notifications to be sent with TLSv1.0 even when configured TLSv1.1/TLSv1.2
- K-12589 Fixed bug that caused an exception in the CDR API if the parameters passed for year, month, and day were incorrect
- K-12748 Fixed bug that prevented changing the account label via API.

Firmware 4.15.0 (29/08/2022)

The main new feature of this firmware is the ability to use Microsoft 365 email boxes as Mail2FAX boxes.

New features

- General
  - K-10036 Added flavuor management for applications developed with the KPE (Kalliope Phone Engine) library
- FAX
  - K-8271 Added ability to use Microsoft 365 email boxes as Mail2FAX boxes

Bug fixes

- CDR
  - K-11015 Fixed bug that marked the outcome of some calls intended for time controls and IVR as failed even though correctly answered
- Call-Center Module
  - K-11282 Fixed bug that caused discrepancies between the talk time displayed in CDR and CDR Call-Center

1.1.3 Firmware series 4.14.x (MR)

Firmware updates from the 4.14.x series are Maintenance Releases, which include all features released in versions 4.13.x.

> **Warning:** To install version 4.14, bootloader 1.1.0 is required.

Firmware 4.14.1 (18/04/2023)

This release fixes performance issues.

Reworking

- K-14375 Decreased workload on CPU in call scenarios that generate many events

Bugfixes

- K-13026 Fixed bug that caused discrepancies between talk time displayed in CDR and CDR Call-Center

Firmware 4.14.0 (27/06/2022)

New Features

- K-9818 Added new events related to call delivery to extensions.

Reworking

- K-10249 Increased security of HTTPS communications with GUI by disabling TLS versions less than 1.2 and less robust cipher suites

Bugfixes

- K-8861 Fixed bug that, in Multi-tenant scenarios, prevented the operation of shared custom selections between tenants belonging to the same tenant group
- K-10549 Fixed bug on notifying mobile apps of click2calls
- K-10752 Fixed bug that, in Multi-tenant scenarios, showed "NO ANSWER" exit code for deleted calls in PBX CDR
- K-10766 Fixed bug on CDR that showed incorrect exit code for calls that had a blank outcome on last call detail
- K-10879 Fixed bug that, in some scenarios, prevented operators from switching pause status via the supervisor panel
- K-10983 Fixed bug that caused the parameters of some events not to be populated
- K-11401 Fixed bug that prevented resetting the state of dynamic queue operators when restoring a backup

1.1.4 Firmware series 4.13.x (TR)

Firmware updates from 4.13.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.12, is the stable release recommended for generic use that does not require features introduced in the TR.

> **Warning:** To install version 4.13, bootloader 1.1.0 is required.

Firmware 4.13.8 (06/05/2022)

This release contains a number of bugfixes.

Bugfixes

- General
  - K-10894 Fixed bug that prevented notifications from working for pbxadmin users of multitenant machines without Enterprise license
  - K-10806 Fixed bug on KalliopeLAM and KalliopeHotel module for machines without Enterprise license

Firmware 4.13.7 (26/04/2022)

The main new feature in this firmware is the ability to include the KCTI Mobile as a queue member while also taking advantage of the wakeup mechanism when the app is in the background.

Feature

- K-9270 Added ability to add KCTI Mobile accounts as queue members
- K-10116 Added Alcatel Lucent Enterprise phones among supported built-in devices
- K-10122 Added support for optional Kalliope Enterprise licenses

Reworking

- K-9911 Added PATCH method for editing atrtributes to the REST API /rest/tenant
- K-10140 Updated template for massive account import.
- K-10170 Modified dialplan to avoid forwarding extension calls to unreachable accounts
- K-10293 Modified dialplan to allow fork2mobile even to extensions without associated accounts and with ring simulator disabled

Bugfixes

- K-8097 Fixed bug on internal im edit error with only hotdesk account associated
- K-10545 Solved problem on deleting secondary accounts Cisco phones
- K-10385 Fixed bug that prevented connection via WEB RTC from browsers that require DTLS1.2 support

Firmware 4.13.6 (08/03/2022)

The main new feature introduced in the firmware concerns new APIs for third-party system integration. In addition, a new automatic alert e-mail was introduced on reaching predefined thresholds of disk occupancy.

New features

- API REST
  - K-8658 Added REST APIs for queue configuration
  - K-8664 Added REST APIs for configuring call groups
  - K-8773 Extended REST APIs for configuring service selections in the numbering plan
  - K-8963 Added REST APIs for IVR configuration
  - K-9234 Added REST APIs for configuring input/output manipulation rules on assigned lines
  - K-9350 Added REST APIs for blacklist / whitelist configuration
  - K-9443 Added REST API for blacklist association to assigned lines
  - K-9023 Added REST API to capture existing voicemail list
  - K-8770 Added REST API for configuring custom selections in numbering plan
- TELEPHONE SERVICES
  - K-9322 Added support for INVITE from Mitel (ex Aastra) phones.
- GENERAL
  - K-8938 Added email alert on exceeding disk occupancy thresholds
- FAX MODULE
  - K-9474 Added the ability to send faxes to multiple destinations

Reworking

- Hotel Form
  - K-9085 Changed charge documentation to thousandths of euro pricing.
- Provisioning
  - K-8949 Modified alerts on errors in internal massive import from .xls file

Bugfixes

- K-9647 Fixed bug on remote driving of Yealink phones with new firmware versions
- K-9309 Fixed bug that did not allow replacing via API an audio file in use.
- K-8945 Fixed bug that caused hot desking accounts to disassociate when editing extensions via API
- K-9191 Fixed bug that could cause an exception on API /rest/accounts
- K-9938 Fixed bug on uploading audio files via API (fixed error in case of files with incorrect format)
- K-9931 Fixed bug that caused an error “405 Method not allowed” on API GET /rest/extension/{exten}

- K-8375 Fixed bug where it could happen that faxes remained in the Dialing state if the call was not completed

Firmware 4.13.3 (14/12/2021)

The main innovation introduced in the firmware concerns the realization of a service monitoring panel where admin and power users can visualize and modify the status of detour (unconditional, on busy, for no answer and on not available), the Do Not Disturb and the busy level of each extension. In addition, a mechanism has been introduced to configure the ringing policies (hunting) of the devices connected to an extension. This feature is configurable only via API, but it will be integrated into the service monitoring panel in the subsequent releases.

New features

- Phone services
  - K-8178 New service monitoring panel (Diversions, DND, Busy Level) for admin and power users has been implemented.
  - K-8546 Added the possibility for admin and power users to modify the operating configurations of the services (from the service monitoring panel)
  - K-8131 Added the option to modify the ringing policy of accounts connected to an extension (only via API)
  - K-8137 Added option to define a label for accounts
  - K-7840 Added wake-up monitoring events for KCTI Mobile
  - K-8352 Added a new placeholder ORIGINAL_CLID among those available for SIP Header customization towards an external line
- Provisioning
  - K-8281 Added new CA Audiocodes to validate phone certificates for HTTPS provisioning service with mutual authentication
- GUI
  - K-8585 New Kalliope logo applied

Reworking

- General
  - K-8358 Accelerated backup import process
- Phone Services
  - K-7972 Updated Firebase protocol for sending notifications to KCTI Mobile
- LDAP
  - K-8193 Modified time limit for LDAP requests to avoid service slowdowns
  - K-7562 Modified LDAP phonebook generation mechanism to reduce configuration application time

Bugfixes

- K-8687 Fixed bug that caused incorrect display of the calling number for calls originated from an extension and delivered through the Fork2Mobile service
- K-8522 Fixed bug that caused all numbers to be displayed (not just the tenant's) when configuring service accessibility reports
- K-8341 Fixed bug that caused unanswered calls recordings to remain in the status "in processing" (instead of "not recorded")
- K-3835 Fixed bug that could cause faxes not to be sent using Mail2Fax service.
- K-5363 Fixed bug that caused calls for which the overflow action was performed not to be displayed for all unregistered tenant accounts
- K-8442 Fixed bug where only calls from the current month were included in reports sent via scheduled tasks
- K-8295 Fixed bug that caused the non-application of filtering in the export of the Call Log
- K-7619 Fixed bug causing incorrect update of provisioning files for extensions with accounts used on different terminals
- K-7982 Fixed bug causing the incorrect display of the outcome of outgoing calls with manipulation of the called number in the Call Log
- K-7377 Fixed bug that could cause incorrect display of KLAM meetings on the calendar

Firmware 4.13.2 (29/09/2021)

The main innovation of this firmware is the introduction of the rebranding module of KalliopeLAM and the addition of new API for the configuration of the central unit

New features

- Kalliope LAM
  - K-7605 KalliopeLAM rebranding module enabled
- REST API
  - K-7028 Implemented REST API for outbound routing rules and classes management
  - K-7526 Implemented REST API for SIP account assignment to extension
  - K-6976 Implemented REST API for reading inbound routing rules (DID)

Reworking

- General
  - K-4382 Added possibility to insert a "valid" hostname as sender of e-mail messages
  - K-7677 Added check on kloggerd execution before stop for storage over maximum quota
  - K-338 Modified remote filesystem mount options to prevent crashing when filesystem is not reachable
- GUI
  - K-7439 Modified display of previous months in logged calls panel

- REST API
  - K-7292 Added ability to delete all time ranges in time control configuration
- KalliopeLAM
  - K-7774 Removed the visualization for months/years on the calendar widget

Bugfixes

- General
  - K-6379 Fixed bug on session db cleanup that could cause the /tmp directory to fill up
  - K-7673 Fixed bug that prevented changing the "busy level" in the default Internal Template definition
  - K-1122 Fixed bug that prevented removing codecs from account templates
  - K-7298 Fixed bug on active calls count in the dashboard widget
  - K-7428 Fixed bug in which Klogger service was not available on panels installed with fw 4.13.0
  - K-7075 Fixed bug of nonpersistence of the login status of the APPs after a restart of the PBX or the kctis service
  - K-7471 Fixed translations in the User Roles panel
  - K-7132 Fixed a bug that caused the generation of an error in the console during the startup of the machine, in case on the PBX there were not configured remote address books to import
  - K-5239 Fixed an issue with file ownership where an exception of type 500 was thrown in case of GUI errors
  - K-7467 Fixed bug on intra-site call counting for Call Admission Control
  - K-7505 Fixed bug on timeout setting for calls received by a group member and transferred to another extension
  - K-7663 Fixed bug on adding new roles in an update for previously defined power user roles

Firmware 4.13.0 (28/06/2021)

The main new feature of this firmware is the introduction of the Kalliope Logger via WEB interface. Thanks to this new feature, you can start the call logging service from the dashboard and then analyze the path of the specific call directly from the Call Log

New features

- General
  - K-6107 Implemented the Kalliope Logger Web service that allows visualizing, starting from the Call Log, the path of a call inside the central unit for analysis and troubleshooting purposes
  - K-4767 Enabled the possibility to activate the VoIP service only on a specific IP address
  - K-6210 Added a new LINKEDID placeholder among those available for SIP Header customization towards an external line
- KCTI Mobile
  - K-6710 Enabled support for the transfer with offer from KCTI Mobile (Android and IOS) - requires KCTI IOS 4.9.0 and KCTI Android 4.8.0

- REST API
  - K-6440 Implemented the REST API for managing time controls and audio files

Reworking

- General
  - K-2455 Modified active call count widget to correctly include calls forwarded to external lines as well
  - K-6199 Added check and confirmation request on tenant deletion

Bugfixes

- General
  - K-6375 Fixed bug that prevented associating two accounts to the same extension in the presence of hot-desking accounts
  - K-6162 Fixed bug that prevented the display of the name of the destination for calls delivered to IVR
  - K-2846 Fixed bug that caused the display of system error messages to all users (and not only to pbxadmin)
  - K-6332 Fixed bug that prevented disabling certificate validation in HTTPS calls of Dynamic Routing service
  - K-6369 Fixed bug that caused the failure of HTTPS notifications with certificate disabling and/or TLS version specification
  - K-2209 Fixed bug causing incorrect counting of active calls exported through SNMP
  - K-4418 Fixed bug that caused calls started in one month and ended in the next month not to be displayed in the Call Log
- REST API
  - K-5997 Fixed bug on KLAM REST API parameters validation.

1.1.5 Firmware series 4.12.x (MR)

Firmware updates from the 4.12.x series are Maintenance Releases, which include all features released in versions 4.11.x.

> **Warning:** To install version 4.12, bootlaoder 1.1.0 is required.

Firmware 4.12.1 (06/08/2021)

This release contains a number of bugfixes, some of which were already released in the previously released TR 4.13.0.

Reworking

- Multitenant Module
  - K-7194: Added check and confirmation request on tenant deletion [Backport K-6199].

Bugfixes

- General
  - K-7283: Fixed a bug that caused a console error to be generated during machine startup if no remote address book was configured on the PBX to be imported
  - K-7273: Fixed a file ownership issue that caused a 500 exception to be thrown in case of GUI errors
  - K-7267: Fixed a bug that caused the "KCTI mobile app" flag not to be displayed in the SIP account edit panel
  - K-7177: Fixed bug that prevented associating two accounts to the same extension when there were hot desking accounts [Backport K-6375].
  - K-7221: Fixed bug that prevented disabling certificate validation in HTTPS calls of Dynamic Routing service [Backport K-6332]
  - K-6660: Fixed a bug that caused the automatic mechanism to update certificates used by Apple Push Services not to work
  - K-7217: Fixed bug that caused notifications via HTTPS to fail with certificate disablement and/or TLS version specification [Backport K-6369].
  - K-6656: Fixed bug that caused system error messages to be displayed to all users (not just pbxadmins) [Backport K-2846].
  - K-7198: Fixed bug that caused calls that began in one month and ended in the next not to be displayed in the Call Log [Backport K-4418].
  - K-7181: Fixed bug that prevented destination name from displaying for calls delivered to IVR [Backport K-7181].
  - K-7169: Fixed bug of APP login status not persisting the following reboot of PBX or kctis service

1.1.6 Firmware series 4.11.x (TR)

Firmware updates from 4.11.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.10, is the stable release recommended for generic use that does not require features introduced in the TR.

> **Warning:** To install version 4.10.0 or later, bootlaoder version 1.1.0 is required.

Firmware 4.11.12 (19/03/2021)

This release fixes two bugs found after the previous release was released.

Bugfixes

- GUI
  - K-5821: Fixed an error where the validation of passwords according to the new policy introduced in 4.11.11 was applied, in the user panel of password change, also to the old password, which caused the need for the admin to change passwords only from the Users and Roles panel (where the old one is not required as it is a reset action).
- HA
  - K-5806: Fixed a bug introduced in firmware 4.11.10 that caused the HA service to crash incompletely when pressing "Disable HA" from the web interface, which could cause errors on the next reboot attempt. The workaround to get back to a clean state was to reboot the node after disabling HA, which is now no longer necessary.

Firmware 4.11.11 (14/03/2021)

Among the main new features of this firmware there is the possibility to create Instant Meetings on the Kalliope LAM platform (a dedicated license is required to use the service), and the modification of the user password validation policies, which now require a minimum length of 12 characters, following the latest security recommendations. New events related to the HA service have been added. Access to the CDR REST API has also been enabled on the secondary node of an HA cluester to be queried without burdening the active node.

New features

- General
  - K-5360: Added real-time call start/end events that can be used for email or Web Service notifications.
  - K-4872: Added security constraints on passwords used (minimum 12 / max 128 characters)
  - K-5092: Added the possibility to configure the use of TLSv1.1 or - TLSv1.2 in Web Service notification actions
  - K-5106: Added the ability to configure the use of TLSv1.1 or TLSv1.2 in Dynamic Routing Service Web Service calls
- HA
  - K-2539: Added the ability to configure some HA service timeouts to minimize the possibility of false positives in the detection of the fault of the other node in the cluster, specifically helpful in case of lack of physical point-to-point connection between the two nodes
  - K-1116: Added events on starting and stopping the HA service, and on starting and ending resource acquisition or release by a node, which can be used for email or Web Service notifications
- Provisioning
  - K-5131: Added new Gigaset CA for phone certificate validation for HTTPS provisioning service with mutual authentication
- KalliopeLAM

- K-5272: Added the possibility to create an instant meeting in the room administration widget

Reworking

- KalliopeLAM
  - K-5269: Added progress indicator when loading available rooms
  - K-5244: Added focus on the title when opening the meeting creation widget
- REST API
  - K-5283: Added ability to run CDR REST API on passive node of an HA cluster

Bugfixes

- General
  - K-5565: Fixed bug that in some conditions caused an error message when deleting gateways/trunks and terminations
- FAX Module
  - K-5095: Matching of sender and recipient email addresses is now case insensitive, and is correctly handled even in case of sub-addressing (e.g. mario.rossi+estensione@miodominio.org)
- KalliopeCTI Mobile
  - K-5394: Fixed bug that caused incoming calls to be ignored when the app was in CTI mode.
- KalliopeLAM
  - K-5406: Fixed bug that caused conferences whose duration was less than a minimum value not to be displayed on the calendar

Firmware 4.11.10 (03/02/2021)

This firmware version contains optimizations and minor graphic fixes for the KalliopeLAM room management module (meeting title alignment, optimization of the current time display on the booked meetings calendar, handing of meetings spanning two or more days, preventative checking of invitees to ensure that it does not exceed the maximum number of participants allowed in the room).

New features

- KalliopeLAM module
  - K-4710: Added the option to include participants from the phonebook
- Provisioning
  - K-5005: Added new CA Yealink for phone certificate validation for the HTTPS provisioning service with mutual authentication
- Call recording
  - K-4798: Added the option to activate unconditional call recording for "dynamic routing" entities to catch DTMF exchanges between the caller and the PBX (N.B.: this requires the DTMF configuration for the inbound line used is "in audio" and not "RFC2833" or "SIP Info")

- Modulo Hotel
  - K-4886: Added new possible values for the room status (maid service)

Bugfixes

- GUI
  - K-1472: Fixed a bug that made it impossible to download or listen to call recordings saved on the network storage
  - K-5089: Extended the extension Company and Department fields from 40 to 255 characters
- KCTI
  - K-5113: Restored the remote control from KalliopeCTI Pro functionality in cases where the phone web interface was reachable from a non-standard port
- Phonebook
  - K-4882: Fixed a bug that caused exporting via LDAP of contacts without the "Company" or "Department" attributes to fail
- High availability
  - K-4262: Fixed a bug that caused playback of customized audio files on the secondary node to fail for tenants created after the secondary node was linked to the primary node
  - K-3297: Fixed a bug that sometimes caused a logout from the GUI during HA activation, causing the cluster status not to update
  - K-2258: Fixed a bug that caused changes to the configuration of the SMTP service made after the secondary node was linked to the primary node not to be propagated to the secondary node
  - K-3293: Fixed a bug that made it impossible to unlink the secondary node from the GUI of the node itself
  - K-1463: Fixed a bug similar to K-3293 that made it impossible to disable HA on the secondary node while it was waiting to receive the configuration from a coordinator node
- Call-Center Module
  - K-4861: Fixed a bug where service accessibility did not return an exception in cases where there was a numbering to which no time check was assigned

Firmware 4.11.7 (06/12/2020)

This version improves user experience for the KalliopeLAM service by introducing a new creation/editing window that allows users to, among other things, view and copy the access URL (both for moderators and for guests) in the conference window.

New features

- KalliopeLAM module
  - K-4103: Changed the creation/editing window for better usability
  - K-4368: Added the option to view and copy the access URL (both for moderators and for guests) in the conference window
  - K-4139: Changed the conference calendar layout
- Hotel Module
  - K-4195: AddedNow displays the guest name for calls received by rooms with active check-in
- Kalliope-Lift Module
  - K-4349: Changed the Esseti protocol timing
- Third-party SIP phone module
  - K-4443, K-4446, K-4459: Graphical revision of secondary line handling for Cisco Unified IP Phones 78xx/79xx

Bugfixes

- GUI
  - K-4656:Fixed a bug that sometimes caused not to display certain panels (CDR, fax list, events). N.B.: The problem occurred after updating Chrome to v87
  - K-4338: Fixed a bug concerning the filtering of the account list panel
  - K-4402: Fixed a bug that caused the blacklist panel not to be displayed for power users with the functionality enabled
  - K-4360: Fixed a bug that prevented the phonebook not to be created for users without an associated extension
  - K-4352: Fixed a bug that sometimes caused a backup not to be restored
- CDR
  - K-3896: Fixed a bug that caused inbound calls transferred with attended transfer to an extension with fork2mobile enabled not to be recorded in the CDR
- LDAP Phonebook
  - K-4332: Fixed a bug that sometimes caused the LDAP service not to be correctly initialized at startup
- Mobile APP
  - K-4510: Fixed a bug that caused missed calls to be displayed twice on KCTI Mobile Android

Firmware 4.11.6 (19/11/2020)

This release contains the updated SSL certificates for Apple push notification services, which are required for the correct functioning of the iOS app.

New features

- Third-party SIP phone module
  - K-4080: Added support for secondary identity handling on Cisco Unified IP Phones 78xx/79xx

Bugfixes

- API REST
  - K-4184: Fixed a bug that caused the GET CallCenterCDR API to fail with a 500 Internal Server Error
- CDR
  - K-3920: Fixed a bug that in certain conditions caused specific calls not to be recorded in the CDR
  - K-3395: Fixed a bug that cause the called number to be saved without a 0 for calls made from the app towards a landline

Firmware 4.11.3 (10/10/2020)

This version adds support for the new KalliopeLAM service, the Kalliope solution for video conferencing. After acquiring a dedicated license, the PBX admin will be able to assign to user groups the management of licensed video conference rooms; through the KalliopePBX web interface the user can easily view the status of each room, create a new meeting adding internal and external participants, and send invitations via email.

Note: the handling of issue K-3770 changes the way the phonebook is accessed via LDAP. Up until version 4.11.2 it was possible to access the dc=extensions,dc=phonebook,dc=<domain> sub-branch,dc=root anonymously, and authentication was only necessary to access the dc=system,dc=phonebook,dc=<domain>,dc=root sub-branch. Starting from this version, access to both sub-branches requires authentication.

New features

- GUI - K-2282: Added support for the KalliopeLAM video conferencing service - K-2473: Added management of closed groups and pickup groups with the option to exceed the predefined limit of 63 and with the option to assign mnemonic names instead of the identification number
- Kalliope-Lift Module
  - K-449: Added Amphitec protocol support

Changes

- LDAP phonebook
  - K-3770: Changed LDAP tree permissions to make it impossible to access tenant information without authentication in multi-tenant scenarios. As a consequence, the dc=extensions DN of each phonebook can no longer be accessed anonymously

Bugfixes

- Phone services
  - K-3758: Fixed a bug introduced in version 4.9.4 that in cases where the blind transfer of a call to an extension failed a forwarding action was performed; this bug also caused the call not to be inserted in the CDR
- GUI
  - K-2804: Fixed a bug that sometimes caused a “502 Bad Gateway” error to be displayed when accessing the web GUI on PBXs with 2GB RAM and certain configuration conditions
- KalliopeCTI
  - K-2354: Fixed a bug that caused the display of changes to the queue configuration not to be updated on the KalliopeCTI client
  - K-3735: Fixed a bug present in versions 4.10.0 and 4.11.0/1/2 where if the firmware was directly installed the chat service did not work for users whose username contained capital letters; the issue did not occur if the firmware was updated from previous versions
  - K-3614: Fixed a bug that under certain conditions caused the CTI server to crash, causing all clients to temporarily be disconnected
  - K-3089: Fixed a bug that sometimes caused and incoming call notification to be received twice if KalliopeCTI was in Free mode with more than one SIP account associated to the extension
  - K-3471: Fixed a bug that caused updates of the CDR on KalliopeCTI to be interrupted in the presence of calls with specific patterns
- Mobile APP
  - K-3380: Fixed the display of caller number of inbound calls in cases where the caller was on a landline (it previously removed the 0 from the city prefix)
  - K-2866: Fixed a bug that caused direct calls to an extension to be automatically refused for extensions with more than one associated SIP accounts
- General
  - K-3804: Restored the automatic execution of planned tasks
  - K-3801: Fixed the error notification mechanism during the evaluation of the disk space used by the tenant to send the email to the PBX manager and not the tenant admins
  - K-3606: Fixed an issue that sometimes made it impossible to restore a backup of size greater than around 200 MB
- CDR
  - K-3797: Fixed an issue that sometimes caused outgoing calls made by an extension which then transferred it to another destination not to be recorded in the CDR
- Rubrica LDAP

- K-3913: Fixed an issue that under certain conditions caused the LDAP phonebook of the PBX not to be populated on startup

Firmware 4.11.2 (12/08/2020)

New features

- API REST
  - K-3306: Added a new API to inject a sequence of DTMF tones into a call, identified by a Linked-id, with the option to send to on the caller's or the callee's channel
- Third-party SIP phone support module
  - K-3065: Added the option to define customized "Cisco Unified IP Phone" devices with a configurable number of function keys to manage equipped devices with an additional Cisco Cp-7914 keypad

Changes

- High availability
  - K-250: Optimized the invocation of periodic jobs on passive nodes of the HA cluster

Bugfixes

- General:
  - K-2963: Fixed a bug that caused pingbacks to fail in VMs, causing the warning banner to appear after the first 24 hours, if version 4.9.9 was directly installed; the issue did not occur if the firmware was updated from previous versions
- GUI
  - K-3121: Fixed an issue that sometimes caused the web interface to time out while displaying the IVR pages that contained a high number of sub-menus among all its levels
- API REST
  - K-2827: Fixed a bug that sometimes caused a malformed backup file to be returned if its size exceeded a certain limit
- Hotel Module
  - K-2874: Fixed a bug that prevented the correct synchronization of of room status on the passive node of an HA cluster
  - K-3010: Fixed a bug that if a room's status was changes via the phone service, failed the status returned by the API not to be updated in cases where Etags were used
- Kalliope-Lift module
  - K-3130: Fixed a bug introduces in version 4.9.6 that caused events concerning alarms not to be recorded, causing notification not to be sent

Firmware 4.11.1 (03/07/2020)

New features

- Kalliope-Lift module
  - K-1709: Fixed support for generic protocols for which reading acquisition is not required
  - K-2860: Added generation of a specific event for the end of a call before the protocol handshake is completed
  - K-2707: Made uniform the events generated by the Ademco Contact ID protocol, adding alarmreceiver.ademco-contactid.* alongside the legacy ademco.contactid.*

Firmware 4.11.0 (08/06/2020)

his version introduces support for a new optional module, extended third-party SIP phone support, concerning the integrated management of phones that implement some functions though non-standard SIP variants. At the moment these include some Cisco Unified IP Phones of the 79xx series that are flashed with SIP firmware, and others that share this property (e.g. 78xx).

The module enables handling of BLF (Busy Lamp Field) keys though the assigned SIP account panel and the generation of the corresponding provisioning file for the phone. Some services can also be controlled through the phone function keys (call pickup on BLF, unconditional call forward). The LDAP phonebook can also be accessed by phones for direct consultation.

Changes

- GUI
  - K-2507: In order to make the extension creation page more clear, when an account is created during extension creation there is now the option to only specify the base attributes (username, secret, and template), and the complete form will no longer be displayed.

1.1.7 Firmware series 4.10.x (MR)

Firmware updates from the 4.10.x series are Maintenance Releases, which include all features released in versions 4.9.x.

Firmware 4.10.2 (18/11/2020)

This release contains the updated SSL certificates for Apple push notification services, which are required for the correct functioning of the iOS app.

Firmware 4.10.1 (09/11/2020)

Note: the handling of issue K-3770 changes the way the phonebook is accessed via LDAP. Up until version 4.11.2 it was possible to access the dc=extensions,dc=phonebook,dc=<domain> sub-branch,dc=root anonymously, and authentication was only necessary to access the dc=system,dc=phonebook,dc=<domain>,dc=root sub-branch. Starting from this version, access to both sub-branches requires authentication.

Changes

- LDAP phonebook
  - K-3770: Changed LDAP tree permissions to make it impossible in multi-tenant scenarios to access tenant information without authentication. As a consequence, the dc=extensions DN of each phonebook can no longer be accessed anonymously

Bugfixes

- Phone services
  - K-3758: Fixed a bug introduced in version 4.9.4 that in cases where the blind transfer of a call to an extension failed a forwarding action was performed; this bug also caused the call not to be inserted in the CDR
- GUI
  - K-3121: Fixed an issue that sometimes caused the web interface to time out while displaying the IVR pages that contained a high number of sub-menus among all its levels
- API REST
  - K-4184: Fixed a bug concerning the GET /callCenterCdr API
- KalliopeCTI
  - K-2354: Fixed a bug that caused the display of changes to the queue configuration not to be updated on the KalliopeCTI client
  - K-3614: Fixed a bug that under certain conditions caused the CTI server to crash, causing all clients to temporarily be disconnected
  - K-3735: Fixed a bug present in versions 4.10.0 and 4.11.0/1/2 where if the firmware was directly installed the chat service did not work for users whose username contained capital letters; the issue did not occur if the firmware was updated from previous versions
- General
  - K-3606: Fixed an issue that sometimes made it impossible to restore a backup of size greater than around 200 MB
  - K-3804: Ripristinato il corretto funzionamento dell'esecuzione automatica dei task pianificati
- CDR
  - K-3797: Fixed an issue that sometimes caused outgoing calls made by an extension which then transferred it to another destination not to be recorded in the CDR
- LDAP phonebook

  - K-3913: Fixed an issue that under certain conditions caused the LDAP phonebook of the PBX bot to be populated on startup
- Mobile app
  - K-3380: Fixed the display of caller number of inbound calls in cases where the caller was on a landline (it previously removed the 0 from the city prefix)

Firmware 4.10.0 (29/07/2020)

Bugfixes

- K-2827: Fixed a bug that sometimes caused a malformed backup file to be returned if its size exceeded a certain limit
- K-2804: Fixed a bug that sometimes caused a "502 Bad Gateway" error to be displayed when accessing the web GUI on PBXs with 2GB RAM and certain configuration conditions
- K-3260: Fixed a bug concerning VM pingback (only present if version 4.9.9 was directly installed)
- K-2850: Fixed a bug concerning the execution of network commands (ping, traceroute) though the Kalliope Logger (only present in version 4.9.9)
- Hotel module
  - K-3217: Fixed a bug that caused the GET REST API /rooms (with ETag) not to be updated when the room status was changed through phone code
- KCTIS
  - K-3089: Fixed a bug that sometimes caused and incoming call notification to be received twice if KalliopeCTI was in Free mode with more than one SIP account associated to the extension
  - K-2812: Fixed a bug that caused the DND status of all extensions to be sent to clients simultaneously, which could cause the clients to be disconnected
  - K-2740: Fixed a bug that caused the a wrong notification to occur on KCTI iOS when the user made a call and cancelled it before it was picked up
- HA
  - K-2874: Fixed a bug that prevented the correct synchronization of of room status on the passive node of an HA cluster

Changes

- HA
  - K-247: Changed periodic cronjobs to execute only on the primary node
  - K-250: Changed periodic cronjobs to check that the resources used are active before execution

1.1.8 Firmware series 4.9.x (TR)

Firmware updates from the 4.9.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.8, is the stable release recommended for generic use that does not require features introduced in the TR.

Firmware 4.9.9 (10/06/2020)

Changes

- iOS App
  - Changed the mechanism for sending notification to iOS terminals so that the correct functioning of the chat and CTI mode can be restored
- Telephone services
  - In cases where an outbound call ends with a 480 response, no attempt to repeat the call on the backup lines will be made according to call routing rules (uniformed in case of a 486 - Occupied response). The backup will still be executed in all other cases.
- Hotel module
  - Added the Hotel Module license to the “4SP” bundle
  - Extended payment profile management to add importing and exporting to xls and the option to create a profile by cloning a preexisting one
  - Extended the Hotel Module APIs to only return changes to the previous status through the use of an ETag header

New Features

- API REST
  - K-2253: Added the API callCenterCdrReport

Bugfixes

- GUI
  - K-2042: Fixed the pagination for the panel for editing members of a call campaign or of a dial-out conference in cases where internal and external numbers were present
  - K-2404: Fixed a bug that under certain conditions caused 500 error during custom language pack creation
- Telephone services
  - K-2068: Fixed a bug that sometimes caused calls on the backup line to fail in cases where additional headers (PAI, PPI, RPID…) were added to the primary line, as these were not deleted before forwarding the call to the backup line
  - K-1975: Fixed a bug that caused the UniqueID in the web service to have the same value as the one in the CDR for callback calls via API
- KalliopeLogger

- K-2548: Fixed a bug introduced in version 4.9.8 that caused the authentication of the used pbxadmin to fail on the KalliopeLogger client

- High Availability
  - K-2510: Fixed a bug that caused a malformed HA configuration file to be generated if version 4.9.7 or later was installed by the bootloader, making it impossible to start the service
- API REST
  - K-2447: Fixed the functioning of the API for deleting call recordings with "linkedid" key
  - K-2496: Fixed validation for the API for exporting a tenant to handle target firmware version with non-numerical revisions (used starting from version 4.7.16)
- FAX Module
  - K-1989: Fixed a bug that prevented faxes and fax reports for previous months not to be accessible from the GUI
- Hotel module
  - K-1279: Fixed the display of the alarm clock in the room widget to display the next deadline and not the last one inserted
  - K-1758: Added the option to assign an unlimited number of rooms for each tenant in multi-tenant nodes
  - K-1283: Fixed a bug that caused all alarm clocks for a room deleted from the list of unanswered alarm clocks if a new alarm clock was created for that room
- Kalliope-Lift module
  - K-2703: Fixed a bug that caused forwarding actions not to be executed if an error occurred in the alarm receivers

Firmware 4.9.8 (26/04/2020)

Changes

- KCTI iOS app
  - Changed the management of notifications sent to KalliopeCTI apps to restore the functioning of incoming calls on the app on phones with iOS 13
- General
  - K-1940: the pbx.extension.missedcall event now contain the caller_name attribute
- CDR
  - K-1519: Added two columns in multi-tenant systems for reporting to geographic number to which the cost of the call will be attributed in cases where this differs from the caller number

Bugfixes

- GUI
  - K-2139: Extended the cache size to avoid blocks
  - K-2053: Fixed a bug that caused external dial-in conference participants not to be displayed is the line commitment code was empty
- Chat
  - K-182: Fixed a bug that caused all authentication to fail indefinitely if there was a burst of requests (e.g. after server restart or a network interruption)
- KalliopeCTI client
  - K-1971: Fixed a bug that caused the external line commitment code to the added in the CDR of desktop clients for outgoing calls, making it impossible to call the number by double-clicking from the CDR

Firmware 4.9.7 (31/03/2020)

New features

- Added an API for the operational management of the Hotel Module, which make it possible to implement on third-party systems all actions currently available through the integrated Receptionist panel (K-1622)
- Added some columns to the PBX CDR in multi-tenant systems to display the single tenant attribution for outgoing calls, distinct from the caller number

Bugfixes

- System
  - K-1889: Fixed a bug that in some PBXs made it impossible to load a new SSL certificate through the web configuration panel
  - K-1897: Restored a caching extension of the web interface, the removal of which (in version 4.9.4) caused slowdowns when accessing the KPBX configuration interface
- CDR
  - K-1651: Fixed a bug that caused calls made by SIP accounts whose username exceeded 40 characters (including the tenant prefix in multi-tenant systems) not to be saved in the CDR
- Multi-tenant
  - K-1863: Fixed a bug that in certain cases following an update from version 4.7.x to 4.8 or 4.9 caused the default tenant extension panel not to be displayed
- Call campaign service
  - K-1846: Fixed a bug that sometimes caused a campaign cancelled while it was blocked due to reaching the configured limits to continue after a restart

Firmware 4.9.6 (16/03/2020)

New features

- Extended the “Kalliope Hotel” module with the [Charges documentation] feature, which makes it possible to configure charges for calls made from rooms and generate a summary of the sustained costs from check-in. The reports persist after check-out and can be accessed through the “Booking log”.
- Added the option to select the dial-out participants of audio conference rooms and the recipients of call campaigns from the phonebook by beginning to dial the number or the name; the system will suggest the matching contacts. N.B.: it is not currently possible to select numbers marked as extensions (K-1233, K-1236)
- Added the generation of new notifiable events to the Dynamic Routing service upon entering the service and every time an input from the caller is gathered (K-1479)

Bugfixes

- Telephone services
  - K-1717: Fixed a bug that caused the blacklist not to work for inbound lines if a match on the called number was specified
- Audio conference service
  - K-1212: Fixed a bug that caused external dial-in participants not to be displayed in the conference status panel (if not already present as dial-out participants)

Changes

- System
  - K-1467: The presence of mounted remote filesystems (NFS o CIFS) caused changes to network configuration to fail; the system now checks for the presence of remote filesystems and requests them to be temporarily disabled before making changes to the network configuration
- Telephone services
  - K-1491: Introduced a loop-mitigation mechanism that prevents a call from being forwarded more than 20 times. This prevents possible performance issues or crashes of the call documentation system in cases where deviations or forwards are intentionally or accidentally programmed to cycle between two or more entities. Once the limit is reached, the call will be automatically terminated.
- FAX Module
  - K-627: Extended the handling and validation of the attributes received by the transmission apparatus when a fax is received, which previously, if malformed, caused the received fax not to be saved
- API Rest
  - K-1498: Extended the CDR API to support POST filtering with additional attributes (e.g. accountcode)

Firmware 4.9.4 (20/02/2020)

New features

- Added a new “Warning Campaign” service, which automatically makes a series of calls to a preconfigured list of of recipients (extensions or external numbers), plays a prerecorded audio message, and optionally gathers the confirmation that each has answered the call/listened to the message.
- Added a new optional “Kalliope Hotel” module dedicated to hotel phone management, which can be activated through an additional license, The features offered by the Hotel module include:
  - Management of phone users for each room, with the option of blocking direct inter-room calls
  - Check-in/check-out service, with registration of the names of the guests of each room with the option to add notes
  - Selective block of external calls for each room through configurable classes
  - “Clean room service”, which automatically marks each occupied room as “dirty” every night and returns them to “clean” status through a phone code or through web GUI
  - Alarm clock service, which allows one or more alarms to be set for each room and displays through web GUI those that have not been answered
  - Receptionist dashboard, a web panel that displays the status (check-in, cleaning status, next alarm) of each configured room, which can be filtered and searched (by building, floor, guest, or note), through which the rooms can be supervised and managed

Note: the Hotel module includes a charge management service that is not available in this version but will be released in an upcoming update (currently planned for version 4.9.6)

- Extended the “Kalliope Lift” module to implement new protocols (in addition to the preexisting Ademco ContactID) used by the alarm systems

Bugfixes

- CDR
  - Kalliope-933: Fixed a bug that caused the outcome of an outbound call that was canceled before being answered to be incorrectly displayed (NOANSWER instead of CANCELED) in cases where a manipulation of the caller number was present in the outbound line
  - Kalliope-925: Fixed a bug that caused manipulated calling and caller numbers not to be displayed in cases where a manipulation was applied to one of the two numbers for an outbound call
  - Kalliope-1153: Fixed a bug introduced in version 4-8-0 that caused calls made through click-2-call services (API or web interface) not to be displayed
- KCTI Mobile APP
  - Added a time-to-live to the notifications sent to the app to prevent calls arrived during connectivity outages from being presented to the app

Changes

- Extended the timeout for outbound faxes from 20 to 30 seconds
- Replaced the REST API documentation bundle; the integrated sandbox through which APIs could be invoked via web interface has been replaced by a Postman collection that integrates the code to automatically add the required authentication header (it is only necessary to set the IP address of the PBX and the username/password credentials of the user who invokes the API)

1.1.9 Firmware series 4.8.x (MR)

Firmware updates from the 4.4.x series are LTS Maintenance Releases, which include all features released in versions 4.7.x.

Firmware 4.8.5 (18/05/2020)

Bugfixes

This version only solves the issue with the REST APIs which occurred in version 4.8.4.

Firmware 4.8.4 (30/04/2020)

Known issues

This version contains an issue in the execution of the REST APIs. We recommend installing version 4.8.5 instead.

Changes

- KCTI iOS app
  - Changed the management of notifications sent to KalliopeCTI apps to restore the functioning of incoming calls on the app on phones with iOS 13

Bugfixes

- General
  - K-2154: Fixed the handling of 480 error when an external call fails so that no attempt to repeat the call on the backup lines will be made
  - K-2157: Fixed a bug that sometimes caused calls on the backup line to fail in cases where additional headers (PAI, PPI, RPID…) were added to the primary line, as these were not deleted before forwarding the call to the backup line
  - K-1839: Fixed a bug that caused the blacklist not to work for inbound lines if a match on the called number was specified
- GUI
  - K-1922: Fixed a bug that in some PBXs made it impossible to load a new SSL certificate through the web configuration panel

    - K-1832: The presence of mounted remote filesystems (NFS o CIFS) caused changes to network configuration to fail; the system now checks for the presence of remote filesystems and requests them to be temporarily disabled before making changes to the network configuration

- Multi-tenant
    - K-1901: Fixed a bug that in certain cases following an update from version 4.7.x to 4.8 or 4.9 caused the default tenant extension panel not to be displayed
- FAX Module
    - K-2205: Fixed a bug that prevented faxes and fax reports for previous months not to be accessible from the GUI
- KalliopeCTI Client
    - K-1971: Fixed a bug that caused the external line commitment code to the added in the CDR of desktop clients for outgoing calls, making it impossible to call the number by double-clicking from the CDR

Firmware 4.8.3 (21/01/2020)

Known issues

See 4.8.0

Bugfixes

- CDR
    - K-1153: Fixed a bug introduced in version 4.8.0 that caused outbound fax calls and calls made through the click-2-call service not to be displayed in the CDR

Firmware 4.8.2 (23/12/2019)

Known issues

Bugfixes

- Multitenant
    - K-994: Fixed a bug present in version 4.8.0 that caused manipulation rules for calling and called numbers on outbound lines assigned to the tenants not to be applied

Firmware 4.8.0 (2/12/2019)

> **Warning:** This update distributes the updated certificates required by Apple's PushKit service. The certificates installed with previous versions will expire in January 2020. To guarantee the correct functioning of the KalliopeCTI mobile app for iOS after that date, it is necessary to update KalliopePBX to this version.

Note: In order to update the firmware to version 4.8.0, it is necessary to update the bootloader to version 1.0.7 (which is recommended even for previous firmware versions).

Known issues

For outbound lines (gateways, trunks, or terminations) with a space in their name (e.g. "Outbound line"), the outcome of outbound calls will always be reported in the CDR as "NOANSWER" even if the call was answered and its "Conversation time" attribute is not null. The current workaround is to remove any spaces in the names of all outbound lines.

New features

- KalliopeCTI (Pro)
  - K-217: added the option when configuring a provisioning device to set the IP and access port of the GUI of the phone used for remote control through KalliopeCTI or Pro.

Bugfixes

- Telephone services
  - K-545: Fixed a bug concerning the group call pickup with invite service that in multi-tenant systems sometimes caused the wrong caller name to be displayed when other inbound calls on other tenants were present
- Kalliope CTI
  - K-135: Fixed a performance issue with the queries used to pass the user CDRs to the KalliopeCTI clients that caused the data to be slow to update on the client when the PBX had a high number of calls (more than a hundred thousand per month)
- FAX Module
  - K-647: Fixed a bug that caused fax log export requests to generate an empty file

1.1.10 Firmware series 4.7.x (TR)

Firmware updates from the 4.7.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.6.0, is the stable release recommended for generic use that does not require features introduced in the TR.

Firmware 4.7.17 (29/10/2019)

Bugfixes

- System
  - Kalliope-549: Fixed a bug (present in versions 4.7.15 and 4.7.16) that caused scheduled jobs for evaluating accessibility statistics (Call Center module) to fail. The script failing caused a log to grow indefinitely, eventually fully occupying the file system. The fix removes the problem and prevents the log from growing indefinitely.

Known issues

Please note that when using this firmware version there may be delays in the display of the CDR in KalliopeCTI Desktop in cases of high telephone traffic (more than 5000 daily calls) while more than 100 KalliopeCTI Desktop clients are simultaneously connected. The telephone engine and web GUI CDR are not affected. This issue will be fixed in the version 4.8.0.

Firmware 4.7.16 (05/10/2019)

New features

- REST API
  - Added a new REST API to export the CDR in "blues" format containing outgoing, local, and incoming calls, which can be filtered by category. The output of this API is different from the preexisting "blues_out" and is used (optionally, as an alternative) by the new version of Kalliope Blue's Connector alongside the new import drivers for Blue's Enterprise

Bugfixes

- GUI
  - 7691: Fixed a bug that caused an exception when opening the import panel for Kalliope v3 backups
- Telephone services
  - 7197: Fixed a bug that prevented forwarding cycles of an extension to itself from being detected
- Dial-out conference service
  - 7134: the application did not notice when a dial-out call to a participant failed (e.g. because it was busy or refused) and did not try to call again even under an automatic invite with repetition policy
- Mobile app
  - 7303: Fixed a bug that caused transfers of calls received by the app to an external number to fail
- Provisioning
  - 7590: Added SNOM D717 to the list of built-in devices
- Kalliope4SP
  - 7572: Extended the timeout for connecting to the license server so that the transfer of billing data accumulated through a long period of time during which the license server was unreachable does not fail

Known issues

Please note that when using this firmware version there may be delays in the display of the CDR in KalliopeCTI Desktop in cases of high telephone traffic (more than 5000 daily calls) while more than 100 KalliopeCTI Desktop clients are simultaneously connected. The telephone engine and web GUI CDR are not affected. This issue will be fixed in the version 4.8.0.

Firmware 4.7.15 (26/07/2019)

Bugfixes

- KalliopeCTI
  - 7586: Fixed a bug that caused the name of the callee not to be displayed on KalliopeCTI for calls to an external number and that made it impossible to redial from the CDR

Known issues

Please note that when using this firmware version there may be delays in the display of the CDR in KalliopeCTI Desktop in cases of high telephone traffic (more than 5000 daily calls) while more than 100 KalliopeCTI Desktop clients are simultaneously connected. The telephone engine and web GUI CDR are not affected. This issue will be fixed in the version 4.8.0.

Firmware 4.7.14 (18/07/2019)

Bugfixes

- KalliopeCTI
  - 7376: Display of missed calls from groups or queues on the CT client

Firmware 4.7.13 (11/07/2019)

New features

- General:
  - Added new formats for the timestamp within notification actions. The %event_timestamp% parameter now includes a format option that produces a customizable string (with millisecond resolution) instead of the epoch.

Bugfixes

- General:
  - 5171: fixed a bug that made it possible for users to change the numbering plan even if heir role only had read permissions
  - 6808: extended the length limit for the secret field for trunks and VoIP terminations from 40 to 128 characters
  - 7054: added alert when saving an extension whose failover action is to forward to voicemail when the extension does not have a voicemail box configured
  - 7319: fixed the validation of the selection of forwards to a parking slot to make it numeric and outside the range of the parking slots
- Call Center module
  - 7405: fixed a bug that sometimes caused the VoIP services to restart when a call served through the automatic callback feature ended
  - 7322: fixed the display of queued calls handled through the the automatic callback feature in the CDR and adjusted the statistics report generation to take this type of call into account
- KalliopeCTI
  - 7444: fixed an anomaly that caused the active frame to persist after the end of a call for call pickups or attended call transfers from a phone of the extension associated with the client
- KalliopeCTI Mobile App (Android/iOS)
  - 7393: fixed a bug that sometimes caused calls not to be received by the app if the phone was in deep sleep mode
- Phonebook
  - 6237: fixed an issue with the display that made it impossible to save lists with more than 11 contacts

Firmware 4.7.12 (06/06/2019)

New features

- General:
  - Extended the audio conference service to include dial-in/dial-out modes and a comprehensive room supervision and management panel with a list of participants and the option to add/remove/mute/unmute users both from the GUI and through APIs
- Multi-tenant:
  - Added the option to set the tenant admin password during creation instead of setting a default password (admin)
  - Added the option to activate an unconditional forward on an "assigned line" level (all associated numberings) to an external number through API; the forward uses the same line on which the call was received

Bugfixes

- General:
  - 6992: The "ignore source port during recognition" flag is now respected for gateways and VoIP domains with TCO or TLS transport
  - 7085: Fixed the preservation of the calling number within the Display-Name when the <displayprefix> attribute was set in the Dynamic Routing service
- Rest API:
  - 6968: Fixed the /rest/outboundLines/voipLine API, which always reported the "Parameter 'domainIp' is missing or malformed" error
  - 7172: Fixed the execution of the API for adding contacts to the phonebook (POST /rest/phonebook/shared) for JSON
  - 7088: Restored the API POST /rest/operation/ufwdWhitelist/{username}, which broke in version 4.7.9
- Provisioning:
  - 7041: Fixed a bug when saving provisioning devices that were associated with multiple SIP accounts (multi-account devices)
- Call Center module
  - 7047: Fixed the calculation of simultaneity of the "service accessibility" statistic, which previously considered the start of call event (inbound and outbound) associated with the answer instead of the effective start of the call
- Multi-tenant:
  - 7152: Fixed a bug with the export of tenant backups when the tenant name contains a backslash ()
  - 7042: Fixed a bug that when the fax lines were configured on multiple tenants caused faxes not to be sent and blocked fax reception in the "Ready for conversion to PDF" status

Firmware 4.7.9 (14/03/2019)

New features

- Implemented REST APIs for managing the phonebook (adding/editing/deleting contacts)
- Added the option to mass assign provisioning devices to the Hot Desking service

Changes

- Generation of all function keys configured independently of the phone model (in order to manage expansion modules)

Bugfixes

- 0006860: Eliminated the Diversion Header containing the redirecting extension for calls redirected towards external numbers (certain VoIP providers would reject the call)
- 0006871: Fixed a bug that sometimes caused the CCBS service not to function within a queue
- 0006975: Fixed a bug that when mass importing extensions from file sometimes caused an error when setting the "force password change" flag
- 0007048: Fixed a bug that sometimes caused the interruption of the fax sending service on multi-tenant systems

Firmware 4.7.8 (25/02/2019)

New features

- Implemented a Mail2FAX service that allows faxes to be sent via email
- Added a new widget to the user dashboard that lets phone function keys be configured.
- Added a work code field in the Call Center CDR export

Bugfixes

- 0006731: Added a check to prevent member to be added to a queue without specifying an account
- 0006762: Added a check to prevent the values of parking slots to contain the "*" character, which would cause calls to be terminated when answered
- 0006778: Fixed a bug that prevented a backup from being restored on a different PBX than the one it was made on if provisioning devices were present
- 0006690: Fixed a bug that caused rebranding not to be reset after restoring a backup with rebranding disabled on a PBX with rebranding enabled
- 0006730: Fixed a bug where the deletion of an account did not remove the account from the list of dynamic members of queues
- 0006757: Fixed a bug that sometimes prevented fax from being sent for archival on remote storage
- 0006752: Fixed a bug that caused the "guided configuration" menu not to be displayed when HA was enabled
- 0006760: Fixed a bug that prevented access control rules from being disabled for paging groups
- 0006837: Fixed a bug that prevented the Call Center CDR from being downloaded from KCTI
- 0006584: Fixed a bug that sometimes prevented call recordings from being archive on remote storage
- 0006832: Fixed the filename generation rule for "Maxwell 2" devices
- 0006761: Fixed a bug that prevented the correct functioning of the %%IPUI%% placeholder
- 0006439: Fixed a bug with ordering by duration of the CDR
- 0006853: Fixed a bug that caused the outbound routing of calls that did not contain the line commitment code
- 0006870: Fixed a bug that caused periodic operator statistic not to be sent to the KCTI if the user had a CC operator role
- 0006559: Fixed a bug that caused restoration from backup to fail when the associated tenant group had already been created

- 0006831: Fixed a bug that caused user configuration (e.g. forwards, pauses) not to be restored hen restoring from a backup on a different node
- 0006809: Fixed a CDR bug that caused calls to queues that use the work code were sometimes erroneously considered answered
- 0006856: Fixed a bug that caused the MIB of configured accounts not to be updated when the tenant had been removed

Reworking

- 0006796: Optimized the backup creation mechanism in multi-tenant systems to reduce execution time
- 0006855: Changed the VoIP domains page to reduce the loading times of the line assignments to tenants section
- 0006798: Changed the assigned line management page to reduce loading times

Firmware 4.7.4 (21/12/2018)

New features

- REST API
  - Added a GET /rest/phonebook REST API for consulting the phonebook
  - Added a GET GET /rest/provisioning/settings REST API for managing provisioning settings
- CDR
  - The user CDR now displays calls to groups or queues that the user belongs to (currently only on the web GUI)
- Call Center module
  - Extended the Call Center report with operator statistics (served calls, calls answered by another operator, missed calls, time spent paused)

Bugfixes

- 0006723: Fixed a bug that under certain conditions caused the TIMEOUT exit reason to the call details when a call was canceled.
- 0006677: Fixed a bug that caused the exit reason for all call details to be overwritten after a call pickup on queue
- 0006546: Fixed a bug that caused phone configuration not to be completely generated when more than 350 tenants were defined
- 0006703: Fixed the filename generation rule for Gigaset Maxwell 2 devices
- 0006764: Fixed an issue with backup restoration when configured FAX instances were present
- 0006765: Fixed an issue with server reachability checks for new installations of firmware version 4.7.3

Firmware 4.7.3 (11/12/2018)

New features

- Introduced integrated FAX module support, which currently offers the option to send/receive faxes via web GUI and receive email notifications for received faxes

Bugfixes

- 0006633: Fixed a bug that causedadditional headers (PAI / PPI / RPID / Call-Info) not to be added when the configuration string contained the " character
- 0006516: Fixed a bug that made it impossible to change the template associated with a device directly from the provisioning device list
- 0006469: Fixed a bug with the overwriting of a cvs file in Dynamic Routing from file
- 0006479: Fixed a bug that when a blacklist entry was edited caused last modified to change for all entries
- 0006618: Fixed a bug that caused scheduled tasks not to be executed after a firmware update
- 0006625: Fixed a bug that caused scheduled tasks not to be executed after a backup was restored
- 0006636: Fixed a bug that caused click-to-call calls to be forwarded as well when call forwarding was active
- 0006623: Fixed a bug that sometimes caused certain extension not to be displayed in the corresponding KCTI panel
- 0006658: Fixed a bug that made it impossible to create new provisioning devices after a backup was restored
- 0006666: Fixed a bug that made it impossible to create new extension templates after the default emplate was edited
- 0006596: Fixed a bug that caused changes to the SNMP configuration not to be applied if the first configuration was done on firmware 4.5.11 or later
- 0006273: Fixed a bug that caused the outbound status of calls answered through fork2mobile to be incorrectly displayed in the CDR
- 0006101: Fixed a bug that caused calls to log into the hot desking service not to be displayed in the CDR

Firmware 4.7.2 (20/11/2018)

New features

- Added support for application rebranding (KCTI Desktop and KCTI Mobile). This feature requires a K4SP or Rebranding license.
- REST API
  - Added a REST API that returns the instantaneous map of current calls (filterable by extension or account)
  - Added a REST API for editing and revoking product licenses assigned to a tenant
  - Added a REST API for configuring notifications and notification actions
- Provisioning
  - Added management for Polycom root CA and HTTPS provisioning
- Misc
  - Added distinctive ringing support for CISCO SPA5xx and SPA3xx phones

- Added the option for the pbxadmin to view the full audit log (which includes the actions of individual tenants)

- Reworking
  - Optimized the mechanism for calculating queue statistics to avoid delays when displaying statistics in the Supervisor Panel
  - Changed the email-type event notification action not to include the serial number of the KPBX (it may be added as a placeholder).

Note: This change affects actions configured before the update; the placeholder corresponding to the serial number is only added to node events in multi-tenant systems during migration so that the notification is the same as the one sent before the update.

Bugfixes

- Telephone services
  - 6548: Fixed a bug that caused click-2-call calls towards 1-digit selection not to function
  - 6575: Fixed a bug that sometimes caused trunks and VoIP terminations not to be registered when the "registration domain" item was set
  - 6620: Fixed a bug that caused COLP updates to fail for calls from an extension to a speed dial
  - 6576: Fixed a bug that prevented the blind transfer of calls answered through the Fork2Mobile service
- GUI
  - 6513: Fixed an issue where the registration state of trunks and VoIP terminations was not displayed when "Registration domain" parameter was not set to null
  - 6639: Fixed a bug that sometime caused thousands of empty rows to be added when uploading a dynamic routing xls file
  - 5634: Fixed a bug that caused a 500 "Internal server error" when trying to order the SIP accounts by the "ACL IP source", "ACL IP Contact", or "SRTP enabled" columns
  - 6574: Fixed a bug that caused a 500 error when integrating the shared phonebook by importing from an xls file with a preesiting contact without the type attribute
  - 6556: Fixed a bug that caused the warning to be displayed twice when mass importing the phonebook
  - 6554: [Multi-tenant] Fixed the validation of the form for saving the assigned lines to handle the case where a prefix manipulation rule is specified omitting the value of the prefix, which previously caused the page to silently fail to be saved
  - 6547: [Multi-tenant] Fixed a bug that caused all tenant limits to be set to zero the moment an expired K4SP license was reactivated

Firmware 4.7.0 (26/10/2018)

New features

- Added the option to generate an event ("pbx.extension.missedcall") for missed calls to an extension. This can be enabled individually for each combination of reason (busy, not answered, not available) and origin (internal call, external call, transferred call). It is also possible to enable notifications (e.g. via email) sent to the %event_param[email_address]% placeholder, which corresponds to the email address linked to the extension.
- Added the option to force users to reset their password when they log in for the first time after the user was created or the admin changed the password.
- Added a REST API for resetting user passwords.
- Added events for to creating a new user ("system.user.create") and changing a password ("system.user.password-change"), which can be linked to the mechanism for sending notifications.
- Added an event for creating a new tenant ("system.tenant.create") [Only relevant to multi-tenant nodes].
- The available scheduled tasks now include sending the CDR extract (on a configurable schedule).
- Added the option to configure a different outbound proxy for each SIP account.

Changes

- Updated Asterisk to version 13.21-cert2.
- Changed the generation of the %event_params[<format>]% of the event parameters within notifications to include the general attributes of the event alongside the specific ones.
- Changed the name of the "Periodic reports" panel under "Scheduled tasks".
- Removed the option to backport the configuration when restarting on the secondary firmware.

Bugfixes

- Phonebook
  - 6573: Fixed a bug that caused a "500 Internal server error" when importing a file containing an entry that already existed in the PBX phonebook with a contact that lacked the "type" attribute (fixed, mobile, etc.)
- GUI
  - 6571: Failed display of active calls (on the "active calls" widget, the panel of the same name, and in notifications to the KCTI client) under specific load conditions
- Call center module
  - 6370: Fixed a bug that under certain conditions caused the pause status of the operators not to be restored after rebooting the machine

1.1.11 Firmware series 4.6.x (MR)

Firmware updates from the 4.6.x series are LTS Maintenance Releases, i.e. stable versions with long term support. The releases in this series have been thoroughly tested before being released to the public and therefore guarantee greater stability.

Firmware 4.6.2 (16/01/2019) - Old stable

Changes

- 0006775: Added a check to prevent firmware updates when the secondary firmware is in use

Bugfixes

- 0006667: Fixed a bug that made it impossible to create new extension templates after the default template was edited
- 0006689: Fixed a bug that caused changes to the SNMP configuration not to be applied if the first configuration was done on firmware 4.5.11 or later
- 0006695: Fixed a bug that caused phone configuration not to be completely generated when more than 350 tenants were defined
- 0006704: Fixed the filename generation rule for Gigaset Maxwell 2 devices
- 0006770: Fixed a bug that caused the %%IPUI%% placeholder not to be generated during the generation of provisioning files
- 0006771: Fixed a bug that caused the "guided configuration" menu not to be displayed what HA was enabled
- 0006772: Fixed a bug that caused an exception when a queue was configured with an unspecified account
- 0006774: Fixed an issue with filtering the CDR by "call duration" and/or "time of billing"
- 0006776: Fixed a bug that sometimes caused call recordings not to be archived on network storage
- 0006779: Fixed a bug that caused an error when restoring a backup containing provisioning devices on a different node than the one on which it was created
- 0006784: Fixed a bug that caused click-to-call calls to be forwarded as well when call forwarding was active
- 0006786: Fixed a bug that made it impossible to change the template associated with a device directly from the provisioning device list
- 0006787: Fixed a bug with the overwriting of a cvs file in Dynamic Routing from file

Firmware 4.6.1 (02/11/2018)

Changes

- Telephone services
  - 6508: Changed the way the code confirmation request is handled when applying Dynamic Routing so that failure to confirm after 3 times is considered a negative response (instead of waiting indefinitely for explicit positive or negative confirmation)
- Audit log

- 6509/10: Added masking (during creation and editing) of the service PIN and the user password

Bugfixes

- Telephone services
  - 5939: Fixed a bug that prevented the blind transfer of calls answered through the Fork2Mobile service
  - 6564: Fixed a bug introduced in version 4.6.0 that sometimes caused trunks and VoIP terminations not to be registered when the "Registration domain" item was set
  - 6549: Fixed a bug that caused click-2-call calls towards 1-digit selection not to function
  - 6500: Fixed a bug that under certain conditions caused the pause status of the operators not to be restored after rebooting the PBX
- Call recording
  - 6538: Fixed a bug introduced in version 4.6.0 that caused the call recording service not to work for calls made by an extension
- GUI
  - 6050: Fixed a bug that caused a 500 error when integrating the shared phonebook by importing from an xls file with a preexisting contact without the type attribute
  - 6517: Fixed an issue where the registration state of trunks and VoIP terminations was not displayed when "Registration domain" parameter was not set to null
  - 6555: Fixed a bug that caused the warning to be displayed twice when mass importing the phonebook
  - 6552: Fixed a bug that caused a 500 "Internal server error" when trying to order the SIP accounts by the "ACL IP source", "ACL IP Contact", or "SRTP enabled" columns
  - 6425: [Multi-tenant] Fixed the validation of the form for saving the assigned lines to handle the case where a prefix manipulation rule is specified omitting the value of the prefix, which previously caused the page to silently fail to be saved

Firmware 4.6.0 (24/09/2018)

New features

- Extended the Kalliope SNMP subagent to collect new information such as the number of active calls, the number of calls since last reboot, etc.
- Extended the /rest/phoneServices/callback/ REST API to manage source and destination as selections (and not necessarily as extensions)
- Xtelsio TAPI for Asterisk integration (allows integration with Estos ProCall)

Reworking

- Changed the way KCTI client requests are handled to improve KCTIS response time
- Changed certified import/upload mechanisms for intermediate CA management
- Changed idletimeout configuration on the LDAP service to prevent blocks due to lack of connections
- The warranty expiration date is now displayed in the list of licenses and the 'Product Information' widget
- Changed the organization of the 'Operating mode' and 'Whitelist' panels

Provisioning

- 0006381: Added new SNOM CA to the ones preloaded on KPBX
- 0006372: Added handling for new Yealink 80:5e:c0:xx:xx:xx MAC addresses on the PNP SIP service
- 0006383: New built-in provisioning device: Snom D385
- 0006347: Added the option to set a hostname in the SNOM redirection server

Bugfixes

- 0006124: Fixed a bug on performed actions in cases of error/timeout in Dynamic Routing from file
- 0006440: Fixed a bug on call transfer for extensions belonging to closed groups
- 0005736: Fixed a bug on phonebook lookup for routed calls on a group/queue
- 0006327: Fixed a bug when editing a password for the Snom redirection server
- 0006365: Fixed a bug when checking available space on remote filesystems
- 0006373: Fixed a bug when archiving call recordings on remote filesystems
- 0006051: Fixed a bug when editing the name of an extension with "Presenting the number below" in the extension template
- 0006355: Fixed a bug on the visibility of the Meetme Applications menu for users with no linked extension
- 0005593: Fixed a bug when displaying multiple user phonebooks
- 0006367: Fixed a bug on FastTransfer for accounts whose usernames contains the character "-"
- 0006329: Fixed a bug when playing audio files whose filename contains the character "&"
- 0005943: Fixed a bug to allow the character "?" to be used in trunk/VoIP termination passwords
- 0006247: Fixed a bug when sending check-syncs when a custom placeholder is edited
- 0006354: Fixed a bug when displaying calls for users with no linked extension in the /rest/cdr REST API
- 0006410: Fixed a bug when displaying anonymous calls in the /rest/cdr REST API
- 0006401: Fixed a bug when filtering calls in the /rest/cdr REST API
- 0006458: Fixed a bug when validating linkedID in the /rest/recordedCall/{linkedId} REST API

1.1.12 Firmware series 4.5.x

Firmware updates from the 4.5.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.4.2, is the stable release recommended for generic use that does not require features introduced in the TR.

> **Warning:** Starting from firmware version 4.5.4, there is a 1GB RAM requirement for VMs. If the firmware is updated or installed on a VM that does not meet the memory requirements, the PBX will fail to start.

Firmware 4.5.17 (02/08/2018)

New features

- Added a Blacklist service on inbound lines; one or more access lists can now be defined for each inbound line (based on the calling number and optionally on the called number), each associated with a specific action (block, forward, etc.) so that both backlists and whitelists can be defined
- Added the option to encode call recording files for each archival path (local or remote). Encoded audio files saved on remote archival paths (share NFS or CIFS) can only be listened to from the Kalliope web interface (or downloaded unencoded via API) and not directly from the file system
- Added REST APIs to manage call recording files; APIs for listing (with GET and POST filtering similar to the CDR API), download, and erasing (with linked key) are defined.
- Added REST APIs to consult the list (with the option to filter messages after a certain date), listen to and erase messages in the voicemail box
- Added a panel (and corresponding REST API) for defining the default template settings
- Added automatic erasure of CDR and Call Center CDR records older than 2 years (records for the current month and the 24 previous whole months are kept)
- Added generation of new queue and member events (pbx.queue.enter, pbx.queue.enqueue, pbx.queue.ringmember, pbx.queue.ringnoanswer, pbx.queuemember.pause e pbx.queuemember.unpause), and changed the pbx.queue.servedcall and pbx.queue.unservedcall events to be generated the moment the event happens instead of the end of the call, as happened previously
- Added storage.quota.exceeded (replacing pbx.filesystem.quota.exceeded, which was removed) and storage.quota.restored events, generated when the storage quota configured per tenant is reached and when it is restored
- Added native Country Code setting to outbound lines to normalize the called number of the extensions

Changes

- Optimized the CTI service to better handle requests from the client in order to reduce response times of sent commands
- Updated the publication mechanism of devices on the new Yealink RPS
- Extended the granularity of event timestamps to microsecond precision

Bugfixes

- Telephone services
  - 6054: In multi-tenant systems, edits to the outbound proxy in the system SIP settings were not applied to single tenant accounts until a tenant "Apply" was executed
  - 6137: Fixed a bug that caused the enable flag for audio conference rooms not to function
  - 6156: Fixed the handling of inbound calls whose calling number is empty (made uniform with anonymous calls)
  - 6260: Restored the correct functioning of call recording on demand from extensions
- GUI
  - 5169: Fixed the mechanism for mass importing extensions when the user password column is omitted; the user is created with the automatically generated password shown on the screen during import file validation
  - 6037: Fixed a bug that caused audio file playback and recording not to function for hotdesking SIP accounts
  - 6161: Extended the validation of text fields in forms invoked via API to prevent special characters from being insterted (r, n, t, v, and f)
  - 6133: [Multi-tenant] Fixed the validation of tenant group edit forms, which allowed extact remote numberings to be defined without specifying their value
- Provisioning
  - 6280: Fixed the regular expression to recognize the MAC address from the URL, which caused the extraction of an incorrect MAC address for CISCO SPA devices
- CTI
  - 6068: Fixed a bug that caused the incorrect display of the calling number for missed call notifications
- Mobile app
  - 6203: Fixed the handling of the SIP account status of the app when the client is not logged in

Firmware 4.5.15 (16/07/2018)

New features

- Extended the Dynamic Routing service to send DTMF sequences and insert pauses within the dynamic component of the response

Changes

- Extended Kalliope mobile app support to make calls without the need to add the external line commitment prefix, in order to best take advantage of the device's phonebook

Bugfixes

- Telephone services
  - 6132: Fixed a bug when the unconditional forward service interacted with closed groups: if three extensions belonged to the same closed group, if extension A called extension B and was redirected to C, the call was blocked as not allowed
  - 6245: Fixed a bug where the caller number was incorrectly set for external calls to an extension redirected to an external number
- Provisioning
  - 6087: Fixed a bug where certain placeholder were incorrectly replaced for custom devices
  - 6149: Fixed a bug that did not prevent the provisioning files generated by the PBX, which are protected from being erased, from being moved
- Call Center module
  - 6151: Fixed an issue with the generation of Call Center reports when one or more queues had names longer than 32 characters

Firmware 4.5.11 (20/06/2018)

New features

- Extended the /rest/operation/service API to handle the new services:
  - BUSYLEVEL: changes the Busy Level on an extension configuration level
  - CFBS, CFNA, CFUN: call forwarding on busy subscriber (BS), no answer (NA), and unavailable (UN) to a selection of the numbering plan, with priority over the values set in the extension configuration
- Extended the user widget to manage the CFBS, CFNA, and CFUN operation from the GUI
- Added a Periodic Report Generation feature (with customizable time span) with the option to send reports to a group of recipients via email
- Added a Forward on All Unreachable service, which routes inbound calls to a failover destination when all SIP accounts are unreachable
- Extended the Dynamic Routing service to carry out authentication through KalliopePBX client certificate
- Added support for the new Kalliope mobile app with integrated phone functionality
- Added the option to downgrade the configuration when rebooting on secondary firmware (feature only available for secondary firmware version 4.5.8 or later)
- Added the option to import configuration backups made with firmware versions older than the one on the PBX (feature only available for firmware version 4.5.8 or later)
- Extended the REST API for generating backups to export a backup for a specific firmware version equal to or older than the current one (feature only available for firmware version 4.5.8 or later)

Changes

- General
  - 5747: Updated Asterisk to version 13.18-cert3
  - 5730: Changed the call generation mechanism for certain services (call pickup with consultation, recording calls and listening to audio files from the terminal, callback service on queues) due to occasional cases of services failing to activate
  - 5627: Extended the SNMP agent to export occupation data of the folder /tmp (OID: ucdavis.dskTable, index 3)
  - 5944: Added a flag to the SIP account settings to identify those that can be assigned to the mobile app
- Provisioning
  - 6010: Changed the predefined settings to disable the service mechanisms of insecure files (TFTP, HTTP, HTTPS without authentication of the requester via client certificate)
  - 5941: Optimized the mechanism for notifying phones when the provisioning file is downloaded
  - 5858: Optimized the search for the provisioning file when the file was directly requested

Bugfixes

- General
  - 5763: Fixed a bug with the formatting of XLS files generated by the system (on demand and periodic reports, exported CDR, and other tables) where files could not be opened with Microsoft Excel (they could be correctly opened with LibreOffice and WPS)
  - 5641: Fixed a bug where the Passive Listening service failed to function for operators whose SIP account contained the character "-"
  - 4863: Fixed occasional issues where a "The controller must return a response (null given). Did you forget to add a return statement somewhere in your controller?" error was presented after login, which required the user to explicitly log out and then log in again
  - 5671,5717: Fixed some issues where edits to the telephone configuration could fail to be applied
  - 5723: Fixed a bug where the backup restoration process was interrupted due to an attempt to insert a duplicate role
  - 5720: Fixed the alidation of the DID configuration form to prevent the submission of an inconsistent configuration (e.g. exact selection without specifying the selection), which generates an exception (500 error) during saving
  - 5702: Fixed a bug introduced in version 4.5.8 where the tenant UUID of outbound lines was not updated when restoring a single-tenant backup, which caused the lines not to be displayed in the rule editing panel
  - 5599: Fixed a bug that, when mass erasing extensions, caused the corresponding voicemail boxes not to be erased
  - 5585: Fixed an error that, when trying to delete an extension that cannot be deleted (e.g. because it is currently part of a group), allowed the change to be applied anyway, generating an inconsistency in the resulting configuration and the generation of an exception when trying to edit or delete the extension
- High availability
  - 5642: Fixed a bug where the pairing between two nodes failed in cases in which firmware was installed from the bootloader more than once on one or both nodes

- API REST
  - 5583: Extended the validation of the /rest/phoneServices/c2c/ API to accept the characters * and # in the "destination" parameter
  - 5802: Added an "id" attribute to the information returned by the /rest/extension/extensionTemplate and /rest/extension/extensionTemplate/{templateName} GET APIs, which is required when creating an extension via /rest/extension POST API
  - 5773: Fixed the /rest/cdr API for restoring the management of JSON format to filter the results
  - 5609: Restored the correct functioning of the CallCenter report generation API, which starting from version 4.5.6 returned a 5 byte file
  - 5688: Extended error handling for cases in which a user with an existing username was created to return, alongside a 400 error, an explicit readable message instead of the SQL error output
- CDR
  - 5644: Fixed the calculation of the billing time in the PBX CDR, which sometimes displayed a negative value (e.g. -0.121) when the call failed immediately
- CTI server
  - 5529: Fixed an issue where waiting time of the oldest call in a queue was incorrectly displayed when the CTI server was temporarily disconnected
  - 5800: Fixed a bug where the CTI service was rebooted after the removal of a tenant with connected clients
  - 5660: Fixed the handling of call transfers wen remote controlling Yealink phones with firmware versions v80 or later, which requires the DTMF tone # to be sent in a separate invocation than the other digits (0-9, *)
- Multi-tenant
  - 5586: Fixed a bug in the PBX backup restoration procedure where personal audio files of the tenants could not be played until the machine was rebooted
  - 5793: Fixed an issue where the caller was not recognized as a remote extension for inbound calls to a tenant when it originated as a failover action on another tenant in the group
  - 5154: Fixed a bug where, after moving an assigned line from one teant to another, caused the DIDs configured for the original tenant to be executed
  - 5648: Extended the validation of the domain name when creating or editing tenants to prevent the use of accented characters, which cause the creation of the corresponding LDAP tree to fail
- Phonebook
  - 5580: Fixed the functioning of the filters on the "email", "organization", and "department" fields in the extensions phonebook

Firmware 4.5.7 (13/02/2018)

New features

- Penalties for queue operators. Starting from this release, penalties can be assigned to members of a queue. This parameter allows calls to be presented to an operator only if all other operators with fewer penalties are busy. A REST API for configuring penalties has been implemented.

- "In progress" messages on IVR menus. This release introduces the option to play "in progress" audio files (before the call is actually answered). The maximum duration of these messages is usually shorter than one minute and depends on the phone line provider.
- Generation of change of service state events. These events can be linked to notifications to receive alerts whenever individual services malfunction/reset.
- Kalliope-Lift module. This module, available through a dedicated license, allows management of alert messages from ADEMCO dialers.
- New built-in provisioning devices. The following devices have been added:
  - AudioCodes 405HD,420HD,430HD,440HD,445HD,450HD
  - Snom D712,D785,D120
  - Gigaset Maxwell 2
  - Yealink SIP-T52S,SIP-T54S,SIP-T56A,SIP-T58A,SIP-T58V

Changes

- Telephone services
  - 0005487: Added a uniqueness check for time check names
- API REST
  - 0005347: Extended the REST API for changing queue operator pause status to operate on all queues
- Call Center module
  - 0005417: Added the boundaries of the custom range to the Call Center CDR

Bugfixes

- General
  - 0005344: Fixed a bug that prevented complete backups from being restored on multi-tenant systems
  - 0005362: Fixed a bug with setting event severity
- Telephone services
  - 0005493: Fixed a bug with enabling and disabling switches from the numbering plan
  - 0005484: Fixed a bug that prevented a paging group from being created
  - 0005435: Fixed a bug that under certain conditions caused a misalignment between the effective pause status of an operator and the one displayed
  - 0005382: Fixed a bug that prevented MeetMe rooms with no audio file from being edited
  - 0005483: Fixed a bug that prevented access to the Whitelist page
  - 0005445: Fixed a bug that prevented click-to-call from working once the busy level was reached
  - 0005391: Fixed a malfunction in Time Check
- Phonebook
  - 0005333: Fixed a bug that prevented speed dials from being added in the phonebook
- Call Center Module

  - 0005428: Fixed a bug where every "Apply" reset the queue statistics (and therefore caused the ring strategies to malfunction)
  - 0005531: Fixed a bug that sometimes caused the incorrect visualization of older calls in the supervisor panel
- GUI
  - 0005342: User dashboard: fixed the links to groups and queues
- Multitenant
  - 0005469: Fixed an error with multi-tenant license activation when gateways with no assigned lines were present
  - 0005412: Fixed an error that prevented notifications for non-default tenants from being sent

Firmware 4.5.6 (12/12/2017)

New features

- Event notification service. This release introduces an engine for managing events, which can be generated both on a node/system level and on a single tenant level (telephone events). The first events to be introduced concern the registration of the outcome of calls in queues (pbx.queue.servedcall and pbx.queue.unservedcall events). A configuration page for the event notification service is also available; it lets users associate to each event (or event class) one or more notification actions (either sending an email or invoking a web service), which are completely configurable with the attributes of the corresponding event.
- "Do Not Disturb" (DND) service. Adds the DND service on an extension level. The DND service implemented on a PBX level differs from the one available on telephone terminals in that it operates on the entire extension and not for the single SIP account configured on the terminal. The DND service operates on calls made directly to the extension as well as those that are presented to the extension as member of a ring group (but not for queues) and ensures that the extension is treated as "unavailable" when choosing a failover action (for direct calls). The DND state can be changed via BLF key (dnd<extension> selection), with which it is also possible to view the state of activation, or via the /rest/operation/service/dnd/<extension> REST API.
- REST APIs for the operation of extension services. Added /rest/operation/service/<service>/<extension> REST APIs in GET/POST and DELETE modes to read, set, and reset the state of the following extensions services: DND (Do Not Disturb), FORKMOBILE (Fork to Mobile), CFIM (Unconditional forward or call redirection). The APIs can be used with API access permissions and authentication by each user (for single extensions) and by users with a Power User role that is enabled to manage the "operating state of the services" (for all extensions).
- REST APIs for the dynamic management of queue operators. Added /rest/operation/queue/<operation> REST APIs to manage pause and add/remove dynamic operators to/from a queue.
- The "Service" and "Queue" widgets have been extended; it is now possible to change the state of the "Unconditional Forward" and "Fork to Mobile" services and change the pause status of one's SIP accounts on each queue with a handy switch.

Changes

- General
  - 5200: Changed the name of the “Diagnostic” menu in “Monitoring” following the addition of the notification management panel
  - 5196: Changed the “None” string in “Select account” in the provisioning device definition and edit panel
  - 5308: Changed the SIP error message returned when CAC limits are reached for an inbound call to the PBX or a tenant from “403 Forbidden” to “486 Busy Here” (note: the message for inbound or outbound calls remains unchanged)
- Call Center Module
  - 5301: Added the option to exclude operator events when exporting the Call Center CDR in detailed format
- Multi-tenant
  - 5276: Optimized execution time when deleting a tenant
  - 5185: Added check to prevent a tenant from using a calling number (for outbound calls) that does not belong to those defined in the assigned line used. If necessary, the calling number will be automatically changed based on the type of numbering present on the assigned line, following this rule:
    * exact selection: sets a specific number
    * range selection: sets the lowest number in the range
    * prefix selection: sets the root of the prefix

Bugfixes

- General
  - 5173: Fixed a bug that caused the generation of a 500 error when deleting the ACL item of a switch
- Phonebook
  - 5305: Fixed a bug that caused periodic importing of remote phonebooks to be suspended after a firmware update until its settings were saved again
- Modulo Call Center
  - 5228: Fixed a bug that caused calls not to be inserted in the CDR upon certain failure outcomes (FULL, JOINEMPTY, or LEAVEEMPTY)
- High availability
  - 5178: Fixed an issue with the synchronization of 4SP licenses and corresponding child licenses

Firmware 4.5.5 (25/10/2017)

New features

- General
  - Added two new widgets to the dashboard to view calls in progress in real time and with a graph, with customizable classification (direction and filter) and time span, which shows statistics on numbers of calls extracted from the CDR
  - Added a "Closed Group" service, which offers the option to restrict the ability to call specific extensions to a list of enabled extensions

Changes

- General
  - 5147: Restored the ability to use the "apostrophe" character in the First Name and Last Name fields for extensions and made uniform the corresponding validation in mass import
  - 5099: Introduced ordering by type and name in the outbound line selection form
  - 5091: Introduced alphabetic order in the tenant selection form
  - 4465: The current firmware version can now be viewed within the "System status" widget in the dashboard
  - 5015: Changed the uniqueness check for SIP template names to be case-insensitive
  - 5083: Added instantaneous duration display in the active call visualization panel
  - 5126: Added a visual indicator of the execution of the CC report generation request
  - 4895: Changed the way call refusal is handled for non-authenticated calls originating from hosts not defined among the configured VoIP domains and gateways from dialplan level to SIP level
- Telephone services
  - 5156: Changed the handling of the 480 response sent from telephones when DND service is enabled to execute the forward action for "not available" rather than "busy"
- Provisioning
  - 5149: Extended MAC recognition from the URL of a request to use a regular expression and not only built-in filename formats
  - 5132: Added the option to configure a custom path on redirection servers
  - 5121: Added management of the "public" subfolder of its provisioning path with anonymous access (not authenticated) even when configuring provisioning access via client authentication through certificate
- Multitenant
  - 5003: Changed the lock management so that the pbxadmin acquiring the lock does not automatically cause all admins of all tenants to lose the lock; it now only does so when changes are applied, and only for tenants involved in the changes made by the pbxadmin
- REST API
  - 5012: Added HTTP caching support through "ETag/If-None-Match" and "If-Modified-Since" headers

Bugfixes

- General
  - 5165: Fixed a bug that caused the incorrect presentation of the calling number for attended transfers of outbound calls made by an extension
  - 5158: Restored the functionality of external API invocation in POST mode or the Basic type authentication
  - 5017: Fixed the handling of the "0" switch
  - 5098: Fixed a bug that prevented the correct handling of more than 128 tenants on a single node
  - 4548: Removed hot desking device accounts from the number of accounts that count towards the license limit
  - 5063: Fixed a bug that caused the addition of the outbound prefix to the calling number of inbound calls to persist, even eliminating it from the PBX or tenant settings
  - 5065: Fixed the display of custom logo (with a rebranding license) when accessing the web GUI through HTTPS protocol
  - 5094: Made uniform the Mime-Type validation when uploading audio files and hold music, which previously caused certain files to erroneously be refused
  - 5096: Fixed a bug that generated an exception when accessing the wizard while an extension with no linked SIP account was present
  - 4992: Fixed a bug that caused the theme selected by rebranding license holders not to be maintained
  - 4993: Removed the audit log from the backup
  - 4984: Fixed the functionality of the "Enabled" filter and all filters operating by selection that executed a substring-type match instead of exact
  - 4874: Fixed a bug that caused manual time synchronization via NTP to fail when the current date was set later than the effective one
- CTI server and applications
  - 5050: Fixed the handling of SIP accounts that contain the "-" character in their username
  - 5168: Fixed the remote control of telephones via KalliopeCTI Pro for hot desking terminals
  - 5175: Fixed a bug that caused attended transfers from KalliopePhone to occasionally fail for extensions with more than one linked SIP account
- REST API
  - 5021: Fixed a bug that prevented backup restoration APIs from functioning in singe-tenant scenarios
  - 5020: Fixed filename validation for backups uploaded via API to require the extension ".bak"
  - 4977: Made the output of CDR download APIs match the one obtained when exporting from the GUI
- CDR
  - 3953: Fixed registration of the reason for a call to exit a queue in case of CCBS
- Call Center Module
  - 5152: Fixed filtering for tenants when generating Call Center CDR reports in multi-tenant scenarios
  - 5089: Fixed the counting of calls served by single operators in the CC report for calls picked up or served by dynamic operators
  - 5118: Fixed a bug that generated an exception when filtering by operator in the Call Center CDR

  - 5088: Fixed the calculation of average conversation time in the CC report
  - 5087: Differentiated the outcome of calls in the queue during closing time, which were previously marked as "Not served"
  - 5085: Fixed a bug that caused the outcome to be set to TIMEOUT instead of ANSWERED_ELSEWHERE in the operator detail after call pickup
  - 5084: Fixed the display of the extension in the "Operator extension" column, which previously showed the account name
- Call recording
  - 5086: Fixed a bug in version 4.5.4 that prevented the normal saving of call recording files
- Provisioning
  - 5140: Fixed the generation of the TFTP and HTTP/S provisioning path when importing a tenant
  - 4702: Changed the mechanism for sending check-sync NOTIFY messages to supported terminals (to force the download of provisioning files), which under certain deployment conditions were sometimes not sent
  - 4626: Fixed a bug that caused check-sync NOTIFY messages to supported terminals to sometimes not be sent to a terminal when the linked account had been edited
  - 5067: Fixed a bug that caused the failed functioning of provisioning file servicing through HTTP and HTTPS when installing firmware version 4.5.4 directly
- Multi-tenant
  - 5028: Fixed "remote extension" type origin recognition inbound to a tenant in case of automatic sharing of extensions among tenants belonging to the same tenant group
  - 4785: Fixed a bug that caused the order of the inbound manipulation rules on the assigned lines of a tenant not to be saved
  - 5145: Fixed a bug that prevented the restoration of audio files in the backup when importing a tenant from a backup

Firmware 4.5.4 (29/08/2017)

Integrates all changes included in version 4.5.3, which was not released to the public.

New features

- General
  - Added a "Diagnostics" > "Active calls" panel with real-time active call display to the PBX, from which each call can be individually terminated
  - Replaced the External APIs application with its Dynamic routing extension, which allows calls to be managed both by invoking an external web service (such as the original application) and by matching the parameters on a XLS/CSV file uploaded to the PBX
  - Added to the "Dynamic Routing" service the option to forward a call to the selection of the numbering plan returned by a web service or retrieved from a local file
- API REST
  - Added a /rest/tenantGroup/{tenantGroupName}/extension API to obtain the list of all extensions defined on the tenants in a tenant group

- Added a /rest/extension/{exten}/services API to obtain the activation state of extension services

Bugfixes

- General
  - 4431: Fixed a bug that made it impossible to halt playback of an audio file of a paging group (in unattended mode with infinite repetitions) if the configuration of the PBX was edited between the starting call and the halt request
  - 4800: Restored the ability to define custom selections with a value that coincides with that of an extension within the tenant
  - 4826: Fixed an issue that caused an irreversible error when viewing the User management page when the number of defined users was greater than 1000
  - 4879: Fixed validation of forms in which forwards to an external number can be configured, as edits to the destination number that consisted of adding or removing the prefix 0 were not saved
  - 4882: Fixed a bug that prevented the deletion of ACL rules in paging groups
  - 4884: Fixed a bug that caused an error to be generated when trying to delete ACL rules in paging groups
  - 4899: Fixed a bug that prevented the details of several entities from being displayed even when the role of the user included read permissions
  - 4129: Fixed the handling of visibility and permissions for Power Users (who can now assign to new users only the base tenant user role or their own)
  - 4937: Fixed Call Admission Control assessment, which previously prevented calls between extensions when only one call was available (and the exclusion of inter-office calls from the count)
- CDR
  - 4846: Changed the CDR filtering logic to prevent the GUI from locking due to running out of system memory when the number of calls per month in the CDR exceeds a few tens of thousands
  - 4862: The string “xxx” is no longer displayed as the Caller name in the anonymized CDR
- KalliopeCTI
  - 4773: Fixed an issue where the outcome of the call was not revealed to the caller when using click-to-call from KalliopeCTI Free (or Pro without remote control) to a busy external number
- Provisioning
  - 4876: Fixed a bug that prevented the provisioning file service through HTTP and HTTPS when version 4.3.9 or later was directly installed (not present when updating from previous versions)
  - 4854: Fixed the handling of edit permissions for provisioning entities (device, template) so that power users can edit those created by the admin and vice versa
- HA
  - 4781: Fixed a bug that allowed PBX firmware updates even when the HA service was active
  - 4782: Added synchronization of the PBX provisioning folder (in multi-tenant scenarios) and the provisioning request record
- Multitenant
  - 4698: Fixed an issue with the duplication check of the account linked to a device when provisioning was performed by the pbxadmin

- 4887: Fixed a bug that caused the paging service in unattended mode to fail to function for tenants other than the default

Firmware 4.5.3 (11/07/2017)

Note: Internal release not available to the public.

New features

- General
  - Added the option to assign the execution of the first configuration wizard to Power User roles
  - Extended ACL functionality for SIP accounts, differentiating them by source IP and Contact and allowing the configuration of more than one subnet per each
- Provisioning
  - Added Patton Smartnode SN4522/24/26/28 JS (multi-port ATA) and Gigaset Pro N720 multicell DECT system to built-in devices
- API REST
  - Added APIs for editing service codes in the numbering plan
- Multitenant
  - Added an option on tenant groups to automatically share custom and extension selections in the numbering plan of all tenants in the group (without requiring remote numbering ranges assigned to single tenants to be explicitly defined); introduced an inter-tenant duplication check for the selections in question to prevent the presence of the same selection on two different tenants in the same group

Bugfixes

- General
  - 4719: Fixed an issue where permissions assigned to a previously defined Power User role could not be edited
  - 4789: Fixed a bug that prevented playback of hold music for MeetMe rooms
  - 4778: Fixed a bug that under certain conditions caused an error when saving remote extensions in the configuration of a trunk
- CDR
  - 4707: Fixed the registration of calling and called numbers for call transfers performed through the SIP REFER method
  - 4704: Fixed the registration of the outcome of a call forwarded to an outbound line in case of failure with CONGESTION as the cause
  - 4703: Fixed the registration of the outcome of a direct call to an extension that was redirected to a service
- Provisioning
  - 4736: The IPUI column will now be displayed during the validation phase when mass importing provisioning devices

- Multitenant
  - 4747: Fixed a bug that prevented KalliopeCTI from remote controlling supported phones for tenants other than the default

Firmware 4.5.2 (29/06/2017)

New features

- General
  - Added the option to create custom "Language Packs", replacing the integrated audio files of a certain language with audio files uploaded by the user
- Provisioning
  - Added function key file generation for Avaya terminals
  - Added placeholders for time and date
- API REST
  - Added APIs for managing roles and extension templates

Bugfixes

- General
  - 4666: Fixed an error that caused the removal of the privacy permissions of a user following edits to that user by the admin
  - 4722: Fixed an issue where the bypass flag of the B/S filter was not respected when a call from a secretary to a boss failed
  - 4729: Fixed an error that caused the audio message for the predefined failover action during the defined time span for a time check to play when a specific failover action for a certain span was defined with no associated audio message
- CTI
  - 4690: Fixed a criticality that under certain conditions caused the CTI server to restart when sending a message to iOS clients
  - 4708: Fixed a malfunction of the mobile phone and SIP icons on the KalliopeCTI Mobile client
  - 4709: Restored the functionality of the click-to-call command towards mobile from the KalliopeCTI Mobile app
- CDR
  - 4658: The name of the called extension is now displayed in the call detail section
- API REST
  - 4682: Fixed a backward compatibility issue with tenant creation APIs
  - 4683: Updated the documentation of the provisioning device creation REST API to reflect the fact that the "priority" parameter for multi-account device management introduced in version 4.5.1 is required
- Provisioning

- 4698: Fixed the duplication check for accounts linked to a device when provisioned by the pbxadmin (in multi-tenant systems)
- 4672: Adapted the model of the XLS file for mass importing provisioning devices to include the IPUI column
- 4674: Fixed an inconsistency that prevented the functioning of provisioning though HTTP/HTTPS for PBXs on which firmware version 4.3.9 or later was directly installed (this bug was not present for PBXs that updated from firmware version 4.3.8 or earlier)

Firmware 4.5.1 (11/06/2017)

New features

- General
  - Added support for licenses for rebranding the web interface
- Queues
  - Added "in conversation" to the reasons for operator unavailability when determining the immediate failover action when placing a new call in a queue
- Provisioning
  - Added multi-account device management (e.g. DECT, IP Channelbank, M-ATA systems)
  - Added function key (BLF) configuration panel for each extension and provisioning for each on all accounts linked to the SIP accounts of the extension
  - Added SNOM D745 to the list of integrated provisioning devices
  - Added the attribute "number of function keys" to provisioning device models
- Hotdesking
  - Added the ability to log into a hot desking terminal on which another extension is already logged in without having to log out first (implicit logout)
  - Hot desking accounts are now displayed in the SIP account list

Bugfixes

- General
  - 4515: Fixed the way assigning a user to an extension after its creation is handled, which previously caused the personal phonebook not to be displayed
  - 4315: Fixed a bug that caused outbound calls to be refused when one of the lines associated to the trunk or termination domain were configured with a concurrency limit of 0 (unlimited)
  - 4628: Fixed the handling of SIP account names that contain the "-" character
  - 4600: Fixed a bug that under certain conditions caused a 500 error when viewing a SIP account template
  - 4601: Fixed an issue with the validation of trunk edit forms that made it impossible to define remote extensions
  - 4569: Fixed an issue with mass importing extensions that caused voicemail box creation to fail
  - 4552: Fixed an issue with the handling of B/S filters for call transfers to a boss on the part of an entity of a different group where the active filter was erroneously bypassed

  - 4527: Fixed an error that prevented outbound calls from being forwarded to the backup line when the max concurrency number of the main line of a routing rule was reached
  - 4528: Fixed a bug that caused the incorrect generation of the links within the hot desking device page
  - 4511: Fixed a slowdown during the "apply" operation after editing a configuration when there was a high number of call recordings contained in the network folders
  - 4501: Fixed a 500 error when viewing the event record
  - 4483: Fixed the rotation of the provisioning request record
  - 4513: Fixed the validation of the external API form, which prevented placeholders from being used in the GETs
  - 4519: Fixed an issue where creating a backup with the same name as an existing one caused the old one to be overwritten and therefore lost
  - 4510: Fixed a bug that prevented the functioning of call limits set on a trunk for inbound calls
  - 4509: Fixed a bug that under certain conditions caused the incorrect generation of inbound routing rules to domains and gateways (DID)
  - 4437: Fixed the handling of edits to remote extension configurations, which previously caused a 500 error
  - 4445: Fixed the handling of audio files with names that contain spaces
  - 4439: Fixed the validation of failover action forms in cases of failed selection of destination entity
  - 4447: Fixed a bug that prevented switches from being deleted or disabled from the time check configuration

- Queues
  - 4490: Fixed a typo that prevented the correct functioning of the "fewestcalls" ring strategy
- CTI Server
  - Several optimizations and minor fixes to improve performance and interoperability with previous versions of the desktop and mobile clients
- Multitenant
  - 4563: Fixed a bug that caused concurrency limits not to be respected for outbound calls from an assigned line of a tenant
  - 4531: Fixed the counting of outbound calls from tenants that were previously counted twice when determining CAC admissibility
- 4SP Module
  - 4479: Fixed the handling of 4SP license activation on PBXs with a previous expired multi-tenant license
- Hot desking
  - 4624: Added a way to handle cases where the login state of an extension and the provisioning state of the terminal are misaligned
  - 4573: Fixed a bug that caused a 500 error when accessing a hot desking panel when no SIP account was available
  - 4544: Fixed a bug that caused the failed deletion of hot desking accounts when deleting an extension with hot desking enabled
  - 4523: Fixed a bug that prevented the regeneration of provisioning files for hot desking accounts
  - 4486: Fixed the display of the available account limit in the hot desking device edit page
- API REST

  - 4574: Added a way to handle cases where a user finds the configuration database locked while invoking APIs
  - 4558: Fixed an issue with restarting the Jabber server that prevented the creation of new tenants via APIs
- Provisioning
  - 4567: Restored the functionality of provisioning template reassignment to a set of devices
  - 4557: Fixed a bug that made it impossible in multi-tenant systems to access the edit panel of a provisioning device created by the pbxadmin if the linked SIP account was deleted by the tenant
  - 4539: Fixed check-sync NOTIFY message sending in multi-tenant systems
  - 4458: Fixed a bug that prevented accounts from being assigned when creating a provisioning device
- KalliopePBX v3 backup importer
  - 4536: Fixed importing of personal contacts in the phonebook, which were previously inserted into the shared phonebook
  - 4537: Fixed extension configuration import to handle timeouts for each extension
  - 4538: Fixed a bug that caused the duplication of entries in imported personal phonebooks

Firmware 4.5.0 (22/05/2017)

New features

- General
  - Added a PBX operating mode that allows certain types of calls to be enabled or disabled. Three operating modes are currently available: full, block outbound calls except for whitelisted numbers, disabled (no outbound calls allowed). In multi-tenant systems this can be set for each tenant.
  - Added a Call Admission Control function that allows the maximum number of calls that can be made from a branch to be set.
  - Added an option to force the failover action for a queue when all operators are busy.
- Provisioning
  - When editing a device/account link, the check-sync NOTIFY SIP message is automatically sent to force the configuration to reload.

Bugfixes

- General
  - 4447: Fixed a bug that prevented switches from being deleted or disabled within a time check
  - 4453: Added validation of forms with failover actions towards unselected entities (e.g. groups or queues)
  - 4454: Added an indication when audio files containing spaces fail to save
  - 4485: Fixed the display of account limits in the Applications -> Hot Desking panel
- CDR
  - 4440: Fixed a bug that prevented the correct functioning of the CDR REST APIs
  - 4401: Fixed a bug that caused only calls displayed on the GUI to be exported

    - 4443: Fixed a bug that caused transferred calls not to be included in reports generated by the Call Center CDR

1.1.13 Firmware series 4.4.x

Firmware updates from the 4.4.x series are LTS Maintenance Releases, i.e. stable versions with long term support. The releases in this series have been thoroughly tested before being released to the public and therefore guarantee greater stability.

Firmware 4.4.2 (12/09/2017)

Bugfixes

- General
    - Fixed a bug that prevented the Enabled/Disabled filter in several lists (e.g. the MeetMe room list) from functioning
    - Fixed a bug that caused outbound calls to be refused when one of the lines associated to the trunk or termination domain was configured with a concurrency limit of 0 (unlimited)
    - Fixed an issue that caused an irreversible error when viewing the User management page when the number of defined users was greater than 1000
    - Fixed the handling of SIP accounts that contain the "-" character in their username
    - Fixed the validation of forms in which forwards to an external number can be configured, as edits to the destination number that consisted of adding or removing the prefix 0 were not saved
    - Fixed an error that caused the removal of the privacy permissions of a user following edits to that user by the admin
    - Fixed an error that caused the audio message for the predefined failover action during the defined time span for a time check to play when the failover action for that interval was defined with no associated audio message
    - Fixed an issue where the outcome of the call was not revealed to the caller when using click-to-call from KalliopeCTI Free (or Pro without remote control) to a busy external number
    - Fixed a bug that under certain conditions caused an error when saving remote extension in the configuration of a trunk
    - Fixed a bug that caused manual time synchronization via NTP to fail when the current date was later than the effective one
    - Fixed a bug that prevented the deletion of ACL rules in paging groups
    - Fixed a bug that caused an error to be generated when trying to delete ACL rules in paging groups
    - Fixed a bug that prevented the details of several entities from being displayed even when the role of the user included read permissions
- Telephone services
    - Fixed an anomaly in the paging service with infinite repetitions of a prerecorded audio file
    - Fixed a bug that prevented playback of hold music for MeetMe rooms
- Hot desking

  - Added a way to handle cases where the login state of an extension and the provisioning state of the terminal are misaligned

- CTI
  - Fixed a criticality that under certain conditions caused the CTI server to restart when sending a message to iOS clients

- CDR
  - The string "xxx" is no longer displayed as the Caller name in the anonymized CDR
  - The name of the called extension is now displayed in the call detail section
  - Changed the CDR filtering logic to prevent the GUI from locking due to running out of system memory when the number of calls per month in the CDR is exceeds a few tens of thousands

- HA
  - Fixed a bug that allowed PBX firmware updates even when the HA service was active
  - Added synchronization of the PBX provisioning folder (in multi-tenant scenarios) and the provisioning request record

- Provisioning
  - Fixed an inconsistency that prevented the functioning of provisioning though HTTP/HTTPS for PBXs on which firmware version 4.3.9 or later was directly installed (this bug was not present for PBXs that updated from firmware version 4.3.8 or earlier)
  - Fixed the handling of edit permissions for provisioning entities (device, template) so that power users can edit those created by the admin and vice versa

- Multitenant
  - Added the option to order inbound mapping rules on assigned lines
  - Fixed a bug that prevented KalliopeCTI from remote controlling supported phones for tenants other than the default
  - Fixed a bug that caused the paging service in unattended mode to fail to function for tenants other than the default
  - Fixed a bug that caused accounts in the paging group list to be incorrectly displayed for tenants other than the default

- API REST
  - Fixed exportation from REST API of the detailed CDR in cvs format

Firmware 4.4.1 (29/05/2017)

New features

- General
  - Added an option to force the failover action for a queue when all operators are busy
  - Added White Label license management

- Provisioning
  - Added built-in device Snom D745

Bugfixes

- General
  - 4446: Fixed a bug that prevented switches from being deleted or disabled within a time check
  - 4512: Fixed a bug with validating external API URL forms
  - 4561: Fixed a malfunction in voicemail box creation when mass importing extensions
  - 4402: Added validation of failover actions towards entity parameters in the IVR menu form
  - 4433: Added an indication when audio files containing spaces fail to save
  - 4448: Fixed a bug in routing management to/from remote extensions
  - 4429: Fixed a bug in the fewestcall ring strategy within a queue
  - 4263: Fixed a malfunction that caused an existing backup to be overwritten when a backup with the same name was created
  - 4407: Fixed a slowdown when reloading VoIP services when too many call recordings are present
  - 4559: Fixed a bug that caused hot desking accounts not to be deleted when deleting an extension with hot desking enabled
  - 4484: Fixed the display of account limits in the Applications -> Hot Desking panel
  - 4492: Fixed a bug with generating links in the Applications -> Hot Desking panel
  - 4581: Fixed a malfunction in the Applications -> Hot Desking panel when the available account limit was reached
  - 4554: Fixed a malfunction of the Boss-Secretary service when transferring a call between secretaries belonging to different groups
  - 4478: Fixed the handling of 4SP license activation on PBXs with an expired multi-tenant license
- CDR
  - 4440: Fixed a bug that prevented the correct functioning of CDR REST APIs
  - 4401: Fixed a bug that caused only calls displayed on the GUI to be exported
  - 4403: Fixed a bug that caused transferred calls not to be included in reports generated by the Call Center CDR
- Provisioning
  - 4562: Fixed a bug with assigning templates to devices after filtering by model
- Multitenant
  - 4499: Fixed a malfunction of LCR rule failover actions in case of failure due to reaching call limit
  - 4520: Fixed a bug that prevented pbxadmin from editing a provisioning device after deleting the associated account
  - 4543: Fixed an issue with restarting the Jabber server that sometimes prevented the creation of new tenants
- KPBXv3 backup importer
  - 4141: Fixed a bug that caused personal contacts to be inserted in the shared phonebook
  - 4140: Fixed a bug that duplicated user phonebooks
  - 4039: Fixed a bug with importing failover timeouts for extensions

Firmware 4.4.0 (04/05/2017)

New features

- General
  - Replaced the G.729 codec module with a recompiled GPL version
  - Added PBX registration on Tribe and association to partners
  - Added alphanumeric Request-URI management for inbound calls
  - Added image files with the Kalliope logo for use in telephone displays
- REST API
  - Implemented handling of POST requests with data in JSON format
  - Added a REST API for listing tenants
  - Added REST APIs for listing, creating, editing, and deleting tenant groups
- Provisioning
  - Added new devices (Yealink T46S, Gigaset Pro Maxwell Basic/3/10)
  - Added a way to handle built-in device/brand names that conflict with the custom ones set by the user
- CDR - The account_code of call_details in CDR and operator_exten are now displayed in the Call Center CDR

Bugfixes

- General
  - 4008: Fixed an anomaly in direct media functionality
  - 4245: Changed the validation of first name/last name fields for extensions to prevent the use of the characters ; ” ‘
  - 4128: Changed user management by Power Users to prevent the creation of users with roles other than their own or tenant user
  - 4271: Fixed a timeout bug when restoring single-tenant backups with a high number of extensions
  - 4369: Fixed a bug in the simultaneous connection of a high number of XMPP clients
  - 4351: Fixed a bug in the Boss-Secretary form validation when there are duplicate secretaries
  - 4284: Fixed an inbound routing bug for remote extensions originating from external lines
- KalliopeCTI
  - 4132: Fixed a bug with changing the web GUI login password that prevented it from being updated on KCTI clients until the service was restarted
  - 4240: Fixed an encoding problem that caused certain characters in notifications from KCTI to KCTI Mobile to be incorrectly displayed
  - 4100: Fixed a malfunction that caused the failed sending of notifications to KCTI Mobile that disconnected immediately after connection
- Multitenant
  - 4122: Fixed an anomaly that sometimes caused the deletion of tenants when new tenant creation failed

- 4278: Fixed an anomaly that caused the GUI to time out when creating/restoring backups of large size (in terms of number of tenants)
- 4180: Fixed an anomaly in the handling of outbound calls to remote extensions among tenants belonging to the same tenant group
- 4312: Fixed a bug that prevented the movement of remote extensions among tenants belonging to the same tenants group
- 4352: Fixed a bug that caused importing of tenant backups with spaces in their names to fail

- Hot Desking
  - 4144: Fixed a bug that prevented the functioning of the hot desking service
  - 4136: Fixed a bug that prevented the creation of provisioning files for hot desking accounts
  - 4347: Fixed a bug where the confirmation audio file failed to play upon logout
- IVR
  - 4157: Fixed the editing panel to prevent audio filed from being selected when selection is disabled
  - 3839: Changed the default viewing mode from tree to list
- REST API
  - 4199: Fixed the display of documentation on api/doc URL
  - 3791: Changed the /rest/extension REST API to return not only the list but also all extension attributes
- Phonebook
  - 4138: Fixed an issue with exporting a phonebook with fields that contain malformed values in xlsx format
- CDR
  - 3998: Fixed an issue with viewing entries where the calling or called number includes the characters , ; " '
  - 4233: Fixed a bug with viewing a call that was picked up from a queue (also on Call Center CDR)
  - 4195: Fixed a bug with viewing calls to a queue forwarded to an extension
  - 4248: Fixed a bug with the value of the account_code in the call_details
  - 4224: Fixed a bug with the value of the source number in the call_details
  - 4254: Fixed a bug with the value of the outcome of unanswered calls to remote extensions in the call_details
  - 4255: Fixed a bug with anonymizing the source number in the call_details for calls from remote extensions
  - 4212: Fixed an issue with viewing the CDR from the panel of a user with a role other than tenant user
- Call Center
  - 4252: Changed Call Center CDR data filtering to use the timestamp for the end of the call and not the beginning
  - 4257: Fixed a bug that sometimes caused the time of the end of a call to be viewed as null in the Call Center CDR operator events (even for answered calls)
  - 4214: Fixed a bug that sometimes caused invalid accounts in the Call Center CDR operator events to be displayed
  - 4190: Fixed an issue with the values of call duration in the Call Center CDR
- Provisioning

  - 4171: Fixed REST API functionality for editing templates to handle the automatic regeneration of configuration files
  - 4143: Fixed the handling of provisioning template owners and device/template association

1.1.14 Firmware series 4.3.x (TR)

Firmware updates from the 4.3.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.2.1, is the stable release recommended for generic use that does not require features introduced in the TR.

> **Warning:** Two bugs have been found in versions 4.3.8 and 4.3.9 that concern call recording (issues #4058 and #4078) and could prevent the regular functioning of this service. Specifically, bug #4078 prevents regular functionality at the beginning of each month. For this reason, if you have updated to one of these versions, we recommend updating to version 4.3.10 as soon as possible. If you have any doubts, please contact technical support through the usual channels.

Firmware 4.3.10 (20/03/2017)

New features

- Provisioning
  - Added support for TENANT_UUID, TENANT_DOMAIN, and TENANT_NAME placeholders
  - Extended MAC address management to cover cases where a device is inserted by both the pbxadmin and the tenant admin
- REST API
- Added an API to allow editing of tenant configuration settings

Bugfixes

- General
  - 4037: Fixed the display of the users panel for the privacyadmin
  - 4038: Fixed a bug that appeared when trying to enable privacy mode on a user that already had it enabled
  - 4080: Fixed a bug that under certain conditions appeared when restoring a backup
  - 4001: Fixed the handling of attended calls to a nonexistent selection, which were previously presented to the transferer
  - 4052: Fixed an issue that caused calls to fail when made to an extension with fork2mobile enabled but whose class does not permit calls to the configured mobile number
  - 4049: Fixed the way closing of voicemail box consultation services through the # key is handled
  - 4050: Fixed the handling of attempts to define a remote LDAP phonebook already defined in the PBX, which caused a 500 error on the GUI
  - 4043: Fixed an anomaly in the GUI when the admin reassigned a conference room previously assigned to a user

- High availability
  - 3911: Fixed an issue where the cluster creation procedure sometimes failed when nodes were unable to reach the DNS servers
- CDR
  - 4017: Integrated FORK2MOBILE event management
  - 4079: Fixed issues with the display of CDR rows when filtering as privacyadmin
- Call recording
  - 4058: Fixed an issue introduced in version 4.3.8 with the generation of metadata files with unconditional call and unanswered call recording, which blocked recordings from being archived
  - 4078: Fixed a bug introduced in version 4.3.9 that prevented call recording at the beginning of the month
- KalliopeCTI
  - 4046: Identified and fixed a memory leak in the kctis server
  - 4086: Fixed the failed presentation of the inbound caller name to CTI clients
  - 4073: Fixed an issue where periodic queue and operator statistics were not updated
- Provisioning
  - 4075: Fixed the management of preconfigured redirection server properties
  - 4076: Fixed the management of redirection servers in multi-tenant systems
  - 4088: Fixed a bug in the provisioning file importer that caused the provisioning file encryption flag to be enabled when the corresponding column was left empty

Other changes

- 3962: Changed the header strings of certain columns in the CDR and Call Center CDR and made uniform the strings with detail outcomes

Firmware 4.3.9 (09/03/2017)

New features

- General
  - Added a SIP stack monitoring system with block/slowdown notification
  - Introduced the option to manage trunks with INVITE authentication even when registration is disabled
  - Added the option to create administration users, other than admin and pbxadmin, that are not linked to extensions
- Provisioning
  - Added the option to enable/disable provisioning and MAC recognition services
- REST API
  - Added a REST API for extracting the Call Center CDR (with details)
- Call Center
  - Added management for Call Center licenses with limits on the number of operator/supervisor users

- Added the option to generate reports after custom intervals of time in the Call Center CDR
- Added more details to the Call Center CDR to completely track operator behavior

Bugfixes

- General
  - 3889: Fixed an error with looking up callers on the user phonebook
  - 3969: Fixed an error with normalizing the international prefix for calls to Italy from abroad
  - 3906: Fixed a malfunction with rotating the web application logs
  - 3981: Fixed an issue with authenticating with Active Directory that allowed anonymous binding
  - 3996: Fixed an error with importing backups with configured authentication methods
  - 3997: Fixed an error with displaying the outcome of backup finalization
  - 3551: Fixed an issue with hold music audio for calls to external numbers
- Call Recording
  - 3888: Fixed the functionality of the flag for disabling recording between extensions
- GUI
  - 3891: Fixed the display of certain values inherited from templates in the extensions panel
  - 3976: Fixed the display of the TCP activation state in the account list
  - 3861: Fixed a display error in the gateway and VoIP domains on KPBXs with an expired VM license
- KCTIS
  - 3845: Fixed a crash caused by the deletion (or deactivation due to license expiration) of a tenant with connected CTI clients
- API REST
  - 3952: Fixed the CDR REST API to return the name of the caller instead of the calling number
  - 3971: Fixed the time filtering on the CDR REST API for requests that include different months
- Multitenant
  - 3882: Fixed a malfunction with the display of redirection servers owned by pbxadmin for tenant admins
  - 3956: Fixed a malfunction that caused voicemail emails to be sent with the default sender
  - 3971: Fixed a malfunction when authenticating XMPP users (chat on KCTI clients)
  - 3984: Fixed an issue with the paging service
  - 4004: Fixed a bug with restoring tenant backups
- HA
  - 3880: Fixed a malfunction with tftp provisioning with switches on secondary node
- Call Center
  - 3869: Fixed an error that prevented the creation of Call Center CDR reports from CTI clients

Firmware 4.3.8 (06/02/2017)

New features

- General
  - Added the ability to receive notifications for inbound calls (with the option to redirect to mobile) for extensions with no linked accounts by simulating ringing on a fake account
  - Added a mechanism for customizing the content of the "From:" SIP header as well as the optional "P-Asserted-Identity:", "P-Preferred Identity:", "Remote-Party-ID:", and "Call-Info:" ones for outbound calls from VoIP terminations and trunks
  - Added the option to start/shut down/restart services to the service state widget, possibly regenerating the corresponding configuration if necessary
  - Added a setting to the service state widget to start certain services (SIP PnP, SMTP, SNMP, TFTP) upon booting
  - Added management of product license codes associated with the annual extension of the PBX update license
  - Added a synchronization mechanism from the license server for previously activated licenses
- Call recording
  - Extended the call recording configuration panel to allow the saving of files with the recording metadata alongside the audio file and optionally enabling sending it via email to a set of configurable destination along with the audio file itself
- Minor changes
  - 3752: Added a maximum lifespan of one hour for call reservation requests, at the end of which the request is automatically canceled

Bugfixes

- General
  - 3751: Fixed the handling of CLIR from VoIP terminations

3833: Fixed the string that enables COLP sending and receiving on VoIP domains that continued to indicate the RPDI term

- 3799: Fixed the display of the user management panel for enabling "Privacy" for the privacyadmin user
- 3804: Fixed the validation of outbound line forms to prevent a concurrency that exceeds than the maximum PBX limit from being assigned
- 3606: Fixed the functioning of the redirect to mobile action requested by KalliopeCTI Mobile for inbound calls to a queue
- 3788: Fixed the ring timeout setting for calls forwarded to an external number by the failover action of an extension, which previously inherited the timeout value for the extension
- 3779: Fixed an error that caused product licenses to fail to activate from the guided configuration wizard

- Provisioning
  - 3781: Fixed an error introduced in version 4.3.7 that in single-tenant systems prevented provisioning file servicing via HTTP and HTTPS (only TFTP worked)
  - 3777: Fixed a javascript error that caused the page to lock when editing provisioning devices

  - 3768: Fixed an error with mass importing provisioning devices using a multi-tenant template file that includes the tenant_uuid column
  - 3557: Fixed the display of errors concerning the insertion of unknown brands or models in the provisioning device import file
- CTI
  - 3776: Fixed an issue where duplicate messages were sent to CTI Mobile client when logging in or out with multiple connected clients for the same user
  - 3741: Fixed an issue where duplicate messages were sent to CTI Mobile client when disconnecting from the client while downloading voicemail audio files
- Rest API
  - 3789: Fixed the CDR data filtering to use the timestamp for the end of the call (which coincides with the time the call is inserted in the CDR) instead of the beginning
  - 3727: Fixed the called number field returned by the CDR API in compat_v3 mode to display the value of the "answered_by" column and not the "called_num" column
- Multitenant
  - 3747: Extended the information on the tenant dashboard to display both the number of assigned lines and the maximum concurrency
  - 3744: Fixed the string in the display of the concurrency assigned to tenants from -1 to unlimited
  - 3743: Fixed the validation of the multi-tenant license activation wizard to make it possible to assign 0 external calls to the predefined tenant
  - 3746: Fixed the validation of the forms for assigning lines to tenants to prevent a concurrency per assigned line that exceeds the tenant concurrency from being assigned
  - 3767: Fixed the failed creation of LDAP users when creating a new tenant
  - 3555: Fixed the handling of attempts to create a second tenant with the same name as an existing one, which previously generated an exception

Firmware 4.3.7 (16/01/2017)

New features

- General
  - Added support for calling line identification restriction (CLIR), with several modes: unconditional for extensions to local and/or external destinations, on request through a service code, on request based on the privacy configuration of the calling terminal
  - Added a completion of calls to busy subscriber service
  - Added a flag to enable P-Asserted-Identity header sending for outbound calls, containing the calling number even when CLIR is set
- Provisioning
  - Added provisioning support for Escene devices
  - Added support for the AES encryption of generated provisioning files, with encryption keys that can be assigned to single terminals (feature available for Escene terminals)
  - Added redirection/provisioning server support for Yealink and Escene

- Multi-tenant
  - Added a "file manager" panel for the pbxadmin user

Changes

- General
  - 3386: Reduced call setup time by optimizing the dialplan
  - 3347: Extended the audit log to include system menu actions
  - 3668: Added protection from deletion/editing though file manager for files generated by the provisioning system
- Multitenant
  - 3681: Extended the logic for assigning lines to allow concurrency overprovisioning to the tenants
  - 3282: Added the first configuration wizard to single tenants
  - 3670: Added a destination tenant recognition service for provisioning requests based on the MAC address of the requester

Bugfixes

- General
  - 3587: Fixed the mass import of extensions from file to allow the insertion of rows with empty "username" attribute
  - 3693: Fixed an issue with viewing and filtering records by months of the previous year
  - 3711: Fixed a display issue in the audio file panel when one or more files are being used as the announcement message for the beginning or the end of call recording
  - 3730: Removed a restriction in inserting a user password when using an external authentication mechanism
  - 3553: The disabled state is now displayed in the switch selection within the time checks
- High availability
  - 3654: Fixed an issue that made it impossible to disable HA on the secondary node
- Provisioning
  - 3722: Fixed the handling of import from xls file when the SIP account is already used by another device
  - 3586: Fixed the mass import template returned in single-tenant systems (without the tenant_uuid column)
- Phonebook
  - 3663: Restored the display of the personal phonebook the user GUI panel
- Rest API
  - 3627: Fixed an issue that caused a 500 error when invoking the CDR API in POST mode
  - 3600: Fixed the CDR API response that when authenticating as privacyadmin still returned anonymized external numbers
- CDR
  - 3706: Fixed the filtering of the call detail of the user CDR

    - 3719: Fixed the display of the "sourcenum" attribute for calls that passed through a time check

- LDAP
    - 3739: Fixed an issue when importing the shared phonebook that caused the duplication of LDAP contacts
    - 3737: Fixed an issue with starting the LDAP server that caused the deletion of the shared phonebook from the LDAP contacts
    - 3672: Fixed the failed lock of the PDAP server on Kalliope Mini that under certain conditions was erroneously started
- Multi-tenant
    - 3766: Fixed an issue with the pbxadmin accessing KalliopeLogger
    - 3666: Fixed the handling of attempts to delete a template used by one or more devices
    - 3674: Restored the functionality of calls (via trunk) to remote extensions of a tenant
- KalliopePBX v3 backup importer
    - 3626: Fixed several issues with importing v3 backups
    - 3657: Fixed an issue with importing a backup under specific conditions of the phonebook present in the backup
    - 3677: Fixed the import of provisioning data after adding the tenant_uuid column
    - 3644: Fixed the import of the unconditional forward service when the values are different from the default
    - 3476: Fixed the import of IVR menu, which made it impossible to add missing selection to imported menus

Firmware 4.3.6 (28/11/2016)

New features

- General
    - Added T.38 passthrough configuration parameters to the SIP settings (error correction mode, max datagram size)
- Provisioning
    - Added the option to define custom placeholders in the provisioning templates
    - Added the option to define custom device models, associating customizable rules for constructing the names of the resulting provisioning files
    - Added the option to manage provisioning templates and devices on a pbxadmin level
- Call Center
    - Added a call reservation on queue service
- Multi-tenant
    - Added the option for the pbxadmin to use the KalliopeLogger software for diagnostic and monitoring calls to and from tenants
- Service provider
    - Added a panel that collects the maximum number of daily and monthly configured extensions per tenant
- XMPP chat/presence

- Added a server module for managing message archival in order to obtain the complete history of a user's conversations on all clients

- REST API
  - Added an API for editing account templates
  - Added APIs for managing new provisioning features (custom placeholder, custom devices, provisioning by pbxadmin)

Changes

- General
  - 3217: Added an indication on the dashboard of which firmware version is running by displaying an R
  - 3265: Optimized configuration reload when applying multiple changes from the GUI or through APIs

Bugfixes

- General
  - 3552: Fixed an error that blocked the finalization of backup restoration
  - 3459: Fixed an error message when activating G.729 licenses while the Digium server was unable to be reached
  - 3457: Fixed a configuration error that made it impossible to disable SNMP agent execution
  - 3431: Introduced an automatic mechanism for resolving the issue where the unreachable state of the VoIP domain subsisted if the system was started while the configured DNS servers were unreachable, making it impossible to make and receive calls
  - 3544: Fixed the handling of the response when the configured busy level is reached so that the "486 Busy Here" message is returned instead of the generic "603 Declined"
  - 3294: Fixed the behavior of Fork2Mobile calls to respect the timeout for unanswered calls configured for the extension, which previously kept ringing even after the timeout was reached, making it impossible to use the forward action set for unanswered calls
  - 3463: Fixed an error introduced in version 4.3.5 that made it impossible to define new roles
  - 3346: Fixed the display of the total number of rows in tables when a search filter is applied
  - 3403: Fixed the management of the time check configuration panel, which generated a 500 error when a switch was enabled while none were selected in the drop-down menu
  - 3410: Fixed the mechanism for sending advance license expiration notifications via email
  - 3402: Fixed the failed presentation of the caller name for calls to remote numbering ranges (extensions)
  - 3321: Fixed the functionality of the manipulation rules of calling and called numbers on outbound lines when multiple range match rules are present
  - 3370: Fixed the handling of empty rows when mass importing files
  - 3368: Fixed the handling of new audio files with the same name as an existing one
  - 3378: Fixed the handling of strings concerning IP addresses in which all 4 octets are 3-digit numbers, for which the automatic format of the cell passes from text to number, losing the "." characters when importing from an XLS file

- Phonebook
  - 3121: Fixed an issue with the misalignment of search filters in the shared and personal phonebooks when the speed-dial service is disabled
  - 3382: Changed the timeouts to avoid "504 Gateway timeout" errors when importing the phonebook from file while the number of contacts exceeded a few tens of thousands; a 50000 contact limit for single imports was added
  - 3136: Removed the speed-dial definition box in the contact configuration panel when speed-dial is globally disabled
  - 3390: Removed the display of mobile numbers linked to extensions from the extension phonebook visible to non-admin users
- Chat/presence XMPP
  - 3513: Fixed an issue that prevented the presence state for users with roles other than base "tenant user" from being displayed
- Call recording
  - 3188: Fixed the handling of multiple sequential recording requests for the same call
- CDR
  - 3411: Fixed the display of calls that use the FastXfer service
  - 3416/3425: Harmonized the exit cause strings and fixed certain outcomes (e.g. outbound line not available, call limit reached)
- KalliopeCTI
  - 3419: Fixed an issue with remote controlling Yealink phones with fw xx.73.xx.xx firmware (with KCTI Pro) that caused multiple DTFM tones not to be sent
  - 3392: Fixed an error that caused the string "xxx" to be sent as the display-name for calls to or from an external number (when the calling/called number was absent from the application phonebook)
  - 3373: Fixed the failed sending to CTI clients of information on Organization or Department of the extension phonebook contacts when they were defined in the template but not defined on an extension configuration level
  - 2558: Restricted the admissible number validation for the click-to-call service to exclusively accept strings containing only digits (0-9) and phone characters (+, *, #)
  - 3294: Fixed the handling of calls from KalliopePhone that, under certain conditions, instead of completing correctly sometimes caused the call by the destination of the transfer to continue towards the failover action for when the transferring extension is unavailable
- Multi-tenant
  - 3345: Fixed the handling of requests for the creation of a tenant with the same domain name as an existing one, which previously generated a 500 error
  - 3443: Fixed an error that caused the import of a tenant from a preexisting backup to fail
  - 3150: Fixed the display of multi-tenant license limits

Firmware 4.3.5 (08/11/2016)

New features

- Added support for the new Kalliope 4 Service Providers license

Changes

- General
  - 3311: Extended the remote API application to support passing the "DNID" (called number) parameter to the APIs and handle the "displayprefix" tag in the response for setting a prefix to the Display Name
  - 3222: Added a domain/gateway reachability state check before sending a call to the outbound line, so that the backup line can be immediately used if necessary
  - 3331: Changed the activation state BLFs for the Boss-Secretary service filters to use "*" instead of "-" as the separator between the ID of the boss and the ID of the secretary
  - 3260: Extended the wizard to manage link.voipvoice.it as well as sip.voipvoice.it for the VoipVoice provider
- KalliopeCTI
  - 3334: Transferred to the clients the state of the BLFs associated to the Boss-Secretary service (information not currently used by the clients)
  - 3314: Optimized the sending of queue statistics to supervisors and operators

Bugfixes

- General
  - 3143: Fixed an issue where, when importing a backup made on a different machine, the outbound lines were not selectable in the outbound routing rules and had to be recreated
  - 3327: Fixed the handling of records with empty "contactType" when importing the phonebook from file
  - 3323: Fixed the validation of the speedDial field when importing the personal phonebook from file
- CDR
  - 3306: Fixed the registration of the outcome of an unanswered outbound call when the backup line is in use, which previously returned the UNAVAILABLE state associated with with the first call attempt
- KalliopeCTI
  - 3330: Fixed the transfer to clients of the Organization and Department information corresponding to extensions when the attributes were defined in the template but not for the extension
  - 3333: Removed notifications to clients of the "mobile number" and "email" information configured for each extension
  - 3312: Fixed the handling of on call service codes when they have been changed from the predefined ones
- Multi-tenant
  - 3222: Fixed the handling of maps assigning operators to queues when multiple tenants are present
  - 3336: Fixed the calculation of the line limits in the tenant configuration

Firmware 4.3.4 (17/10/2016)

New features

- Added a mechanism for automatically registering SNOM terminals (defined among the provisioning devices) on the redirection server of the producer

Changes

- Client KalliopeCTI
  - 3057/3160: The client (version 4.2.1 or later) now displays the activation state of the on-demand call recording service and the activation state of the recording following the press of the corresponding key

Bugfixes

- General
  - 3191: Fixed an issue with saving extensions that caused a 500 error unless all failover actions were redefined from the ones from the template
  - 3255: Fixed the display of trunk and termination status on a domain when multiple trunks and/or terminations are present
  - 3257: Fixed the display of the reachability status of a domain when the IP or hostname is edited
  - 3247: Fixed the functioning of the search filter for the current month in the audit log
  - 3224: Restored the functionality of the Fork2Mobile feature for direct calls to the members of a ring group, which has not been working since version 4.1.1
  - 3199: Fixed an issue where inbound calls that passed through a time check were terminated when a button was pressed while the closing message was playing
  - 3220: Fixed a validation issue when importing contacts from files with empty speeddials
  - 3176: Made uniform the error response (a generic "invalid credentials" message) when logging into the web GUI with the wrong credentials, as a disabled or a non-existent user, in order not to give away information on the existence or lack thereof of a specific username
  - 3179: Fixed the behavior of the PBX for direct calls to a disabled extension, which did not proceed in the numbering plan to a custom selection but were instead terminated
  - 3144: Any spaces at the beginning or the end of a field will now be removed when saving phonebook contacts
  - 3164: Any spaces at the beginning or the end of attributes of contacts imported from XLS/CSV files, which previously caused validation to fail, will now be removed
  - 3156: Fixed an issue where hold music failed to play when transferring an external call in a queue
- KalliopePBX v3 backup importer
  - 3183/3184/3185/3246: Extended 3.12.3 backup validation checks to handle entities (extensions, groups, queues, etc.) with duplicate keys
- Multi-tenant
  - 3208: Fixed an issue with backup restoration in multi-tenant mode where the procedure failed during finalization

- Rest API
  - 3205: Fixed an issue where creating extensions via REST API did not correctly generate the configuration of BLF keys
- Registro chiamate
  - 3154: Made uniform the columns available in the web panel and those resulting from exporting to CSV/XLSX/XML/JSON

Firmware 4.3.3 (30/09/2016)

New features

- Added queue configuration options to mark paused operators or operators with an unregistered terminal as not available for service in order to execute the failover action without having to wait in the queue

Changes

- 2995: Changed the phonebook import format to conform with the one used when exporting
- 2993: Increased the maximum number of extensions that can be created via configuration wizard from 10 to 100
- 3011: Changed the name of the headers of certain columns in the CDR
- 3114: Changed certain strings in the extension panel
- 3125: Reduced the wait between activating the paging service in "unattended" mode and the start of the call from 3 seconds to 1 second
- 3074: Added new Yealink telephone models to the list of provisioning devices

Bugfixes

- KCTI
  - 3137: Fixed the remote control of Yealink phones, which in cases with two or more calls on the terminal acted only on the call on the first line
- General
  - 3111: Fixed an issue where new device provisioning templates were not displayed when assigning templates to a terminal
  - 3132: Fixed an issue with the mechanism for mass import from Excel files, which generated a 500 error when empty rows were present
  - 3130: Fixed an error with field validation when mass importing provisioning devices when rows with a note field contained a numeric value
  - 3110: Fixed the mechanism for resetting failover actions, which previously prevented entities (extensions, queues, groups, etc.) from being deleted while in use as the destination of a failover action
  - 3112: Fixed a bug that prevented edits to custom selections in the numbering plan from being saved
  - 3104: Fixed an issue where under certain conditions the write lock for the web interface was incorrectly generated

  - 3086: Fixed an issue where the main partition of the PBX was filled when call recording on demand was enabled for queues but unconditional call recording was disabled
  - 3076: Fixed the functionality of the electronic lock service
- Multi-tenant
  - 3101: Fixed the display of the status of accounts in the dashboard of tenants other than the default
  - 3098: Fixed the notification message for reaching the maximum number of extensions assignable to a tenant
  - 3081: Fixed an error that prevented the creation of the TFTP folder of a tenant contextually to the creation of the tenant itself until the PBX was rebooted
- KalliopePBX v3 backup importer
  - 3090: Fixed the handling of ring groups with duplicate extensions with the same priority
  - 3088: Fixed the handling of voicemail boxes

Firmware 4.3.2 (15/09/2016)

New features

- Introduced a new service: Hot Desking
- Added the option to configure an outbound proxy for communication with SIP accounts

Bugfixes

- KCTI
  - 3058: Fixed an issue where certain notifications were not forwarded from telephones to the linked client
- General
  - 3049: Fixed an issue with the handling of batch actions on extensions and SIP accounts (e.g. deleting 500 or more extensions), which previously caused the execution of the command to fail

Firmware 4.3.1 (06/09/2016)

New features

- Introduced a new service: Paging
- Extended notifications sent to the KalliopeCTI clients to optimize call management

Bugfixes

- Multi-tenant
  - 2953: Fixed the display of the tenant admin dashboards, which in version 4.3.0 were visible to the pbxadmin
  - 2989: Fixed an error introduced in version 4.2.1 that caused the incorrect routing of inbound calls from external lines for multi-tenant systems
- KalliopePBX v3 backup importer
  - 2972: Added a way to handle multiple references to the same audio file, not present in the backup (e.g. dismissed IVR menus), which caused an exception when inserting the duplicate entry in the database
  - 2974-2976: Fixed the handling of errors during import; now if an error occurs, a banner will apear to restore the consistency state of the internal databases before restarting import
  - 2985: Fixed an error when importing KCTI usernames what prevented authentication unless each usernames was redefined
- CDR
  - 2951: Fixed the handling of calls presented to multiple terminals, which in some cases were registered as not answered even when they were
  - 2957: Fixed the failed registration of direct calls to an extension sent to voicemail box
  - 2964: Fixed the handling of calls to extensions, groups, and queues that are canceled by the client, which were previously registered as not answered
- General
  - 2954: Restored the maximum size for audio backup files uploaded to the web interface to 64 MB, which was erroneously reduced to 1 MB with the web server replacement in version 4.3.0
  - 2978: Fixed an issue that under certain conditions caused an exception when accessing the queue panel, limited to Mini devices
  - 2979: Added user dashboard configuration to configuration backups
  - 2981: Fixed the handling of provisioning devices when an extensions and accounts are deleted, which previously generated an exception
  - 3007: Changed the authentication mechanism of Action-URIs used for remote controlling Yealink phones (from query-string to HTTP Basic Authentication) for v73 phones with firmware version x.73.0.50 or later and phones of the v80 series or later
  - 3009: Restored the functionality of the Distinctive Ringing feature (currently available for SNOM, Yealink, and Gigaset Pro DExx0 devices)
  - 3017: Fixed the functionality of the cancellation code for the call redirection service, which, when digits other than the cancellation code were present, set up a new redirection to that destination
  - 3018: Changed the mechanism for calls to the mobile number linked to and extension to omit the confirmation request for fastXfer and redirection from KalliopeCTI Mobile (confirmation requests are kept for fork2mobile and click-to-call for mobile)
- API Rest
  - 3016: Fixed an issue where REST APIs did not respect the user activation flag and the use of external authentication providers

Firmware 4.3.0 (09/08/2016)

> **Warning:** A bug has been found in version 4.3.0 that, in multi-tenant systems, causes the incorrect routing of inbound calls from external lines, preventing their reception and causing the inbound call to be terminated with the "603 Declined" response. For this reason, we recommend not updating multi-tenant systems to this version and instead directly updating to version 4.3.1 or later.

New features

- Call Center Module - Call Tagging (work Code)
- Dynamic user-customizable dashboard
- Web server and GUI reworking to increase execution speed and web socket support

Bugfixes

This release includes all bugfixes from versions 4.2.0 and 4.2.1

1.1.15 Firmware series 4.2.x (MR)

Firmware updates from the 4.2.x series are LTS Maintenance Releases, i.e. stable versions with long term support. The releases in this series have been thoroughly tested before being released to the public and therefore guarantee greater stability.

Firmware 4.2.7 (16/02/2017)

Malfunzionamenti corretti

- General
  - 3671: Restored the personal phonebook in the user web GUI
  - 3694: Fixed an error that caused CDR/audit logs to be displayed for the current year instead of the one selected
  - 3731: Fixed an anomaly when mass importing extensions with empty usernames
  - 3881: Fixed a bug with the rotation of web application logs
  - 3862: Fixed a display error with the list of gateways with an expired VM license
- Phonebook
  - 3772: Fixed an error that caused contacts imported from file to be duplicated when published to LDAP
  - 3773: Fixed an error that caused the publication of the shared phonebook to LDAP to fail when the machine was rebooted
- KPBXv3 Backup Importer
  - 3678: Fixed an error when importing conditional forwards
  - 3676: Fixed an error when importing IVR menus
- REST API

  - 3790: Changed data filtering of the CDR to use the timestamp for the end of the call and not the beginning
- KCTI
  - 3274: Fixed certain issues with remote controlling phones in KCTI Pro mode

Firmware 4.2.6 (20/12/2016)

Bugfixes

- General
  - 3458: Fixed an issue with the configuration where it was impossible to disable SNMP agent execution
  - 3460: Fixed an error message when the Digium server was unreachable for G.729 license activation
  - 3462: Fixed the mechanism for sending advance license expiration notifications via email
  - 3465: Fixed the role deletion action to not require the configuration lock
  - 3430: Introduced an automatic mechanism for resolving the issue where the unreachable state of the VoIP domain subsisted when the system was started while the configured DNS servers were unreachable, making it impossible to make and receive calls
- Phonebook
- 3122: Fixed an issue with the misalignment of search filters in the shared and personal phonebooks when the speed-dial service is disabled
- Call recording
  - 3450: Fixed the handling of multiple sequential recording requests for the same call
- XMPP chat/presence
  - 3494: Fixed an issue that prevented the presence state for users with roles other than base "tenant user" from being displayed
- KalliopeCTI
  - 3625: Fixed the handling of calls from KalliopePhone that, under certain conditions, instead of completing correctly sometimes caused the call from the destination of the transfer to continue towards the failover action for when the transferring extension is unavailable
  - 2558: Restricted the admissible number validation for the click-to-call service to exclusively accept strings containing only digits (0-9) and phone characters (+, *, #)
- API REST
  - 3628: Fixed a bug with CDR REST APIs in POST mode
- KPBXv3 Backup importer
  - 3619: Fixed an error that under certain conditions caused importing to fail
- Multi-tenant
  - 3441: Fixed an error with KPBXs with a multi-tenant license that prevented KalliopeLogger from connecting as pbxadmin
  - 3604: Fixed the handling of multi-tenant licenses with unlimited channels
  - 3629: Fixed the handling of edits to a tenant from the GUI, which caused certain inconsistencies in the display of the account state and the connection to KCTI clients

Firmware 4.2.5 (14/11/2016)

Changes

- General
  - The dashboard now shows the current firmware version
  - 3331: Changed the activation state BLFs for the Boss-Secretary service filters to use "*" instead of "-" as the separator between the ID of the boss and the ID of the secretary
  - 3296: Extended the validation of 12-digit IP addresses when importing to xls file to handle cases where cells were automatically converted from text to number

Bugfixes

- General
  - 2614: Fixed an issue with creating time checks and adding a switch while leaving the selection empty
  - 2752: Fixed the behavior of Fork to Mobile calls to honor the ring timeout set for the extension
  - 3243: Fixed the functionality of the audit log panel filtering
  - 3244: Fixed the display of trunk and termination registration state when more than one is present on a given domain
  - 3258: Fixed an issue with the display of the reachability state of a VoIP domain when the address or the hostname is edited
  - 3295: Fixed the validation of mass import files with empty rows
  - 3310: Fixed the handling of the creation of audio files with the same name as an already existing one
  - 3324: Fixed the validation of the SpeedDial column when importing the personal phonebook from file
  - 3369: Fixed an issue with the handling of the manipulation rules of calling and called numbers on outbound lines when multiple range match rules are present
  - 3328: Extended the validation of import of phonebook contacts from file to handle rows with empty ContactType attribute
  - 3354: Fixed an issue where restoring a backup made on a different machine made it impossible to select outbound lines from the backup in the routing rules
  - 3391: Fixed the display of the number of rows in the provisioning panel when a filter is active
  - 3413: Fixed the failed display of the caller name for calls to remote extensions
- KCTIS
  - 3253: Fixed an issue with displaying the presence of dynamic operators after rebooting the PBX
  - Several CTI server corrections concerning the handling of queues and notifications in multi-tenant systems and the presentation of the caller name and number to the client
- CDR
  - 3266: Fixed the display of calls using the FastTransfer feature
- Importer backup v3
  - 3236: Fixed certain conditions that could generate an exception (500 error)
- Multitenant

- 3337: Fixed the calculation of line limits in tenant edit
- 3376: Fixed validation issues when creating a tenant with an already existing domain
- 3363: Extended the uniqueness check for tenant domain names in case-insensitive mode

Firmware 4.2.4 (10/10/2016)

Changes

- Importer backup v3
  - 3138/3182/3186: Extended 3.12.3 backup validation checks to handle entities (extensions, groups, queues, etc.) with duplicate keys
- CDR
  - 3169: Made uniform the columns available in the web panel and those resulting from exporting to CSV/XLSX/XML/JSON

Bugfixes

- General
  - 3192: Fixed an issue with saving extensions that caused a 500 error unless all failover actions were redefined from the ones from the template
  - 3147: Changed the error code (from 603 to 486) sent in response to a call to an extension when the busy level or the call limit has been reached
  - 3223: Restored the functionality of the Fork2Mobile feature for direct calls to the members of a ring group, which stopped working in version 4.1.1
  - 3221: Fixed an issue where inbound calls that passed through a time check were terminated when a key was pressed while the closing message was playing
  - 3168: Any spaces at the beginning or the end of a field will now be removed when saving phonebook contacts
  - 3165: Any spaces at the beginning or the end of attributes of contacts imported from XLS/CSV files, which previously caused validation to fail, will now be removed
  - 3067: Fixed the handling of Direct Media activation for calls to extensions linked to multiple SIP accounts when some of them do not have Direct Media enabled
  - 3158: Fixed an issue where hold music failed to play when transferring an external call in a queue
  - 3180: Fixed the behavior of the PBX for direct calls to a disabled extension, which did not proceed in the numbering plan to a custom selection but were instead terminated
  - 3177: Made uniform the error response (a generic "invalid credentials" message) when logging into the web GUI with the wrong credentials, as a disabled or a non-existent user, in order not to give away information on the existence or lack thereof of a specific username
- Multi-tenant
  - 3173: Fixed an issue with backup restoration in multi-tenant mode where the procedure failed during finalization
- Rest API

- 3204: Fixed an issue where creating extensions via REST API did not correctly generate the configuration of BLF keys

Firmware 4.2.3 (30/09/2016)

Changes

- 2868: Changed the phonebook import format to conform with the one used when exporting
- 2992: Increased the maximum number of extensions that can be created via configuration wizard from 10 to 100
- 3010: Changed the name of the headers of certain columns in the CDR
- 3119: Changed certain strings in the extension panel
- 3080: Added new Yealink telephone models to the list of provisioning devices

Bugfixes

- General
  - 3133: Fixed an issue with the mechanism for mass importing from Excel files, which generated a 500 error when empty rows were present
  - 3131: Fixed an error with field validation when mass importing provisioning devices when rows with a notes field contained a numeric value
  - 3083: Fixed the mechanism for resetting failover actions, which prevented entities (extensions, queues, groups, etc.) from being deleted while in use as the destination of a failover action
  - 2827: Fixed a bug that prevented edits to custom selections in the numbering plan from being saved
  - 3024: Fixed an issue where the main partition of the PBX was filled when call recording on request was enabled for queues but unconditional call recording was disabled
  - 3072: Fixed the functionality of the electronic lock service
  - 3067: Fixed the handling of Direct Media for multiple SIP accounts linked to the same extension with different activation states
- Multi-tenant
  - 3099: Fixed the notification message for reaching the maximum number of extensions assignable to a tenant
  - 3087: Fixed an error that prevented the creation of the TFTP folder of a tenant contextually to the creation of the tenant itself until the PBX was rebooted
- Importer backup v3
  - 3089: Fixed the handling of ring groups with duplicate extensions with the same priority
  - 2997: Fixed the handling of voicemail boxes

Firmware 4.2.2 (06/09/2016)

Bugfixes

- Multi-tenant
  - 2987: Fixed an error introduced in version 4.2.1 that caused the incorrect routing of inbound calls from external lines for multi-tenant systems
- KPBXv3 Backup importer
  - 2971: Added a way to handle multiple references to the same audio file, not present in the backup (e.g. dismissed IVR menus), which caused an exception when inserting the duplicate entry in the database
  - 2975: Fixed the handling of errors during import; now if an error occurs, a banner will be presented to restore the consistency state of the internal databases before restarting import
  - 2986: Fixed an error when importing KCTI usernames that prevented authentication unless each usernames was redefined
- CDR
  - 2937: Fixed the handling of calls presented to multiple terminals, which in some cases were registered as not answered even when they were
  - 2957: Fixed the failed registration of direct calls to an extension sent to voicemail box
  - 2952: Fixed the handling of calls to extensions, groups, and queues that are terminated by the client, which were registered as not answered
- General
  - 2938: Fixed an issue where the status of the SIP account was sometimes not updated
  - 2977: Fixed an issue that under certain conditions caused an exception when accessing the queue panel, limited to Mini devices
  - 3008: Restored the functionality of the Distinctive Ringing feature (currently available for SNOM, Yealink, and Gigaset Pro DExx0 devices)
  - 3019: Fixed the functionality of the cancellation code for the call redirection service, which, when digits other than the cancellation code were present, set up a new redirection to that destination
- API Rest
  - 3015: Fixed an issue where REST APIs did not respect the user activation flag and the use of external authentication providers

Firmware 4.2.1 (02/08/2016)

> **Warning:** A bug has been found in version 4.2.1 that, in multi-tenant systems, causes the incorrect routing of inbound calls from external lines, preventing their reception and causing the inbound call to be terminated with the "603 Declined" response. For this reason, we recommend not updating multi-tenant systems to this version and instead directly updating to version 4.2.2 or later.

Bugfixes

- 2925: Fixed a malfunction when starting the HA service

- 2911: Fixed DND setting on SNOM phones with KCTI Pro in remote control mode
- 2906: Fixed a malfunction when disabling an IP address
- 2904: Fixed an issue with provisioning phones through HTTP/HTTPS
- 2901: Fixed an issue with updating the status and mobile number linked to the Fork2Mobile service
- 2882: Fixed the display of assignable/available accounts for KPBX in multi-tenant mode
- 2875: Fixed the handling of the outbound identity for calls to an external number made through a custom selection in the numbering plan
- 2767: Fixed the data filter for archived months for the CDR and the Call Center CDR

Firmware 4.2.0 (15/07/2016)

This version introduces a tool for converting configurations from V3 toV4 and includes a promotional KalliopeCTI Phone license. Several bugs have also been fixed, making this a stable MR release.

New features

- Added a promotional KalliopeCTI Phone license to the system
- Added a migration feature to convert configuration from KalliopePBX V3 by importing a backup made from version 3.12.3 or later

Bugfixes

- 2747: Fixed the handling of call recording for unserved calls in in a queue, which remained "elaborating" indefinitely
- 2787: Fixed an issue that prevented the correct resolution from the phonebook of the caller name for inbound calls to queues and groups
- 2790: Fixed an anomaly when exporting the shared phonebook from file when there are contacts with multiple phone numbers
- 2799: Restored the functionality of the phone code to control the state of the fork2mobile feature
- 2807: Fixed an anomaly when using multiple ENUM rules within the same routing rule, which caused only the first to be executed
- 2815: Fixed the handling of the presentation of the connected line identifier to KCTI clients and to phones in cases where the identifier changes during the call (e.g. attended transfer, call pickup)
- 2823: Fixed an issue where the busy failover action was not executed for calls to an extension that had reached the busy level or concurrent call limit
- 2830: Fixed an issue where under certain conditions the audit log was not displayed
- 2834: Fixed in issue where under certain conditions the PBX event record was not populated
- 2836: Fixed an issue where tenant admins failed to authenticate on KalliopeLogger
- 2837: Fixed an issue where the transferer identity was incorrectly set for call transfers from the calling extension, which caused the call not to return if the transfer failed
- 2840: Fixed an issue where outbound routing class was not assigned to remote extensions
- 2846: Fixed an issue where the fastXfer failed to function when commanded by the calling extension

1.1.16 Firmware series 4.1.x (TR)

Firmware updates from the 4.1.x series are Technology Releases, which introduce new features; though they have been tested, they likely contain bugs that emerge under specific configurations or use conditions. The latest Maintenance Release, version 4.4.0, is the stable release recommended for generic use that does not require features introduced in the TR.

Firmware 4.1.7 (05/07/2016)

This version introduces the Boss-Secretary service with modes from 1 to N and extends the user authentication logic on the web interface and the KCTI clients to use external authentication sources such as external LDAP servers and Active Directory domains.

Nuove funzionalità

- Added the Boss-Secretary service, with the option of having groups with multiple secretaries and managing filters through BLF keys
- Added the option of authenticating users for CTI and web access on Active Directory domains or external LDAP servers
- Extended the remote LDAP phonebook importer to support estos MetaDirectory

Bugfixes

- 2619: Restored the functionality of speeddial, both from the shared phonebook and from personal phonebooks
- 2658: Restored the functionality of the call pickup with invite service (direct and group)
- 2744: Fixed the sending of COLP while the calling phone is ringing in cases where the destination is an extension linked to multiple SIP accounts
- 2750: Fixed the handling of the deletion of voicemail messages saved to the PBX after being deleted from the extension
- 2753: Removed the catch-all action from the numbering plan in cases where outbound calls are configured without a commitment prefix
- 2755: Fixed a malfunction of the "forward to an external number" extension failover action
- 2758: Fixed a bug that, after updating the firmware, prevented access to voicemail messages saved to the PBX with the previous firmware version
- 2760: Fixed the handling of refused or unanswered calls in click-2-call
- 2761: Fixed a bug that prevented call transfers from mobile for calls received via the fork2mobile service
- 2762: Fixed a bug that prevented call transfers from mobile for calls redirected to mobile via KalliopeCTI

Firmware 4.1.6 (23/06/2016)

This release extends the call recording service to outbound calls made by extensions, makes the persistent record of edits to the configuration available for consultation, and introduces a number of small changes and corrections.

This release also introduces the first configuration wizard, an assisted procedure for generating an initial base configuration that includes the creation of the accounts and extensions, a ring group and a queue, and an outbound line (starting from a list of included preset ones), including the inbound routing. With this feature, setting up KalliopePBX is even quicker and simpler.

New features

- Added configuration wizard
- Added a mechanism to automatically generate and evaluate the strength of the SIP secrets of accounts and users
- Added the user call detail record (CDR)
- Added a REST API for click-2-call
- Added the option to generate reports from the Call Center CDR from the panel of the same name
- Extended the functionality of fastXfer from extension to mobile to group calls
- Added an action to mass delete devices in the provisioning panel
- Added the option to download custom audio files
- Added the option to export the system and personal phonebooks and the call records to several formats (xls, csv, etc.)
- Integrated XMPP server to manage CTI presence and advanced instant messaging features available on the KalliopeCTI desktop and mobile applications
- Added a visualization panel for the audit log, which contains all edits to the PBX configuration of a user
- Added a call recording feature for outbound calls made by an extension, on demand and/or unconditional

Bugfixes

- 2669: Inserted an explicit line in the numbering plan to indicate the forwarding of the call to the outbound routing rules
- 2674: Restored the correct functionality of the "Overwrite failover" option in the time span of the time checks
- 2677: Fixed the CDR and Call Center CDR to display the full numbers to Privacy users
- 2691: Fixed the "state" filter in the extension list
- 2704: Fixed the handling of the "any" selection in the ENUM rules
- 2708: Fixed an issue where the wrong hold music class played for calls in queues from external numbers
- 2712: Fixed a display issue with the custom codecs in the SIP account edit panel
- 2717: Fixed an issue where priorities after the first could not be deleted from ring groups
- 2719: Added Snom D345 to provisioning devices, which was missing if the PBX was updated from versions previous to 4.0.8
- 2720: Restored the functionality of the BLF key for changing the activation state of the fork2mobile service
- 2724: Fixed an issue where the CSR SSL failed to generate

- 2734: Added the option to delete the profile image
- 2736: Fixed an issue where the codecs of an account could not be reset to the default value of the template after configuring a custom set
- 2746: Fixed the display of the SIP account identifiers in multi-tenant systems to include the prefix of the tenant
- 2748: Fixed an issue where devices could not be deleted from the provisioning panel if they had been inserted by mass importing when indicating a non-existent template and account

Firmware 4.1.5 (31/05/2016)

This release fixes with several bugs present in the previous versions and introduces two important features: SIP URI number support through ENUM queries, and the ability to import contacts from LDAP phonebooks with the option to schedule syncing and enable or disable the presentation of the resulting contacts to CTI clients.

New features

- Graphical redesign of the extensions and SIP accounts panels to add visibility to the configuration options borrowed from the template or overwritten by a specific setting
- Added ENUM support to the routing rules, with the option to make calls directly on the destination domain or use one of the configured trunks
- Added a service for periodically importing contacts from remote LDAP phonebooks, with the option to configure for each whether to present the corresponding contacts to CTI clients and the user web pages
- Added an action to forward a call to the numbering plan, keeping the inbound selection, to the list of outbound actions of the time checks
- Extended the attended transfer service to add shuttle and three-way-conference features

Bugfixes

- 2660: Fixed an issue with the HA module where resources were incorrectly reacquired by the primary node when the connectivity between the two nodes of the cluster was interrupted and restored
- 2574: Changed the refresh timing of the HA state page to avoid cases where, if the DNS failed to resolve, the update time exceeded the refresh time, causing GUI slowdown
- 2666: Fixed an issue where under certain conditions the capture of the diagnostic track failed to start
- 2665: Fixed the formatting of the response of the CDR REST API for requests in detailed csv format
- 2655: Fixed an issue where under certain conditions an exception was generated when restoring configuration backups
- 2652: Fixed an issue with remote controlling phones from KalliopeCTI Pro and Attendant Console when the phone was linked to a SIP account different from the extension number
- 2650: Fixed the presentation of the calling number for click-to-call calls originating from the GUI for recording and listening custom audio files
- 2646: Disabled the request of the options concerning the DNS when the PBX web interface is configured in DHCP mode (the DNS must always be statically specified)
- 2648: Reduced the timeout for sent SIP requests to speed up the use of the backup line when necessary
- 2403: Extended the display of validation errors for forms based on templates

- 2624: Fixed an issue with the default hold music where hold music failed to play for attended transfers or cases where an inbound call from an external line was put on hold
- 2632: Fixed an issue with the handling of the selection of the group pickup service where edits to the selection were not recognized by the PBX on a telephone level
- 2631: Restored the correct functionality of the cancellation code for attended transfer
- 2635: Fixed the display of the SIP account creation panel, which did not show the audio/video codec values inherited from the template
- 2613: Fixed an issue with the handling of the SNMP agent configuration which caused the binding of the service only on the loopback interface of the PBX
- 2639: Fixed an issue where audio was absent for outbound calls from toll-free numbers
- 2623: Fixed an issue where the Fork2Mobile service failed to function on all previous 4.1 versions
- 2603: Fixed the anonymization of calling numbers in the Call Center CDR for inbound calls from external lines

Firmware 4.1.4 (10/05/2016)

This version does not introduce new features but fixes some bugs present on previous 4.1.x versions in order to make way for the release of the MR 4.2.0 version.

Bugfixes

- 2597: Fixed a bug that sometimes caused a loop within the numbering plan for custom range selections
- 2599: Fixed a bug that prevented the correct functioning of the remote controlling of SNOM and Yealink phones through KalliopeCTI Pro
- 2595: Fixed an issue that under certain conditions caused an exception when generating the provisioning files when mass importing devices for provisioning
- 2587: Fixed an issue that in multi-tenant systems sometimes generated an exception when backup restoration was finalized
- 2590: Removed cross-validation between extension numbers and numebring plan selections, which prevented the creation of a custom selection with the same number as an existing extension (and vice versa)
- 2577: Fixed an issue where in systems without a multi-tenant license the File Manager showed the wrong direct download path, as it contained the UUID of the tenant
- 2589: Fixed an issue where after listening to a custom audio file via browser the file did not play during calls
- 2588: Restored the correct generation of the configuration for DIDs that point towards the “External API” application
- 2586: Fixed a bug that prevented the deletion of a tenant for systems with a multi-tenant license
- 2582: Fixed the validation of outbound line forms, which made it impossible to reduce the value of the maximum number of calls
- 2581: Fixed a bug that prevented the correct restoration of backups in multi-tenant systems
- 2580: Fixed an issue where the pbxadmin was unable to log in during updates
- 2579: Fixed an error with the handling of calls that pass through time checks that contain switches
- 2593: Fixed a bug that prevented the deletion of assigned lines when deleting a tenant (only in multi-tenant systems)

Firmware 4.1.3 (02/05/2016)

This version introduces the option to add audio files to the PBX by recording from a phone terminal (in addition to uploading prerecorded files). It also adds a special user for the authenticated access of the LDAP tree of the phonebook (its credentials can be inserted in the phone provisioning templates to make it unnecessary to manually insert those of each single PBX user in the phone)

New features

- Created the "phonebook" user with a dedicated role with read-only permissions for the web panel and the LDAP tree of the system phonebook
- Added a feature to the GUI to add audio files by following a guided procedure to record from the phone terminal (this feature also includes the option to play audio files on the system by request on a specific terminal and directly from the browser)

Bugfixes

- 2573: Fixed a bug that caused a 500 error on the GUI when mass importing extensions where two or more lines contained the same username (including empty ones)
- 2534: Fixed a bug that prevented the owner of a conference room from viewing and editing the room settings via web if their role did not have write access to the conference room panel
- 2533: Fixed a bug that under certain conditions caused a 500 error when saving edits to a conference room
- 2536: Fixed a bug that prevented users with the "Tenant user" role from accessing the shared system phonebook via the web GUI
- 2575: Fixed an issue where it was impossible to create accounts on KPBXv4 Mini
- 2576: Fixed an issue where the outbound prefixes in the calling number of an inbound call were not reconstructed

Firmware 4.1.2 (28/04/2016)

This version adds several features, including a call recording service (available for inbound or local calls to queues, groups, or extensions; call recording for outbound calls will be added in the next firmware release) on local or network storage, and support for the optional Call Center module.

Nuove funzionalità

- Added server-side pagination of the extensions and accounts panels to more quickly and efficiently handle amounts of entities larger than one thousand
- Added support for the optional Call Center module:
  - Call Center CDR
  - Supervisor/operator roles
  - Dynamic handling of operators from the supervisor panel and KalliopeCTI client
  - Dynamic handling of the pause status from KalliopeCTI clients and through phone codes
  - Listen/Whisper/Barge services for supervisors
  - Automatically pauses operators who fail to answer a call in the queue before ring timeout

- Added a call recording service (unconditional and on request) for inbound or local calls to queues, groups, or extensions
- Network storage management that assignable to tenants and with quote support
- Added a column for the mobile number linked to the extension in the template of the file for mass importing extensions
- Changelog extension for registering account management actions
- Added privacy user and admin roles to manage access to call recordings

Bugfixes

- 2504: Fixed a bug that caused a GUI error when trying to delete a gateway
- 2505: Fixed a bug which caused the time check bypass option not to be respected for calls to queues or groups
- 2515: Extended the validated of gateway forms to prevent the use of spaces or special characters in the identifier
- 2516: Fixed a bug when selecting the failover action for local or external calls to an extension
- 2527: The secret field in configuration of the SIP accounts of terminations and trunks is now masked
- 2529: Fixed an issue where the indicator on the GUI of the registration state of terminations and trunks associated to VoIP domains for which a port other than 5060 is specified remains gray even when registration is okay
- 2530: Restored the correct backup functionality
- 2531: Removed the parameter for the authentication request of calls from VoIP domains
- 2537: Fixed some issues with deleting and editing gateways created with version 4.1.1
- 2538: Fixed a bug that caused inbound calls to be refused if the external line commitment prefix was removed
- 2539: Restored the correct assignation of audio file language for external calls
- 2541: Fixed an issue where the PBX would drop a call instead of playing the error message when indicating a non-existent conference room
- 2542: Removed the display of the mobile number linked to an extension in the integrated phonebook
- 2548: Fixed the generation of the TFTP path between single and multi-tenant 4.1 systems
- 2549: Fixed the handling of forwards to the numbering plan with selection request to the caller via DTMF (e.g. post-selection on IVR menu)
- 2550: Restored the functionality of the service for playing an audio message on request to the interlocutor of a queue operator
- 2551: Fixed the functionality of the "no answer" filter in the CDR panel
- 2553: Restored the functionality of the extension deletion button
- 2564: Restored the correct functioning of failover actions for calls to queue or groups when a time check is enabled

Firmware 4.1.1 (06/04/2016)

This version is the first in the 4.1 series released to the public. It introduces multi-tenant support (which can be activated through a license), KalliopeCTI Mobile support, and other features described in the changelog below.

New features

- Added the "Remote API" PBX application to invoke external web services, which can be used, for example, to integrate the PBX dialplan with logic derived from queries to external databases
- Added the option to activate AMI (Asterisk Manager Interface) access, limited to "call" and "originate" permissions, specifying credentials and an access ACL
- Added support for a multi-tenant license (available only for KPBX-V4R+ physical devices and KPBX-V4-ESX virtual machines). With this license, multiple "virtual" PBXs can be placed within the same PBX, each with its own numbering plan and independent from the others.

1.1.17 Firmware series 4.0.x

Firmware 4.0.10 (27/05/2016)

Other than a few bug fixes, this version extends the functionality of the attended transfer service by adding the following features:

- Shuttle: during the transfer, dialing *9 will switch the paused interlocutor with the active one;
- Three-way conference: during the transfer, dialing *3 will convert the transfer to a conference call with both interlocutors.

New features

- Extended the attended transfer service to add shuttle and three-way conference features

Bugfixes

- 2654: Fixed an issue where phone numbers were not resolved from the shared phonebook
- 2653: Fixed an issue with remote controlling phones from KalliopeCTI Pro and Attendant Console when the phone was linked to a SIP account different from the extension number
- 2643: Restored the correct functionality of the cancellation code for attended transfer
- 2640: Fixed an issue where audio was absent for outbound calls from toll-free numbers

Firmware 4.0.9 (10/05/2016)

Bugfixes

- 2567: Fixed an issue where the LCR rule backup failed to function when the call-limit was set on a line
- 2476: Fixed an issue that generated a 500 error when mass importing contacts from the phonebook
- 2477: Forced the maxdatagram setting to 1400 bytes to solve compatibility issues with some T.38 lines
- 2596: Fixed a validation issue with the GUI where it was possible to delete LCR classes even if they were assigned to extensions
- 2562: Fixed the handling of forwards to queues and ring groups with integrated time check bypass
- 2547: Randomized license verification in order to avoid overloading the server, causing verification to fail
- 2598: Fixed an issue where phones could not be remote controlled from CTI Pro
- 2447: Fixed an issue where the failover action for unanswered external calls was not executed

2486: Fixed an error with the custom range selections in the numbering plan

Firmware 4.0.8 (04/03/2016)

New features

- Added a flag to automatically regenerate the device configuration file when saving provisioning templates
- Extended heartbeat communication to all interfaces for which HA is active
- Added a NAT helper feature to the SIP settings
- Added the option to export the shared system phonebook via LDAP
- Added REST APIs to back up and restore configuration; CDR exporting in summary format, extended and v3 compatibility
- Added SNMP v1/v2c read support
- Added an option to change the language of messages when entering a conference room
- Added phone credential management to activate remote remote control through KalliopeCTI Pro

Bugfixes

- 2455: Fixed group call form validation
- 2458: Restored the correct functionality of fast transfer from extension to mobile and vice versa (fastXfer) for direct calls to an extension

Firmware 4.0.7 (15/02/2016)

Bugfixes

- 2419: Fixed several issues concerning the voicemail service (consultation access, MWI indicator functionality, email sending)
- 2427: Voicemail boxes are now completely deleted upon deletion of the linked extension
- 2417: Fixed some issues with username validation when mass importing extensions
- 2423: Fixed a display error after deleting multiple selected extensions
- 2418: Fixed LCR class handling to prevent their creation without LCR rules
- 2395: Fixed an issue when restarting the backup node of a High Availability cluster

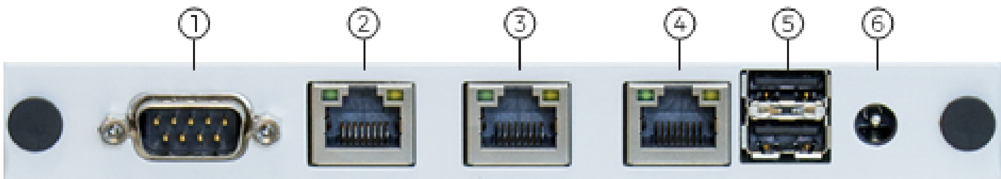
CHAPTER

TWO

GETTING STARTED

2.1 Getting Started

2.1.1 Mini/Lite



1. COM1 (Console 115200 8/n/1)
2. Gigabit Ethernet (eth0)
3. Gigabit Ethernet (eth1)
4. Gigabit Ethernet (eth2)
5. USB 2.0
6. Power (12v, 1A)

Installation

1. Connect KalliopePBX to your network using the eth0 interface.
2. Optional: connect KalliopePBX's COM1 serial port to your PC using a serial cable (not included - settings: 115200 8/n/1).
3. Connect the included power cord to KalliopePBX and plug it into a power socket (100-240V AC, 50-60 Hz). KalliopePBX will boot automatically.
4. Open http://192.168.0.100:10080 on your browser to register. KalliopePBX will require access to https://license.kalliopepbx.it; the network configuration can be edited from the "settings" menu. In case the error UNSAFE_PORT appears follow this procedure:

Resolution of ERR_UNSAFE_PORT problem with Chrome and Firefox browsers

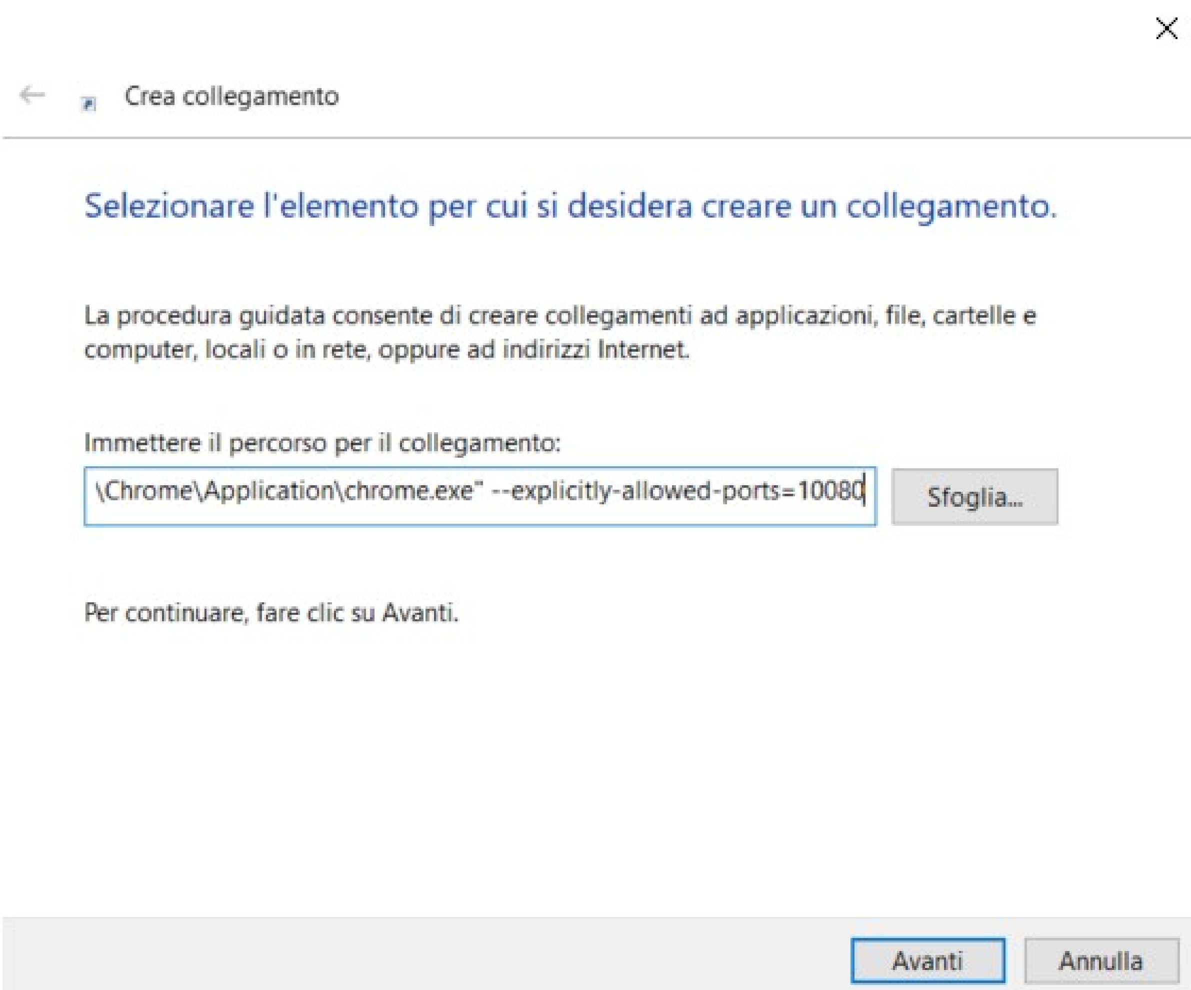
Resolution of ERR_UNSAFE_PORT problem with Chrome browsers

To launch Kalliope as “Restore Console” in the latest versions of Chrome you need to enable access to port 10080. To enable access to the Recovery Console, a new shortcut has to be created and it has to point directly to the Chrome executable, you can create it by right-clicking on the desktop and then clicking “New Shortcut”. The two possible paths depending on the installation you have done are:

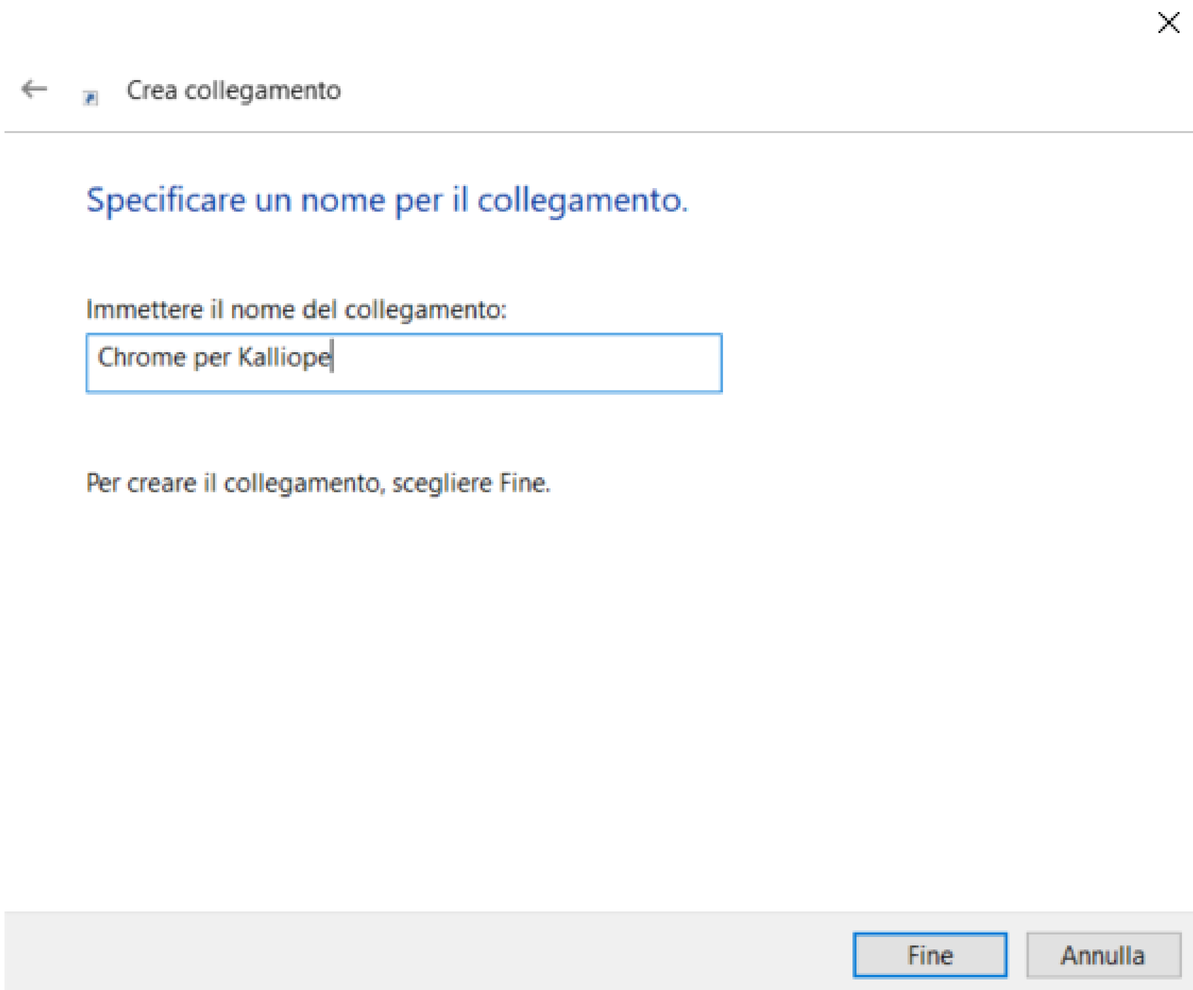
```
"C:\Program Files\Google\Chrome\Application\chrome.exe"
"C:\Program Files (x86)\Google\Chrome\Application\chrome.exe"
```

After making sure where the application is located on your PC, insert one of the following lines into the path of the new shortcut according to your Chrome installation:

```
"C:\Program Files\Google\Chrome\Application\chrome.exe" --explicitly-allowed-ports=10080
"C:\Program Files (x86)\Google\Chrome\Application\chrome.exe" --explicitly-allowed-ports=10080
```

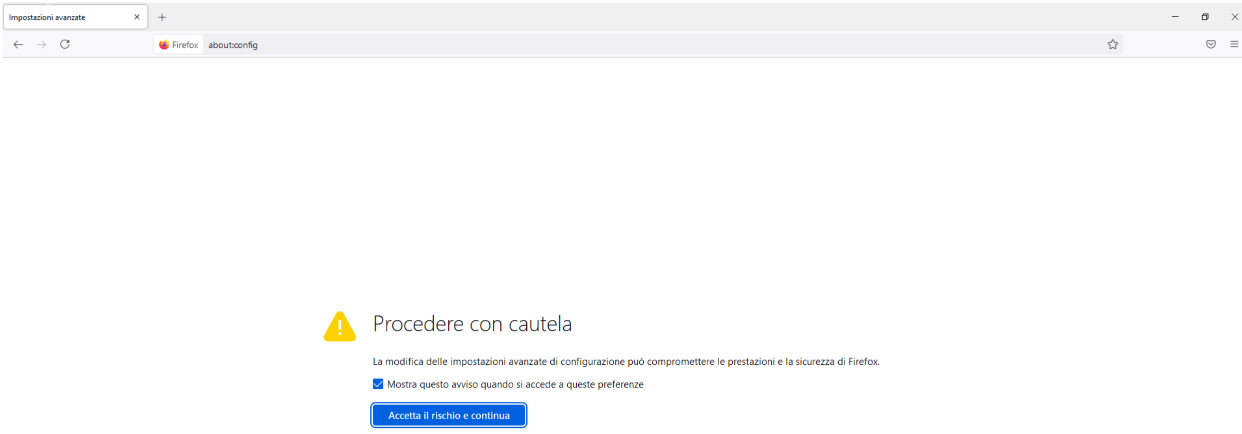


At this point click on “Next” and enter the name you want, then click on “Finish”. From now on, every time you need to access the GUI in “Restore Console” mode you have to use the shortcut you created. In some cases you may need to close all open Chrome instances before opening this one.

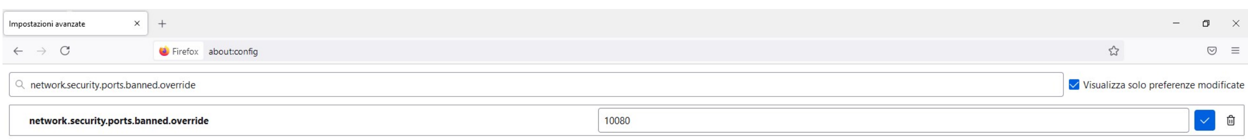


Resolution of ERR_UNSAFE_PORT problem with Firefox browsers

To launch Kalliope as "Restore Console" in the latest versions of Chrome you need to enable access to port 10080. Open Firefox and type about:config in the address bar:



Click on "Accept the risk and continue". Tick "Show only modified preferences" and in **Search preference name** write **"network.security.ports.banned.override"** select "String" and click on the + button.



At this point you can access the Kalliope Restore Console with Firefox.

5. After registering, choose and install the desired firmware from those available. KalliopePBX will require access to the address https://updates.kalliopepbx.it. The process may take a few minutes, depending on the size of the firmware and your own network speed.
6. Follow the firmware installation progress; once it's over, restart KalliopePBX and access the configuration interface by opening http://<Configured_ip> (default http port 80) on your browser. The default credentials are admin/admin.

2.1.2 V4R/V4R+

KalliopePBX v4 Rackmount (Hw rev. 3)

The KalliopePBX v4 rackmount appliances (in the different -R, -R+, -R-OPT-FO and -R-OPT-FO+ variants) in hw rev. 3 are based on a headless hardware platform that provides:

- one Gigabit Ethernet network interface equipped with an 8-port hardware switch, all Gigabit Ethernet autosensing (ETH0)
- two single-port Gigabit Ethernet network interfaces (each dual mode RJ45 and SFP 1GbE) (ETH1 and ETH2).

Note: When GBIC is used in the SFP slot of one of these interfaces, if the network cable is left plugged into the RJ45 port, the latter will take precedence.

Note: In the case of using GBIC 100 Mbps, the system does not recognize them automatically and an enabling operation by our technical support is required.

Once the device is connected via the supplied power supply, the system starts up automatically and within a few seconds you will be able to access the KalliopePBX.

The first access to KalliopePBX is via one of the switch ports on the ETH0 interface, pointing the browser to the default address http://192.168.0.100:10080.

Note: Some browsers have started to block access to pages on ports other than the standard ones by default. If an error indicating "ERR_UNSAFE_PORT" is displayed, it is necessary to unblock port 10080, using the procedure given on page *Resolution of ERR_UNSAFE_PORT*

As with VMs and other physical equipment, the IP address of KalliopePBX can be changed by connecting a PC configured with an IP address on the 192.168.0.0/24 subnet to the ETH0 interface and accessing the "Network Settings" panel from the Tools menu, accessible at the top right of the Kalliope GUI.

Alternatively, it is possible to access the KalliopePBX console (operating only in "recovery" or "recovery console" mode, in which the system is thus booted on bootloader and not firmware) and perform configuration using the built-in CLI (accessed with the default manager/manager credentials). Unlike previous hardware versions, access to the KalliopePBX console is not directly through the serial port on the device, but through the VNC service.

In this hardware revision, in fact, KalliopePBX runs as a virtual machine within a KVM-based virtualization environment. Access to KalliopePBX is therefore mediated by the native operating system running on the physical machine, through which some simple configuration tasks can be performed, as detailed below.

In case the KalliopePBX web interface can be accessed with the default network settings (192.168.0.100/24), the initial activation procedure follows the standard one common to other versions, and consists of the following steps:

1. Configuring the IP address, default gateway and DNS via web GUI and re-accessing the newly assigned IP address. Verification of correct date/time setting and possible forced synchronization using system NTP servers (or personal ones possibly configured).
2. KalliopePBX VM license activation. In this case, unlike traditional VMs, it is not necessary to enter an activation key because this will be available on board the device, so activation consists only of pressing the corresponding button from the web interface
3. Product Registration. Similarly to other Kalliope systems, this operation unlocks the possibility of proceeding with the installation of firmware, and determines the beginning of the expected warranty period on the apparatus and access to updates

4. Update bootloader, if necessary, and install firmware. After the firmware installation is complete, it will be necessary to reboot from the KalliopePBX GUI, and once the boot is complete, redirection to the login page (on standard port 80) will be automatically performed

In case you cannot access the default IP 192.168.0.100 and you need or want to proceed with its modification using the KalliopePBX CLI, you still need to have access to the device's native operating system in order to configure VNC access to the KalliopePBX console. Specifically, once you have configured an IP to the native operating system, based on the directions in the next section, simply start a VNC client (e.g., TightVNC Viewer) by configuring the connection to the host <IP_OS_native:5900>, to gain access to the KalliopePBX CLI on which you can login (using the wired user manager/manager)

Access to the apparatus' native ATOS operating system

Access to the device's native operating system (ATOS) may be necessary to configure its network settings in order to gain access via VNC to the KalliopePBX console, as well as to be able to perform forced shutdown and reboot of KalliopePBX where necessary.

Access to ATOS can be through the serial ("Console") port on the front of the device (with 115200/8/N/1 settings), or through ssh access. Similarly to the out-of-band management systems of many servers (iLO for HP, DRAC for Dell, etc.) ATOS has its own network stack, which by default is configured as a DHCP client on the ETH0 interface (with a MAC of the 00:D0:D6:xx:xx:xx family, as listed on the label on the device).

The login credentials are listed on the label on the apparatus. The user is "manager" and the password is uniquely assigned to each apparatus (later changeable after accessing ATOS).

Note: this manager user is unrelated to the manager user you use to access the Kalliope CLI in bootloader mode.

Once the connection has been made via serial console, or via ssh (if the IP acquired by ATOS is known, e.g., thanks to the logs of the DCP server that may be present in the network), proceed with the login using the user "manager" and the password given on the label on the device:

```
login as: manager
manager@192.168.23.60's password: *******
ATOSNT Remote CLI

CTRL+d to exit

Init Command Line Interface...
ATOS Version: 7.2.7 (mcccfvitbkovsc)
ATOS Date: 06/10/2022 13:34
ATOS License: NFVSBC
Hardware: XV8800 - 4C8R120D - 2518D
Product Code: 708190501
Serial Number: xxxxxx
MAC Address: 00:D0:D6:xx:xx:xx
BARCODE: AE70819050110204xxxxxx

User name :manager
Password :*******
<manager> logged at MANAGER level
KPBXv4_kvm>
```

By typing "?" followed by the enter key, the system presents the available configuration items and commands:

```
KPBXv4_kvm>?

Available nodes:
                        system
                        interfaces
                        dns
                        nfv
Available commands:
up                      Move one step up from the current node
top                     Back to the root of the tree
quit                    Exit from CLI session
set                     Set node options
add                     Add a new option
del                     Remove an added option
show                    Show 'KPBXv4_kvm' settings
help                    Help of item
info                    Show the system informations
date                    Show or setting system date and time
save                    Save configuration data
restart                 Restart device
ping                    Send an ICMP ECHO request
tracert                 Display a trace of packet
show-logging-level      Show logged level

password                Set user/others password on node KPBXv4_kvm\system>>
```

Below are the commands for displaying the current settings, then followed by the device commands, with which to make changes to the ATOS configuration.

Note: all commands have online help, simply complete the current command with the "?" character to have the system inform you of all the possible options available for that command. For example:

```
KPBXv4_kvm>set interfaces vswitch-0 ip ?

Set command parameters:
ip address          [address]             Current value: 0.0.0.0
default router      [defaultrouter]       Current value: 0.0.0.0
dhcp client         [dhcp-client]         Current value: on
```

In addition, the "Tab" key performs the completion of the entered command:

```
KPBXv4_kvm>set interfaces vswitch-0 ip dh<tab>
```

produces:

```
KPBXv4_kvm>set interfaces vswitch-0 ip dhcp-client
```

In the case of multiple autocompletion possibilities, repeated pressing of the "Tab" key causes it to cycle through all possible options.

ATOS status information display commands

Display assigned IP address and interfaces status

```
KPBXv4_kvm>show interfaces status

Show status of KPBXv4_kvm interfaces
INTERFACES      IPV4-ADDRESS    MAC-ADDRESS        STATUS     PROTOCOL  STATUS-DETAILS  ␣
↪                           VRF
eth0            unassigned      00:D0:D6:xx:xx:xx  up         up        operational/
↪running/bundled                global
eth1            unassigned      00:D0:D6:xx:xx:xx  down       down      operational/not␣
↪running                    global
eth2            unassigned      00:D0:D6:xx:xx:xx  down       down      operational/not␣
↪running                    global
tap-vdev0       unassigned      02:09:C0:17:8C:24  up         up        operational/
↪running/bundled                global
tap-vdev1       unassigned      02:09:C0:CE:2D:08  up         up        operational/
↪running/bundled                global
tap-vdev2       unassigned      02:09:C0:FA:F0:9E  up         up        operational/
↪running/bundled                global
vswitch-0       192.168.23.60   00:D0:D6:xx:xx:xx  up         up        operational/
↪running/ip4 address assigned  global
```

In this output, interfaces eth0-1-2 are the physical ports of the equipment (eth1 and eth2 are “not running” since there are no network cables connected), tap-vdev0-1-2 are logical ports used internally to interconnect the network interfaces of the Kalliope VM with the physical ports. The vswitch-0 is the network access interface to ATOS, which in this example received the address 192.168.23.60 via DHCP.

DNS server display set to ATOS

```
KPBXv4_kvm>show dns work
Show of KPBXv4_kvm dns
Enable                 : on
Level of log           : 1
Max retries            : 3
Timeout retries (sec) : 20
Local resolution       : preferred-v4

LIST OF DOMAIN NAME SERVERS
VRF                  : global
Domain Name          : anydomain
Interface            : vswitch-0
Primary address      : 8.8.8.8
Secondary address    : 4.4.4.4
Priority             : 1
```

DNS preset in ATOS are 8.8.8.8 and 4.4.4.4

NTP server display for ATOS time synchronization

```
KPBXv4_kvm>show system timesync work
Show of KPBXv4_kvm system timesync
Level of log                 : 1
Enable                       : on
Sync frequency (sec)         : 300
GMT offset (min)             : 60
Daylight saving time period  : last Sun Mar 02:00 last Sun Oct 03:00
Local IP Address             : 0.0.0.0
Local IPv6 Address           : ::
VRF                          : global

LIST OF SERVERS
Server                                  Type
it.pool.ntp.org                         sntp
```

KalliopePBX VM operating status display

```
KPBXv4_kvm>show nfv kpbxv4 status

Show status of KPBXv4_kvm nfv kpbxv4
Status     : on
UUID       : 614c20db-8219-4804-b27b-ca9afe91398c
Huge pages : 4096
Nic model  : virtio-net-pci
```

Device and configuration change commands

Warning: To make the following commands persistent upon reboot, it is necessary to execute the "save" command after the changes are complete.

Stopping/starting the KalliopePBX VM.

```
KPBXv4_kvm> set nfv kpbxv4 off
KPBXv4_kvm> set nfv kpbxv4 on
```

IP address management interface vswitch-0 of ATOS

Disabling/enabling DHCP client (**attention**, if you are connected via ssh and not via serial, disabling the DHCP client causes the address to be released and then disconnected!)

```
KPBXv4_kvm>set interfaces vswitch-0 ip dhcp-client off
KPBXv4_kvm>set interfaces vswitch-0 ip dhcp-client on
```

Setting static IP address

```
KPBXv4_kvm>set interfaces vswitch-0 ip address ?

  ip address  [aa.bb.cc.dd[/0-32]]

  Current value:    0.0.0.0
  Default fw value: 0.0.0.0
```

example:

```
KPBXv4_kvm>set interfaces vswitch-0 ip address 192.168.55.200/24
```

Setting default gateway (for possible remote access):

```
KPBXv4_kvm>set interfaces vswitch-0 ip defaultrouter ?

default router  [aa.bb.cc.dd]

Current value:    0.0.0.0
Default fw value: 0.0.0.0
```

example:

```
KPBXv4_kvm>set interfaces vswitch-0 ip defaultrouter 192.168.55.1
```

Change DNS servers used by ATOS (used for resolution of configured NTP servers)

In this case, the "del" command is used to remove the configured DNS set, and the "add" command is used to add a new set of servers:

```
KPBXv4_kvm>del dns SERVER anydomain 1
```

This command removes the priority 1 set (the only one present by default) with which DNS with IPs 8.8.8.8 and 4.4.4.4 are associated. Information about the defined priorities and their associated servers can be obtained with the command "show dns work"

To add a new set of DNS servers, we instead use the command

```
KPBXv4_kvm>add dns SERVER anydomain 8.8.8.8 4.4.4.4 1
```

Edit NTP servers used by ATOS

Again, the "add" command is used to add a new server and the "del" command is used to remove the one to be deleted:

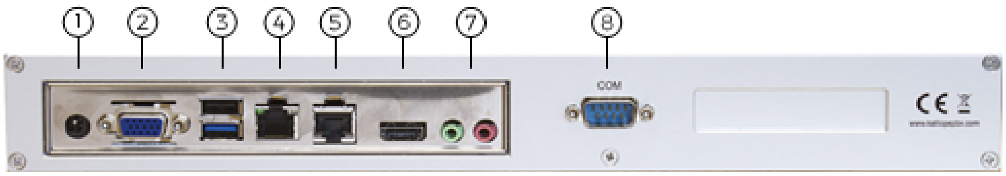
```
KPBXv4_kvm>del system timesync it.pool.ntp.org
```

```
KPBXv4_kvm>add system timesync ntp1.inrim.it
```

If desired, multiple NTP servers can be added, without necessarily deleting the existing one.

KalliopePBX v4 Rackmount (Hw rev. 1 e 2)

Port description



- Power (12v, 5A)
- VGA (unused)
- USB 2.0/USB 3.0
- Gigabit Ethernet (eth0)
- Gigabit Ethernet (eth1)
- HDMI (unused)
- Audio jack (unused)
- COM1 (Console 115200 8/n/1)

First installation

- Connect KalliopePBX to your network using the eth0 interface.
- Optional: connect KalliopePBX's COM1 serial port to your PC using a serial cable (not included - settings: 115200 8/n/1).
- Connect the included power cord to KalliopePBX and plug it into a power socket (100-240V AC, 50-60 Hz).
- Turn on KalliopePBX by pressing the power button in front; should it be unexpectedly disconnected from the power supply, it will automatically reboot once power is restored.
- Open http://192.168.0.100:10080 on your browser to register. KalliopePBX will require access to https://license.kalliopepbx.it; the network configuration can be edited from the "settings" menu. In case the error UNSAFE_PORT appears follow this procedure: Click here
- After registering, choose and install the desired firmware from those available. KalliopePBX will require access to the address https://updates.kalliopepbx.it. The process may take a few minutes, depending on the size of the firmware and your network speed.
- Follow the firmware installation progress; once it's over, restart KalliopePBX and access the configuration interface by opening http://<Configured_ip> (default http port 80) on your browser. The default credentials are admin/admin.

2.1.3 Virtual Machine

VMWare

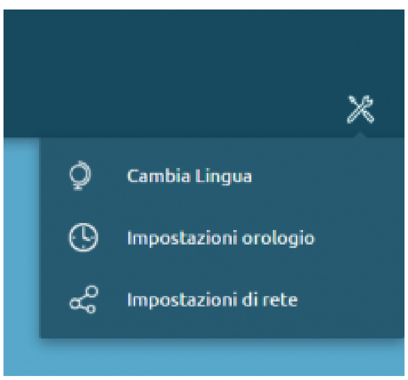
1. Download the KalliopePBX virtual image file. KalliopePBX VM is available in VMWare ESXi 5.x environment or later. The two images differ for the maximum number of virtual CPUs that are visible from the kernel (up to 8 for the second, and over 8 up to a theoretical maximum of 128 for the first) and for the size of the system disk, which for the first is 120GB while for the second is limited to 40GB (values that can be increased with a simple procedure executable from the VMWare console).

Image	Download link
KalliopePBX v4 (supports over 8vCPU)	https://areaclienti.vianova.it/drive/download/BeWHIH36hbF12cM1/ (KalliopePBXv4_esxi5_128vcpu_120GB.ova)
KalliopePBX v4 (supports a maximum of 8vCPU)	https://areaclienti.vianova.it/drive/download/bmOBoFRXDaSlcH2K/ (KalliopePBXv4_esxi5_bl-1.0.5-7156.ova)

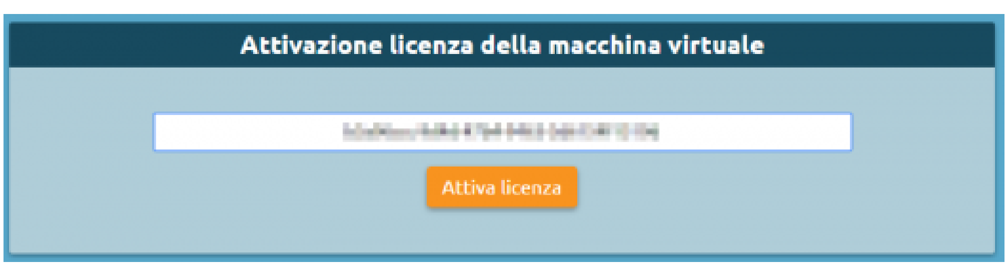
2. Import the downloaded image file into the virtual environment. Upon booting for the first time, the VM will start in recovery mode, i.e. with only the bootloader and no phone firmware.
3. Access the predefined address 192.168.0.100:10080 from the GUI. In case UNSAFE_PORT error appears follow this procedure, (*click here*)

Note: A console access is available (with manager/manager credentials, accessible only in recovery mode) that allows you to make the change of the IP address of the eth0 interface, in case you cannot access the default viq web browser address.

4. Go to the Tools menu (click on the tools icon) in the top right to change your network configuration so that the PBX can access the internet and reach the license server.



5. Activate the KalliopePBX VM license with the activation key you were provided.



6. Once the license has been activated, click on "Register product" to access the Recovery console.

7. Follow this procedure to update the bootloader to the latest version, (click here)

8. Once the update is finished, click "Restart bootloader" to restart the machine. The machine will always restart in recovery mode.

9. Click "Find updates" again to download the list of available firmware updates and select the version you want.

10. Click on "Install on the primary partition (p5)". KalliopePBX will warn you that this will reset the partition erasing all data and ask for confirmation.

Since this is the first installation, confirm the request.

Note: Installing firmware from the recovery console will restore the machine to factory conditions. This can be useful when you need to completely reset a machine.

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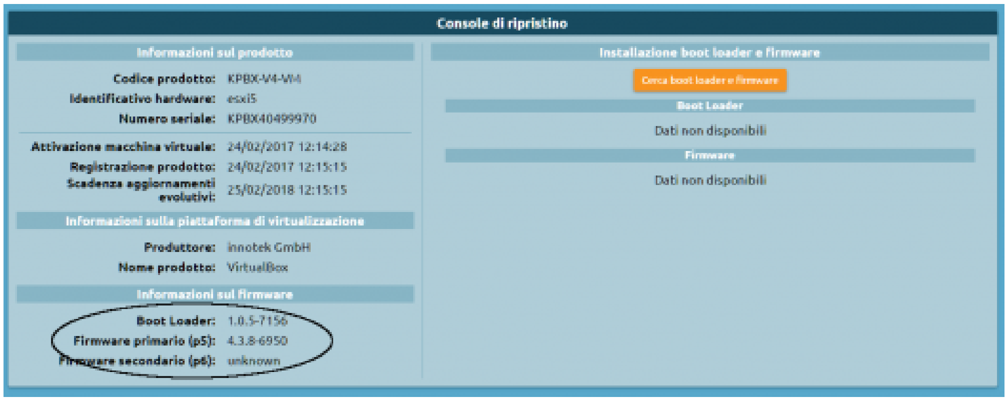
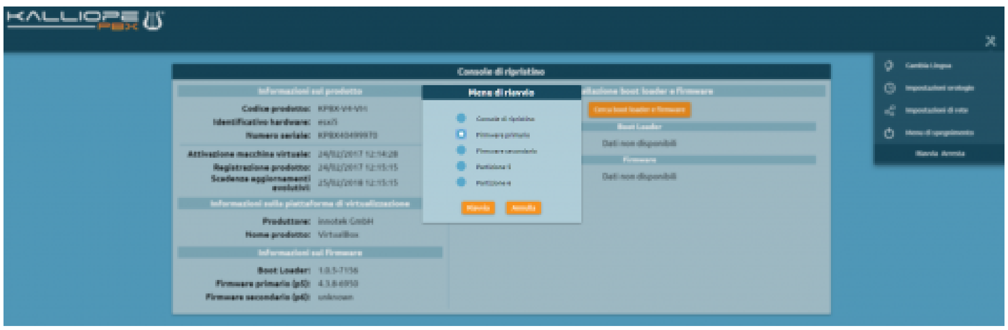
11. Once the firmware installation is finished, verify the firmware version number from the recovery console.



12. Open the tools menu and click on the "Shutdown menu -> Restart" item, select "Primary firmware" from the restart menu, and click "Restart".



13. After restart, you will be redirected to the main login (on port 80).

The default login credentials are

- Username: admin
- Password: admin

You are advised to change the password immediately after restarting.

Promox

Update - 10/05/2020: A new Kalliope image for Proxmox VE environment is now available, which overcomes the limitations of the previous image to use a maximum of 8 vCPUs and to have a default disk size of 22GB. The new image, prepared on Proxmox VE version 6.1 environment and having a 120GB disk, handles up to a maximum of 128 vCPUs

Note: It is not possible to upgrade the VM with 8vCPUs to the one that supports more than 8, so in case of new installations it is recommended to install the new image anyway, even if it will be assigned a number of vCPUs equal or lower than 8

To install KalliopPBX on Proxmox environment you first need to download the two files: .vma.lzo and .log in the folder: /var/lib/vz/dump/

Image	Download Link
KalliopePBX v4 (max 128 vCPU)	https://areaclienti.vianova.it/drive/download/6qmKASR4EfpHbG11/ (vzdump-qemu-100-2020_05_10-23_03_40.vma.lzo - 1,4 GB) https://areaclienti.vianova.it/drive/download/F9zgxpv8UAhcJerD/ (vzdump-qemu-100-2020_05_10-23_03_40.log)
KalliopePBX v4 (max 8 vCPU)	https://areaclienti.vianova.it/drive/download/0hh0XCroc9AOsFzt/ (vzdump-qemu-200-2017_06_28-10_06_39.vma.lzo - 1,0 GB) https://areaclienti.vianova.it/drive/download/vcZYocwlyIeqoHHd/ (vzdump-qemu-200-2017_06_28-10_06_39.log)

Once copied the two files related to the image you want to install inside the /var/lib/vz/dump folder of the host machine (keeping the original names, indicated above), the restore command must be executed in a new VM: .. code-block:: console

qmrestore vzdump-qemu-<versione>.vma.lzo <nnn> –storage <storage_name>

Where:

- **<nnn>** is the id to be assigned to the VM (must not be already assigned to other VMs in the Proxmox node/cluster).
- **<storage_name>** is the name of the storage in which you restore the VM to (normally it is "local" or "storage").

The VM is now operational and you can boot it and access the console to configure the network address.

Note: Before starting the virtual machine, it is necessary to change the VM settings by enabling the "KVM hardware virtualization" flag, which is disabled in the exported image, otherwise performance will be extremely slow. The new image already has this flag enabled by default, so it is no longer necessary to change this setting; however, it is recommended to check its actual value.

Note: The VM is deployed with a generic vCPU model, which emulates a Pentium 4 processor, for maximum compatibility. If you use the VM on a single node, or you have homogeneous Proxmox VE nodes in terms of physical CPUs, you can change the processor model to the "host" value, which directly remaps the functions of the physical processor, thus making available to the OS of the VM all the hardware extensions present on the server CPUs. In this regard, refer to the Proxmox documentation page, section "CPU Type": https://pve.proxmox.com/wiki/Qemu/KVM_Virtual_Machines#_emulated_devices_and_paravirtualized_devices

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After this, you can operate as with other VMs, i.e. by accessing with manager/manager credentials, and a wizard will let you configure the network settings.

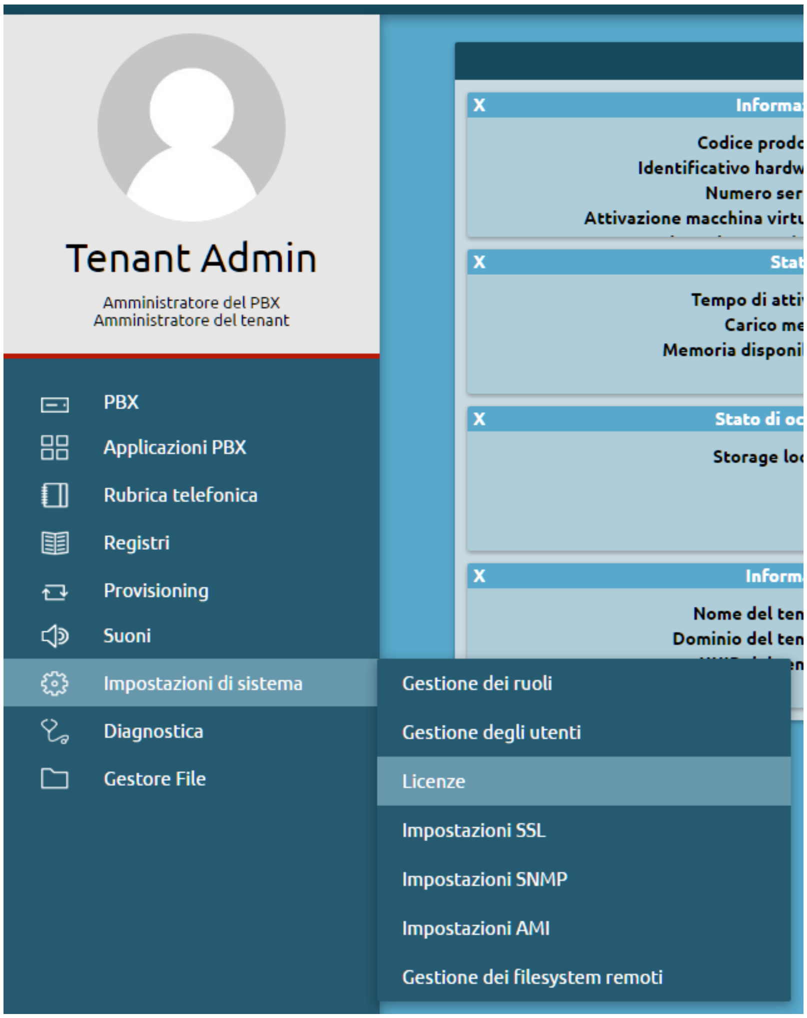
Once on the network, activation and configuration is done via web interface (initially on port 10080, and after installation of the phone firmware on standard port 80). In case UNSAFE_PORT error appears follow this procedure (*click here*)

The VM must be able to reach in HTTP our license and update servers (license.kalliopepbx.it and updates.kalliopepbx.it), which currently both resolve the IP address 77.72.27.4.

2.1.4 Multitenant

To activate a Kalliope Multitenant license, follow these steps:

1. If you wish to activate the license on a new machine, begin by following the instructions for activating a KalliopePBX VM license with the license key you were provided. If you wish to activate a Kalliope Multitenant license on a KalliopePBX VM that was previously set up, skip to step 2.
2. Log into the KalliopePBX administration GUI with "admin" credentials and open the license management page from the main menu.



3. Click on "Activate new license" and insert the "KalliopePBX Multitenant" license key.

Click on "Activate".

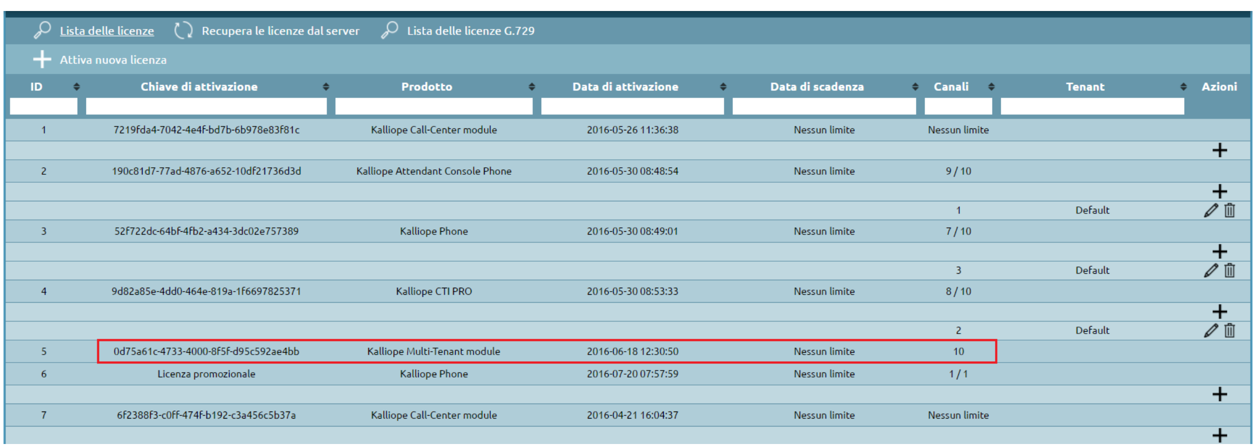
4. After activating the license, you will need to log in again to unlock multitenant functionality. Before restarting you will be asked which limitations, if any, to set for the "default" tenant, i.e. the preexisting one set up during firmware installation (see step 1).

5. To manage tenants, log in with the following credentials:

- Username: pbxadmin
- Password: admin

You are advised to change the password for the user "pbxadmin" immediately after login.

6. Once you have logged in as pbxadmin, you can view the new license from the Licenses menu.



During the activation of a multitenant license, KalliopePBX executes a series of internal procedures; some are automatic, while others require the admin to input the necessary parameters.

When operating in its default single tenant mode, KalliopePBX is already a multitenant system with a single predefined tenant enabled (associated to the "default" domain); the pbxadmin and the tenant admin (see "Users and roles") are simply combined into a single user, "admin".

When a multitenant license is activated, the two users are made distinct, separating the admin's configuration privileges (the operation of the single tenant) from those of the new user "pbxadmin". The users of a single tenant are now identified by the domain of the tenant they belong to (in this case, "default", so the admin becomes admin@default, and so on) so they can have the freedom to pick any username within the tenant.

At the same time, the external SIP lines (gateways, domains, trunks, terminations) and the tenants are decoupled, as the former are controlled by the pbxadmin and their visibility to the tenants needs to be mediated. This introduces the concept of an "assigned line", a numbering or numbering range attributed to an outbound line, which is assigned to the exclusive use of a single tenant.

Upon the activation of a multitenant license, should there be any inbound/outbound lines already configured on the PBX, assigned lines will be created (one for each outbound line) and attributed to the "default" tenant. In this phase, it is possible to specify which numbering to present to the tenant, in order to reserve others for use by other tenants.

Assigning lines to the tenants works similarly to configuring DID on the inbound lines of a single-tenant system; the difference is that the destination will not be an entity such as an extension, queue, group, or other, but rather an entire tenant. Routing to the final destination will be carried out according to the configuration of the tenant, following the concept of DID associated to a single assigned line.

> **Warning:** During the conversion from single-tenant to multitenant lines, it is important to consider that all inbound manipulation rules (to a gateway or a domain), which in a single-tenant scenario were implemented before applying the DID routing rules, will remain associated to the inbound lines and will not be transferred to the tenant. It follows that the rules for assigning lines to tenants MUST be implemented by configuring the resulting selections AFTER the inbound manipulation rules.

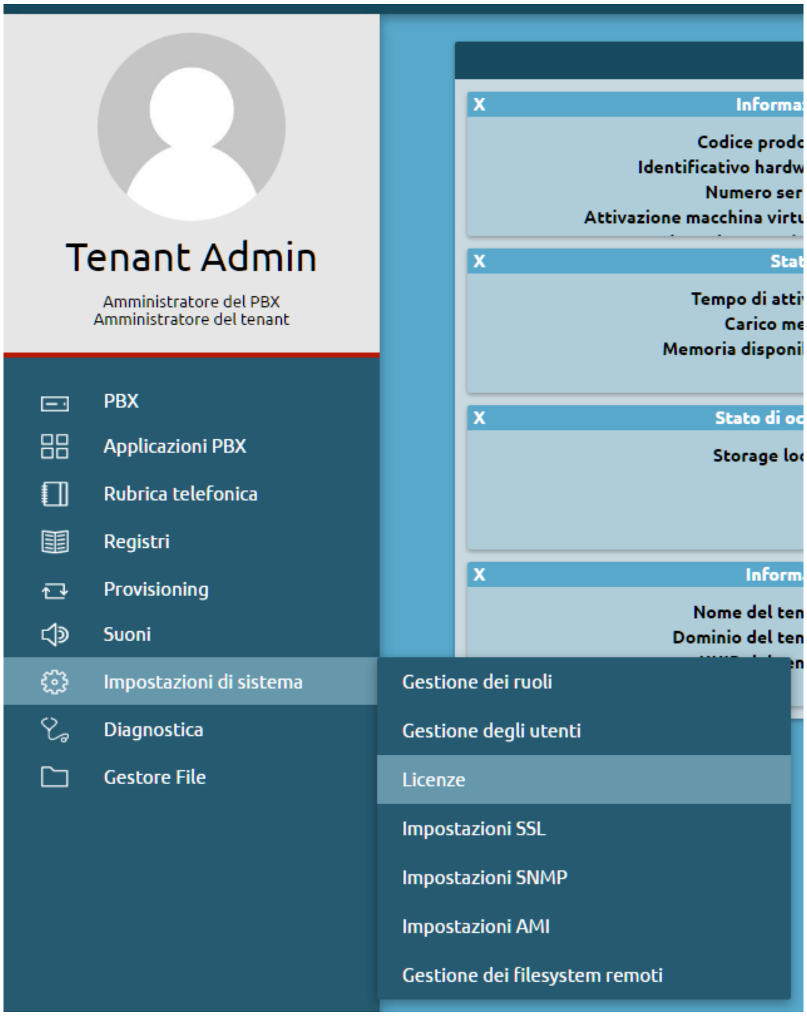
Similarly, any outbound rule will be applied on a PBX level after the call has left the tenant and has been attributed to a specific outbound line.

Within each tenant there are further manipulation tables (both inbound and outbound) associated to the single assigned line attributed to it, allowing both the pbxadmin and the admin to implement the necessary modifications to calling and called numbers during the various phases of call management.

2.1.5 Kalliope4SP

To activate a Kalliope4SP license, follow these steps:

1. If you wish to activate the license on a new machine, begin by following the instructions for activating a KalliopePBX VM license with the kpbx-v4-vm-null (cod. KPBX-V4-VM-NULL) license key that was provided along with the KalliopePBX Starter Kit for Service Provider (cod.KPBX-V4-4SP-SK) license. If you wish to activate a Kalliope4SP license on a KalliopePBX VM that was previously set up, skip to step 2.
2. Log into the KalliopePBX administration GUI with "admin" credentials and open the license management page from the main menu.



3. Click on “Activate new license” and insert the “KalliopePBX Starter Kit for Service Providers” license key. Click on “Activate”.

4. After activating the license, you will need to log in again to unlock multitenant functionality. Before restarting you will be asked which limitations, if any, to set for the “default” tenant, i.e. the preexisting one set up during firmware installation (see step 1).

5. To manage tenants, log in with the following credentials:

- Username: pbxadmin
- Password: admin

You are advised to change the password for the user "pbxadmin" immediately after login.

6. Once you have logged in as pbxadmin, you can view your license from the Licenses menu, valid for 12 months with no limitations, as per the Kalliope4SP offer.

ID	Chiave di attivazione	Prodotto	Data di attivazione	Data di scadenza	Canali	Tenant	Azioni
1	Licenza promozionale	Kalliope Phone	2017-02-24 11:41:50	Nessun limite	0 / 1		
					1	Default	
2	29928278-352d-49da-ba08-dd76ceeb1ec5	Kalliope For Service Provider Starter Kit module	2017-02-24 13:17:30	2018-02-24 13:17:30	N/D		
3	29928278-352d-49da-ba08-dd76ceeb1ec5-1	Kalliope Multi-Tenant module	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
4	29928278-352d-49da-ba08-dd76ceeb1ec5-2	Estensione linee ingresso/uscita	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
5	29928278-352d-49da-ba08-dd76ceeb1ec5-3	Estensione interni	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
6	29928278-352d-49da-ba08-dd76ceeb1ec5-4	Estensione account	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
7	29928278-352d-49da-ba08-dd76ceeb1ec5-5	Kalliope Call-Center module	2017-02-24 13:17:30	2018-02-24 13:17:30	N/D		
							+
8	29928278-352d-49da-ba08-dd76ceeb1ec5-6	Kalliope CTI PRO	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
							+
9	29928278-352d-49da-ba08-dd76ceeb1ec5-7	Kalliope Phone	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
							+
10	29928278-352d-49da-ba08-dd76ceeb1ec5-8	Kalliope Attendant Console CTI	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
							+
11	29928278-352d-49da-ba08-dd76ceeb1ec5-9	Kalliope Attendant Console Phone	2017-02-24 13:17:30	2018-02-24 13:17:30	Nessun limite		
							+

CHAPTER

THREE

KALLIOPE ADMINISTRATION GUIDE

3.1 Kalliope Administration Guide

3.1.1 Basic concepts

Basic Features

- SIPv2 (UDP, TCP, TLS, and WebSocket; RTP and SRTP)
- Supported audio codecs (with transcoding): G.711 (A.law, u.law), G.726, GSM, G.722 (wideband), G.729, Opus
- Supported video codecs (passthrough, no transcoding): VP8 H.264, H.263+, H.263, H.261
- Fax support (audio or T.38 passthrough)
- Busy Lamp Field
- ENUM support
- Access control list for extensions (ACL)
- SNMP (v1/v2c) read access support (Net-SNMP daemon)
- LDAP support (both client and server)

Extensions and SIP accounts

Extensions are the primary telephone entities. An extension is a logical entity identified by its number, which is used as the identifier for all calls made by the extension and is the number dialed by other users who wish to contact the extension. Each extension has a number of attributes that define its permissions (for outgoing calls from the extension) and behavior (for incoming calls to the extension), as well as a few identity records (first and last name of the person to whom the extension is assigned, its organizational unit, and so on).

SIP accounts are "service" entities. They are the credentials (username and secret) that need to be configured on a SIP terminal (hardphone or softphone) so that it can authenticate to the PBX. Authentication is performed by the devices with two procedures: "SIP registration" and the execution of a new call.

The relationship between extensions and SIP accounts is one-to-many: each extension can be linked to multiple SIP accounts, which all behave as the same telephone entity in terms of identity, presentation, permissions, etc.

It is also possible to create a unique user for each extension; different permissions and roles can be granted to these users, in order to allow them to access their personal web page, perform some administrative or configuration tasks, use the KalliopeCTI applications (desktop or mobile), and invoke the available REST APIs. Please see the users and roles page for more details.

SIP registration and multiple devices per extension

SIP registration informs the PBX of the current location of a SIP account, i.e. the IP address and port (plus the protocol, e.g. UDP, TCP, TLS, or WebSocket) where the SIP account can be reached when the PBX needs to send it a message (e.g. an INVITE related to an incoming call). Registration is performed by the device at boot time (if the account is correctly configured) and then refreshed periodically before its validity expires; each periodic registration requires the repetition of the authentication procedure. The lifetime of the registration is established during the registration procedure itself. The device inserts a "Proposed Expiry" value (in seconds, usually defaults to 3600) in its REGISTER request; upon successful authentication, the PBX responds with a "200 OK" message that notifies the actual registration lifetime to the device, which must then send a new registration before the timeout expires (usually this new registration is performed at about half time, to allow for re-transmission in case of failures). If the registration timeout expires without the reception of a registration refresh, then the location of the account is discarded by the PBX and the calls destined to that account will fail due to it being "unavailable".

KalliopePBX stores a single location for each configured SIP account; if the same SIP account is configured on multiple devices, all active at the same time, the periodic registration messages from each continuously change the location stored in KalliopePBX. A call to the SIP account is therefore presented only to the device that registered last. It is however possible to have multiple devices which behave as a single extension by defining one SIP account for each of the desired devices, and linking all these SIP accounts to the same extension.

Extension attributes and templates

Each extension has its own set of attributes that describe its identity and behavior. Some of these attributes are specific to each device and have to be individually configured, while others can be common to all or a subset of extensions. The former include the extension number itself (which must be unique within the PBX, or within each tenant for a multi-tenant PBX) as well as personal details such as first and last name, email address, and the personal PIN code used for authentication when accessing certain PBX services. The latter include call limits and permissions, and the failover actions to be executed on a failed call to the extension, based on the origin of the call and the cause of failure.

To ease the management of these common attributes, KalliopePBX introduces the concept of an extension template: a collection of attributes and settings that can be assigned to multiple extensions. Defining multiple templates (with different settings based on the extension type) reduces the number of settings that need to be specified for each single extension and helps quickly edit the same setting for all the extensions that share the same template simply by changing the value of the setting in the corresponding template.

In the configuration panel of each extension, it is possible to override any setting inherited from the associated template if a specific exception is needed. The overridden settings are not affected by changes to the template.

SIP account attributes

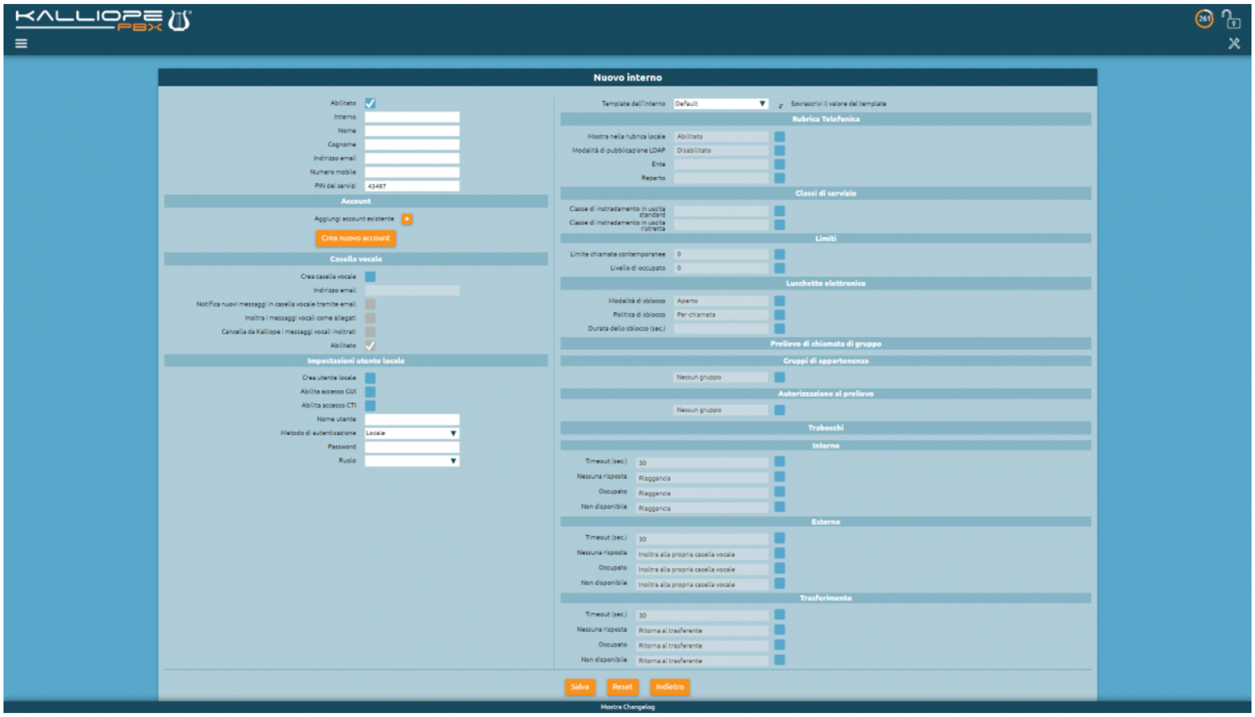
As with extensions, SIP accounts have some specific attributes (mainly the username, which must be unique within the PBX, and the SIP secret), while others may be common to a "class" of accounts. These include supported transport protocols, media, or codecs, the authorized ACL, and others; similarly to extensions, SIP account templates can be used to define classes of SIP accounts with common settings.

Extensions configuration

The extensions page contains the attributes of each KalliopePBX user. The main attribute that identifies each user is the telephone extension. If multiple devices (Accounts) are linked to the same user, these will share the telephone identity defined on this page. This means that, for example, all calls to an extension will be presented to all devices linked to the user, and all calls made from any of these devices will be made under the same telephone identity.

To configure extensions just open the operating menu and click on PBX > Extensions and Accounts. To create a new extension proceed by clicking on "Add new extension".

- **Enabled**: Lets you disable an extension without losing its configuration.
- **Extension**: The internal phone number linked to the user.
- **Name**: Part of the name displayed to other users and shown in the phonebook.
- **Last name**: Part of the name displayed to other users and shown in the phonebook.
- **Email address**: Displayed in the phonebook.
- **Mobile number**: Displayed in the phonebook and used for Fork2Mobile and FastTransfer services.
- **Service PIN**: The code needed to access telephone services that require authentication (voicemail, switches, paging, electronic lock).



Account

Parameter	Description	Value
Add existing account	Lets you link an existing SIP account to the extension	Account
Create account	Lets you create a new SIP account to link to the extension	Account

Voicemail

Parameter	Description	Value
Create voicemail box	Lets you create a voicemail box for the account	Yes / No
Email address	The address to which new message notifications are sent (optionally with audio file attached)	xxxxx@domain.yy
Email notifications for new messages in voicemail box	If this option is enabled, the user will receive email notifications for new voicemail messages.	Yes / No
Forward audio messages as attachments	If this option is enabled, audio files containing recorded messages will be sent as email attachments	Yes / No
Delete forwarded messages from Kalliope	If this option is enabled, forwarded messages will be deleted from KalliopePBX once the email is sent and will no longer be accessible from the phone and the KalliopeCTI mobile app	Yes / No
Enabled	Lets you enable and disable voicemail without losing its settings and recorded messages	Yes / No

Local user settings

Parameter	Description	Value
Create local user	Lets you create a local user for KalliopePBX in order to enable GUI or CTI access	Yes / No
Enable GUI access	Allows the user to access the web GUI as a standard tenant user. The role of the user can be changed on the users management page	Yes / No
Enable CTI access	Allows the user to use the KalliopeCTI clients. In order to use KalliopeCTI Pro or Phone, you will need to add the license from the users management page	Yes / No
Username	Username used for logging into the GUI or KalliopeCTI	Alphanumeric
Password	Password used for logging into the GUI or KalliopeCTI	Alphanumeric

Template

Parameter	Description	Value
Extension template	The template that contains the default parameters to use for the selected extension type. All attributes on the page will be changed to the default values, but it is possible to overwrite them if necessary	Extension template

Phonebook

Parameter	Description	Value
Show in local phonebook	Choose whether or not the extension is shown in the local phonebook	Yes / No
LDAP publishing mode	How the extension is published to LDAP, among the available options. The general LDAP publishing rule is set in the LDAP settings page	Disabled / LDAP publishing rule / Show the number below / LDAP publishing rule applied to the extension below
Custom LDAP extension	Extension to which the LDAP publishing rule is applied. This field will only be shown if the option LDAP publishing rule applied to the extension below is selected	Numeric
Custom LDAP number	Phone number linked to the user in the LDAP phonebook. This field will only be shown if the option Show the number below is selected	Numeric
Organization	Used when publishing the phonebook (corresponds to the organization attribute when publishing to LDAP)	Alphanumeric
Department	Used when publishing the phonebook (corresponds to the organizationUnit attribute when publishing to LDAP)	Alphanumeric

Service classes

Parameter	Description	Value
Standard outbound routing class	The routing class applied to the user when the electronic lock is disabled. If the unlock mode is set to Open, this will be the class applied to all calls	Outbound routing class
Restricted outbound routing class	The routing class applied to the user when the electronic lock is enabled. If the unlock mode is set to Open, this will never be applied	Outbound routing class

Limits

Parameter	Description	Value
Concurrent call limit	The maximum number of allowed concurrent inbound and outbound calls on all accounts linked to the extension. Setting this limit to 1 will prevent the extension from accessing services such as attended transfer as the call on hold waiting to be transferred will still be considered active	Numeric (0 = no limit)
Busy level	The number of calls on all accounts linked to the extension after which the user is considered busy (the PBX will not present the call to the user's devices and answer with a 486 Busy Here SIP Message). Setting this limit to 1 for a single account will prevent inbound call notifications even if call waiting is enabled on the device	Numeric (0 = no limit)

Electronic Lock

Parameter	Description	Value
Unlock mode	The unlock mode for the extension. **Open** –> The electronic lock is always disabled. **Code** –> Code – The electronic lock can be disabled with the unlock code specified in the numbering plan. **Password** -> The electronic lock can be disabled with the unlock code followed by the service PIN for the extension.	Open / Code / Password
Unlock policy	The unlock policy for the extension. **Per call** –> The lock must be disabled before making each call. **Automatically block after the number of minutes below** –> The lock will be automatically enabled after the specified duration. **Automatically block after the number of minutes below** –> Once the lock is disabled, it will remain so until enabled again by the user.	Per call / Automatically block after the number of minutes below / Unlocked until locked by the user
Unlock duration (sec.)	Length of time during which the lock is disabled. Only applicable if the unlock policy is Automatically block after the number of minutes below.	Numeric

Group call pickup

Parameter	Description	Value
Group membership	List of groups that this extension belongs to (calls to this extension can be picked up by any extension authorized to pick up calls from one of these groups).	Pickup groups
Pick up authorization	List of groups from which this extension is authorized to pick up calls (the extension can pick up calls to any extension that belongs to one of these groups)	Pickup groups

Failover

Parameter	Description	Value
Extension	Failover action on calls from an extension (including remote extensions)	
External	Failover action on calls coming from external numbers	
Transfer	Failover action on call transfers	
Timeout (sec.)	Length of time after which the failover action will be executed in case of no answer	Numeric
No answer	A call is considered not answered after the timeout time has passed.	Hang up / Custom selection / Ask for selection / External number / Extension / Group / Queue / Checktime / IVR / Voicemail / MeetMe room
Occupied	The extension is considered occupied if it has reached the Busy Level set for the extension or if the terminal sends a 486 Busy Here SIP Response	Hang up / Custom selection / Ask for selection / External number / Extension / Group / Queue / Checktime / IVR / Voicemail / MeetMe room
Not available	The extension is considered not available if the terminal is not registered, unreachable at an IP level, or if the terminal sends a 480 Temporarily Unavailable SIP Response	Hang up / Custom selection / Ask for selection / External number / Extension / Group / Queue / Checktime / IVR / Voicemail / MeetMe room

Account configuration

In the Account panel are defined the SIP credentials that can be used by a device to register and make/receive calls through the KalliopePBX. To these credentials are associated attributes to increase security and changes in the behavior of the KalliopePBX in terms of signaling and audio streams to be associated to a specific device. These attributes are defined at account level and not at extension level because two accounts associated to the same extension but to different devices may have different requirements.

Example: I can associate to an extension an account used on a physical phone and one used on a softphone. While for the physical phone I can use codecs with higher bandwidth consumption e.g. G711a for the softphone that is used for example in teleworking I can choose to use codecs such as G729 that optimize the use of bandwidth.

To configure accounts just open the operating menu and click on PBX > Extensions and Accounts. To create a new account, click on "Accounts" in the top bar and then on "Add SIP Account".

- **Enabled**: Lets you disable an account without losing its configuration.
- **KCTI Mobile App**: Lets this account be used with the KalliopeCTI mobile app, enabling push message sending for call signaling.
- **Username**: The username used for the SIP authentication of the device.
- **Password**: The password used for the SIP authentication of the device.
- **Account template**: The template that contains the default parameters to use for the selected account.
- **Enable registration verification**: When this setting is enabled, KalliopePBX will verify that the call setup request (SIP INVITE) comes from the same IP port as the registration request (SIP REGISTER).
- **Enabled address**: The IP address or subnet from which KalliopePBX accepts registration and call setup requests.
- **Enabled subnet mask**: Completes the ACL information on base IP for registration and call setup request.
- **Enable NAT**: When this setting is enabled, KalliopePBX will ignore IP addresses in the SIP and SDP headers and always answer from the IP address and port from which it received the request. This setting must be enabled only for devices that are one NAT behind KalliopePBX and do not solve the NAT traversal issue (through STUN / ICE / ALG SIP).
- **Enable direct media**: This setting lets you establish audio flows between two PBXs in direct visibility conditions (with no NAT). If this setting is enabled, services that require RTP flow monitoring (e.g. call recording, call transfer and parking with KalliopePBX service codes) will be disabled.
- **Enable SRTP**: This option lets you enable RTP encryption support. Since keys are exchanged within SIP / SDP messages in plaintext, it is best to use SRTP along with signaling encryption through TLS.

Outbound proxy settings

Parameter	Description	Value
Outbound proxy address	Lets you set the IP address/hostname of the outbound proxy	Alphanumeric
Outbound proxy port	Lets you set the port of the outbound proxy	Numeric
Outbound proxy protocol	Lets you set the protocol used to communicate with the outbound proxy. You can only set protocols that have been enabled in the SIP settings	UDP / TCP / TLS / WS / WSS

Transport settings

Parameter	Description	Value
Enable UDP transport	Lets you enable the UDP transport protocol for SIP signaling. This setting is not available if UDP transport is not enabled in the SIP settings	Yes / No
Enable TCP transport	Lets you enable the TCP transport protocol for SIP signaling. This setting is not available if TCP transport is not enabled in the SIP settings	Yes / No
Enable TLS transport	Lets you enable the TLS transport protocol for SIP signaling. This setting is not available if TLS transport is not enabled in the SIP settings	Yes / No
Enable WebSocket transport	Lets you enable the WebSocket (HTTP) transport protocol for SIP signaling. This setting is not available if WebSocket (HTTP) transport is not enabled in the SIP settings	Yes / No
Enable secure WebSocket transport	Lets you enable the secure WebSocket (HTTPS) transport protocol for SIP signaling. This setting is not available if secure WebSocket (HTTPS) transport is not enabled in the SIP settings	Yes / No

Audio codec

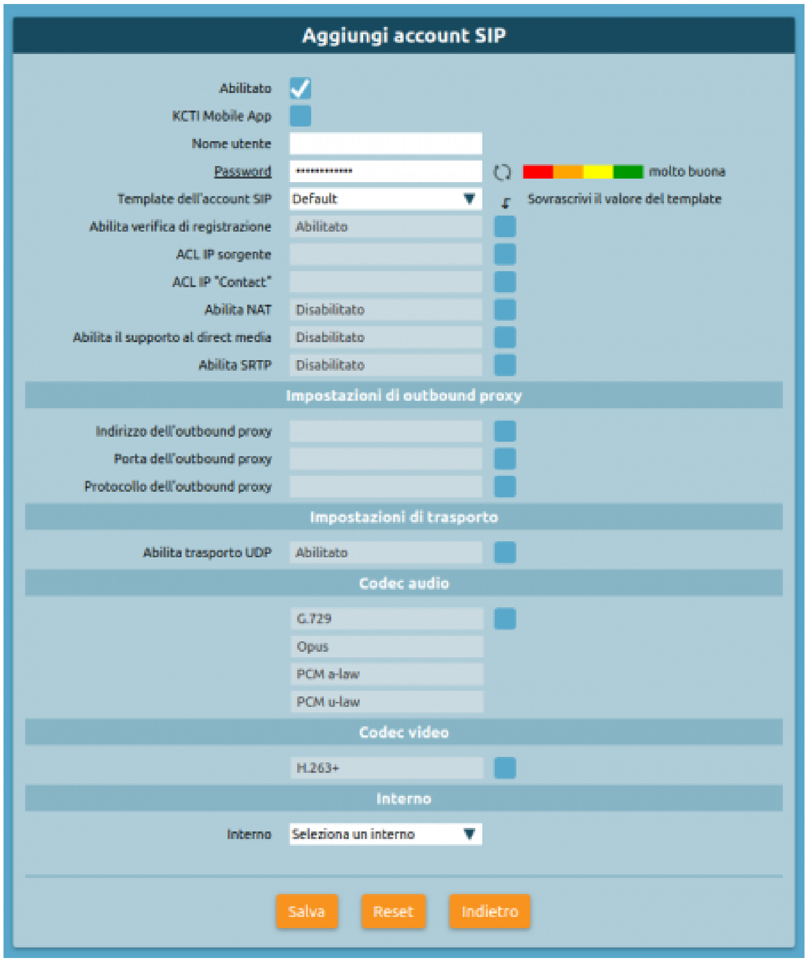
Parameter	Description	Value
Add codec	This section lets you select and organize the audio codecs usable by the account (which will be inserted into the SDP media description)	PCM a-law / G.722 / G.726 / G.729 / GSM / Opus / PCM u-law

Video codec

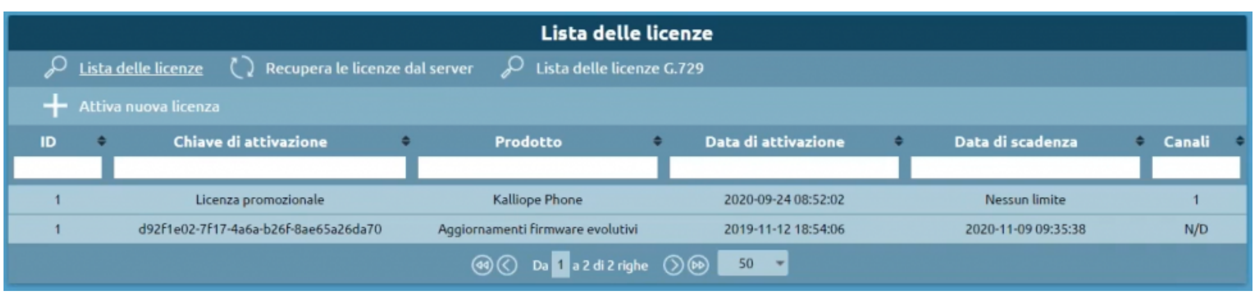
Parameter	Description	Value
Add codec	This section lets you select and organize the video codecs usable by the account (which will be inserted in the SDP media description)	H.261 / H.263 / H.263+ / H.264 /VP8

Extension

Parameter	Description	Value
Extension	The extension to which the SIP account is linked	Extension



Licenses



The license page has three sections: Kalliope licenses, Retrieve licenses from the server and G729 licenses.

List of licenses

This section shows a list of already activated licenses with the following information:

- ID
- Activation key
- Product
- Activation date
- Expiration date
- Channels

By clicking on **Activate new license** you can access a page where you can insert a valid activation key.

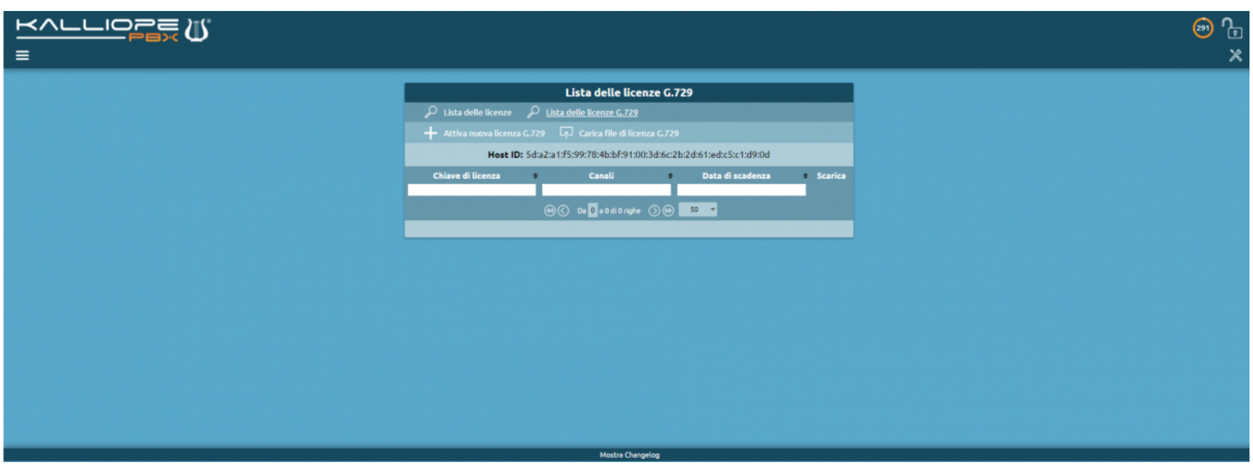
In this section you can add licenses for: Kalliope Multi-Tenant, KalliopeCTI Pro, KalliopeCTI Phone, Kalliope Attendant Console CTI, Kalliope Attendant Console Phone, Kalliope Call Center, Upgrade Mini to Lite.

Retrieve licenses from the server

Here you can view which licenses have been previously activated on a given serial. This section is divided into:

- Product licenses updated on the server
- Product licenses that can be imported automatically
- Product licenses that can be imported manually

List of G729 licenses



Similarly, the G729 licenses section shows a list of already activated licenses with the following information:

- License key
- Channels
- Expiration date
- Download

By clicking on Activate new G729 license you can access a three-step activation procedure:

1. Insert G729 activation key
2. Accept the terms and conditions
3. Insert personal information

Once you have provided the required information, you can click on Activate to complete the process.

Outbound and inbound lines

"Outbound and inbound lines" are all the SIP lines through which the PBX can make and receive calls to and from external numbers, i.e. not an internal service or extension (local SIP account).

Calls to external numbers

Calls to external numbers are not forwarded directly to the outbound lines, but are presented to the outbound routing engine. This engine decides whether the calling user/entity is authorized to perform the call (based on the destination number) and which outbound lines can be used.

Calls can reach the outbound routing engine from the numbering plan or directly as a failover action of a previous destination (e.g. an incoming call to an extension can be forwarded to an external number in case of no answer). In both cases, the requested outbound call has two associated parameters: the outbound identity and the outbound routing class.

Outbound identity

The outbound identity is the extension number used to derive the CLID for outbound calls (according to the corresponding calling number manipulation table). The Outbound Identity can be explicitly set for failover actions, while it is automatically assigned for call smade from a SIP account or for transferred/forwarded calls:

- **Calls made by a SIP account**: the outbound identity is set to the extension linked to the SIP Account.
- **Calls forwarded by a device** (telephone-driven call forwarding): same as above.
- **Calls forwarded using the KalliopePBX (unconditional) call forwarding service**: the outbound identity is set to the forwarding extension number.
- **Transferred calls** (using KalliopePBX star-codes or telephone functions): the outbound identity is set equal to the transferring extension number.

In all these cases, if the caller requests to present itself as anonymous (according to the different CLIR supported methods), the outbound identity retains the extension number throughout all the lifetime of the call, and the actual calling number restriction is performed when placing the call to the outbound line (or to the destination SIP accounts for local calls).

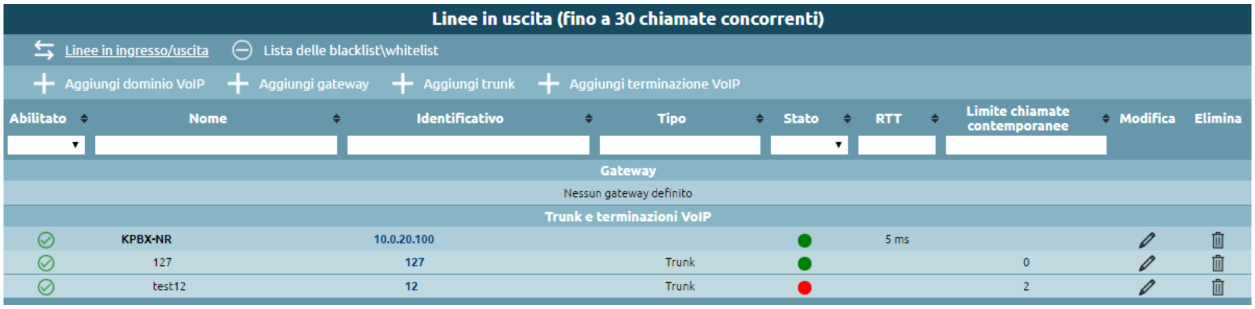
Outbound routing class

The outbound routing class defines the actual handling of the call, i.e. whether or not it is allowed, and if so the sequence of outbound lines to be used to perform the call.

Except when explicitly set by a failover action, the choice of outbound routing class is automatically derived from the outbound identity.

Configuration

Lines can be configured in the VoIP Gateways and Domains panel.



The "Gateways and VoIP Domains" screen collects the configuration of all input/output lines from the PBX.

KalliopePBX supports both physical gateways (which interconnect the internal telephone network to analog, ASDN, or GSM lines) and VoIP terminations and trunks, using the standard SIP protocol.

It is also possible to configure multiple gateways and VoIP terminations or trunks simultaneously. Through this page you can:

Add a VoIP domain or edit an existing one

For every VoIP domain, you can set up any combination of VoIP terminations and trunks.

For outbound lines, it is possible to use each as an independent line. For inbound lines, all calls from that domain will be handled by the "VoIP domain" entity, which contains all the routing rules related to the numbering of the trunks and terminations that belong to it.

The following table lists the configurable parameters for a VoIP domain.

Parameter	Description	Value
Enabled	Lets you disable a VoIP domain without losing its configuration	On / Off
Name	Mnemonic name assigned to the VoIP domain	Alphanumeric

SIP settings

Parameter	Description	Value
Server IP address	The address or hostname of the VoIP domain	Hostname / IP address
Server port	The port used by the VoIP domain. This can be omitted if the hostname is specified, as KalliopePBX will automatically acquire the port with an SRV query. If specified anyway, KalliopePBX will use it regardless of the SRV query	Numeric
Enable insecure port	Let you enable recognition of the origin peer of the base call based only on IP address, ignoring the source port. This setting is useful when there is a firewall between KalliopePBX and the gateway that alters the source port of SIP messages, preventing calls from correctly entering the system	Yes / No
Enable SRTP	Enables SRTP support for this VoIP domain.	Yes / No

Transport settings

Parameter	Description	Value
Transport type	Lets you choose between SIP with UDP/TCP/TLS transport or SIP with WebSocket/Secure WebSocket transport for this domain	SIP / WebRTC
Preferred transport	Drop-down menu that lets you choose the preferred transport among those enabled. Selecting a disabled transport will automatically enable it	UDP / TCP / TLS / WS / WSS
Enable UDP transport	Enable unencrypted UDP transport. Only available if "Transport type" is set to SIP.	Yes / No
Enable TCP transport	Enable unencrypted TCP transport. Only available if "Transport type" is set to SIP	Yes / No
Enable TLS transport	Enable encrypted TLS transport. Only available if "Transport type" is set to SIP	Yes / No
Enable WebSocket transport	Enable unencrypted WS (Web Socket) transport. Only available if "Transport type" is set to WebSocket	Yes / No
Enable WebSocket transport	Enable encrypted WSS (Secure Web Socket) transport. Only available if "Transport type" is set to WebSocket	Yes / No

Advanced settings

Parameter	Description	Value
Extract number from "To:" header	Extract the called number from the "To:" header instead of the "Request-URI". Required by some VoIP providers	Yes / No
DTMF mode	Choose how DTMF tones are sent to this gateway, among the modes provided (RFC 2833, SIP Info, in audio). By default, this will be set to the predefined system mode	RFC 2833 / SIP Info / In audio
Respect RPID	Choose whether or not the Remote Party ID (RPID) or P-Asserted-Identity (PAI) header must be respected for calls from this domain	System default / Enabled / Disabled
RPID sending mode	Choose whether to show the identity of the calling party through P-Asserted-Identity (PAI) or Remote-Part-ID (RPID), in order to update the connected line identification presentation (COLP)	System default / Disabled / Remote-Part-ID / P-Asserted-Identity

Audio codec

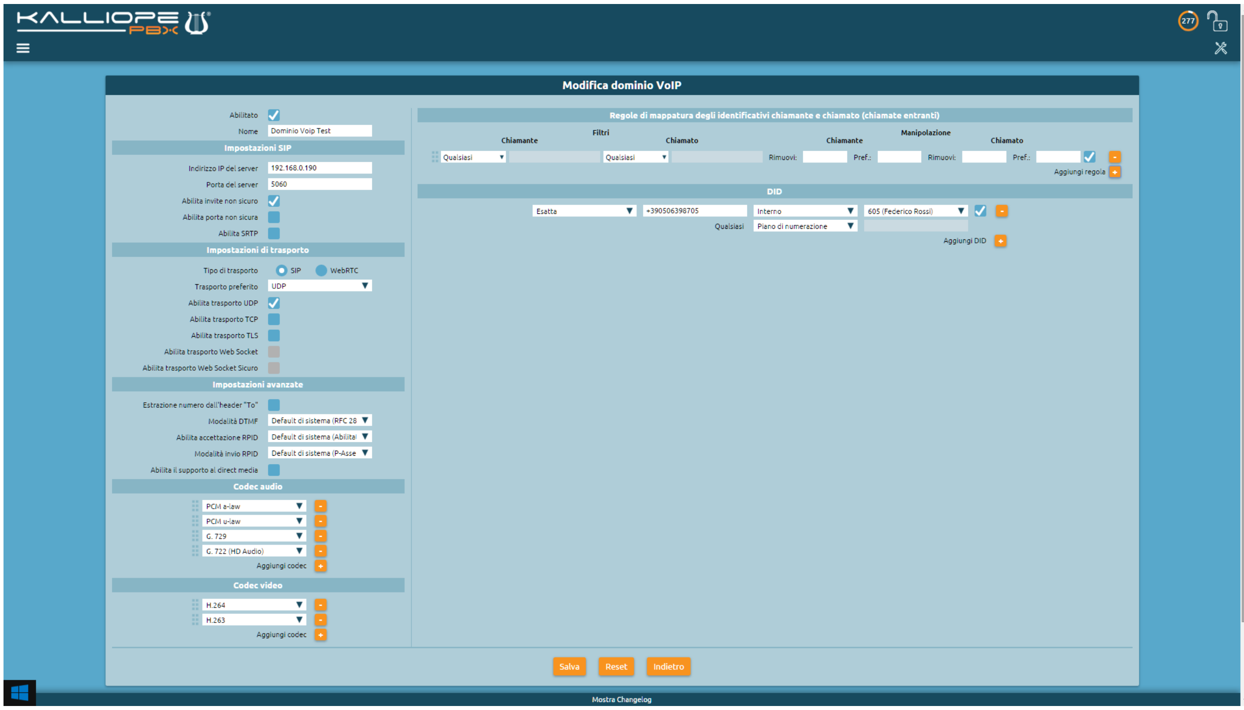
Parameter	Description	Value
	Allows you to choose the types of audio Codecs	PCM a-law / G722 (HD Audio) / G.726 / G.729 / GSM / Opus / PCM u-law

Video codec

Parameter	Description	Value
	Allows you to choose the types of video Codecs	H.261 / H.263 / H.263+ / H.264 / VP8

Caller and called identifier mapping rules (incoming calls)

Parameter	Description	Value
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Manipulation	Remove + Pref.	Numeric
Caller Manipulation	Remove + Pref.	Numeric



Add a physical gateway or edit an existing one

The following table lists the configurable parameters for a gateway.

Parameter	Description	Value
Enabled	Lets you disable a gateway without losing its configuration	On / Off
Name	Mnemonic name assigned to the gateway	Alphanumeric

SIP settings

Parameter	Description	Value
Identifier	Gateway identifier	Alphanumeric
Server IP address	The address or hostname of the gateway	Hostname / IP address
Server port	The port used by the gateway. This can be omitted if the hostname is specified, as KalliopePBX will automatically acquire the port with an SRV query. If specified anyway, KalliopePBX will use it regardless of the SRV query.	Numeric
Enable inbound registration		Yes / No
Do not request authentication for calls from this peer		Yes / No

Transport settings

Parameter	Description	Value
Transport type	Lets you choose between SIP with UDP/TCP/TLS transport or SIP with WebSocket/Secure WebSocket transport for this domain	SIP / WebRTC
Preferred transport	Drop-down menu that lets you choose the preferred transport among those enabled. Selecting a disabled transport will automatically enable it.	UDP / TCP / TLS / WS / WSS
Enable UDP transport	Enable unencrypted UDP transport. Only available if "Transport type" is set to SIP	Yes / No
Enable Web RTC transport	Enable Web RTC transport protocol	Yes / No

Advanced settings

Parameter	Description	Value
Simultaneous call limit	Number of simultaneous calls allowed	Numeric
DTMF mode	Choose how DTMF tones are sent to this gateway, among the modes provided (RFC 2833, SIP Info, in audio). By default, this will be set to the predefined system mode	RFC 2833 / SIP Info / In audio
COLP sending mode		System default / Disabled / Remote-Part-ID / P-Asserted-Identity
COLP acceptance mode		System default / Disabled / Enabled

Audio codec

Parameter	Description	Value
	Allows you to choose the types of audio Codecs	PCM a-law / G722 (HD Audio) / G.726 / G.729 / GSM / Opus / PCM u-law

Video codec

Parameter	Description	Value
	Allows you to choose the types of video Codecs	H.261 / H.263 / H.263+ / H.264 / VP8

Caller and called identifier mapping rules (incoming calls)

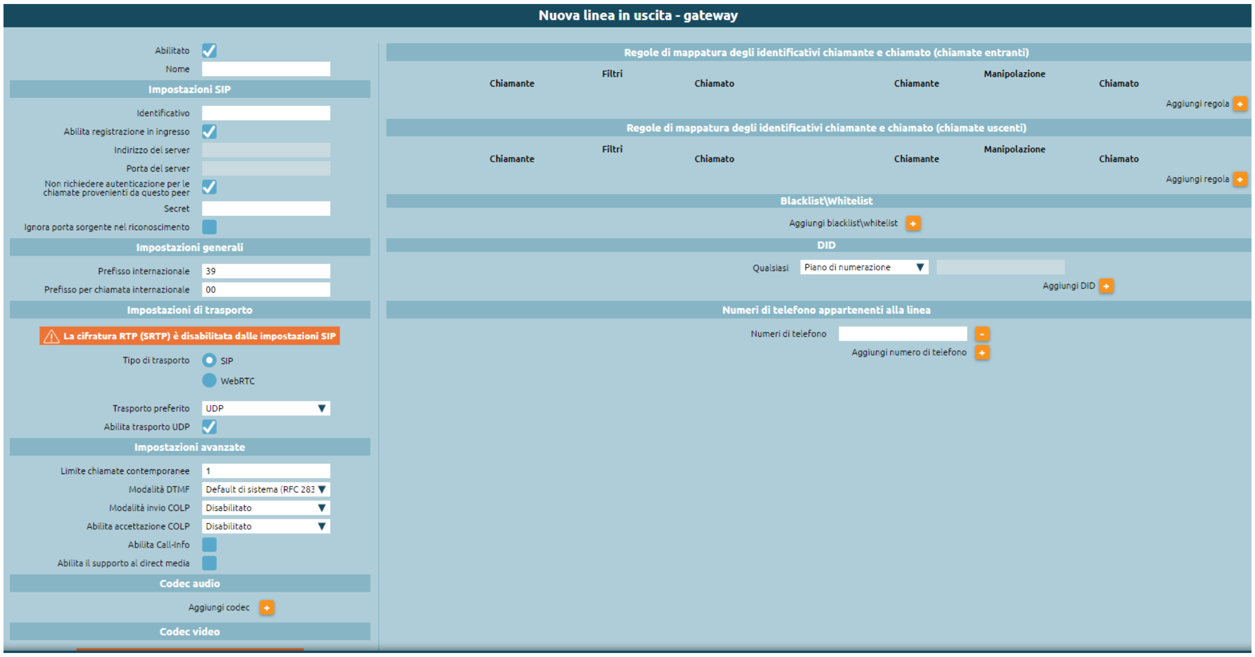
Parameter	Description	Value
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Manipulation	Remove + Pref.	Numeric
Caller Manipulation	Remove + Pref.	Numeric

Caller and called identifier mapping rules (outgoing calls)

Parameter	Description	Value
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Manipulation	Remove + Pref.	Numeric
Caller Manipulation	Remove + Pref.	Numeric

Telephone numbers belonging to the line

Parameter	Description	Value
Phone numbers		Numeric



Add a VoIP termination or edit an existing one

For every VoIP domain, you can set up any combination of VoIP terminations and trunks.

For outbound lines, it is possible to use each as an independent line. For inbound lines, all calls from that domain will be handled by the “VoIP domain” entity, which contains all the routing rules related to the numbering of the trunks and terminations that belong to it.

The following table lists the configurable parameters for a VoIP termination.

Parameter	Description	Value
Enabled	Lets you disable a VoIP termination without losing its configuration.	On / Off
Name	Mnemonic name assigned to the VoIP termination	Alphanumeric

SIP settings

Parameter	Description	Value
VoIP domain	Select the hostname of the VoIP domain to which you wish to associate the termination	Hostname
Source user	Name of the source	Alphanumeric
Username	Username with which the trunk is accessed	Alphanumeric
Secret		Alphanumeric

Transport settings

Parameter	Description	Value
Transport type	Lets you choose between SIP with UDP/TCP/TLS transport or SIP with Web-Socket/Secure WebSocket transport for this domain	SIP / WebRTC
Preferred transport	Drop-down menu that lets you choose the preferred transport among those enabled. Selecting a disabled transport will automatically enable it	UDP / TCP / TLS / WS / WSS
Enable UDP transport	Enable unencrypted UDP transport. Only available if "Transport type" is set to SIP	Yes / No
Enable Web RTC transport	Enable Web RTC transport protocol	Yes / No

Advanced settings

Parameter	Description	Value
Simultaneous call limit	Number of simultaneous calls allowed	Numeric
DTMF mode	Choose how DTMF tones are sent to this gateway, among the modes provided (RFC 2833, SIP Info, in audio). By default, this will be set to the predefined system mode	RFC 2833 / SIP Info / In audio
COLP sending mode		System default / Disabled / Remote-Part-ID / P-Asserted-Identity
COLP acceptance mode		System default / Disabled / Enabled

Video codec

Parameter	Description	Value
	Allows you to choose the types of video Codecs	H.261 / H.263 / H.263+ / H.264 / VP8

Header SIP

Parameter	Description	Value
Set the user part of the From URI as		Caller number after manipulation / Caller number before manipulation / Authentication username / Custom.
Sets the display name in the From header to the user part of the From URI		Yes, except to remote extensions / No (keep the original display name) / Yes

Show available placeholders

Parameter	Description	Value
P-Asserted-Identity / P-Preferred-Identity / Remote-Party-ID / Call-Info		

Caller and called identifier mapping rules (outgoing calls)

Parameter	Description	Value
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Manipulation	Remove + Pref.	Numeric
Caller Manipulation	Remove + Pref.	Numeric

Telephone numbers belonging to the line

Parameter	Description	Value
Phone numbers		Numeric

Add a VoIP trunk or edit an existing one

For every VoIP domain, you can set up any combination of VoIP terminations and trunks.

The following table lists the configurable parameters for a trunk.

Parameter	Description	Value
Enabled	Lets you disable a trunk without losing its configuration	On / Off
Name	Mnemonic name assigned to the gateway	Alphanumeric

SIP settings

Parameter	Description	Value
VoIP domain	Select the hostname of the VoIP domain to which you wish to associate the trunk	Hostname
Identifier	Trunk identifier	Alphanumeric
Username	Username with which the trunk is accessed	Alphanumeric
Secret		Alphanumeric
Registration validity		Numeric
Source domain (registration)		Alphanumeric
Source domain		Alphanumeric

Transport settings

Parameter	Description	Value
Transport type	Lets you choose between SIP with UDP/TCP/TLS transport or SIP with WebSocket/Secure WebSocket transport for this domain	SIP / WebRTC
Preferred transport	Drop-down menu that lets you choose the preferred transport among those enabled. Selecting a disabled transport will automatically enable it	UDP / TCP / TLS / WS / WSS
Enable UDP transport	Enable unencrypted UDP transport. Only available if "Transport type" is set to SIP	Yes / No
Enable Web RTC transport	Enable Web RTC transport protocol	Yes / No

Advanced settings

Parameter	Description	Value
Simultaneous call limit	Number of simultaneous calls allowed	numeric
DTMF mode	Choose how DTMF tones are sent to this gateway, among the modes provided (RFC 2833, SIP Info, in audio). By default, this will be set to the predefined system mode	RFC 2833 / SIP Info / In audio
COLP sending mode		System default / Disabled / Remote-Part-ID / P-Asserted-Identity
COLP acceptance mode		System default / Disabled / Enabled

Audio codec

Parameter	Description	Value
	Allows you to choose the types of audio Codecs	PCM a-law / G722 (HD Audio) / G.726 / G.729 / GSM / Opus / PCM u-law

Video codec

Parameter	Description	Value
	Allows you to choose the types of video Codecs	H.261 / H.263 / H.263+ / H.264 / VP8

Header SIP

Parameter	Description	Value
Set the user part of the From URI as.		Caller number after manipulation / Caller number before manipulation / Authentication username / Custom.
Sets the display name in the From header to the user part of the From URI.		Yes, except to remote extensions / No (keep the original display name) / Yes

Show placeholders available

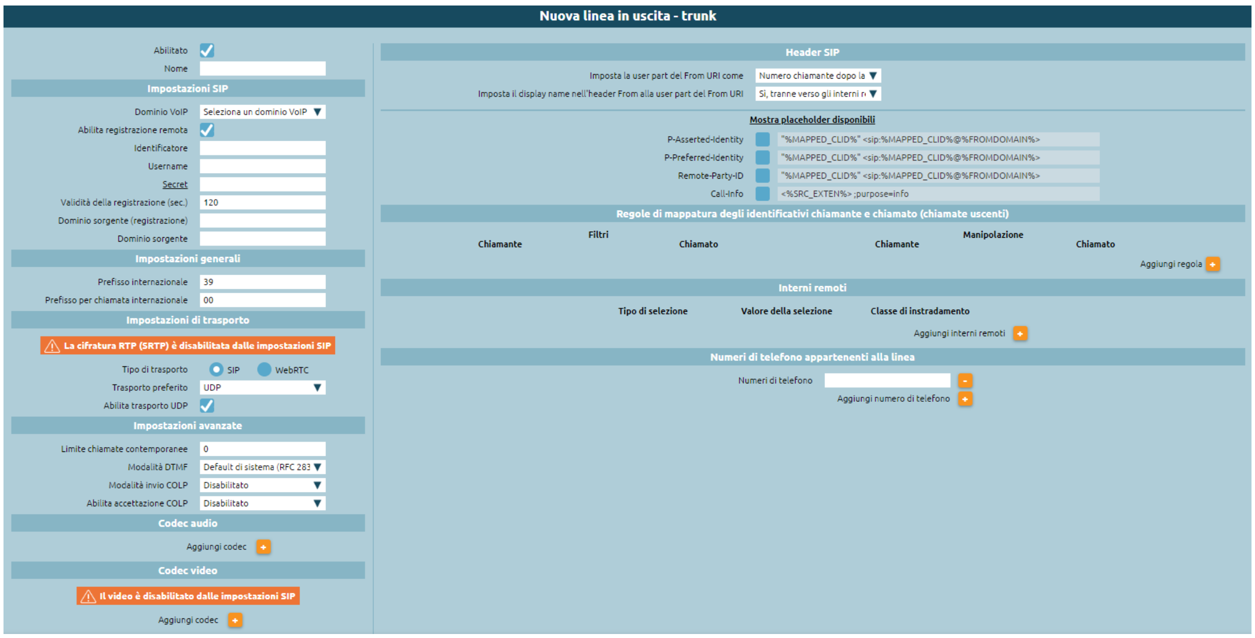
Parameter	Description	Value
P-Asserted-Identity / P-Preferred-Identity / Remote-Party-ID / Call-Info		

Caller and called identifier mapping rules (outgoing calls)

Parameter	Description	Value
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Filters	Any / Exact / Prefix / Range	Numeric
Caller Manipulation	Remove + Pref.	Numeric
Caller Manipulation	Remove + Pref.	Numeric

Remote extensions

Parameter	Description	Value
Type of selection		Exact selection / Range of selection / Prefix selection.
Selection value		Numeric
Routing class		



The difference between VoIP terminations and trunks is due to the fact that with the former every registration/authentication account corresponds to a single phone number, while with the latter it is possible to use a range of numbers with the same authentication credentials, which usually share a common root.

Note: To create a VoIP termination or trunk, it is necessary to first create a VoIP domain to link it to.

The following table shows the columns in the list of outbound lines.

Column	Description	Value
Enabled	Shows whether the outbound line is enabled or disabled	Enabled / Disabled
Name	The name assigned to the line	
Identifier	Unique identifier assigned to the line. For VoIP terminations or trunks, this is the username for authentication	
Type	If it is not a physical gateway, this specifies the type of line	Trunk / VoIP terminal
State	**For physical gateways with inbound registration disabled** the reachable/unreachable state shows whether or not the peer responds to SIP OPTIONS messages. If registration is enabled, it shows whether or not registration was completed successfully on the part of the gateway. **For VoIP domains** the reachable/unreachable state shows whether or not the peer responds to SIP OPTIONS messages. **For VoIP terminations and trunks with remote registration enabled** the reachable/unreachable state shows whether or not the registration was successful. If remote registration is disabled, the static state is shown. The suspended state will only be shown if an element has been added but not yet configured.	Reachable / Unreachable / Suspended / Static
RTT	Round-Trip Time of a SIP packet between PBX and gateway or PBX and VoIP domain/server of the operator.	Value in ms
Show	Visible if lock has NOT been acquired. Clicking the icon will show the line settings in read-only mode	Magnification icon
Edit	Visible only if lock has been acquired. Clicking the icon will open the line modification page	Pencil icon
Delete	Visible only if lock has been acquired. Clicking the icon will delete the line	Trash icon

Users and roles

Users

Access to the KalliopePBX GUI (as well as CTI services, LDAP phonebook, etc.) is granted to users. There are two kinds of users: built-in and custom users. Built-in users include administrative and service users, whose roles are usually predefined and not modifiable, whereas custom users are additional users that can be created and assigned to custom roles.

Each user has one or more associated access permissions among GUI, CTI, and API.

- **GUI**: GUI access means that the user can log into the KalliopePBX web interface; GUI access also grants the user permission to access the integrated LDAP server.
- **CTI**: CTI access allows the users to use Kalliope applications (CTI, Logger, Supervisor Panel) which connect to the PBX using the CTI socket and protocol.
- **API**: API access allows the users to invoke the KalliopePBX REST APIs available at http[s]://<PBX IP>/rest/ (see REST API).

Built-in users

The first example of the built-in user is admin (whose default password is “admin”), used to access the GUI after the first firmware installation. This is the primary technical figure and is commonly used to perform the system configuration. Additional users may have the rights to perform configuration tasks, but they can be limited to specific GUI panels only, according to their granted Role.

The following table lists the built-in users along with their access permissions. (Note: (+) means that this access permission is assigned and cannot be revoked; (-) indicates that the consent can be granted or not.)

Username	Access permissions	Notes
admin	GUI (+), CTI (+), API (+)	This is the main technical user. They have full privileges on PBX configuration both for system (network, network services) and telephony (entities, services, etc.). They have full access to logs and records, but they have some limitations regarding aspects related to the privacy of the users. Firstly, they cannot see the external telephone numbers in the CDR in full, but are only able to view them with last three digits replaced by “xxx”; secondly, the “admin” user does not have access to Call Recording configuration and files, which is limited to “privacyadmin” user (and delegated users).
privacyadmin	GUI (-), API (-)	This user has full access to the external telephone numbers of the CDR, and is the only one who can configure call recording authorization. They can also access call recording records, download and listen to the recorded calls, as well as grant other users “privacy” permissions, which gives them access to full numbers in CDR and to the list of recorded calls and the corresponding files.
phonebook	GUI (-), API (-)	This user has read access to the KalliopePBX phonebook. It has to be explicitly enabled from the “System Settings” -> “Users Management” panel, assigning it a password and the required access permissions. N.B.: GUI permission also grants the right to access the integrated LDAP server, where the KalliopePBX phonebook is published (according to the settings in “Phonebook”->”LDAP Settings” panel). The “phonebook” user is mainly useful to have a single identity (configurable through provisioning) used by telephones to access the KalliopePBX phonebook using LDAP.
click2call	GUI (-), API (-)	This user is useful when using third party applications to send click-to-call commands (using the REST API /rest/phoneServices/c2c/{dest_exten}/{source_exten}) to KalliopePBX using a single user with limited privileges

Multitenant

During Multitenant license activation, the PBX and the tenant entities, bundled under a single administrative entity, are separated and a new built-in user **pbxadmin** is created (with default password “admin”).

Management of the PBX as a system is granted to the new “pbxadmin” user, who has both GUI and CTI permissions, whereas the “admin” user retains control of the telephone service configuration for the tenant. Since multiple tenants can be created, each with its own “admin”, it is necessary to extend the username to specify the relevant tenant domain. The predefined existing tenant domain is “default”, so the predefined built-in users become admin@default, privacyadmin@default, etc.

For each new tenant created (e.g. with domain “sampledomain”), several new users are generated, namely admin@sampledomain, privacyadmin@sampledomain, phonebook@sampledomain, and so on.

The admin@default and admin@sampledomain users are completely independent and each one can only manage their own tenant.

Note: If a user does not specify the domain when logging in (e.g. uses "admin" instead of "admin@somedomain"), then it is assumed to belong to the default domain and authentication is performed accordingly.

Custom users

Additional users can be created. Currently, custom users must be associated with an **Extension**. Custom users can be created in the "Edit Extension" panel, defining a unique username (within the tenant) and assigning GUI, CTI and/or API access permissions. By default, all custom users are created with the standard "Tenant User" role, but a different one can be selected among those available. As detailed below, roles are managed in the "System Settings" -> "Roles Management" panel, where different access permissions (none/list/read/write) can be assigned for each panel of the GUI, allowing the admin to delegate some configuration tasks to selected users.

Users configuration

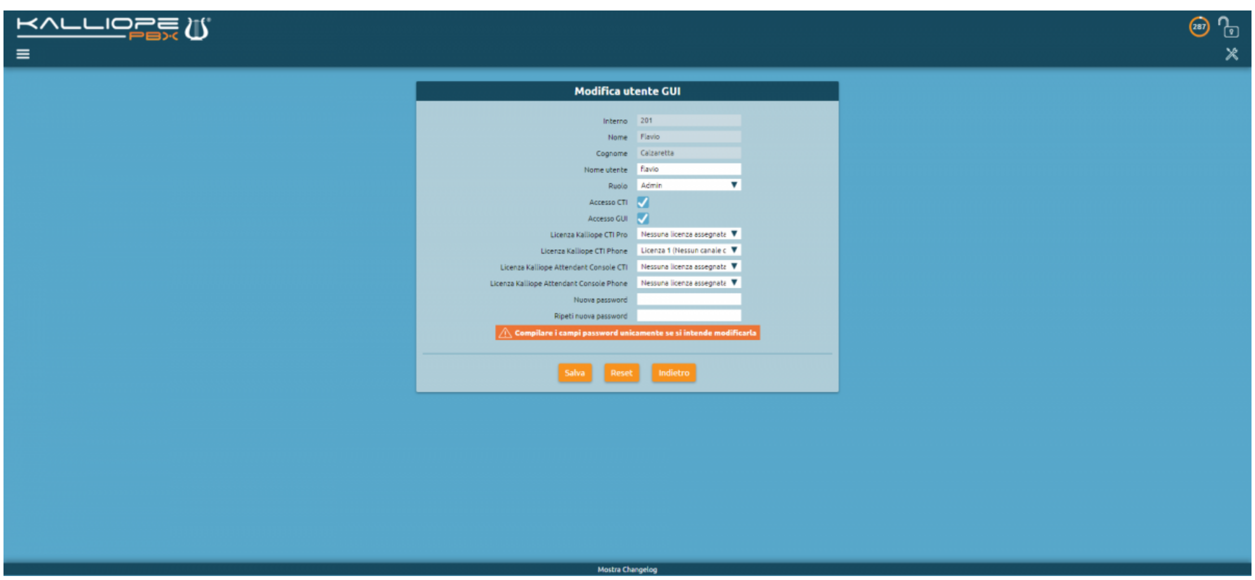
During the creation of an extension, the create local user box is selected and a new GUI user is automatically created with the credentials set during creation.

To edit and manage these users, you need to access the GUI users management in the System settings menu.

Through the users configuration page, you can:

- edit the credentials (username and password) necessary to access the GUI and the clients;
- assign a role and the relative read/write permissions;
- enable/disable access to the GUI and the clients;
- assign the following licenses: KalliopeCTI Pro, KalliopeCTI Phone, Kalliope Attendant Console CTI, Kalliope Attendant Console Phone.

Once created, custom users cannot be edited from the "Edit extension" panel, but they appear in the "System Settings" -> "Users Management" panel, along with the built-in ones.



User authentication

User authentication is performed with a password check, using one of the two available authentication methods.

The first method is "Local Authentication": the user password is handled by the PBX, and its hash is stored in the internal database for authentication. This is the only available authentication method for the "admin" user.

KalliopePBX can also authenticate users with external services; the supported external authentication services are Microsoft Active Directory and LDAP servers. External authentication services are defined on a per-tenant basis, so they need to handle usernames of the form "user@tenant_domain".

Roles

Each user is assigned a role, which determines their permissions in terms of access to the various panels. Since their permissions are fixed, built-in users have built-in roles (currently not assignable to custom users).

Custom users by default have the "Tenant User" (or simply "User") role, which is built-in and not modifiable. This role grants the user the right to access their own CDR and the local, shared, and personal phonebooks.

Additional roles ("Power User" roles) can be created and assigned to the custom users. Each role has a priority attribute (an integer value between 1 and 99; standard users have priority 0, whereas tenant admin has 100) which is used to resolve contention of the Configuration Lock when multiple users need to perform configuration operations on the PBX. Users can acquire the Configuration Lock even if it is currently held by another user, provided that their role priority is higher than the one of the user currently holding the lock. Note that the action of acquiring the lock currently held by another user drops all the pending changes made by the first user.

Roles configuration

To configure a role, you must first set a priority from 0 to 99. Users with higher priority can acquire the lock from power priority users, and unsaved changes will be lost.

Custom roles can be configured by selecting the level of access to each panel from those available:

- "none": the user cannot access the panel and the link to the panel will not be displayed in the navigation menu (direct access to the panel URL is also blocked)
- "list": the user has read access to the panel with the list of related entities (for example, the extension list) but cannot access the details of each item or perform actions on them
- "read": the user can access both the list panel and those of the individual entries, but only in read mode
- "write": the user has full read/write access to the related entities

The following table lists the configurable parameters for each role.

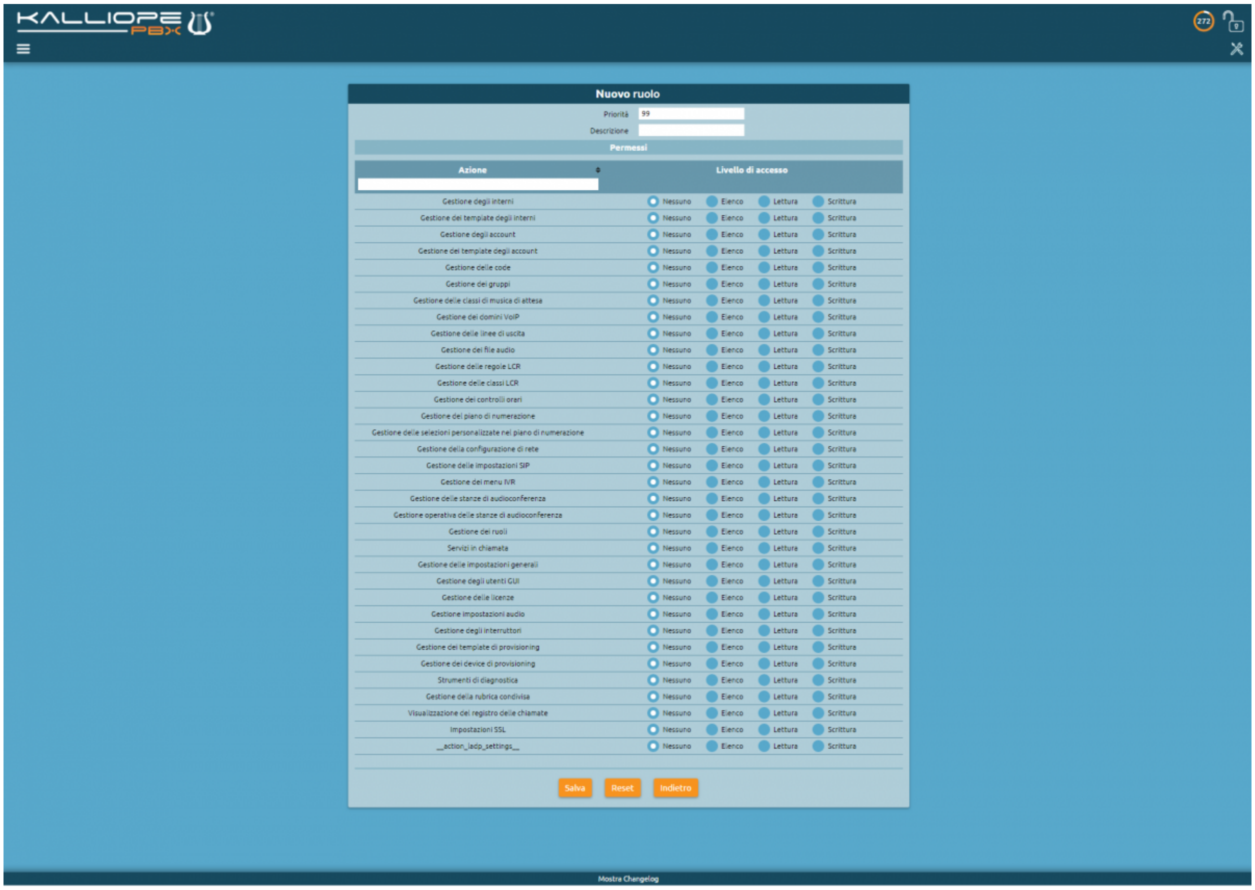
Parameter	Description	Value
Priority	Priority assigned to the role	Numeric (from 0 to 99)
Description	Role identifier	Alphanumeric

Permissions

Parameter	Description
Extension management	Enable users to manage extensions with the selected permissions
Extension template management	Enable users to manage extension templates with the selected permissi
Account management	Enable users to manage accounts with the selected permissions

Table 1 – continued from previous page

Parameter	Description
Account template management	Enable users to manage account templates with the selected permissio
Queue management	Enable users to manage queues with the selected permissions
Ring group management	Enable users to manage ring groups with the selected permissions
Music on hold class management	Enable users to manage music on hold classes with the selected permis
VoIP domain management	Enable users to manage VoIP domains with the selected permissions
Outbound line management	Enable users to manage outbound lines with the selected permissions
Audio file management	Enable users to manage audio files with the selected permissions
LCR rule management	Enable users to manage LCR rules with the selected permissions
LCR class	Enable users to manage LCR classes with the selected permissions
Checktime management	Enable users to manage time checks with the selected permissions
Numbering plan management	Enable users to manage the numbering plan with the selected permissi
Management of the custom selections in the numbering plan	Enable users to manage custom selections with the selected permissio
Network configuration management	Enable users to manage network configuration with the selected permis
SIP setting management	Enable users to manage SIP settings with the selected permissions
IVR menu management	Enable users to manage IVR menus with the selected permissions
Audio conference room management	Enable users to manage audio conference rooms with the selected perm
Audio conference room operation management	Enable users to manage audio conference room operation with the sele
Role management	Enable users to manage roles with the selected permissions
On-call service management	Enable users to manage on-call services with the selected permissions
General setting management	Enable users to manage general settings with the selected permissions
GUI user management	Enable users to manage GUI users with the selected permissions
License management	Enable users to manage licenses with the selected permissions
Audio setting management	Enable users to manage audio settings with the selected permissions
Switch management	Enable users to manage switches with the selected permissions
Provisioning template management	Enable users to manage provisioning templates with the selected perm
Provisioning device management	Enable users to manage provisioning devices with the selected permiss
Diagnostic tool management	Enable users to manage diagnostic tools with the selected permissions
Shared phonebook management	Enable users to manage shared phonebook with the selected permissio
Call detail record viewing	Enable users to view the call detail record.
SSL setting management	Enable users to manage SSL settings with the selected permissions
LDAP setting management	Enable users to manage LDAP settings with the selected permissions



3.1.2 Service configuration

Traditional telephone services

Call Hold and Music on Hold

To put the current call on hold, you can work on the corresponding key of the phone terminal. The key can be physical (normally indicated with “Hold”) or dynamic in according to the specific phone terminal.

Call Pickup (direct and with invite)

Description

This service lets a PBX user pick up a direct call to another extension.

If an extension is ringing and the user cannot or does not want to answer, another user can answer the call from their phone rather than from the other extension.

Direct pickup is applied to all calls to an extension, including those received as part of a ring group or a queue.

KalliopePBX offers two ways of picking up calls:

- **Direct**: the user who picks up the call has no information on the calling party.

- **With invite**: the user who picks up the call can check the identity of the calling party before choosing whether to complete the transfer or not.

Direct pickup is performed by dialing the direct pickup code (by default **) followed by the number of the extension which is receiving the call. Conversation with the original calling party will begin directly.

Pickup with invite is performed by dialing the pickup with invite code (by default #*) followed by the number of the extension which is receiving the call. The user who is picking up the call will first receive a hang up signal; once they have done so, they will receive a call with the caller ID of the original calling party. The user can now choose whether to complete the transfer (by answering the call) or not (by refusing the call). The original extension will stop ringing only once the transfer is complete.

Configuration

The service can be enabled/disabled in the PBX -> Numbering plan page. The service codes can be changed in the PBX -> Numbering plan page.

Interoperability

When using this service, it can be useful to have a key (with Busy Lamp Field) that lets you check the state of the extension which is receiving a call.

For monitoring, KalliopePBX sends SIP NOTIFY messages to communicate changes of state. The phone must send a SIP SUBSCRIBE message to request this information.

This operation is normally executed by configuring a BLF-type function key.

Apart from monitoring the state of the extension, with many phones it is possible to pick up the call with the same key. In this case, the pickup code must be specified.

Examples

On SNOM –> Through the web GUI, you can configure function keys with: .. code-block:: console

> Account: select the account from the drop-down (if only one account is configured on the phone, it will be the first in the list) Type: BLF Value: <extension>|**

Or you can directly edit the configuration file or the template: .. code-block:: console

> <fkey idx="%%id%%" context="%%line_id%%" label="" perm="">blf sip:<interno>@%%KPBX_IP_ADDRESS%%;user=phone|**</fkey>

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
<fkey idx="0" context="1" label="INTERNO103" perm="">blf sip:103@192.168.23.190;
↪user=phone|**</fkey>
```

On YEALINK -> Through the web GUI, you can configure DSS keys with:

```
Type: BLF
Value: <extension>
Line: The line associated with the account (Line 1 if the account is the only one on the␣
```

(continues on next page)

(continued from previous page)

```
↪phone)
Extension: **
```

Or you can directly edit the configuration file or the template: .. code-block:: console

> memorykey.%%id%%.line=%%line_id%% memorykey.%%id%%.value=<extension> memorykey.%%id%%.type=16 memorykey.%%id%%.pickup_value=**

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
memorykey.1.line = 1
memorykey.1.value = 103
memorykey.1.type = 16
memorykey.1.pickup_value=**
```

For pickup with invite, you need to replace the direct pickup code (**) with the pickup with invite code (#*).

Calling Line Identification Restriction (CLIR)

Description

This service lets a user hide their caller ID when making a call. This way the called party will not be able to obtain the number of the caller, even if their device is CLIP (Calling Line Identification Presentation) enabled.

Obscuring caller ID is not always possible and may be subject to regulatory restrictions. Calls to emergency services will ignore CLIR through the Calling Line Identification Restriction Override (CLIRO) service, and some types of calls (e.g. commercial/marketing calls) are not allowed to hide the identity of the calling party.

This service is applied to calls between extensions as well as outbound calls. In order for the service to function correctly for outbound calls, outbound gateways must be configured correctly and/or the VoIP provider must support CLIR.

The service can be enabled/disabled on a call-by-call basis, and the default behavior can be configured for each extension.

The user can obscure their caller ID by dialing the CLIR "Setup" code (by default *671) followed by the number to call. If the user wishes to show their caller ID, they must dial the CLIR "Remove" code (by default *670) followed by the number to call.

Configuration

In order to use the service, it must be enabled in the numbering plan (where you can also edit the service codes).

Whether the service is enabled or disabled, as well as the default behavior, must be specified for each extension (this can be done through extension templates). Other than the default behavior for calls between extensions and calls to external numbers, the settings for each extension include the option to hide the caller ID if the calling device sends a Privacy: id SIP Header.

Interoperability

When CLIR is enabled, SIP signaling is modified as follows:

- the user part of the From URI is replaced with anonymous (e.g. sip:anonymous@<telephone_ip_address>:<telephone_port>);
- the user part of the Contact Header is replaced with anonymous (e.g. sip:anonymous@<ip_address_registrar>:<registrar_port>);
- the Privacy: id SIP header is added.

Calls between extensions do not require additional configuration, while outbound calls might, depending on the type of outbound line.

- VoIP Provider

Even if CLIR is enabled, it is still necessary to send the identity of the calling party to the provider. If this doesn't happen, the provider will normally reject the outbound call. This information can be sent through P-Asserted-Identity Header or Remote-Party Header.

These headers must be enabled in the configuration of the outbound trunks (Gateways and VoIP Domains -> Trunk). The accepted format for the headers is usually the one suggested in the configuration page. If there are issues, you should check the required configuration with your service provider.

To prevent these headers from being overwritten, it is necessary to disable COLP in the outbound trunk configuration.

Group Call Pickup (direct and with invite)

Description

his service lets any PBX user pick up a direct call to any extension that belongs to a pickup group from which the user is authorized to pick up calls.

It is possible to configure which groups a user belongs to and which ones they are authorized to pick up calls from.

If there are multiple inbound calls, call pickup will be executed on the last one received by any of the groups from which the user is authorized to pick up calls.

Group call pickup is applied to all calls to an extension, including those received as part of a ring group or a queue.

KalliopePBX offers two ways of picking up calls: - Direct: the user who picks up the call has no information on the calling party. - With invite: the user who picks up the call can check who the calling party is before choosing whether to complete the transfer or not.

Direct pickup is performed by dialing the direct pickup code (by default *9). Conversation with the original calling party will begin directly.

Pickup with invite is performed by dialing the pickup with invite code (by default #9).

The user who is picking up the call will first receive a hang up signal; once they have done so, they will receive a call with the caller ID of the original calling party. The user can now choose whether to complete the transfer (by answering the call) or not (by refusing the call). The original extension will stop ringing only once the transfer is complete.

Configuration

The service can be enabled/disabled in the PBX -> Numbering plan page.

The service codes can be changed in the PBX -> Numbering plan page.

Ring groups and the corresponding call pickup authorization can be specified in the extensions page or though the extension templates.

Interoperability

When using this service, it can be useful to have a key that lets you make a quick call to the service code.

Examples

On SNOM -> Through the web GUI, you can configure function keys with:

```
Account: select the account from the drop-down (if only one account is configured on the␣
↪phone, it will be the first in the list)
Type: Speed Dial
Value: *9
```

Or you can directly edit the configuration file or the template:

```
<fkey idx="%%id%%" context="%%line_id%%" label="" perm="">speed *9</fkey>
```

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

On YEALINK -> Through the web GUI, you can configure DSS keys with:

Or you can directly edit the configuration file or the template:

```
memorykey.%%id%%.line=%%line_id%%
memorykey.%%id%%.value=*9
memorykey.%%id%%.type=13
```

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
memorykey.1.line = 1
memorykey.1.value = *9
memorykey.1.type = 13
```

For pickup with invite, you need to replace the direct pickup code (*9) with the pickup with invite code (#9).

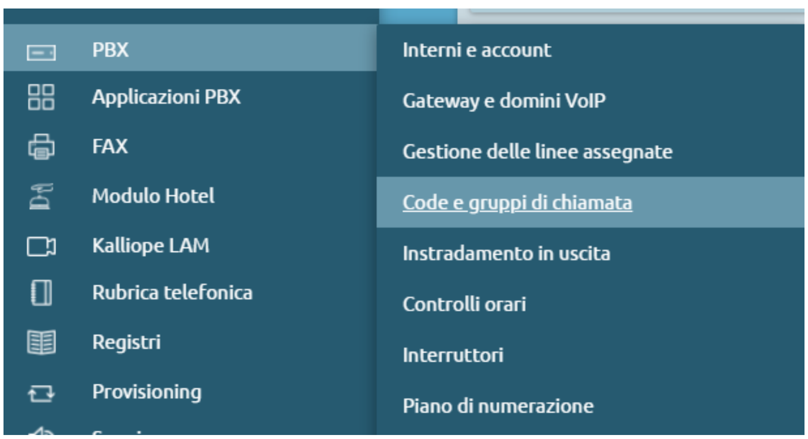
Wait queues (ACD)

Description

The Automatic Code Distribution (ACD) service, also known as the "waiting queue" service, provides professional telephone reception by keeping callers waiting until an operator becomes available. The incoming call is taken over by the central unit that presents the caller with a series of information through audio files, hold music, and puts the callers in a queue to distribute them to the various operators of the queue (or members) based on specific commitment policies that can be configured within each queue. The "Queues" are a mechanism analogous to the Ring Groups, from which they differ for the possibility of defining in a more refined way the ringing strategy and for the sorting of the incoming calls, which are queued and served with a FIFO (First In First Out) policy towards the members of the queue. Each queue can be associated with an arbitrary number of members (accounts), who will serve the calls addressed to it. It is important to remember that an operator can be busy on more than one queue at the same time. In case of concomitance of calls waiting on more than one queue, the service will present to the operator the call coming from the queue with the highest priority.

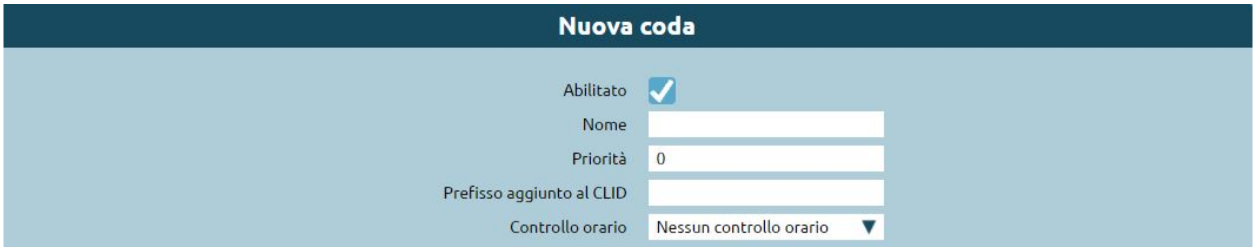
Configuration

To access the service just click on "PBX" > "Queues and Callgroups". We'll be on the page dedicated to the list of configured queues with the main parameters. By clicking on "Add new queue", we can move on to the configuration of the new queue..



There are the following fields:

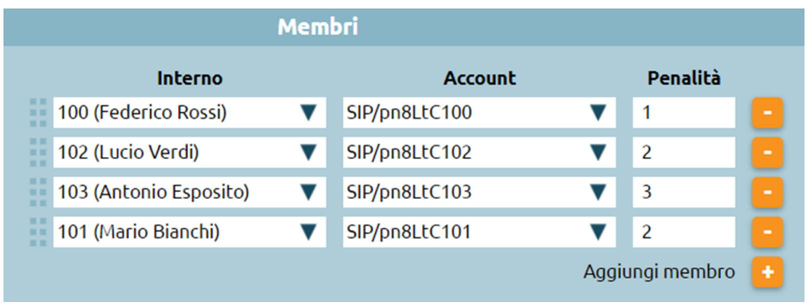
- Enabled: a button that allows to select the queue as active/inactive
- Name: you can insert the name you want to give to the queue (e.g. "Assistance")
- Priority: numerical value that allows you to define which queue should have priority over the others, the higher the value, the higher the priority
- CLID name prefix: the prefix is added to the caller ID (CLID) and allows to indicate on the display of the telephone terminal the specific queue from which the call comes (e.g. "ASS" for "Assistance")
- Checktime: it is possible to choose the time controls already previously configured



Members

We can defines the list of operators assigned to this queue. To add them, just click on the "Add Member" button, you can choose between several extensions and select account for each one. If the extension has more than one account assigned, you can choose on which account (terminal) to commit the extension. The Penalty is a value that can be assigned to each operator: the lower the value, the greater the chance of being engaged by the queuing engine. The lower the penalty, the more the operator is a holder of the service of that queue.

The distinction based on penalties means that only the available operators with the lowest penalty will go to be committed. Take, for example, the following list of members of a queue:



Only the available operators with the lowest penalty will ring. In the specific case of this example, if extension 100 should not be already busy, unregistered or paused, only it will ring. Otherwise, extensions 102 and 101 would ring simultaneously. If, on the other hand, the latter were busy, unregistered or paused, extension 103 would ring, even though it has a higher penalty.

Note: If an operator with the lowest penalty does not answer, also if it is available, the queuing engine will not pass the call to the operators with the next higher penalty but will continue to make the operator with the lowest penalty ring. In the specific example, if extension 100 is available, the queue will always let it ring until it answers, without scaling to extensions 102 and 101 with a higher penalty.

Queue Parameters

- **Auto pause enabled**: feature that automatically pauses an operator who does not answer within their timeout or rejects a queue call.

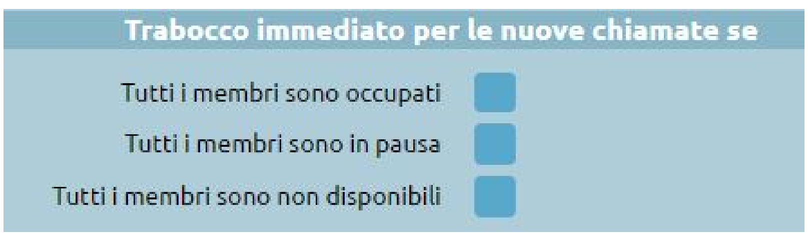
The operation of this option varies depending on the ringing strategy one goes for; in the case of ringall it is not influential: automatic pause is not triggered. Similarly, automatic pause is not triggered if the caller drops out while an operator is ringing. The concept of automatic pause is introduced for callcenter features that allow the exchange to define the roles of supervisor and queue operator. Should a user be a queue operator, he or she has, among the options, the option of "pausing." "Pause" means a user who is registered, available and free at the telephone level, but still cannot be engaged by the queue. The paused/unpause status is tracked within the central unit's logs, so that the operator's pause time is controlled. Pause is handled by codes that we have seen in the "Enumeration Plan" section, and by typing in the codes-customizable-you can pause or remove yourself from the pause. - **Ring Strategy**: drop-down menu that allows you to define the commitment policies of the operators on the queue, i.e. the way the system distributes the incoming calls in the queue to the various operators. Drop-down menu choices:

- **Ringall**: the call arrives at the same time to all free operators in the queue, taking into account the penalty values.
- **Linear**: the call is forwarded to the first free operator according to the sequence in which the members of the queue have been defined (from top to bottom) ignoring the concept of penalties which only applies to ringall
- **Less recent**: assigns the call to extensions that have not answered the call for the longest time. The queuing engine of the central unit takes into account the last time the operator served a call from the queue and, consequently, assigns it exclusively to the extension that has not answered the call for the longest time
- **Fewestcalls**: assigns the call to the operator who has answered the least number of calls that is calculated on the current day, unless the central unit restarts
- **Round robin memory**: distributes calls with round robin mode among available operators and remembers the last one that tried to call
- **Round robin ordered**: like RRMemory, but the order of the operators given in the configuration file is respected
- **Random**: assigns the call randomly among the available operators
- **Ring timeout**: indicates how many seconds the terminal of the engaged operator should ring (e.g. 15 sec). During these seconds, if the conditions change (another operator becomes free and we are in a ringall policy) the queue will not present the call to the operator who has just become free in the meantime, but will include him in the pull of the callable operators only at the expiration of the next ring timeout
- **Retry interval**: the engaged operator rings for a specific number of seconds, there will be a wait of e.g. 5 seconds and then it will go to choose the next operator according to the selected ringing strategy; it will ring again for 15 sec. During the 5 seconds the caller is in the queue and no operator rings.
- **Rest interval**: indicates the time (sec.) for which the operator who has just ended the call service (related to the queue), will not be engaged by the queue
- **Announce sound file**: option that allows to select a previously loaded audio file. When the operator answers the call, before being connected to the caller, he will listen to an audio file that will remind him which queue the call came from. This kind of option is useful for multiservice call centers that answer on behalf of third parties, for the operator - sometimes - it is functional to present himself to the caller
- **Report hold time enabled**: this option allows you to tell the operator how long the caller has been on hold.
- **Ring in use enabled**: allows to activate an option if a queue is particularly important, for example related to a high priority service. When engaging operators you ignore

- **Max waiting time (sec.)**: parameter that monitors the maximum waiting time beyond which an overflow action is triggered (see below).
- **Max length**: parameter that monitors the maximum number of users allowed in the queue beyond which an overflow action is triggered (see below). (e.g. by entering 0 there are no limitations)
- **Failover**: We can perform an audio file and then an failover action, which can be chosen in the drop-down menu

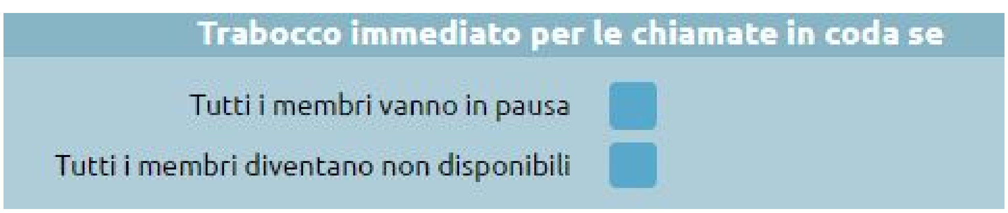
Immediate failover for new calls if

If a caller enters one of the possibilities listed in the image below, an overflow can be triggered immediately, even if the maximum waiting time and the maximum number of users in the queue have not been reached.



Instant overflow for calls in queue

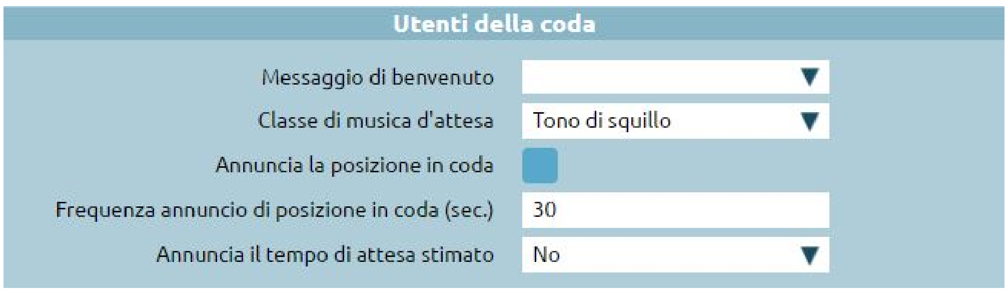
Immediate overflow for calls already in queue for a specific amount of time and in case of the possibilities listed in the image below



Queue users

- **Welcome sound file**: you can select a previously loaded audio file
- **MOH class**: the caller, after the welcome message, can listen to a hold music (previously loaded) or the normal ringing tone
- **Periodically position announce enabled**: option that allows the caller to be informed about his queue position.
- **Periodically position announce frequency (sec.)**: insert a number that indicates every how many seconds to update the caller of his queue position
- **Estimated waiting time announce**: calculated on the basis of the queuing engine statistics, you can choose whether to announce it every few seconds as the queue position announcement frequency, only once, or not to announce it.

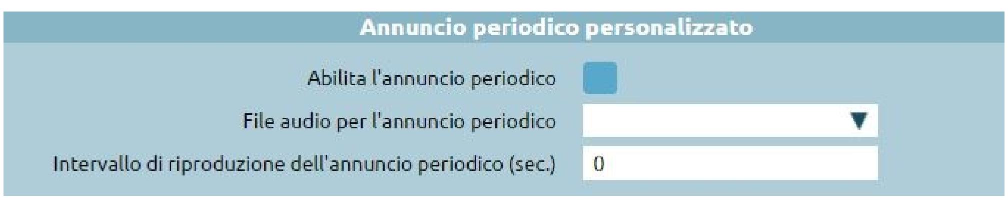
This message is only triggered if the average waiting time is more than two minutes.



Custom Periodic Announce

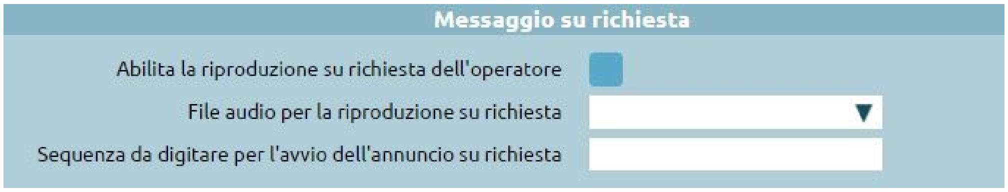
This service can be used to communicate specific procedures to the caller. It is used to offer more frequent information in request to a specific queue

- **Periodic announce enabled**: you can select to enable the announcement.
- **Periodic announce sound file**: you can select a previously loaded audio file.
- **Periodic announce interval (sec.)**: insert the time (in sec.) after which the periodic announcement will start cyclically



On demand message

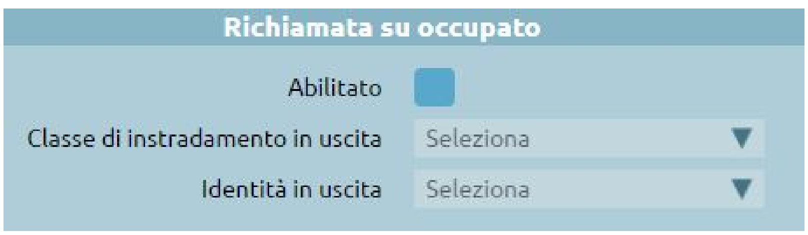
- **On demand play enabled**: it is possible to select the engage of the message on demand
- **On demand play sound file**: it is possible to select a previously loaded audio file
- **On demand play digit**: sequence that the operator in conversation on this queue will have to type to trigger the loaded audio file. (E.g. the operator makes the customer listen to certain conditions by typing a customizable code)



Queue callback

Service enabled in case the callcenter license is assigned on the central unit. It allows the caller on hold not to remain physically on the phone for all the time before being taken over by an operator, but it gives the possibility - by pressing the key 5 - to remain virtually in the queue. A system IVR is executed, asking the caller if he wants to be called back on the current number or if he wants to be called back on another number (to be specified). The user remains virtually in the queue and his position does not change, at his turn the central unit calls the available operator who will tell the central unit to trigger the callback to the user who had booked.

- **Callback enabled**: it is possible to select the callback on busy
- **Outbound Call Routing class**: e.g. Italy no cell phones, if a cell phone number is inserted, the number will not be called back.
- **Outbound identity**: s a call from the central unit to outside, you can treat the outgoing number as if the caller was an extension already configured



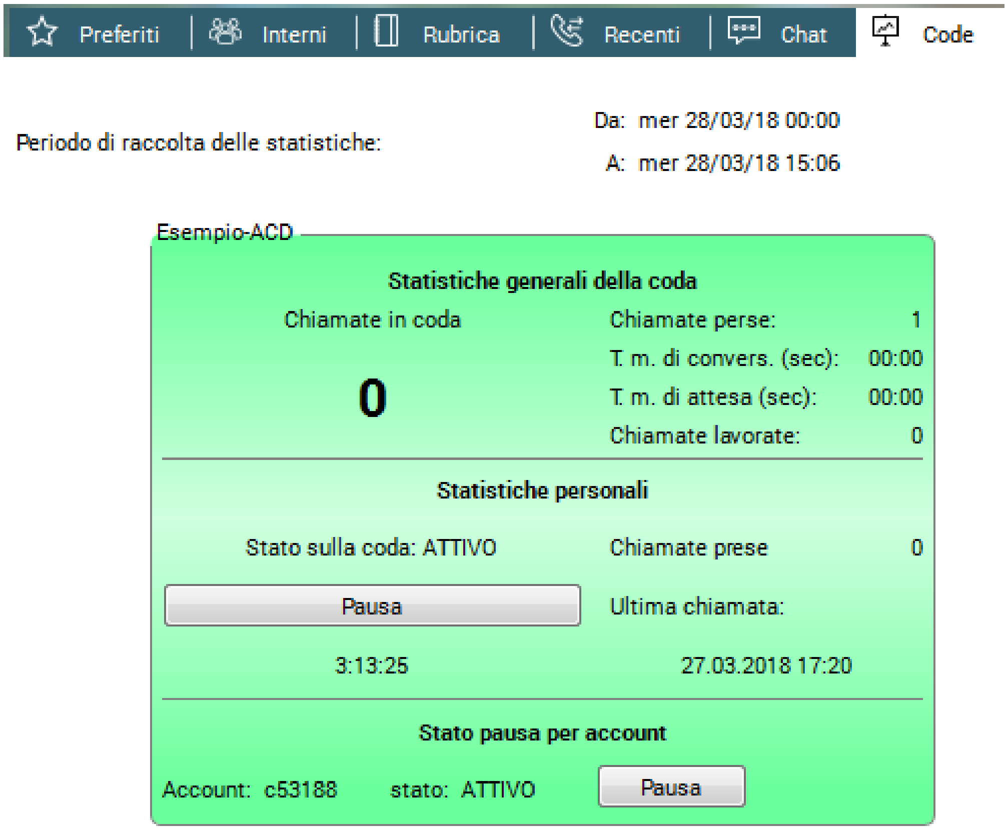
CTI integration

You can make use of certain features of the ACD service, such as adding dynamic operators, pausing operators, and accessing the supervisor panel, directly from Kalliope CTI and Kalliope Attendant Console.

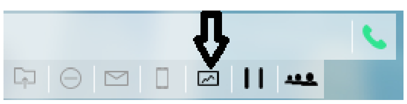
You can, for example: - activate the Call Center license from System settings -> LIcenses; - in multi-tenant systems, assing the license to the desired tenant; - assign a role to a user.

Upon opening Kalliope CTI, you will have available:

- a new tab allowing you to view the queues you belong to and change your pause status if necessary;



- for supervisors, the supervisor panel.



Numbering Plan

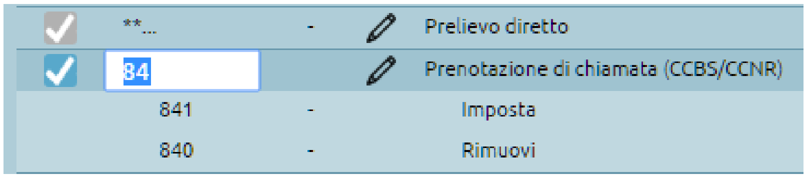
Description

The internal numbering plan rules the routing of a call internally to the KalliopePBX The Numbering Plan is committed by calls generated by an extension, and also by calls coming from outside. To discriminate the different permissions associated with these two types of calls, the Numbering Plan has two columns, one with the enabled selections for calls originating from extensions (local and remote) and one for those originating from an outside line. The selection is matched on the numbering plan according to the displayed order, routing the call according to the following matching priority:

1. Services
2. Custom selections
3. List of extensions
4. Remote extensions
5. Other selections

Services

By clicking on the pencil icon, you can access the page for editing codes and enabling/disabling various services. If a service is enabled but has no assigned code, it will remain inactive.



These are the available services:

- Group call pickup
- Call pickup
- Completion of calls to busy subscriber
- Echo service
- Voicemail
- Audioconference
- Electronic lock
- Unconditional forward
- Fork to Mobile
- Switches
- Passive listening (SPY service)
- Work code
- Queue pause
- Call parking

We report below the default values of the services that can be activated at numbering plan level.

Description	Selection (default)
Call pickup	*9
Direct pickup	**<num.interno>
Group call pickup with invite	#9
Direct call pickup with invite	#*<num.interno>
CLIR (Set)	*671
CLIR (Remove)	*670
Call reservation - CCBS/CCNR (Set)	841
Call reservation - CCBS/CCNR (Remove)	840
Echo Service	800
Voicemail	801
Audioconference	802
Electronick lock (Lock)	850
Electronick lock (Unlock)	851
Unconditional forward (Lock)	811
Unconditional forward (Unlock)	810
Fork to mobile (Switch)	*50*

Description	Selection (default)
Fork to mobile (Enable)	*501
Fork to mobile (Disable)	*500
Fork to mobile (Check status)	*509
Switches (Switch)	*51*<id>
Switches (Abilita)	*511<id>
Switches (Disable)	*510<id>
Director-Secretary (Switch)	*52*<int.dir>[*<int.segr>]
Director-Secretary (Set)	*521<int.dir>[*<int.segr>]
Director-Secretary (Remove)	*520<int.dir>[*<int.segr>]
Paging groups (Login)	*53<id>
Hot Desking	*401
Call parking (Extension)	888
Call parking (First slot)	890
Call parking (First slot)	899
Personal Speed Dial	#0<speed dial>
System Speed Dial	#<speed dial>
Pause on queue (Imposta)	*81
Pause on code (Rimuovi)	*80
Work code	<cod.commessa>*<dest>
Passive listening (SPY service)	<codice><Interno da ascoltare> Accessible only from an extension assigned to a supervi

Custom codes

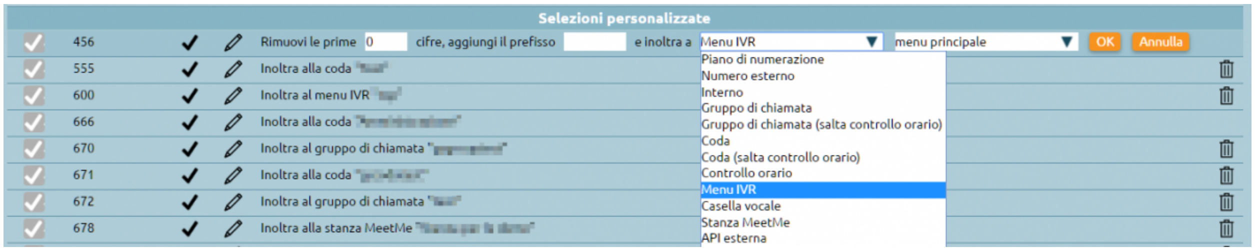
The second part of the numbering plan lets you customize routing based on specific codes or numbering ranges.

You can edit the code (by removing a certain number of digits and adding a prefix) before forwarding it to one of the destinations selectable on the contextual drop-down menu.

You can choose one of the following destinations:

- An external number
- An extension
- A ring group (with or without considering any time checks)
- A queue (with or without considering any time checks)
- A time check
- An IVR menu
- A voicemail box
- A specific audio conference MeetMe room
- An external API

You can edit these settings by clicking on the pencil icon on the right; the page lists the defined custom forwarding rules and lets you edit, delete, and create them. Note that the rules will be automatically ordered as they are saved, with exact ones first and then prefix- or range-based ones.



Checktime and Switches

Description

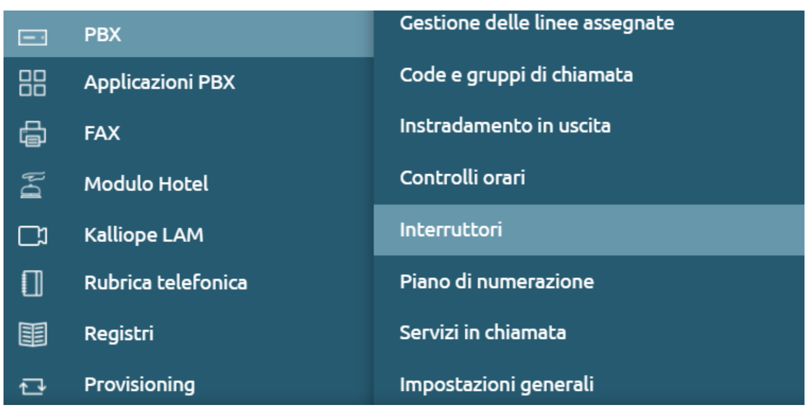
Checktimes are a mechanism to manage call routing on a time and manual basis.

- **On a time basis** because they are based on the definition of time slots encountered in sequence, for each of which it is possible to define the playback of a particular audio message and the subsequent forwarding of the call to a specific destination.
- **On a manual basis** because the checktimes can use switches that are a particular element of the PBX, a flag with two states (on / off), controlled by a code that can be typed by telephone.

Usually, traditional PBX solutions often refer to the day/night service, i.e. the service that offers a courtesy message at office closing times, in this case, the checktime service allows many more choices. It is a transit entity that can be connected at different points in the call flow, since checktime can appear in the call forwarding selection drop-down menu when certain events occur.

Switch Configuration

Let's start by first configuring the switches through **PBX > Switches**.



The switch can have two states, on/off, it is a service that can be managed through the typing of a code entered by telephone or programmed on a function key of a terminal - since the PBX exposes a service Busy Lamp Field with the display of the state - to change a switch and check through the field lamps if the switch is on/off. We are on the Switch List page, clicking on "Add Switch" we can configure it.

- **Enabled: Enable or disable a switch without losing its configuration
- **Name**: Identifier of the switch
- **Number**: Numeric ID of the switch to be used with the enable/disable/switch code.

Switch Access Control

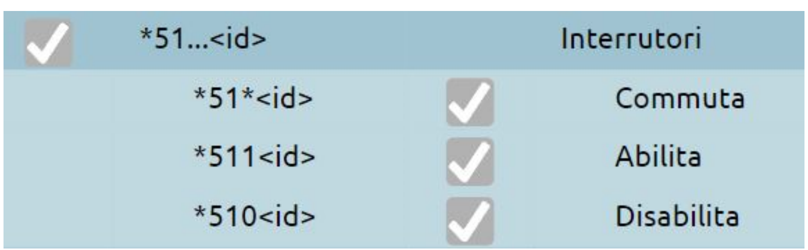
Each switch provides an ACL on an internal basis for piloting and then possible authentication

- **Extension**: Select the extension that is enabled to change the status of the switch
- **PIN Type**: Select the authentication mode of the extension, which can be "None / Custom / Extension Services PIN"
- **PIN Value**: Enter custom PIN, only if the previous value is set to Custom

n PBX > Numbering plan we can find the page for enable/disable/switch the switches.

e.g. To switch, digit on the telephone terminal:

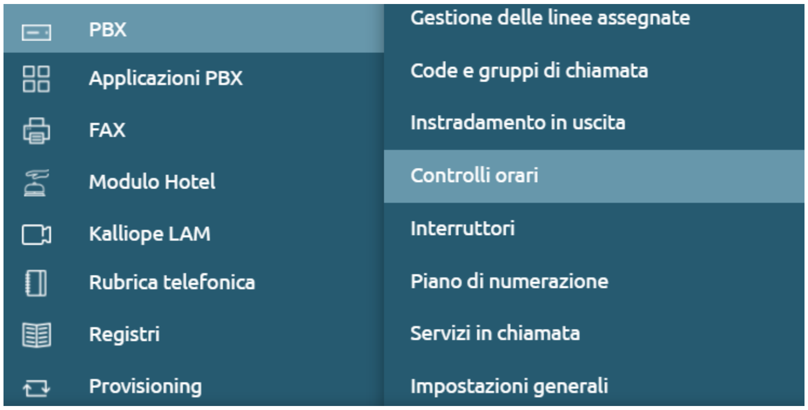
- *51*1, to go to change the state of switch 1, where 1 is the id of the switch
- *5111 to force its opening, where 1 is the id of the switch
- *5101 to force it to close, where 1 is the id of the switch



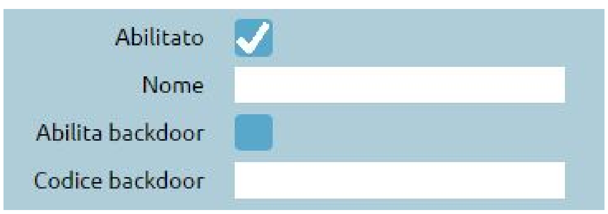
If the switch is turned off and you try to enter the disable code, it will remain off. If a PIN has been set for switch management, you will be prompted to enter the PIN or password after entering this code for the switch action to complete correctly.

Configuration of Checktime

We proceed by going to PBX > Timetable checks and clicking on "Add new Checktime"



- **Enabled**: Allows you to disable a checktime without losing its configuration
- **Name**: Identifier of the checktime
- **Backdoor enabled**: code that allows bypassing the checktime, it will be applied to the incoming call the routing foreseen by the scenario outside the time slots, that is "Overflow outside the time slots"
- **Backdoor code**: Allows defining the backdoor code to be used



If the backdoor code is typed, it will be forced to forward the overflow outside the time bands. This service is used to set a checktime for the opening/closing of some offices. It is helpful in case you want to leave to the employees or to the subjects with particular privileges, the possibility to bypass the limitations of the checktime, typing the code. For example, to communicate with an office of your company, even though the office is closed to the public.

Checktime Periods

You can indicate the opening time slots or the closing time slots of the company. We define standard checktime periods, during the time slots you can take all calls and turn them to the in periods failover which presents in the drop-down menu the various choices. The drop-down menu also presents the item "Return to parent" which allows you to go back to the entity that brought us to this checktime. See groups/code: call groups provided the ability to define a checktime, i.e. a service access policy. If we include this particular checktime within the configuration of a group or queue, to allow it to continue accessing the queue with which it is associated, we should select this item. This allows you to use the same checktime in multiple groups and queues.

Special situations: holidays We can establish during a given day, to proceed with different actions. For example, we can overwrite the failover that we have configured as standard and configure a different one. The graphical arrangement of the information has an important sense: the choices are made from top to bottom, so the more particular rules should be inserted at the top and then the general ones at the bottom.

Out periods failover: Failover action outside of time checktime periods

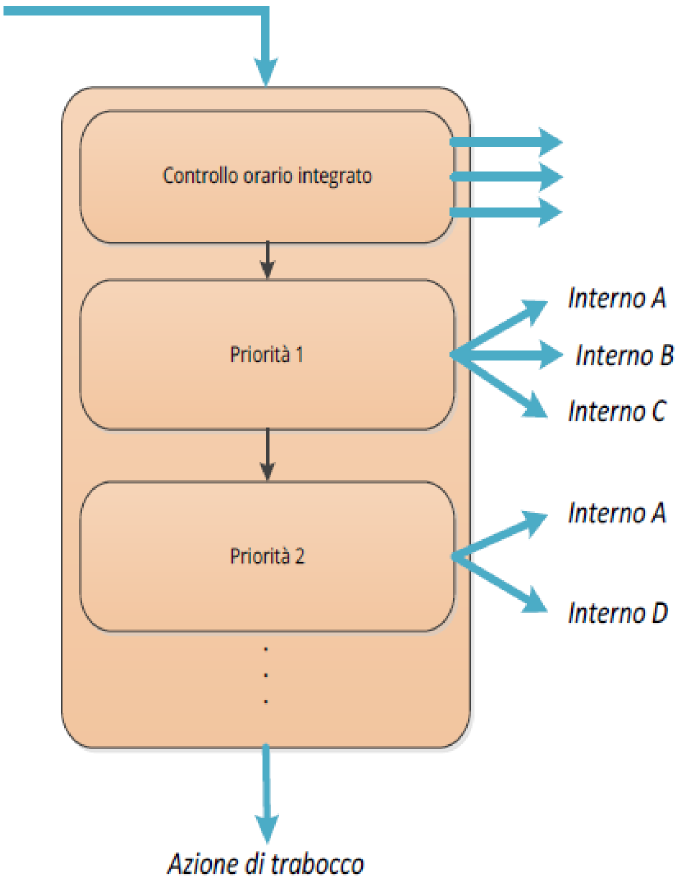
Switches

Allows you to add manual day/night service. Selecting a previously created switch, such as in the example "Force Office Close", if it is on and therefore I wanted to force the office to close, I would have to send the closing sound file and an action, e.g. "Hang Up". If it was closed and we didn't want to force close, we could select "Continue", i.e. proceed to the next switch. Selecting "Force open office" will force the office open regardless of when we have configured the time control. If the switch is on, the same action configured in the failover will be performed during checktime periods. If neither of the two switches is on and involves a forcing, the checktime will behave according to the time slot configuration we have defined. The backdoor code only acts on the behavior of the time slots, if there is a manual open/close forcing, the backdoor code will not work.

Ring groups

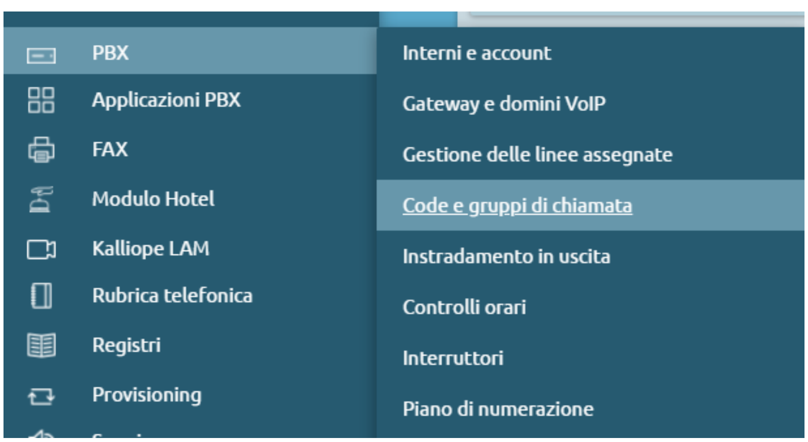
Description

The Call Group service is a call reception service that offers the caller an ordered sequence of extension priorities. The call will be distributed to one or more extensions in parallel. The call group is a grouping of extensions, distributed on different priorities with simultaneous ringing, which identifies the departments of an office.

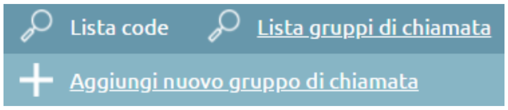


Configuration

We can click on "PBX > Queues and callgroups".



If we click on "Call groups list" we find the list of already configured groups. To move on to the configuration, we click on "Add new callgroup".



The configuration includes the following fields:

- **Enabled**: checkbox to enable or disable the group

Name: insert the name you prefer - **Clid name prefix**: prefix added to the caller ID (CLID) that allows indicating on the display of the telephone terminal the custom code of origin of that particular group of calls (e.g. "ASS" for "Assistance") - **Checktime**: it is possible to define each callgroup its checktime to differentiate the behaviors according to days, months, etc. - **Moh class**: you can choose whether to present to the caller the ringing tone or one of the classes of hold music configured on the central unit - **Failover**: at the end of the consultation of all priority groups it is possible to define a failover action, so in case of no answer of any extension belonging to this group, it is possible to play one of the audio files that we will have set and then to perform one of the actions reported in the drop-down menu

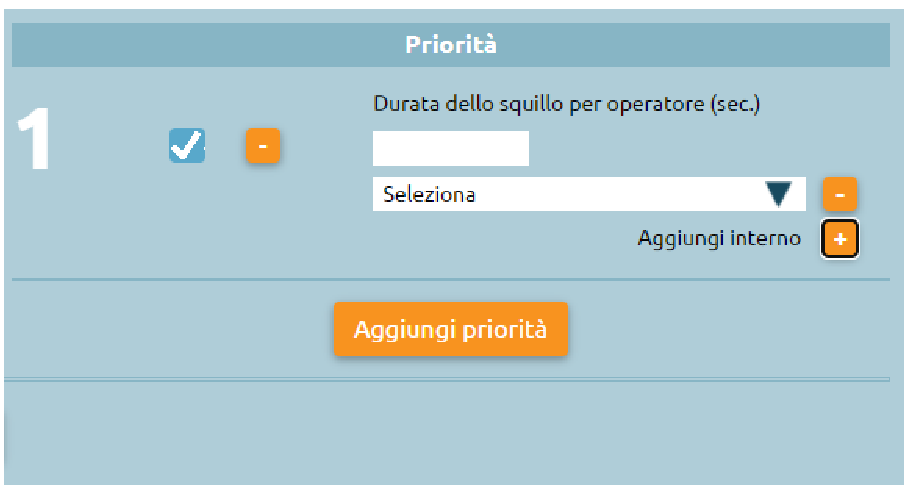
Priorities

We can click on "Add priority" to proceed to configuration.

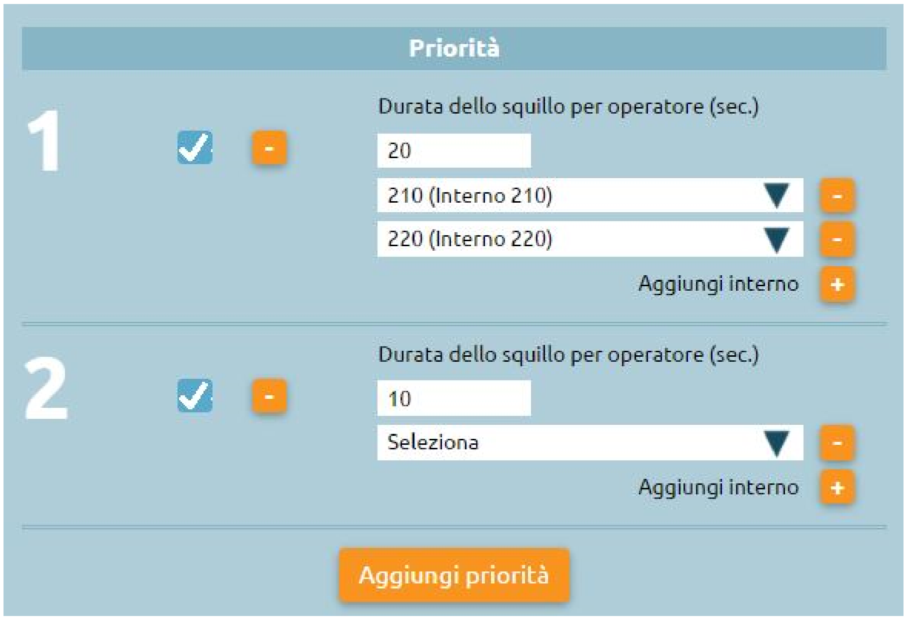
- **Ring timeout (sec.)**: value that defines the ring duration of this priority 1 subgroup
- **Add Extension**: we define the extensions belonging to the group of priority 1

Adding another group of priority 2 (clicking on "Add Priority"), sequentially to group 1, at the moment the ringing seconds of the first group pass, we proceed with the second one.

The groups are executed sequentially, if no subgroup answers the call, it will be diverted to the configured overflow.



Failover actions related to individual interns involved in the groups



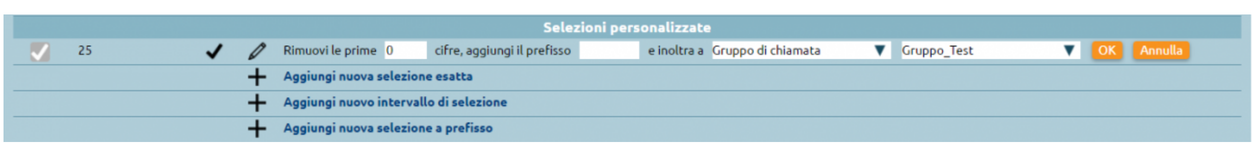
If the call was directed to the group and not the individual extension, individual extension failover actions are not performed. If extension 210 has a failover configured on its extension for no answer to an external number, when this extension rings not because it is called directly, but because it belongs to the calling group, its overflow will not be considered. The same way, the central unconditional forwarding settings for individual extensions belonging to the group, are not taken into account. Suppose you configure a forwarding on the terminal set up with an account of a specific extension. In that case, it will be correctly executed because it is not managed by the central unit, but by the terminal. If the Fork2Mobile service is configured on the extension - to make the phone associated to the specific extension ring - the call is made in parallel also on the mobile associated with that extension.

In summary:

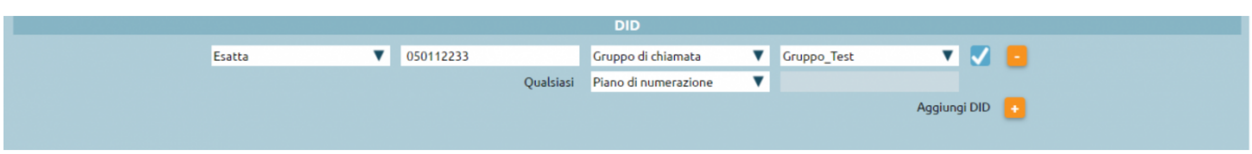
- Failover actions for individual extensions will NOT be executed
- Activating the “unconditional forward” service on the extensions will NOT cause the call to be forwarded
- Activation call forward on one of the terminals of an extension will cause the call to be forwarded
- The Fork2Mobile service of individual extensions that belong to the group will be activated for calls made to the group

Two methods can be used to forward an incoming call to the call group:

- define in the numbering plan a custom selection associated with the call group



- define in a VoIP gateway or domain a numbering that has as destination the call group



Unconditional Forward

Description

The unconditional forward service lets you divert a direct call to an extension to a different destination (local or external). This service can be used to divert calls to a colleague's number or any other selection from the numbering plan (including ring groups or queues, or you own mobile phone when you're away from your workstation).

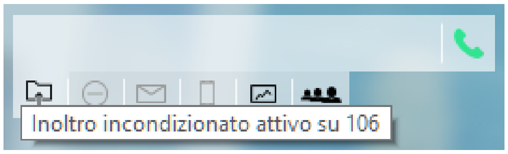
Note: The service only applies to direct calls to the extension and not to calls originating from ring groups the extension belongs to or queues of which one or more accounts linked to the extension are members.

Every time a user activates the service, they must specify the number to which the call will be forwarded. KalliopePBX will forward the inbound call to this number. In case of failure, the failover action will be the one associated with the destination and not the original extension.

A call forwarded to an extension will show the original caller, while one forwarded to an external number will follow the outbound routing rules associated to the original extension, and the number displayed will be the one used by the extension when calling the public phone network.

Unconditional forwarding can be enabled/disabled in three ways:

- **From the phone**: the service can be enabled by dialing the activation code (by default 811) followed by the number to which forwarding should be programmed. KalliopePBX will confirm the activation by playing the "Saved" audio file. If the user wishes to forward a call to an external number, the number must be preceded by the prefix for outbound calls. Similarly, the user can deactivate the service by dialing the deactivation code (by default 810). KalliopePBX will confirm the deactivation by playing the "Thank you" audio file. These codes can only be used on a device linked to the extension on which the service is to be enabled/disabled.
- **From KalliopeCTI Desktop (all modes)**: you can find the unconditional forward icon under the number dialing box. Clicking the icon will enable the service, shown by the icon changing to black color. Clicking on the icon again will disable the service. Hovering the cursor on the icon will display the number to which calls will be forwarded.



- **From KalliopeCTI Mobile**: tap the settings icon on the lower right, then the forward icon. When the service is enabled, the icon will change and show the number to which calls will be forwarded. Tapping the icon again will disable the service.

When this service is enabled, the extension on which it is active will not receive the call; only the destination of the forward will ring. Other forwards associated to the extension will therefore not be applied.

Configuration

The unconditional forward service is always globally enabled.

Activation/deactivation codes are managed from the numbering plan.

Interoperability

When enabling/disabling this service through a phone, it can be useful to have a key (with Busy Lamp Field) that lets you check the state of the service.

For monitoring, KalliopePBX sends SIP NOTIFY messages to communicate changes of state. The phone must send a SIP SUBSCRIBE message to request this information.

This operation is normally executed by configuring a BLF-type function key. The object that needs to be monitored is ufwd<extension>.

Examples

On SNOM

- Through the web GUI, you can configure function keys with:

```
Account: select the account from the drop-down (if only one account is configured on the␣
↪phone, it will be the first in the list)
Type: BLF
value: ufwd<extension>
```

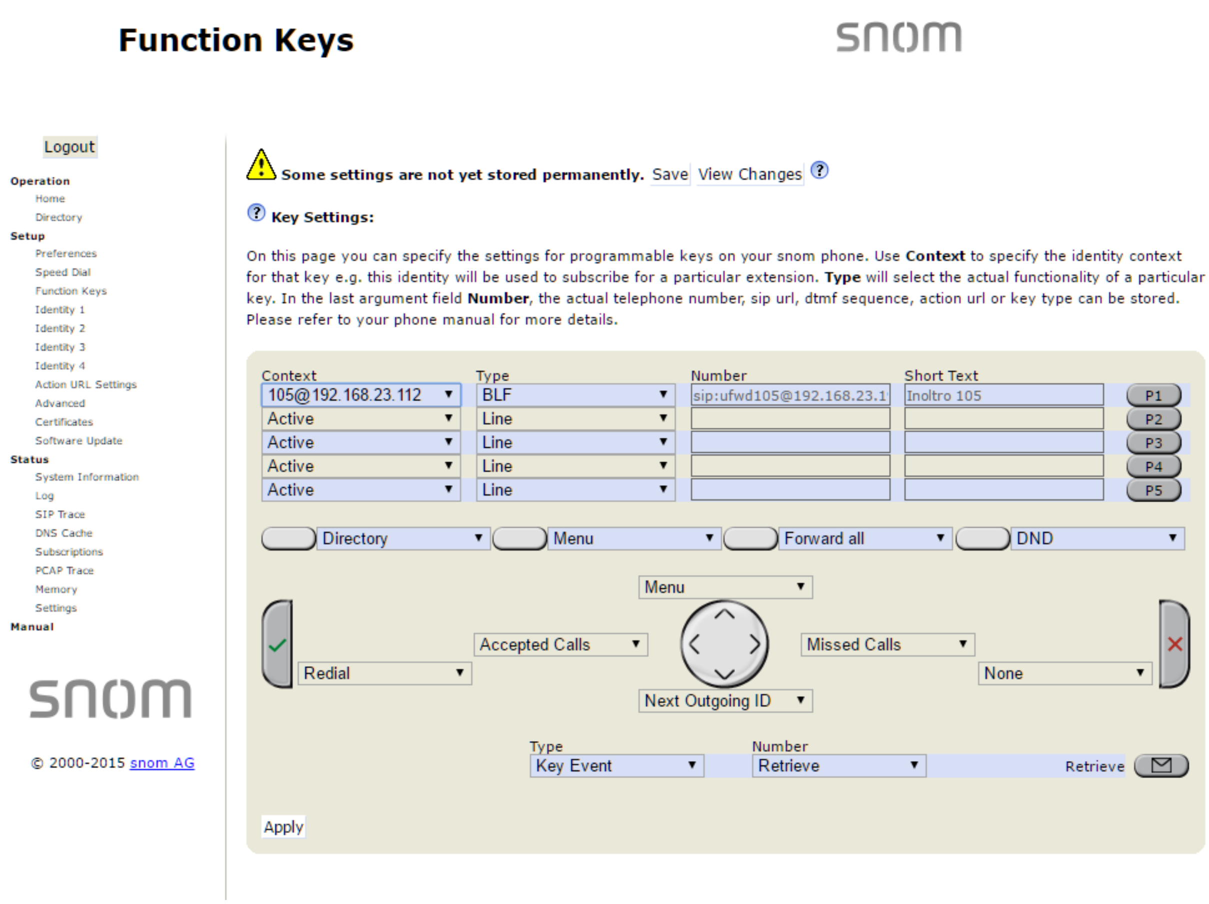
- Or you can directly edit the configuration file or the template:

```
<fkey idx="%%id%%" context="%%line_id%%" label="" perm="">blf sip:ufwd<interno>@%%KPBX_
↪IP_ADDRESS%%;user=phone</fkey>
```

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
<fkey idx="0" context="1" label="Stato Inoltro 105" perm="">blf sip:ufwd105@192.168.23.
↪112</fkey>
```



On YEALINK

- Through the web GUI, you can configure DSS keys with:

```
Type BLF
Value: ufwd<extension>
Line: The line associated with the account (Line 1 if the account is the only one on the phone)
```

- Or you can directly edit the configuration file or the template:

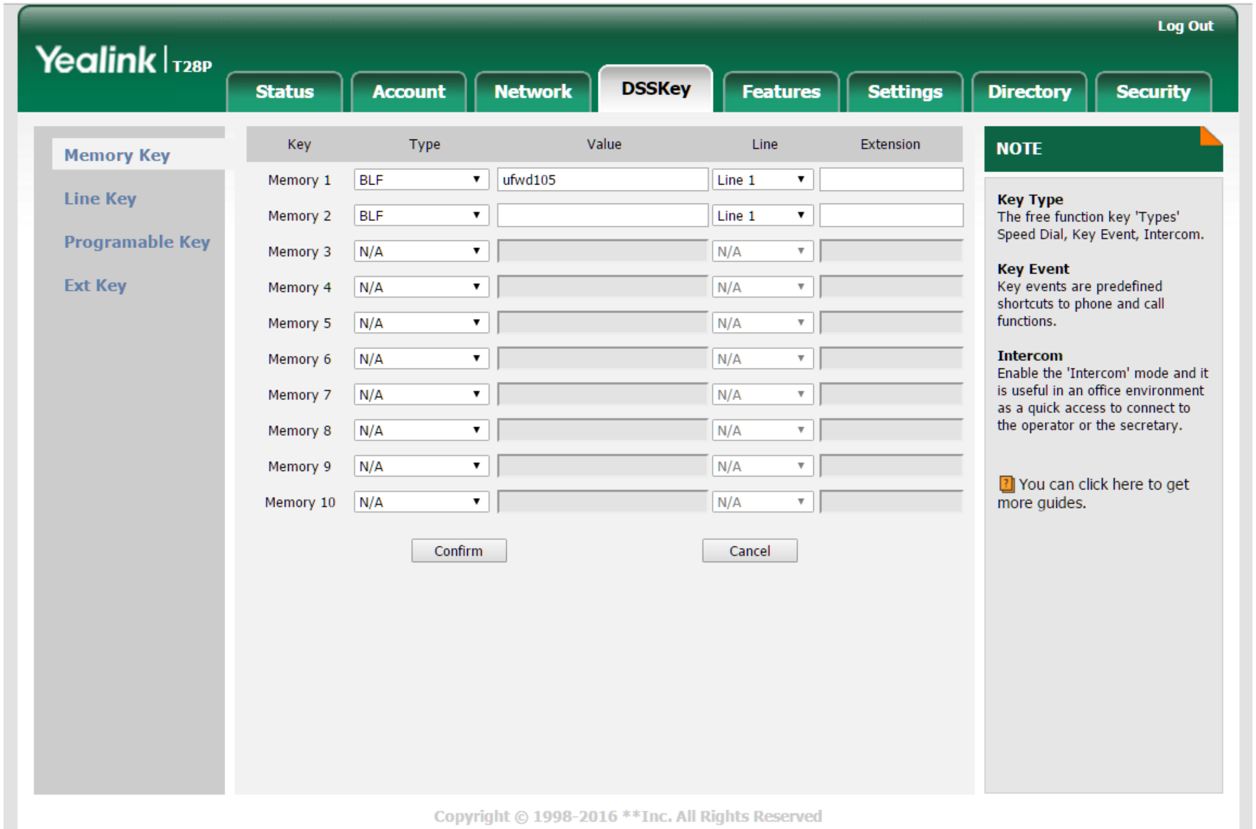
```
memorykey.%%id%%.line=%%line_id%%>
memorykey.%%id%%.value=ufwd<extension>
memorykey.%%id%%.type=16
```

where %%id%% is the ID of the key to configure

and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone)

Example:

```
memorykey.1.line = 1
memorykey.1.value = ufwd105
memorykey.1.type = 16
memorykey.1.pickup_value = %NULL%
memorykey.1.xml_phonebook = %NULL%
```



Call Parking

Description

This service offers the option to move an ongoing call into a parking slot.

When a call is in the parking slot, the interlocutor will be put on hold while the other user can resume the call from any phone (not just the one they parked the call from). The default configuration has 10 parking slots (890-899). If the call is not resumed after 90 seconds, it will automatically be presented to the device (not the extension) that parked it.

Call parking is performed by dialing the configured on-call service code (by default #8). KalliopePBX will reply with the number to call in order to resume the call. The user who parked the call can now hang up and resume conversation by dialing the number provided by KalliopePBX. This call parking mode requires Direct Media to be disabled for the ongoing call. If Direct Media is enabled, a call can be parked out by transferring it to an extension used for the call parking service (by default 888). In this case, once the attended transfer has been executed, KalliopePBX will reply with the number to call in order to resume the call. The user can now hang up and resume conversation by dialing the number provided by KalliopePBX.

Configuration

The service can be enabled/disabled in the PBX -> On-call services page.

The service code can be changed in the PBX -> On-call services page.

The number of parking slots and the corresponding extensions can be configured in the PBX -> numbering plan page.

Interoperability

When using the call parking service, it can be useful to have a key (with Busy Lamp Field) that lets you view whether or not each slot is occupied and, if necessary, resume the parked call.

For monitoring, KalliopePBX sends SIP NOTIFY messages to communicate changes of state. The phone must send a SIP SUBSCRIBE message to request this information.

This operation is normally executed by configuring a BLF-type function key.

The object that needs to be monitored is the extension that corresponds to the parking slot. Other than monitoring the state of occupation of the slot, it is also possible to resume the call by pressing the corresponding function key.

On SNOM

- Through the web GUI, you can configure function keys with:

```
Account: select the account from the drop-down (if only one account is configured on the␣
↪phone, it will be the first in the list)
Type: BLF
value: ufwd<extension>
```

- Or you can directly edit the configuration file or the template:

```
<fkey idx="%%id%%" context="%%line_id%%" label="" perm="">blf sip:<interno>@%%KPBX_IP_
↪ADDRESS%%;user=phone</fkey>
```

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
<fkey idx="0" context="1" label="Parking slot 890" perm="">blf sip:890@192.168.23.190</
↪fkey>
```

On YEALINK

- Through the web GUI, you can configure DSS keys with:

```
Type BLF
Value: <extension>
Line: The line associated with the account (Line 1 if the account is the only one on the␣
↪phone)
```

- Or you can directly edit the configuration file or the template:

```
memorykey.%%id%%.line=%%line_id%%
memorykey.%%id%%.value=<extension>
memorykey.%%id%%.type=16
```

where %%id%% is the ID of the key to configure

and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
memorykey.1.line = 1
memorykey.1.value = 890
memorykey.1.type = 16
memorykey.1.pickup_value = %NULL%
memorykey.1.xml_phonebook = %NULL%
```

Echo Sevice

Description

You can access this service by calling the number specified in the numbering plan (by default 800). After a short message, you will hear your own audio played back to you, letting you verify the correct functioning of your terminal as well as estimate the delay introduced by the network.

Speed Dial

Description

This service lets you create short numbers linked to specific contacts (from either the shared or the personal phonebook).

To quickly call a number, you have to dial the service code (by default # for system speed dial and #0 for personal speed dial) followed by a previously configured code. KalliopePBX will then contact the number associated to the code. For external numbers, it will also add the prefix for outbound calls (specified in the PBX -> General settings page).

Configuration

The service can be enabled/disabled in the PBX -> On-call services page.

The service code can be changed in the PBX -> On-call services page.

Speed dial can be configured by editing the contact profile in the shared phonebook (for system speed dial) or in the personal phonebook.

Speed dial codes can be assigned only to numbers from the local shared contacts, not from remote contacts.

Interoperability

You can also use the Speed Dial function keys provided by your phone. In this case, unlike with speed dials handled through KalliopePBX, the speed dial value will have to include the outbound call prefix (specified in the PBX -> General settings page).

Attended Call Transfer and 3-Way Conference

Description

This service lets a user (transferer) consult another PBX user before transferring a call with a third party (transferred). With this service, the transferer can regain control of the call whenever they want.

If the addressee of the attended transfer does not answer the call (busy, not available, not registered), the failover action set for that specific extension in the Extensions page is executed.

Once consultation has begun, the transferer can switch between two users or convert the transfer into a three-way conference.

An attended transfer is performed by dialing the configured on-call service code (by default #4). KalliopePBX will play audio instructions and ask for the extension number of the addressee. The transferer will then dial the desired extension number and press the # key or wait for default timeout (2 seconds). In the meantime, the transferred party will hear the hold music configured in the Music on Hold page. During consultation with the other user (even if they have not answered yet or if a failover action is executed) the transferer can at any moment cancel the transfer by dialing the necessary code (by default *0).

Once the addressee has answered and is talking to the transferer, the following situations might occur:

- the transferer decides to cancel the transfer by dialing the corresponding code (by default *0);
- the transferer switches between the transferred party and the addressee by dialing the corresponding code (by default *9);
- the transferer converts the transfer into a three-way call by dialing the corresponding code (by default *3);
- the addressee refuses the transfer by hanging up, and the original conversation is automatically restored;
- The addressee accepts the call, the transferer hangs up, and conversation begins between the transferred party and the addressee.

As for every on-call service, attended transfer requires Direct Media to be disabled for the ongoing call. If Direct Media is enabled, a blind transfer can be executed by pressing the specific function key made available by the specific phone model. In this case, failover actions on transfer will not be executed.

Configuration

The service can be enabled/disabled in the PBX -> On-call services page.

The service codes can be changed in the PBX -> On-call services page.

Blind Call Transfer

Description

This service allows a user (transferer) to directly transfer a call with a third party (transferred) to another PBX user. When blind transfer is used, the transferer loses control over the call as soon as they begin the transfer.

If the addressee of the blind transfer does not answer the call (busy, not available, not registered), the failover action set for that specific extension in the Extensions page is executed.

The default failover action for a blind transfer is Return to transferer: the call is returned to transferer and the prefix "R:" is added to the CallerID to notify that it is a return of a call transfer.

As for every on-call service, blind transfer requires Direct Media to be disabled for the ongoing call. If Direct Media is enabled, a blind transfer can be executed by pressing the specific function key made available by the specific phone model.

A blind transfer is performed by dialing the configured on-call service code (by default #4). KalliopePBX will play audio instructions and ask for the extension number of the addressee. The transferer will then dial the desired extension number and press the # key or wait for default timeout (2 seconds). In the meantime, the transferred party will hear the hold music configured in the Music on Hold page. When the addressee answers, they will immediately begin conversation with the transferred party, and the blind transfer will be complete.

Configuration

The service can be enabled/disabled in the PBX -> On-call services page.

The blind transfer service code can be specified in the PBX -> On-call services page.

Voicemail

Description

This service lets a KalliopePBX user receive voice messages even when they are unable to answer a call.

Each voicemail box is linked to an extension. Enabling a voicemail box does not mean that calls to that extension will be sent to voicemail, simply that a voicemail box exists for the user. Whether or not a call is forwarded to voicemail depends on the configured failover actions for the user (as well as other failover/routing rules).

Once a message has been recorded, the user can be notified in various configurable ways:

- a LED that lights up whenever a message is in the voicemail box (MWI – Message Waiting Indicator);
- a notification email;
- an email containing the audio file of the message in the voicemail box.

Whenever a caller is redirected to the voicemail box, before leaving a message they will hear an “audio guide” comprised of two parts:

- a greeting message that corresponds to the audio file: “the extension XXX is currently unavailable/busy” (depending on the reason the call was forwarded to voicemail);
- an instructional message corresponding to the audio file: “please leave a message after the tone, then hang up or press #”.

After the message, the caller will be able to record their message.

Note: The system will not record messages shorter than 3 seconds.

To listen to messages in their voicemail box, the user must dial the voicemail service code (by default 801) followed by their extension number. KalliopePBX will then ask them to input their password, i.e. the service PIN of their extension.

If the user does not dial their extension number, KalliopePBX will play an audio message prompting them to do so.

Once the user has been authenticated, the audio guide will present the options to listen to, delete, or archive messages.

The user can also consult their voicemail box directly from KCTI Mobile. In this case, if the option to send audio messages via email is enabled then “Delete forwarded messages from Kalliope” must not be selected.

Configuration

The service can be enabled/disabled in the PBX -> Numbering plan page.

The service code can be changed in the PBX -> Numbering plan page.

Enabling extensions to use a voicemail box and configuring the voicemail itself is done through the PBX -> Extensions page.

In this page you can also enable email notifications for new messages, audio file forwarding, and audio file deletion.

You can also enable and configure voicemail boxes by importing auto provisioning files for extensions. The columns to fill in are the following:

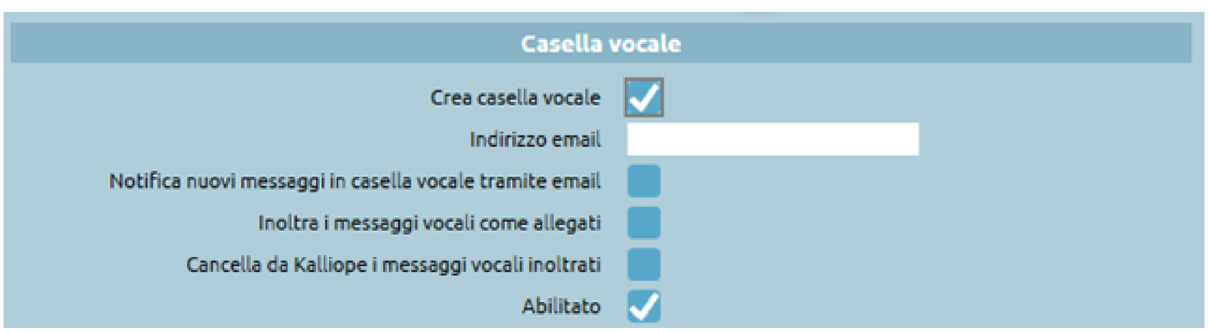
- voicemailEmailAddress: email address;
- notifyEnabled: enable email notifications;
- attachEnabled: forward messages as email attachments;
- deleteEnabled: delete forwarded messages from Kalliope;
- voicemailEnabled: enable voicemail box;

As for the customization of the voice guide, the first part is editable through the phone by contacting the voicemail service code and following the instructions given by the "voice guide. The KalliopePBX after authenticating the user prompts you to , press zero to access the mailbox options and then press 4 to record your temporary greeting message (N.B. you can also record a one-second blank message without voice if you do not want to hear anything). The first part is editable through the phone by calling 801 (or related number modified by the numbering plan) and following the instructions (it will first ask for the voice mailbox (i.e., extension), then for the password (of the voice mailbox), then press zero for the mailbox options, and finally press 4 to record your own tentative greeting message (N.B. you can also record a one-second voiceless blank message if you don't want to hear anything). The second part of the message currently cannot be deleted or edited.

Note: Our roadmap includes the option to configure this message as well. Once this has been implemented, changes will be logged in the changelog and this paragraph will be updated.

Configuration example

Voicemail can be enabled/disabled in the extension configuration page.



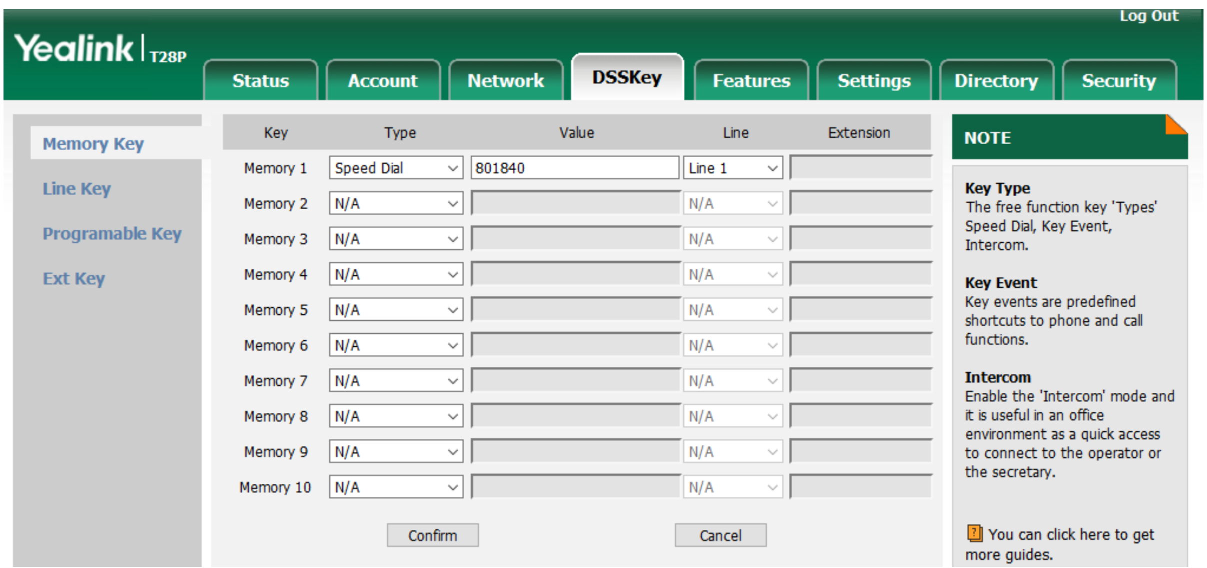
Once enabled, voicemail is available as a failover action for the extension.

Interoperability

You can set a function key to directly access the voicemail box. This can be done simply by setting a Speed Dial key with the voicemail service code followed by the extension number as a value (e.g. if the service code is 801 and the extension is 840, the value should be 801840).

Once the button has been pressed, the phone will dial this number, and the user will be immediately asked to dial their password to access their voicemail box.

An example for a Yealink T28P phone is shown below.



Advanced phone services

Audio conference

Note: The audio conference service has greatly changed starting from firmware version 4.7.12, which introduces "dial-out" mode and a web panel for supervising and monitoring an audio conference room in real time, displaying the status of the participants and execute management actions such as expelling or inviting participants, disabling audio input for one or more participants, etc.

The audio conference service provided by KalliopePBX lets you configure multiple rooms with different settings. These rooms can be independently managed and monitored by individual PBX users (including those without admin privileges).

For each conference room, you can enable "dial-in" and/or "dial-out" access.

With "dial-in" access, participants access the service by calling a give number and join the conference room after inserting an access PIN (sent through DTMF tones).

Conference rooms can be accessed in different modes:

- **access with interactive room selection**: the audioconferencing service has associated a selection within the numbering plan, the default value of which is 802; calling this selection from an extension a guide voice prompts to enter the number of the room to connect to. Depending on the room configuration, the guide voice may be prompted to enter an access PIN.
- **direct access to a specific room**: a call is made to the selection associated with the service concatenated with the room number (e.g., 8021234, in case the service code is 802 and you want to access room 1234). Again, the guide voice will prompt you to enter the access PIN, if provided for the room.

In both these cases, you can allow rooms to be accessed by external callers as well by directing DID on inbound lines or forwards (in the numbering plan, through IVR menus, or from other telephone entities) towards the "Audio Conference Service" and selecting "Ask for room number" or one of the existing rooms.

With "dial-out" access, the PBX will call the configured participants when the room is opened from the GUI.

A room can be configured to have both "dial-in" and "dial-out" participants; a web panel allows the room manager to monitor the status of the room and its participants, with the option to execute management actions such as muting and unmuting one or more participants, expelling a participants from the room, or closing the room (consequently expelling all participants).

Room configuration

Conference rooms are configured in two phases. In the first phase the room is defined by assigning it an identity (i.e. a room number) and settings certain accessory parameters. The second phase concerns the operational settings. In this phase you can set the access mode of the room (dial-in and/or dial-out), set the behavior when a new participant joins the room (e.g. whether of not to require an access PIN) or configure a list of internal or external contacts that the PBX should call when a room with dial-out access is opened.

Creating and configuring an audio conference room

his service can be configured in the "PBX applications" -> "Audio conference service" page. the panel shows the list of existing rooms with their main attributes. To add a new room or edit and existing one you will need to acquire a lock.

To edit an existing room, click on the pencil icon; to add a new one click on "Add new room" above the list of rooms. In both cases a new page will open, in which you will need to set the following parameters:

General settings

- **Enabled**: click the checkbox to enable or disable to room. If a room is disabled it cannot bu used, but its configuration will remain.
- **Number**: the primary ID of the room. Must be numeric. It is used by the system to identify the room, for rooms with dial-in access this is the number that participants must dial when prompted by the automated message or immediately following the service code.
- **Name**: the name assigned to the room. It does not have an operational use; it is used in selection menus when configuring forwards to the room.

Dial-out settings

These settings are needed if you wish the room to have dial-out access. If the room only has dial-in access, these parameters do not need to be set.

The two parameters are identity and outbound routing class and are used by the PBX when calling external numbers. The outbound routing class is used to determine the routing of the call, while the identity is used to set the calling number.

Users enabled to edit the operational settings

This is a list of users who, independently of the permissions given by their role, can edit the operational settings of the room. These settings include the access PIN, the wait music played when only one participant is present, dial-in and dial-out settings. These privileges to not include monitoring and piloting in real time, which can be assigned to different users.

Operational configuration

Once a room has been created you can edit its operational settings by selecting the "Room operational configuration" tab, which lists the existing rooms with a summary of their main operational parameters.

Each room can be "closed" or "open". The operational settings of a room can only be edited (by clicking on the pencil icon at the end of the row) when the room is closed. A room can be opened in two modes, manual and automatic.

In manual mode, a room is opened by an enabled user (see below) by clicking on the corresponding button beside the edit button. Automatic mode is only available for rooms with dial-in access, and happens when any extension joins the room.

The operational configuration page is divided into various sections; the first contains the general settings, which are:

- **Language**: the language of the various audio prompts (e.g. the PIN request or the message played when a new user joins the room).
- **Admin PIN**: the PIN required to enter the room as an admin; note that there may be multiple admins in a single room. if the corresponding flag is enabled, when all admins have left the room all participants will be automatically expelled.
- **PIN**: the access PIN required to enter the room as a standard participant.
- **Announcements enabled**: if this flag is enabled, users who enter the room will be required to say their name so that their arrival and departure can be announced to the other participants.
- **Expel users when the last admin leaves the room**: if this flag is enabled, all participants will be automatically expelled once every admin has left the room.

- **Enable mixing optimization**: optimize performance and audio quality by not transmitting audio from silent participants (Silence Suppression, or Talker Optimization) through the use of VAD (Voice Activity Detection). If enabled, background noise will be reduced, but some brief clipping may occur when a user begins talking, as typical with VAD, since the PBX will need to determine that the audio has exceeded a certain intensity threshold before the user is considered active. Mixing optimization does not significantly impact system performance since any gains from having to decode and mix fewer audio streams are balanced out by the load required by VAD, while the greater part of the performance is due to codifying the resulting audio stream for all participants, which is independent of the number of active participants.
- **Wait music file**: you can set the room to play wait music when only one participant is present; click on "Choose file" to select the audio file you wish to use.

The Dial-in section only contains the checkbox to enable the service. If the service is disabled, users will not be able to access the room by calling the audio conference service; they will only be able to access the room if called by the system in dial-out mode.

The Dial-out section contains the following parameters:

- **Enable Dial-out**: the flag that enables dial-out mode. In order to enable dial-out access for external participants, you will need to set the identity and routing class in the room configuration page.
- **Maximum number of attempted calls per participant**: the max number of times the system will attempt to call each participants; the calls may fail for various reasons (the user may be busy or unreachable, or reject the call). If the invite policy for the user is set to "automatic with repetition" the system will attempt to call again until the max number is reached.
- **Enable audio file playback for complete/incomplete room and corresponding file selection buttons**: with dial-out access, you can set an audio file to play different audio files depending on whether the room is "complete" or "incomplete" based on the presence of all participants marked as "required". This feature is useful for rooms without supervision to let participants know if someone is missing.

Next is the list of dial-out users who will be called by the PBX following the set policy.

Click on the + icon to add a new participant; a new row will appear in which you can specify the participant, choosing between "Extension" or "External". In the former case, the "Contact" field will be a list of the extensions in the PBX; in the latter, you will have to insert the number of the participant, preceded by the external line commitment prefix. In this case, you can also set the name of the participant, which will be displayed in the in the room management/supervision page. For each participant, you can set one of three invite policies: automatic with or without repetition, or manual. See the section on the management/supervision page below for details.

With the automatic invite policy, the room will automatically call the participants the moment the room manager, after opening the room, activates the invite mechanism. The calls are made in parallel; for external participants, it is necessary to ascertain that the call was answered by a person and not an automated system (courtesy messages or voicemail) so the recipient of the call will need to accept the invite by dialing "1". If the tone is received, the PBX will include the participant in the conference, otherwise it will act as if the call had not been answered and end it after a timeout.

If a call fails, the status of the participant will be set to "out of the room" if the invite policy is set to "automatic without repetition", and if they are marked as "required" the incomplete room audio file will play if enabled. If the invite policy is set to "automatic with repetition, the system will attempt further calls until reaching the set maximum. Participants with a manual invite policy will only be called if the action is selected for each individual participant. If the call fails, further attempts will only be made manually.

The final section of the panel lets you add users authorized to operate and manage the room in addition to those authorized to configure the room (because of their role - for which write permissions are necessary - or because specified in the room configuration page).

Note: If dial-in access is enabled, extensions associated to users enabled to manage the room can enter the room in can enter the room as administrators without enterin the PIN.

Users authorized to manage the room will see, if the room is closed, the “Open room” button (play icon) to manually open the room and, if the room is open, the “View room” button (magnifying glass icon) to access the supervision and management panel.

Room supervision and management

Each conference room can be, at a given instant, in one of these three states: “closed,” “open,” or “open and active.” The transition between these 3 states can occur automatically or manually, according to a specific state machine.

As mentioned above, opening a conference room can be done in manual or automatic mode. In manual mode, one of the users enabled to manage it explicitly commands its opening by clicking on the “Open Room” button (play icon) In the case of a dial-in enabled room, the room is automatically opened the moment an extension enters it. In both cases, the status icon in the room operational configuration panel turns into a lens, and the ability to make changes to the operational configuration is inhibited; in order to make changes to the operational configuration of the room, it must first be closed (by entering the room management panel). The “open and active” (or more briefly “active”) status indicates that the dial-out invitation service for participants is active for that room (for those characterized by an automatic invitation policy)

The user, by clicking on the “View status and manage room” icon, accesses the room supervision and management panel, which is divided into 3 sections.

The first section (“Room Information”) shows the name and number of the room, and its status, which can be “Open” or “Active”; in the first case the room is operational but automatic invitations are stopped, while in the second case the control panel takes care of making automatic calls to participants with an automatic invitation policy, and repeating the invitation in case one of those participants leaves for any reason. Next to the status is an X-shaped button that allows you to eject all participants and return to the “closed” room status.

The second section (“Dial-out”) shows the status information related to this room mode, and the buttons that can be used to command its operations. The first flag “Automatic Dial-out” indicates whether the automatic participant invitation function is active or inactive; if the invitation service is inactive, you can start it by clicking on the adjacent “Play” icon; if the service is active, the “Stop” button allows you to eject all participants and stop the automatic invitation mechanism.

The following status indicator shows whether the room is complete or incomplete, based on the presence of the participants who are marked as mandatory. If even one of the mandatory participants is outside the room (unless it has been placed in the “Suspended” state-see below for possible states of room participants) then the room is considered incomplete, otherwise it is in the complete state. In each of the two states a background audio file can be played in the conference room to inform the participants of the condition.

At the bottom of the panel is a list of conference room participants in table form; for each participant there is the following information and a set of actions that can be performed (depending on the nature and state of the participant):

- **Name**: the identity of the extension of the name assigned when adding the participant to the room.
- **Selection**: extension of external number.
- **Call policy**: the policy used when inserting the participant to the room. With dial-out access, this can be manual or automatic with of without repetition. Dial-in participants are also included under manual mode.
- **Required**: participants with this flag are considered necessary when evaluating the completeness status of the room. Disabled for dial-in participants.
- **Dynamic**: determines whether a participant is static (dial-out participant defined in the room operational configuration) of only present in a temporary capacity (either as a dial-in participant or dynamically added through the “Add dial-out participant” button in the dial-out section). When the room is closed, all temporary participants

will be deleted and when the room is next opened automatically only static participants will be called; if the room is opened due to a dial-in participant joining, that user will be present in dynamic mode.

- **Direction**: determines whether the participant is dial-out or dial-in. Dial-in participants will remain included in the list if they end the conversation, are removed from the GUI, or the room is stopped without being closed. You can click on the phone icon uner the "Actions" column to have the room call the user, effectively turning them into a dynamic dial-out participant with manual invite.
- **Status**: each participant can be:
  - **Out of the room**: the user is not participating in the conference; this is the default status when the room is open but not active.
  - **In the room**: the user is participating in the conference.
  - **Invited**: KallipePBX is calling the participant to invite them. If the call fails the room will keep calling them if the policy is set to repetition or set the user as "out of the room". If the user answers and accepts the invitation, the user will be set to "in the room".
  - **Suspended**: the user is temporarily excluded from the room. They will not receive invites and they will not be considered required.

The available actions for each user depend on their status.

- If the user is "out of the room" and has a manual call policy, the room invite service is not active, or the maximum number of call attempts has been reached, the "Invite" (phone icon) and "Suspend" (stop icon) will be available. The former will call the user to the room (with repetitions if configured) and the latter will suspend them.
- If the user is "suspended", the "invite" action will be available.
- If the user is "in the room", the "Hang up" action will be available, which will end the call for the user and suspend the automatic invite policy if set.

A "Mute" action is also available for single or all participants in the room. To mute a single participant, click on the microphone icon under the "Actions" column. To mute everyone in the room use the "Disable all microphones" and "Enable all microphones" buttons present above the list of users.

Blacklist on inbound lines

Note: Introduced in version 4.5.17

Description

This service lets you define specific routings for inbound calls based on the calling and called numbers.

For each pair of numbers, you can choose whether or not to play an audio file (even "in progress") and define the failover action to carry out (among the routing actions available on KPBX).

The pairs of numbers to which the same routing policy should apply can be grouped in blacklists to use on different inbound lines.

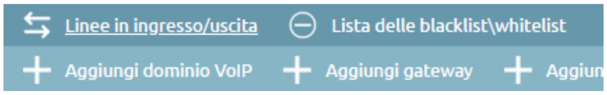
The blacklists are applied to inbound calls, so in a single-tenant system they can be linked to the gateways and VoIP domains. In multi-tenant systems, the blacklists are managed by the tenant admins and are linked to the assigned lines.

You can also link a list of blacklists to each inbound line. These blacklists will be applied in a cascade after the manipulation rules and before the DID rules associated to the line.

Configuration

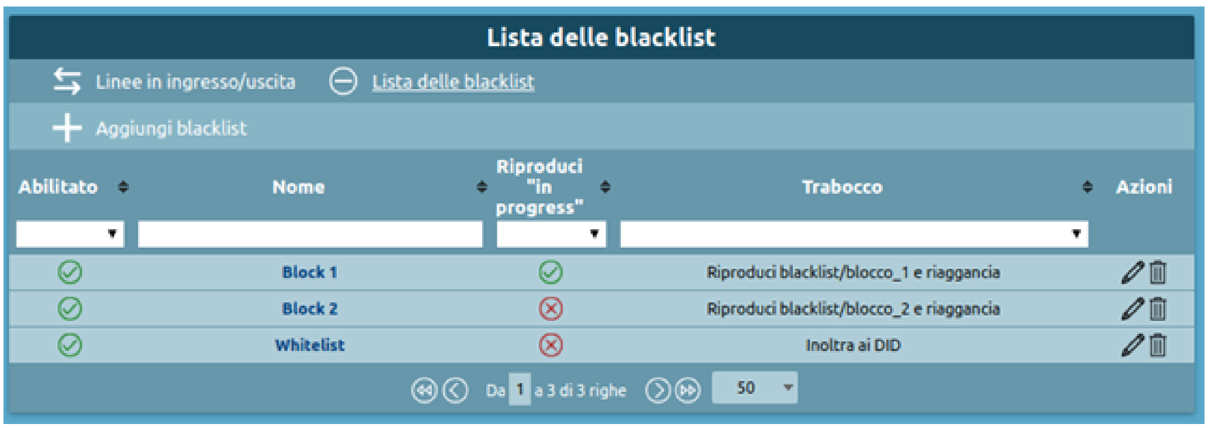
To configure the service, you must first define the blacklist and then associate it to an inbound line.

The blacklist configuration can be set in the PBX -> Gateways and VoIP domains -> List of blacklists page.



Note: In multi-tenant systems, the service can be configured in the PBX -> Assigned line management -> List of blacklists page.

In this page you can create new blacklists by clicking on "Add blacklist" or edit and existing one.



The main fields of the form for defining the blacklist are the following:

- Enablement
- Name
- Failover
- Rules

Rules can be ordered, but, as with the notes, this is purely for organizational purposes. Since they all have the same failover action, the first match will trigger the action.

L'associazione della blacklist alla linea di ingresso viene effettuata su gateway e domini VoIP nel pannello PBX -> Gateway e domini VoIP -> Linee in ingresso / uscita.

The association of the blacklist to the incoming line can be configurated on gateway and VoIP domains in PBX panel -> Gateway and VoIP domains -> Incoming/Outgoing lines .

In multi-tenant systems, this can be configured in the PBX -> Assigned line management -> Assigned lines page.

In either type of system, you can assign blacklists by clikcing on "Add blacklist" and adding all necessary blacklists.

You can also disable all association of the blacklist with the inbound line so that its rules will not be applied.

You can also reorder the blacklists within the inbound lines, so as to implement both blacklist and whitelist mechanisms.

Call Campaign

Note: This service is available for firmware version 4.9.4 or later.

Description

KalliopePBX's "Call Campaign" service lets you execute a session of automated calls towards a group of recipients, playing an audio message and gathers confirmation that each recipient has listened to the call.

On a general level, this service lets you configure call campaigns through a web panel, each with the following settings:

- **list of participants (internal and external)
- **audio message
- **maximum number of simultaneous external calls
- **whether or not to collect confirmations
- **number of call attempts for each recipient

A user with the required permissions can start (instantaneously or with a delay), suspend, or stop a call campaign, or view its status.

A campaign will end the moment, for each configured recipient:

- the recipient answers the call and, if required, provides confirmation that they have answered and/or listened to the message (this is the "success" condition);

the number of call attempts for the recipient has been reached without satisfying the previous condition.

In the case that all recipients have been successfully reached (or the minimum number set for that specific campaign), the status of the campaign will be set to "completed"; otherwise, once the maximum number of call attempts have been reached for all recipients, it will be set to "incomplete". In the latter case, the user may assign further call attempts for one or more of the recipients that have not been successfully reached, or mark the campaign as permanently closed, setting its status to "terminated".

The web panel allows the user to manage individual campaigns in real time (with the ability to view the status for each recipient) as well as access a log with the details of concluded campaigns: this log displays the events and parameters relative to the execution of the campaign and all, so that they can be consulted even if the settings have been changed or the campaign has been deleted.

Configuration

The call campaign services can be configured on the "PBX Applications" -> "Call Campaign" panel. Users may view the panel and its tabs if their role has the required permissions.

The call campaign panel has three tabs:

- **Models list and Campaigns list** (associated to the "Call Campaign Management" permission);
- **General settings** (associated to the "Call Campaign general settings management" permissions)

In the "General settings"' users you can configure the global limits of the service, i.e. the maximum number of simultaneous active campaigns and the maximum number of simultaneous calls (internal, external, or overall) between all active campaigns. the parameters that can be set are:

- **Maximum number of simultaneous active campaigns**: the maximum number of campaigns that can set as "active" at the same time. This parallelism concerns the effective execution of the calls, so for instance so with a parallelism of 1 a campaign can be started even if another one is already being executed. The campaigns will be ordered according to their "priority" attribute, and if two campaigns are started at the same moment, the system will make the calls for the campaign with the highest priority (within the simultaneity constraints set both generally and for that campaign). The empty value means unlimited.
- **Maximum number of simultaneous calls (total/internal/external)**: these three values determine the maximum number of calls that the system can make among all active campaigns. The empty value means unlimited.

Note: The sum of the values set for internal and external calls may exceed the total limit, but in this case the effective number of overall calls cannot exceed it (e.g. with the values 10/8/6, there can be at most 8 internal calls, 6 external calls, but no more than 10 calls in total; if at one moment 7 internal calls are active, only 3 external one can be made)

Every time a call is terminated, whether successfully or not, or when a new campaign is started, the call scheduler (which determines whether or not a new call may be made and towards which recipient among all those for all campaigns) operates according to the current limits.

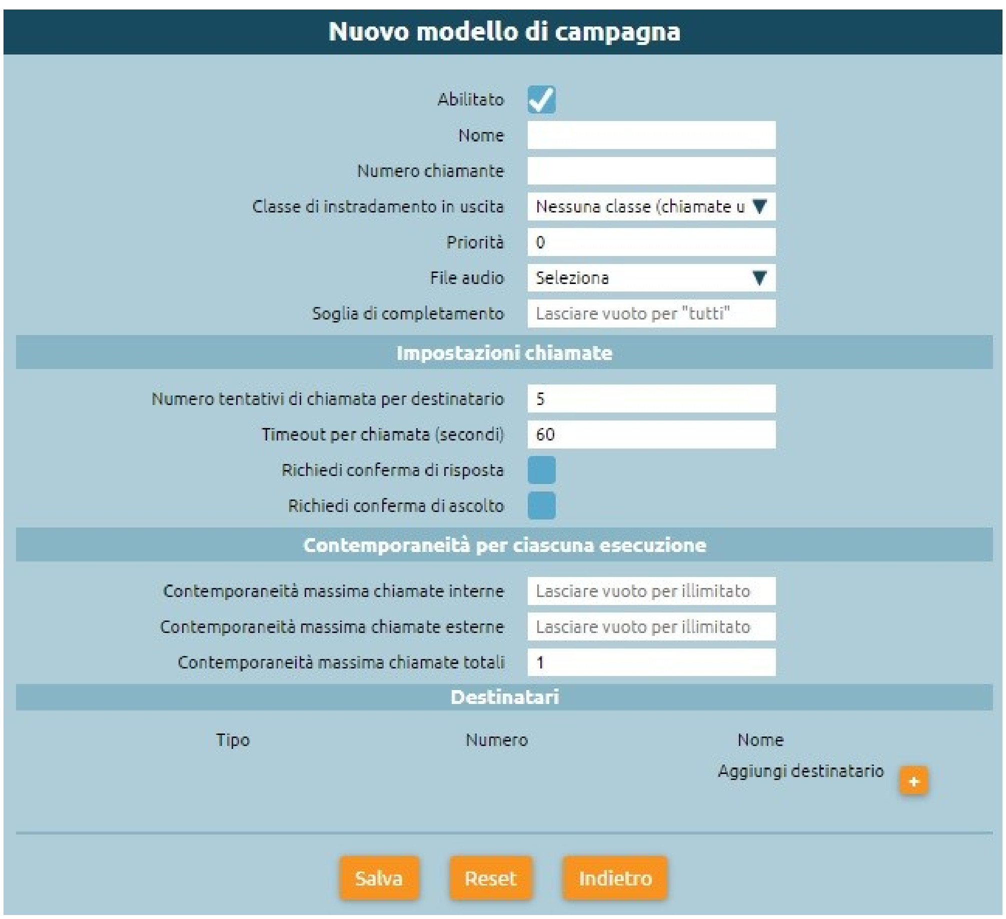
To start a campaign, you must first set a Campaign model with the parameters that determine the behavior of the active campaigns and then start a campaign from that model (or set it to start at a later time). Multiple campaign may be started from the same model at different times, changing a subset of settings such as the audio file to be played or the list of recipients.

Models can be created from the "Models list" panel by clicking on "Add a new campaign model".

The panel for creating a new model or editing an existing one contains the following parameters:

- **Enabled**: this checkbox determines whether the model is enabled or disabled. If a model is disabled new campaigns may not be started or scheduled from it.
- **Name**: the name of the campaign. Can contain alphanumeric characters, spaces, dashes, and underscores. This attribute is used as the Display-Name for calls made by the campaign.
- **Caller number**: the number used by the system to make the calls (both internal and external). For external calls, the effective caller number will be derived from this attribute based on the manipulation rules of the outbound line used.
- **Outbound Call Routing class**: determines the routing of external calls. If the class does not allow one or more of the configured recipients, those calls will fail; no preemptive checks will be made.
- **Priority**: the value used by the scheduler to make the calls if more than one campaign is active at the same time.
- **Sound file**: the audio file played to each recipient.

- **Completion threshold**: the number of recipients that must be successfully reached in order to to consider the campaign complete.
- **Number of call attempts per recipient**: the maximum number of unsuccessful call attempts that will be made towards each recipient. Once this number has been reached for a recipient, the system will not make any new attempts and the recipient will be definitively considered "not reached". The user that manages the campaign may add new call attempts. This value cannot be unlimited (default 5).
- **Call timeout (seconds)**: the ring timeout (in seconds) for each call
- **Request answer confirmation**: this checkbox determines whether or not the system must require the recipient to dial 1 to confirm that they have actually answered before playing the message. This applies to both internal and external calls and ensures that the call was not sent to an answering machine or a voicemail box.
- **Request listening confirmation**: this checkbox determines whether or not the system must require the recipient to dial 9 after the message has been played to confirm that they have actually listened and understood it. If 9 is not dialed, or if another button is dialed, the message will be repeated. If the call ends before confirmation is given, the system will consider the recipient to be "not reached", and if call attempts are still available a new one may be made depending on the policies of the scheduler.
- **Maximum number of simultaneous total/internal/external calls**: these three values determine the maximum number of calls that the campaign cam make. The empty value means unlimited. The sum of the values set for internal and external calls may exceed the total limit, but in this case the effective number of overall calls cannot exceed it.
- **Recipients**: the list of internal or external recipients. For contacts inserted manually, the user can specify a name and an email address along with the number. The list can be ordered; the scheduler will make calls according to the order if other parameters are equal.



Once the model has been created, it will appear in the “models list”, from which an enabled user can:

- **Create any number of models**.
- **Delete one or more existing models** by clicking on the thrash icon. Models from which one or more campaigns have been started or scheduled can be deleted, as the necessary parameters have already been copied to the campaign configuration. The user will still be alerted of the presence of such campaigns through a confirmation popup.
- **Clone an existing model** to create a new one with inherited settings.
- **Edit the configuration of an existing model**; models from which one or more campaigns have been started or scheduled can be edited after clicking a confirmation popup.
- **View the list of recipients** of the campaign.

Starting and managing a call campaign

After configuring a call campaign model, you can start a new campaign, immediately or with a delay.

You can create a campaign by clicking on the “Start” icon in the “Actions” column of the Models List. A new panel will open in which you can set the name of the campaign and change the audio file or the list of recipients if necessary.

Finally, you can configure the start of the campaign in three modes:

- **Immediate**: the campaign will start immediately after you press “start”;
- **Delayed: select the delay** (in minutes) after which the campaign will start;
- **Set schedule**: specify the time and date on which the campaign will start.

Upon confirmation, selecting by pressing the "Start" button, the new campaign (started or scheduled) appears in the "Campaign List" panel. This panel is a record of past and also current campaigns, through which you can view the instant status of the campaign, the progress of calls, as well as some configuration parameters. The campaigns view shows the following information:

- The following configuration parameters: Name, Model, caller number, priority, ring timeout, maximum call attempts per recipient, completion threshold, answer and listing confirmation, simultaneity.
- Start time.
- Total number of recipients.
- Successfully reached recipients: the recipients that have been reached by the campaign according to its settings.
- Current calls: the number of calls that are currently active.
- **Status: the status of the campaign, which can be one of the following**
  - Scheduled: at the time of activation. The status will change to "Active" if the starting mode is immediate, or at the moment set in the configuration if the starting mode is immediate is delayed.
  - Active: calls towards the recipients are currently being made.
  - Paused: can be activated by the user by clicking the "Pause" button. New calls will not be made; current calls will not be interrupted.
  - Blocked: the number of simultaneous calls has been reached.
  - Incomplete: all possible calls towards the recipients have been made but at least one has not been successfully reached.

If the status of a recipient changes to “Failed” to “Idle” (because the user added new call attempts)the the campaign will switch to “Active” mode again. Alternatively, the user can permanently close the campaign by clicking the “STOP” icon.

- Terminated: the campaign has ended even if some recipients have not been successfully reached. If a campaign is “Incomplete”, the user can switch it to this status.
- Completed: this status is automatically reached the moment all recipients have been successfully reached.

Campagne di avviso - Lista campagne

Lista modelli Lista campagne Impostazioni generali

Nome	Nome modello	Numero chiamante	Priorità	Timeout (sec.)	Tentativi massimi	Soglia di completamento	Richiedi conferma di risposta	Richiedi conferma di ascolto	Contemporaneità massima chiamate interne	Contemporaneità massima chiamate esterne	Contemporaneità massima chiamate totali	Programmazione avvio	Stato	Destinatari totali	Destinatari contattati con successo	Chiamate in corso	Azioni
avviso1	Campagna	1212	0	60	5	0			-1	-1	1	2020-01-31 09:48:29	Completata	1	1	0	
avviso2	Campagna	1212	0	60	5	0			-1	-1	1	2020-01-31 12:14:15	Completata	1	1	0	
avviso3	Campagna	1212	0	60	5	0			-1	-1	1	2020-01-31 12:53:13	Completata	2	2	0	
avvioso4	Campagna	1212	0	60	5	0			-1	-1	1	2020-01-31 12:54:54	Completata	2	2	0	
avv5	Campagna	1212	0	60	5	0			-1	-1	2	2020-01-31 12:56:08	Completata	2	2	0	
avviso12	Campagna	1212	0	60	5	0			-1	-1	2	2020-02-05 10:23:41	Incompleta	2	1	0	
avviso11	Campagna	1212	0	60	5	0			-1	-1	2	2020-02-05 10:24:20	Completata	1	1	0	
avviso10	Campagna	1212	0	60	5	0			-1	-1	2	2020-02-05 10:25:03	Incompleta	2	1	0	
avviso14	Campagna	1212	0	60	5	0			-1	-1	2	2020-02-05 10:26:58	Incompleta	1	0	0	
avviso13	Campagna	1212	0	60	5	0			-1	-1	2	2020-02-05 10:26:21	Completata	1	1	0	
avviso16	Campagna	1212	0	60	5	0			-1	-1	2	2020-02-05 10:31:03	Incompleta	1	0	0	
avviso21	campagna 2	2222	1	60	5	0			-1	-1	1	2020-02-05 10:45:45	Completata	1	1	0	
avviso23	campagna2	1111	0	60	5	0			-1	-1	1	2020-02-05 10:50:00	Terminata	1	1	0	
avviso22	campagna 2	2222	1	60	5	0			-1	-1	1	2020-02-05 10:50:00	Terminata	1	0	0	

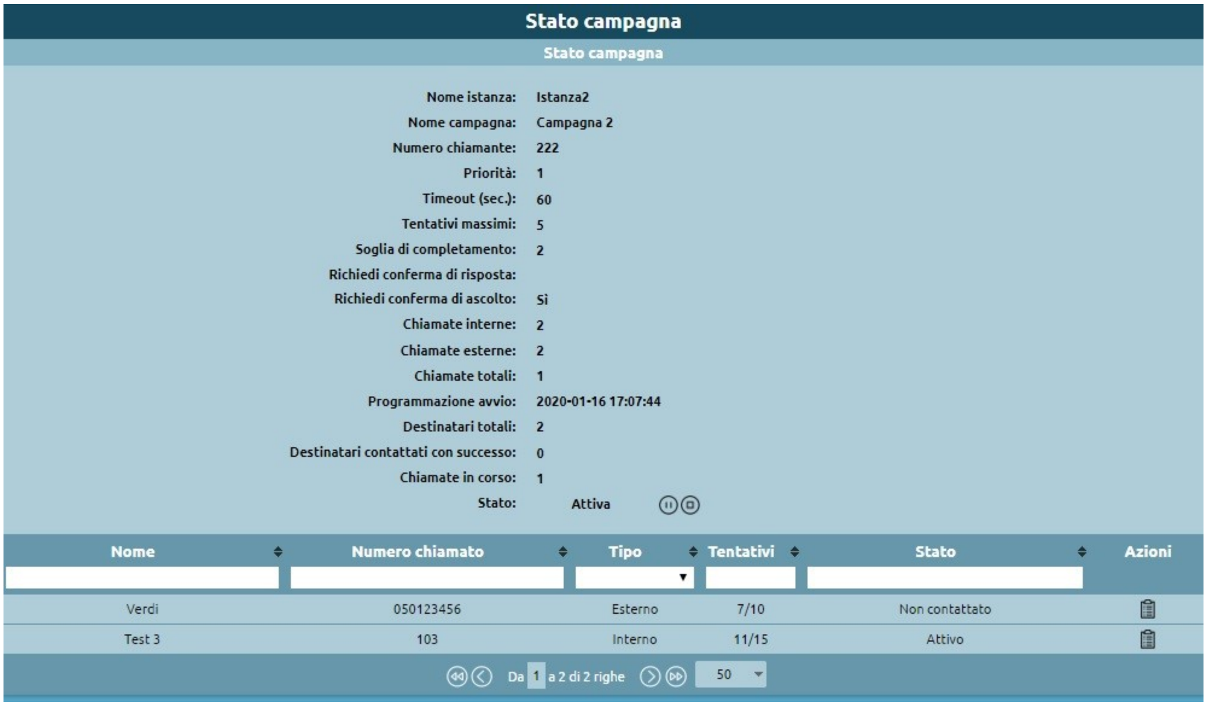
Da 1 a 14 di 14 righe 50

There is also a “Campaign status” panel that displays all the data regarding the status and history of a campaign.

This panel, which can be accessed by clicking the magnifying glass icon, displays the list of recipients and the current status for each: time of last call, outcome, number of call attempts, timestamp of last answer or listening confirmation, list of call attempts and corresponding events and timestamps. Within this panel you can execute actions regarding the campaign (pause and terminate) and the recipients (stop call attempts, add more call attempts, etc.).

The panel shows the state changes of a recipient, which are driven by the initiation and termination of calls directed to it, and can cause the state of the campaign to which they belong to change:

- **Idle**: inactive, available to be called (if the campaign is in Active state).
- **Active**: a call is in progress
- **Contacted**: has been successfully contacted, and required acknowledgments have been given (answered and listened to, if provided)
- **Failed**: several call attempts equal to the maximum expected number have been made, without being successfully contacted. From this state, the user can assign further call attempts to that recipient, which then returns to the “Idle” state again



Call scheduling logic

The logic by which the system decides on the execution of a new call to a recipient belonging to a certain campaign depends on several parameters:

- Maximum simultaneity of campaigns
- Maximum simultaneity of total calls (total/internal/external)
- Maximum simultaneity of calls per campaign (total/internal/external)
- Priority of campaigns
- Start timestamp of each campaign
- Timestamp of last call attempt to each recipient (unsuccessful)

Campaigns active at a given time are sorted by priority, and in case of a tie, precedence is given to the campaign with an earlier start timestamp. Starting with the highest priority campaign, the system generates as many calls until the maximum simultaneities (of the campaign and overall) are reached. Recipients who have not been called previously are called first, according to the order in which they are listed in the campaign configuration. If a concurrent limit has been reached, such as for internal, the system will continue to select recipients from that campaign, choosing only from external ones. When you run out of recipients, who have never been called, if there is still room for more calls for that same campaign, the system will switch to calling back previously called recipients, starting with those who have been called furthest back in time. When a new recipient cannot be selected from this campaign, the scheduler repeats the evaluation on the second campaign (by priority/start time), until the overall call limits are reached, or the available recipients are exhausted. Each time a call ends, a new campaign is started, or when global service limits are changed, the scheduling algorithm is run again to determine whether one or more new calls can be made.

Fast Transfer

Description

The Fast Transfer service allows an extension to transfer an in progress call to a mobile number linked to the extension. This service is useful for users who wish to continue a call on their mobile if they need to leave their workstation.

When this service is enabled, KalliopePBX calls the previously configured mobile number and once communication has been established it bridges the original call with the new one to the mobile phone.

The user can then restore communication on their extension (as with the first transfer, KalliopePBX will first call the extension then interconnect the call channels).

A user who wishes to transfer a call to their mobile or vice versa must simply dial the fast transfer code (by default **). After the user answers, the call will end on the original device and continue on the new one.

Configuration

For global enablement of the Fast Transfer service, it is necessary to enable the Fast Transfer Code (from/to mobile) in the On-Call Services panel and, if necessary, change the code to be used.

Instead, to enable the service for the specific extension, it is sufficient to enter the mobile number in the extension definition panel.

Fork to Mobile

Description

The Fork2Mobile service forwards calls to an extension to a mobile number linked to the extension. This service is useful for users who wish to receive calls even when they are away from their workstation. This service is available for direct calls and calls to groups, but not for calls to queues.

When this service is enabled, KalliopePBX presents the inbound call not only to the accounts linked to the extension but also to a predefined mobile number. Once communication with the mobile phone has been established, KalliopePBX first verifies whether the call has actually been answered (and not, for example, sent to voicemail) and then bridges the original inbound call with the new one to the mobile phone. The other devices will stop ringing once the call is answered. The call to the mobile number follows the outbound routing rules associated to the extension and the caller ID will be the one shown whenever the extension makes calls to the public network. The call to the mobile phone will not therefore include any information on the original calling party.

However, if the KalliopeCTI Mobile app is active and connected, KalliopePBX will send the original calling number to the app, which then displays it to the user. This way, the user can see who is calling them before choosing whether or not to answer.

The difference between this service and unconditional forward to a mobile device are two:

- with Fork2Mobile the mobile number is preconfigured and cannot be changed when activating the service;
- with Fork2Mobile all accounts linked to the user will ring at the same time as the mobile phone until one terminal answers, while with unconditional forward the call will be diverted exclusively to the mobile phone.

The Fork2Mobile service can be enabled and disabled in three ways:

- **From the phone**: the service can be enabled by dialing the activation code (by default *501). KalliopePBX will confirm the activation by playing the "Saved" audio file. Similarly, the user can deactivate the service by dialing the deactivation code (by default *500). KalliopePBX will confirm the deactivation by playing the "Thank you"* *audio file. There are also codes to invert (by default *50*) and to verify the state of the service (by default *509). These codes can only be used from a phone linked to the extension on which the service is being enabled/disabled.

- **From KalliopeCTI Desktop (all versions)**: if Fork2Mobile is enabled for the extension, you can find the mobile icon under the number dialing box. Clicking the icon will enable the service, shown by the icon changing to black. Clicking on the icon again will disable the service. Hovering the cursor on the icon will display the mobile number used by the service.

- **From KalliopeCTI Mobile: tap the Mobile tools icon on the lower right, then the shaking mobile icon. When the service is active the icon will change to white shaking mobile. Clicking on the icon again will disable the service. If the service has been disabled for the extension, tapping the icon will not change its state.

When the service is enabled and the user answers from their mobile phone, they will hear the following audio message: "Press 1 to accept the call."

If the user presses 1 they will answer and other calls will be canceled.

If the user presses any other number the call to the mobile phone will be concluded and the other terminals will keep ringing.

If the user does not press anything after 15 seconds (e.g. if the call was transferred to voicemail on the mobile device), the call will be hung up.

Configuration

Fork2Mobile is always globally enabled. Specific extensions can be enabled by inserting the mobile number in the configuration page for that extension.

The service codes can be enabled/disabled/changed in the numbering plan.

Interoperability

When enabling/disabling the service from the phone, it can be useful to have a key (with Busy Lamp Field) that lets you verify and invert the state of the service.

For monitoring, KalliopePBX sends SIP NOTIFY messages to communicate changes of state. The phone must send a SIP SUBSCRIBE message to request this information.

This operation is normally executed by configuring a BLF-type function key. The object that needs to be monitored is forkm<extension>. The same key will also be set up to call the code to enable/disable the service (by default *50*).

Examples

On SNOM

- Through the web GUI, you can configure function keys with:

```
Account: select the account from the drop-down (if only one account is configured on the phone, it will be the first in the list)
Type: BLF
value: forkm<extension>
```

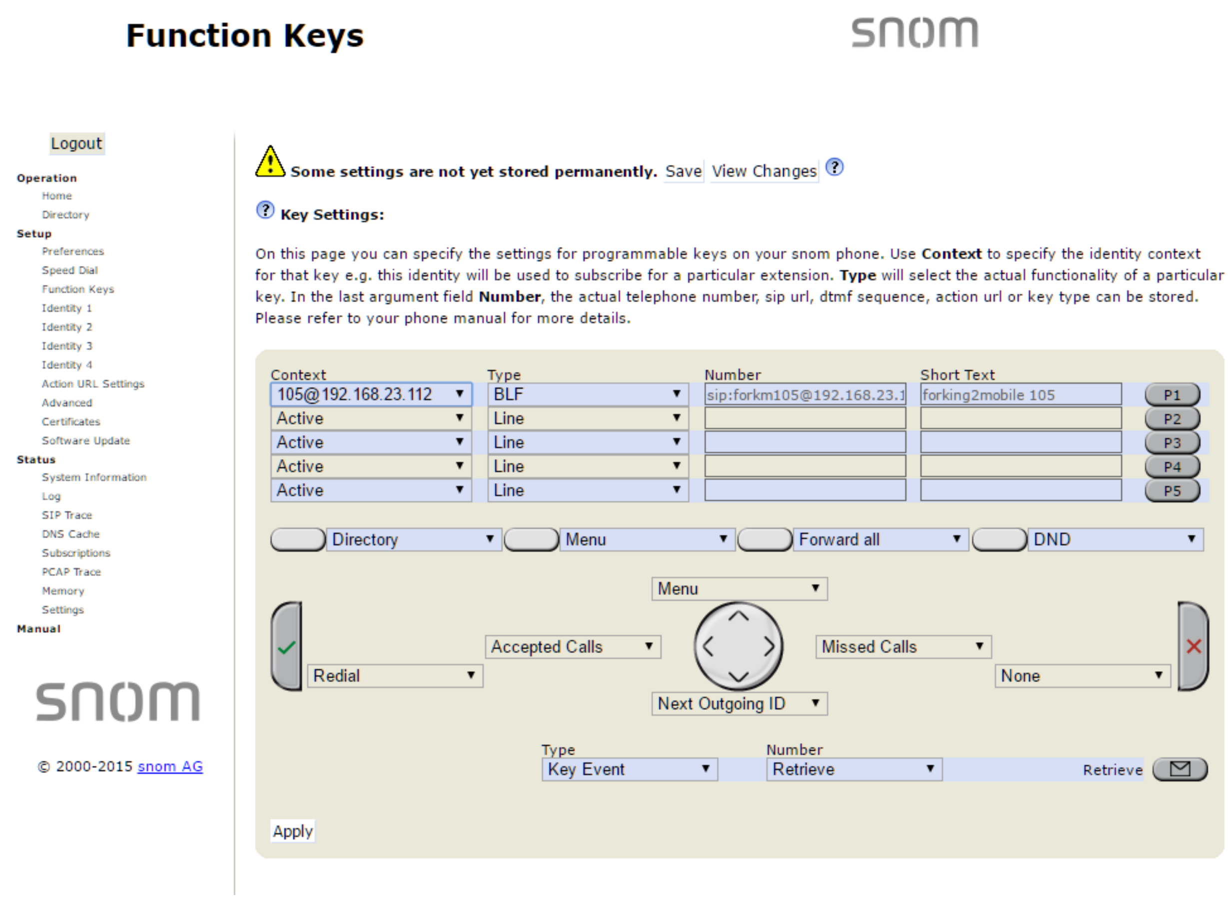
- Or you can directly edit the configuration file or the template:

```
<fkey idx="%%id%%" context="%%line_id%%" label="" perm="">blf sip:forkm<interno>@%%KPBX_IP_ADDRESS%%;user=phone</fkey>
```

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
<fkey idx="0" context="1" label="forking2mobile 105" perm="">blf sip:forkm105@192.168.23.112</fkey>
```



On YEALINK

- Through the web GUI, you can configure DSS keys with:

```
Type BLF
Value: forkm<extension>
Line: The line associated with the account (Line 1 if the account is the only one on the phone)
```

- Or you can directly edit the configuration file or the template:

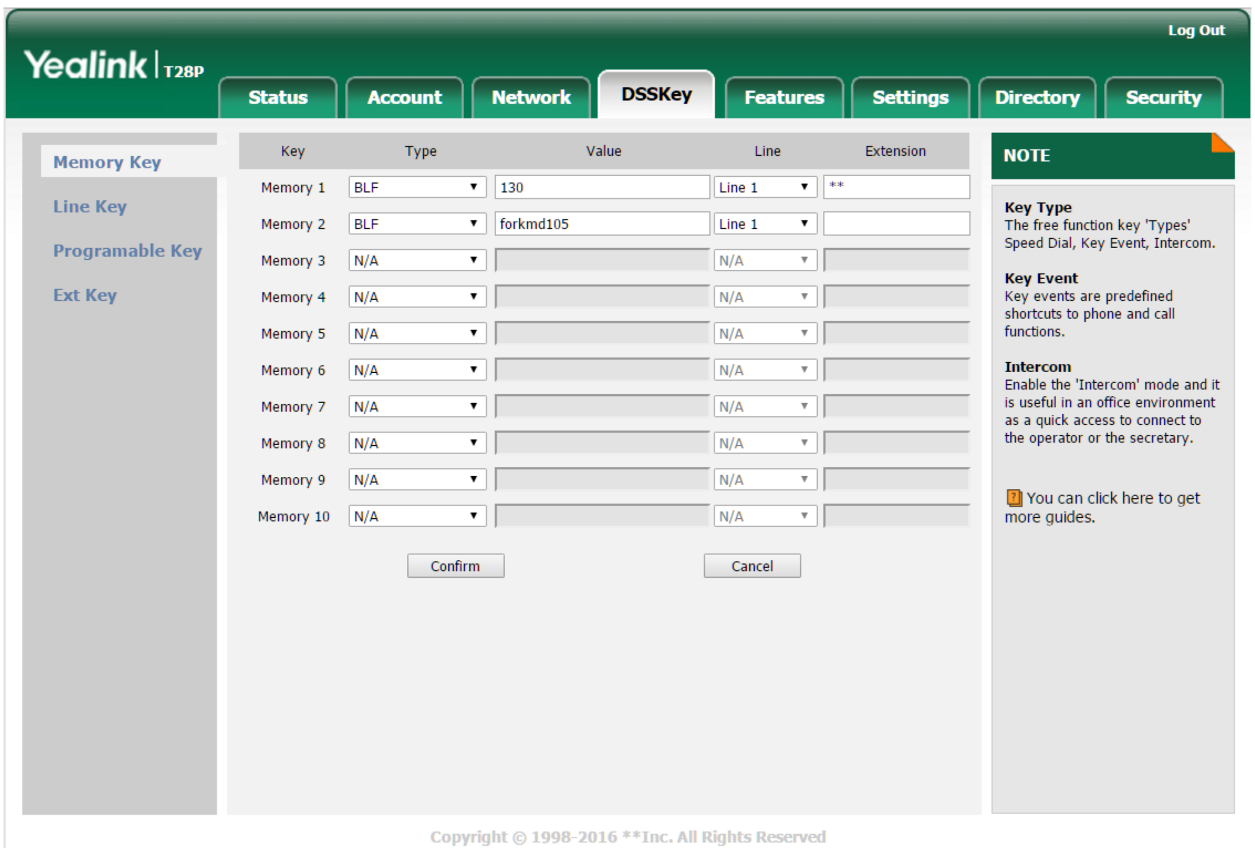
```
memorykey.%%id%%.line=%%line_id%%>
memorykey.%%id%%.value=forkm<interno>
memorykey.%%id%%.type=16
```

where %%id%% is the ID of the key to configure

and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
memorykey.2.line = 1
memorykey.2.value = forkm105
memorykey.2.type = 16
memorykey.2.pickup_value = %NULL%
memorykey.2.xml_phonebook = %NULL%
```



Closed extension groups

Note: Introduced in firmware version 4.5.5

Description

This service lets the admin of the PBX restrict the ability to make calls to one or more extensions to a list of enabled extensions.

If an extension belongs to a closed group, it can be contacted only by the users authorized to make calls to the group.

Configuration

The service can be configured in the Extension configuration panel under the Closed Extension Groups section. In this section you can set which groups the extension belongs to and which groups it is authorized to make calls to.

Example

You can set that extension 101 belongs to group 1 and is authorized to call groups 1 and 2. Or that extension 102 belongs to group 2 and is authorized to call only group 2.

If extension 101 calls extension 102, the call will be correctly established.

If extension 102 calls extension 101, the call will not be established, since 102 is not authorized to make calls to the group that 101 belongs to.

Hot desking

Description

The Hot Desking service allows the telephone identity of a KalliopePBX user to be linked to any enabled phone. This way the user can receive calls to their extension and keep their phone properties (such as calling number, routing rules, failover policies) and a single call record regardless of the terminal they use.

The service is accessed by logging in with the service PIN of the user.

If a user is already logged into the Hot Desking service on another terminal, the other terminal will be immediately disconnected, while all terminals that are statically linked to the user will remain registered to them.

Inbound calls will therefore be simultaneously presented to all static terminals as well as the single Hot Desking terminal.

The Busy Level and number of concurrent calls for each user is specified in the Extensions page.

A user can log into the Hot Desking service by calling their extension from a terminal that is not linked to any extension. KalliopePBX will then ask for their password and the user must input their service PIN followed by # (if # is not pressed the system will still accept the PIN 5 seconds after the last number has been dialed). The system will confirm the login by playing the "Logged in" audio file. The phone will be dynamically reconfigured and the display will show the user information.

The user can log out by dialing the Hot Desking service logout code (by default *400).

Requirements

To enable the Hot Desking service on a terminal, the configuration must be internally handled through auto provisioning, and the terminal must support configuration updates through SIP NOTIFY.

Many phone manufacturers (such as Snom, Yealink, Gigaset, Polycom, etc.) allow this.

For a full list of certified phones and the relative firmware versions, please see the Interoperability section.

Configuration

The Hot Desking configuration involves several different parts of the KalliopePBX web GUI; the main configuration is done through the Hot Desking page reachable from the PBX applications page, but certain settings will need to be edited in the Extensions page and the numbering plan.

Before configuring the service you must first insert the device (including model, MAC address, and IP address) in the provisioning page without linking it to a SIP account. You can omit the template, as it will be configured in the Hot Desking page.

From the Hot Desking page you can set the desired devices (among those defined in the provisioning page but not linked to a SIP account) to Hot Desking mode. The system will automatically generate a SIP account (hotdesk-<mac>) used in the configuration of the terminal in order to allow the call required to log in.

For each Hot Desking enabled extension, the corresponding flag must be enabled (this can be done via template). A dedicated SIP account will be automatically generated (hotdesk-<extension>); this will be used to generate the configuration of the terminal used after login. This account must also be linked to a SIP account template.

Note: The Hot Desking flag and the SIP account template are configuration parameters present in the extension template, and they can be overwritten.

Hot Desking page description

The configurable parameters needed to enable a device to use the Hot Desking service are:

- **provisioning device**: the terminal that needs to be enabled from those defined in the provisioning page;
- **provisioning template**: the template to be used to generate the provisioning file for the terminal;
- **SIP account template**: the template used for the automatically generated SIP account and for the configuration of the terminal.

Enabling extensions

Devices can be enabled to use the Hot Desking service through the extension template, enabling all extensions that use the template, or by overriding the values from the assigned template.

The configurable parameters needed to enable a device to use the Hot Desking service from the template page are:

- **The flag that enables the use of hot desking**
- **The SIP template to assign to the hot desk accoun**t: the SIP template for the automatically generated SIP account, used after a user logs into the Hot Desking terminal.

These parameters are also present in the Extensions page with their template override flags.

Enabling the service

In order to use the service, Hot Desking must be enabled in the numbering plan.

Interoperability

The service has been tested with SNOM and Yealink terminals.

Brand	Series/Model	Firmware
Snom	D3x5	8.9.3.40
Snom	D745	8.9.3.40
Snom	D765 / D725 / D715 / D710	8.7.5.35
Snom	3xx	8.7.5.35
Snom	7xx	8.7.5.35
Snom	8xx	8.7.5.35
Yealink	T19P / T20P / T21P / T22P / T26P / T28P / T32G / T38G / VP530	v70
Yealink	T19P E2 / T21P E2 / T23P / T23G / T27P / T29G / T40P / T41P / T42G / T46G / T48G / VP-T49G	v80

Other models are being tested. Should they prove to be compatible, they will be added to the above table.

In order to prevent Yealink phones from rebooting after reprogramming, you will need to add the following line to the configuration template:

```
sip.notify_reboot_enable = 0
```

Outbound routing (ACR)

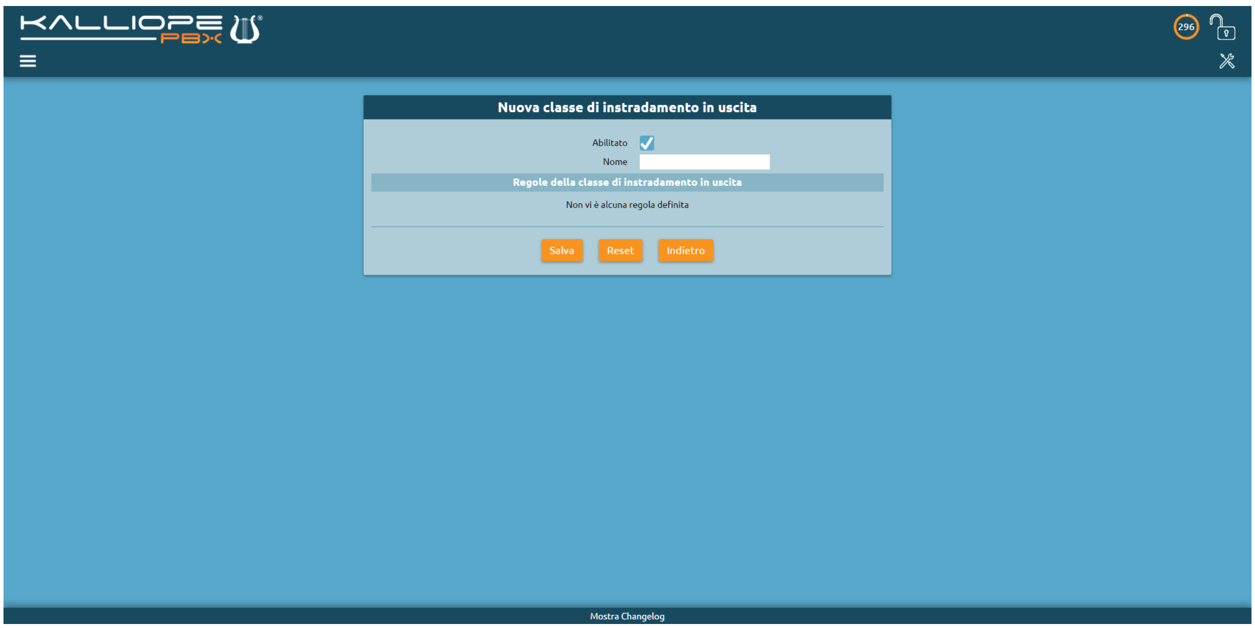
This section details all the settings needed to set up a way to make calls to external numbers.

Outbound routing classes

An outbound routing class is a set of outbound routing rules to be verified in order to establish a routing policy applied to a call from an extension to an external number. The rules are matched in the same order as they are shown in the GUI, from top to bottom.

If the match is valid, the corresponding action will be executed and no further rule will be examined. For this reason it is essential that the rules are arranged from the most particular to the most general. Therefore, it would be wrong to dispose first a general rule that, for example, allows to call all numbers without distinction, since this would lead to the cancellation of all the others that potentially contain particular differentiating features.

You can reorder the rules by clicking and dragging the corresponding icon to the desired place.



The following table lists the configurable parameters for each outbound routing class.

Parameter	Description	Value
Enabled	Lets you disable an outbound routing class without losing its configuration.	Yes / No
Name	Outbound routing class identifier	Alphanumeric

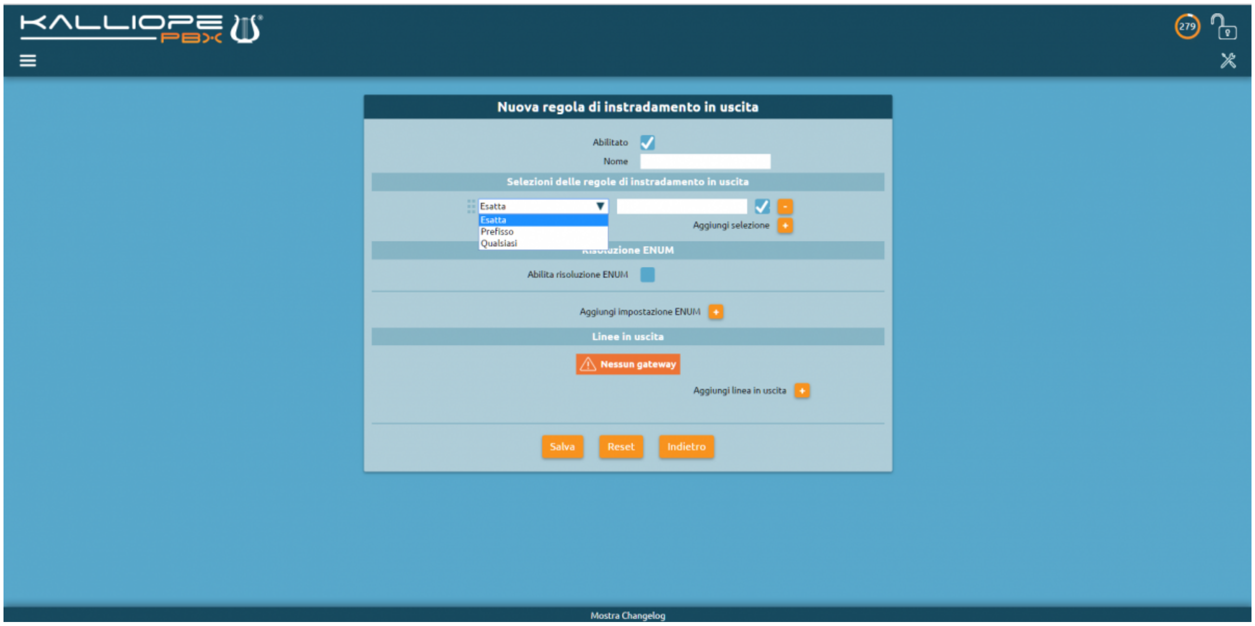
Outbound routing class rules

Parameter	Description	Value
Add rule	Lets you select the configured outbound routing rules and insert them in the order in which they will be checked by the PBX	Outbound routing rule

Outbound routing rules

And outbound routing rule has two components.

- A set of conditions on the called number. The verification may be carried out on a specific phone number ("exact selection") or on a prefix ("prefix"), or else the rule may be validated on any called number ("Any").
- An ordered list of outbound lines on which the PBX will attempt to route the call. Should an error occur on the first outbound line, the PBX will attempt to route the call on the next. The error conditions correspond to a 5xx or 6xx SIP message or a SIP timeout (32 seconds by default). For example, if the PBX receives a 486 Busy Here SIP message, it will make no further attempts to route the call to the next lines.



The following table lists the configurable parameters for each outbound routing rule.

Parameter	Description	Value
Enabled	Lets you disable an outbound routing rule without losing its configuration.	Yes / No
Name	Outbound routing rule identifier	Alphanumeric

Outbound routing rule selection

Parameter	Description	Value
Enable ENUM resolution	Lets you enable ENUM resolution for a specific rule	Yes / No
Add ENUM setting	Lets you specify which search domains (defined in the ENUM settings) must be verified	ENUM settings

Outbound lines

Parameter	Description	Value
Add outbound line	Lets you create a list of outbound lines to use for matched selections	Outbound lines

ENUM settings

An ENUM setting has two components.

- A set of search domains on which to make DNS queries.
- A list of rules to apply if the ENUM server gives a positive response. The behavior may be different depending on the hostname returned in the SIP URI.

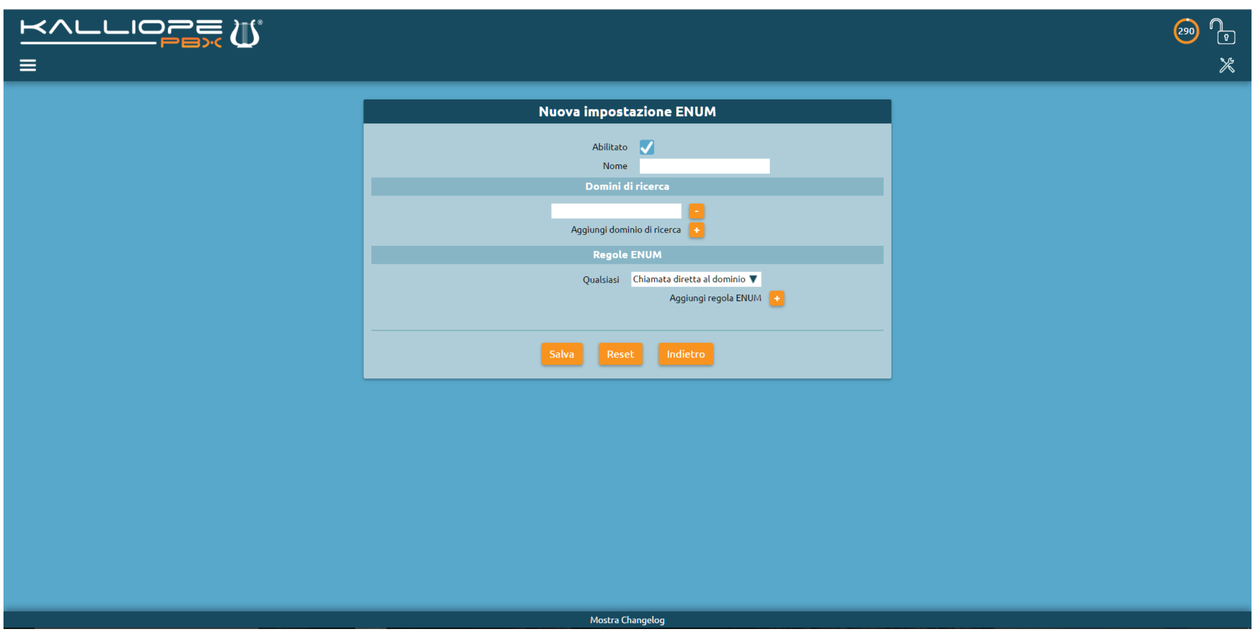
Parameter	Description	Value
Enabled	Lets you disable an ENUM setting without losing its configuration.	Yes / No
Name	ENUM setting identifier.	Alphanumeric

Search domains

Parameter	Description	Value
Add search domain	Lets you specify which search domains must be verified for this setting.	Domain name

ENUM rules

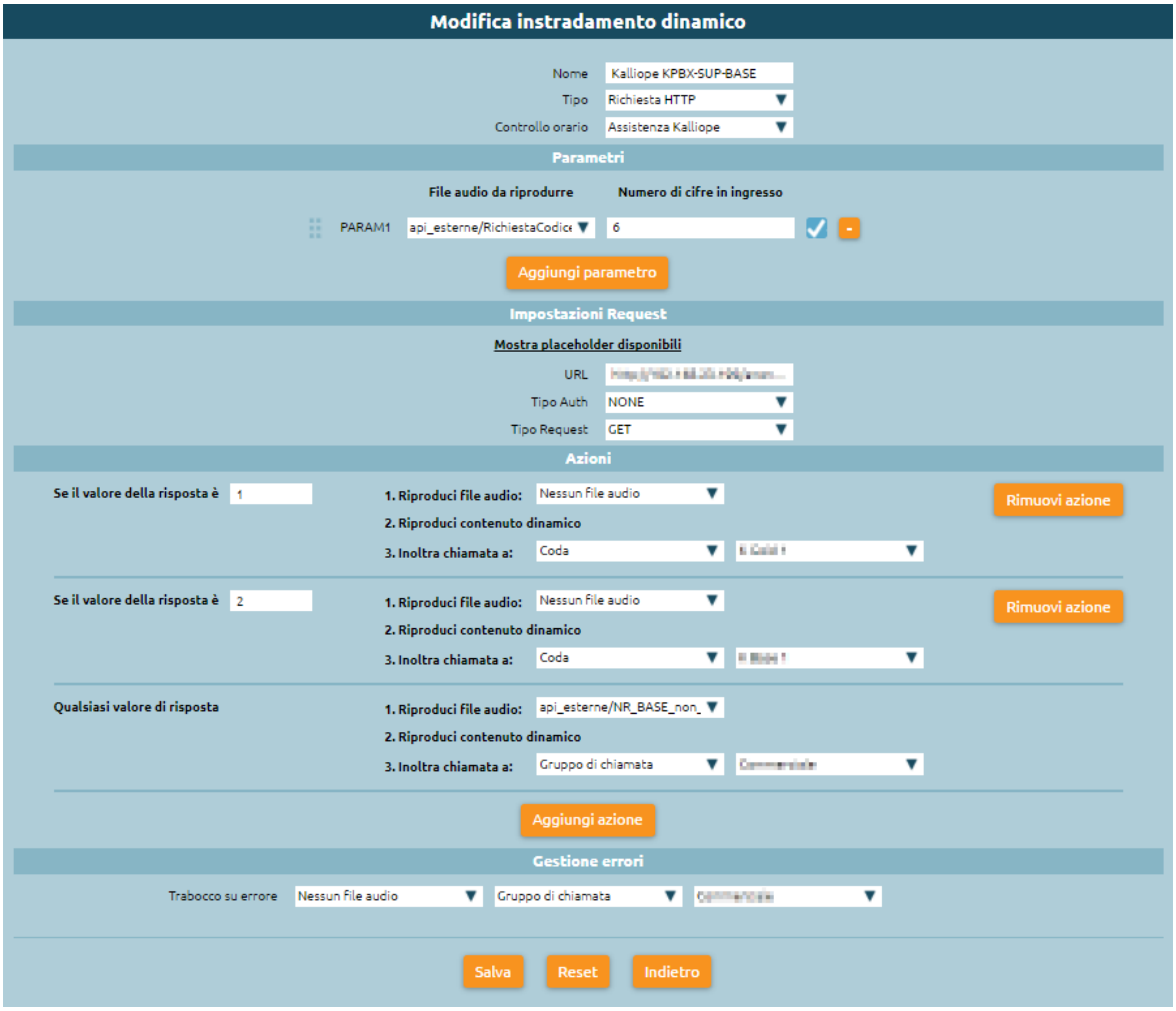
Parameter	Description	Value
Add ENUM rule	Lets you specify a rule to use for outbound call routing if the DNS query on a search domain has a positive result. The outbound line can coincide with the domain returned by the query (direct call to the domain); it is also possible to force a specific outbound line for on-net routing (for example, when it is necessary to use authentication credentials).	Hostname + outbound line / direct call to domain



Dynamic Routing

This service enables inbound call routing based on the response of an HTTP API invoked on an external web server or upon matching one or more parameters specified in a file uploaded to the PBX.

This is done through a series of prerecorded voice prompts asking the caller to input numeric parameters.



Configuration

The service can be configures in the PBX applications -> Dynamic routing page.

During configuration, the following parameters must be specified:

- Name: routing identifier.
- Type: HTTP request (request to an external web server) / Extension (local file).
- Checktime: the time check that must be verified before routing. For dynamic routing, the failover action must be "Return to previous level".

- Parameters: each numeric parameter can be linked to a voice prompt (such as an input request) and, optionally, it is possible to specify the maximum number of digits.

If this value is not specified, the system will assume that the user had finished dialing the parameter after 5 seconds or upon pressing #. If the user fails to input a digit, the prompt will play again, for a maximum of 3 times. The maximum number of parameters is 5.

It is also possible to enable confirmation requests through a checkbox; if enabled, the system will repeat the inputted digits to the caller and ask them to confirm by pressing “1”. If confirmation is not given, the system will repeat the request to insert the parameter.

Request settings

HTTP request The following settings need to be specified for HTTP requests:

- URL: the URL to which the request must be made (HTTP and HTTPS are both supported).
- **Auth type: the type of authentication used by the web server. Can be one of three values:**
  - NONE: no authentication.
  - BASIC: Basic HTTP Authentication; in his case, you will need to specify the authentication credentials (username and password).
  - Client certificate integrated in the PBX (available with firware version 4.5.9 or later): request authentication is performed by comparing the identity of the requester through a unique client certificate. This certificate is integrated in each KalliopePBX, signed by CA Kalliope, and its CN is the serial number of the PBX (e.g. CN=KPBX40412345).
- Request Type: the request method (GET and PST are supported). If POST is used, you can specify the format of the body of the request and the corresponding Content-Type.

The passage from parameters to API is executed through placeholders inserted in the URL or contained in the POST. The recognized placeholders are:

- %CALLER_NUM%: the number of the caller;
- %DNID%: the called number;
- %PARAM1%, . . . , %PARAM5%: the 5 parameters that the caller is asked to insert through the interactive audio menu;
- %UNIQUE_ID%: the unique ID of the call under which the call can be found in the CDR.

For example, with a GET request, the URL might be:

http://www.myserver.com/api?arg1=%CALLER_NUM%&arg2=%PARAM1%&arg3=%PARAM2%

For a POST request, the body might be:

```
<?xml version="1.0"?>
 <parameters>
     <caller>%CALLER_NUM%</caller>
     <param_01>%PARAM1</param_01>
     <param_02>%PARAM2</param_02>
 </parameters>
```

Internal

With an internal request, the file must be uploaded to the TFTP Access folder through the File Manager.

The supported file types are .xls/.xlsx/.ods/.csv and the file may or may not contain columns headers. If it does not, column mapping must be specified, specifying for each field the exact location in the file. It is important that the columns are numbered starting from 0.

The source file must follow this template:

```
| callerNum | calledNum | param1 | param2 | param3 | param4 | param5 | response |␣
↪newCallerNameFull | newCallerNamePrefix | newCallerNum|}
```

Actions

This section contains the configuration of the routing rules to execute based on the response from the web server or file matching.

HTTP request

With an HTTP request, the system expects a "200 OK" type response from the web sever, the body of which must contain an XML text as follows:

```
<?xml version="1.0"?>
 <response>
     <message>
         <elem>digit:1</elem>
         <elem>number:200</elem>
         <elem>alpha:c234</elem>
         <elem>audio:custom/%TENANT_UUID%/sounds/misc/pluto</elem>
         <elem>number:201</elem>
         <elem>digit:123</elem>
         <elem>audio:custom/misc/pippo</elem>
     </message>
     <displayprefix>text</displayprefix>
     <value>105</value>
 </response>
```

Note: The string %TENANT_UUID% must be replaced with the actual value of the UUID specified in the "information" widget on the dashboard.

The <message> and <displayprefix> tags (available with firmware version 4.3.5 or later) are optional; <message> specifies the dynamic component of any audio prompt that is set to play to the caller, while <displayprefix> lets you change the Display Name of the call by adding in front of it the prefix specified in the tag itself.

The <value> tag is required and indicates the API response based on which the actions to be executed are defined.

Actions can be defined as follows:

- Play a static audio file uploaded to the system;
- Play dynamic content;
- Forward the call to a new destination or hang up.

The dynamic content may be comprised of a series of concatenated messages based on the sequence of <elem> tags present in the <message> response. The following table lists the possible <elem> formats and the corresponding additions to the dynamic content:

<elem>...</elem>	Example	Firmware	Description	Example output
digit:{digit_sequence}	digit:1234		Plays the individual digits (numerical, from 0 to 9) of {digit_sequence}	Plays the audio message "one two three four"
number:{number}	number:1234		Plays {number}	Plays the audio message "One thousand two hundred thirty four"
alpha:{alphanumeric_sequence}	alpha:1a2b3c4d		Plays the individual alphanumeric characters of the string {alphanumeric_sequence}	Plays the audio message "One a two b three c four d"
audio:{audio_file}	audio:custom/misc/test		Plays the audio file present on KalliopePBX identified by the name {audio_file} (including the path, as displayed in the "Sounds" -> "Audio files" page)	Plays the audio message custom/misc/test
dtmf:{dtmf_sequence},{duration},{intertone_pause}	dtmf:123w*,350,200	4.3.5	Plays DTMF tones (according to the mode specified for the channel from which the call originates) specified in {dtmf_sequence}; each tone lasts {duration} milliseconds (expressed as an integer) and separated by a pause of {intertone_pause} milliseconds. The allowed characters are the valid DTMF digits (0-9,*#,a-d,A-D) plus "w" and "W" to indicate a pause of 500 or 1000 milliseconds. **N.B.**: the inter-digit pause is inserted even for the "w" and "W" characters.	Plays the following DTMF tones, each for 350 milliseconds: "1", 200 ms pause, "2", 200 ms pause, "3", 200 + 500 + 200 ms pause, "*"
pause:{duration}	pause:450	4.5.12+	Inserts a pause of {duration} milliseconds	Inserts a 450 milliseconds pause

Routing rule

The forwarding action to be executed. In addition to normal actions, it is also possible to forward the call to the numbering plan, using the value contained in the response as the selection.

Extension

With internal requests, the system will verify the calling number, called number, and parameters on the file and return the value that will be used to choose which actions to execute from the corresponding response column.

If there is more than one row that matches the data, the action will be the one corresponding to the row with the most matches (best match).

In this case, dynamic content is not available and the available actions are therefore comprised of the following:

- Play a static audio file uploaded to the system
- Routing rule

At this point the forward action is defined. In addition to normal actions, it is also possible to forward the call to the numbering plan, using the value contained in the response as the selection.

If the parameters are incorrect, the call comes from a number not specified in the file, or no response value is specified, then the error management action will be executed.

It is also possible to change the name and number shown on the display of the called phone by inserting values in the newCallerNameFull, newCallerNamePrefix, newCallerNum fields.

Specifically, newCallerNameFull will replace the name of the caller, newCallerNamePrefix will be added before the calling number, and newCallerNum will replace the calling number.

- Error management: if the request to the web server returns an error or the file matching is unsuccessful, the call will be handled as specified in this section.

Example of dynamic routing

callerNum	calledNum	param1	param2	param3	param4	param5	response	newCallerNameFull	newCallerNamePrefix	newCallerNum
102									simona	111
103		111	22				100			
		123	123	333	123	123	100			321

After the exact selection for dynamic routing has been created in the numbering plan, an extension calls this number.

If the call comes from 102, the system will send response 200 and execute the associated action.

If the call comes from 103, an audio message will ask for the parameters; if the inputted values are correct the system will send response 100 and execute the associated action.

If the call comes from an unspecified extension, an audio message will ask for the parameters; if the inputted values are correct the system will send response 100 and execute the associated action.

In all other cases, the error management action will be executed.

Multilevel IVR

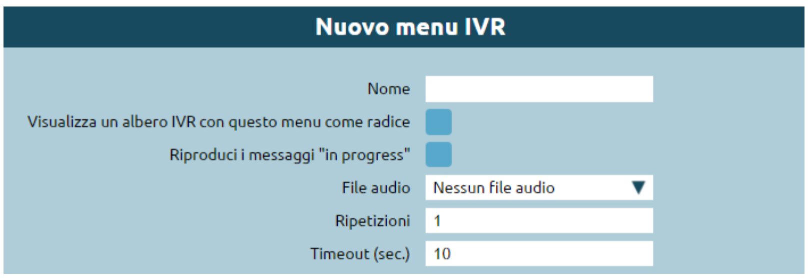
Description

The Interactive Voice Response allows the caller to be routed to a specific service according to the DTMF input.

Configuration

You can reach the IVR Menu selecting PBX Applications.

It's possible to create an arbitrary number of IVR menu that can be drop-down menu or indipendent menu.



- **Name**: you can fill this field with the IVR name you've chosen
- If you want to create drop-down menus you can choose to **Display the IVR tree visualization with the current menu as tree root**.
- The **"Play messages "in progress""** checkbox plays the free audio message (triggered from SIP 183 Session Progress) before the dispatch of SIP message 200 OK that would trigger the beginning of the calling and the billing as a result. This feature is recommended for companies that have a toll-free number associated with the IVR, because the company itself pays the cost of the call and in this way can save precious seconds for each call received. The availability to accept the Session Progress SIP 183 message depends on the operator to which the central unit is connected and normally has a maximum duration of 59 seconds.
- The **Sound File** selection allows you to select a pre-recorded track that contains various options the client can choose using the numeric keypad.
- The **"Repetitions"** field can be filled with the number of time the audio file will be executed. You can't choose 0 because that means the file audio won't be executed.
- **Timeout (sec.)** indicates the selection wait at the end of the audio message. The system is on hold for the indicated number of seconds before performing a second repetition (if there is any) or triggering the default action (field below).

Default Action

The default action is preset as "go to: Hangup" but there are other options in the drop-down menu. You can choose to play an audio file in case the user didn't select anything so the user can notice the error and be forwarded on another list options.

Selections

In the Selections part you can associate the number typed by the user to a specific option in the drop-down menu. Some examples:

- **Hang-up**: the call is over after the audio repetitions.
- **External number**: the call is routed to an external number that can be a cell phone or a number that doesn't belong to the PBX.

You can choose the CLID that is used to identify the phone number of the calling user.

- **Extension**: the call is transferred to an extension number that you can chose using the box on the right.
- **Callgroup**: the call can be reached from more than one extension, the available one can answer an inbound call. For more informations visit the Ring Groups page.
- **Callgroup (bypass checktime)**: time control on the call groups can be ignored using this option.
- **Queue**: incoming calls don't engage one or more destinations, but are placed in a queue. The call is removed from the queue when there are operators available to serve the customer. For more informations visit the ACD Service page.
- **Queue (bypass checktime)**: time control on the queues can be ignored using this option.
- **Checktime**: is a way to manage the routing of calls on a time/manual basis. The time-based process consists of defining time slots that are detected in order: if current date and time match with one of these, the call is forwarded, otherwise a general action is performed. The manual mode consists in on/off switches that can be controlled thanks to a code typed from telephone and/or BLF key. For more informations visit the Checktime page.
- **IVR menu**: allows you to select an existing IVR menu that will be linked to the current one, this creates a cascading menu.
- **Voicemail**: allows a user to receive voice messages when they're unable to answer a call (if busy or unavailable). For more informations visit the Voicemail page.
- **MeetMe Room**: it's possible to log into an audio conference service in which internal or external people are connected to. You can choose to ask the room number (PIN) or to directly send the user to the room.
- **Dynamic Routing**: this service allows to manage the call whether by invoking an external web service (like the original application) or by finding the parameters on a XSL/CSV file loaded on the PBX. For more informations visit the Dynamic Routing page.
- **FAX instance*: allows you to select a FAX instance that is considered a physical FAX that can be accessed by one or more users. For more informations visit the Fax Module page.
- **Numbering plan – custom selection**: allows you to insert a selection that has already been configured in the numbering plan.

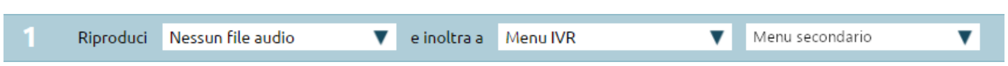
- **Numbering plan – ask for selection**: an audio file that asks for selection will be played. This allows the caller to choose which extension wants to reach. The possibilities are listed in the audio file.
- **Allow digit exten enabled**: allows the caller to type a number - if he already knows it - without listen to the possibililities. The number can be composed of several digits, so the IVR service will wait until the user finish, to avoid directing the customer to the wrong selection.

IVR submenu (in cascade)

To create a cascading IVR menu have to create another menu by proceeding in the same way. Before doing that, save the previously created IVR menu using the Save button. After that, you will be in the main screen. To add another menu, press Add IVR Menu.

After configuring and saving the second menu you can find both on the main screen.

Then, editing the Main Menu you can select the option forward to: IVR Menu and choose the secondary menu in the last box on the right.



Back on the main page you can distinctly see the connection between the two menus by selecting the Tree View.

Electronic Lock

Description

The Electronic Lock service lets a user block calls from a device to certain external numbers. Calls to other extensions (including remote extensions) or PBX services are always accessible even from locked extensions.

A locked phone can only make calls enabled in the “restricted” routing class, while an unlocked one can make any call enabled in the “standard” routing class.

There are different unlock modes, and the duration the hone remains unlocked depends on the selected policy.

The electronic lock is configured per extension, so all accounts (devices) linked to the user will share electronic lock status.

The unlock modes are:

- **Open**: the electronic lock is always disabled, and the standard outbound routing class is always applied.
- **Code**: the electronic lock can be disabled or enabled with the unlock code specified in the numbering plan (by default 850 and 851 respectively).
- **Password**: the electronic lock can be disabled or enabled with the unlock code specified in the numbering plan (by default 850 and 851 respectively) and the service PIN of the extension linked to the device.

The possible unlock policies are:

- **Per call**: the lock must be disabled before every call.
- **Automatically block after the number of minutes below**: The lock will be automatically enabled after the specified interval.
- **Unlocked until locked by the user**: Once the lock is disabled, it will remain so until enabled again by the user.

A user who wishes to make a call on a device linked to a locked extension must follow this procedure.

1. call the unlock service (by default 850);
2. if the unlock mode is set to Password, the PBX will instruct the user to input their service PIN with the “Password” audio prompt;
3. the user must now dial their PIN followed by #;
4. the PBX will confirm that the lock has been disabled by playing the “Saved” audio file;
5. if an error occurs, the PBX will play the “Login error” audio file and hang up.

To lock the extension again (if the policy does not involve automatic lock) the user must follow this procedure:

1. call the lock service (by default 851);
2. the PBX will confirm the lock by playing the "Thank you" audio file.

Configuration

To configure the electronic lock it is first necessary to define the outbound routing classes that must be applied when the lock is enabled and disabled.

Afterwards, the following parameters need to be set in the Extensions page:

- Standard outbound routing class
- Restricted outbound routing class
- Unlock mode
- Unlock policy
- Unlock duration (sec.)

Examples

- International calls can only be made with a PIN: in this scenario, the restricted class allows all calls except international ones, which are allowed by the standard class. The unlock mode is "Password" and the unlock policy is "Per call".
- Calls can only be made with a code: the restricted class only allows emergency calls while the standard class allows all calls. The unlock mode is "Code" and the unlock policy automatically re-enables the lock after 30 seconds.

Completion of calls to busy subscriber (CCBS)

Description

With this service, a caller attempting to reach a busy extension can instruct the PBX to call them back once the other party is available. This service is exclusive to extensions calling other extensions.

If a user wishes to use this service, they must dial the CCBS code (disabled by default) within 20 seconds of the end of the call. KalliopePBX will confirm by playing the "Saved" audio file. When the busy user is free, the PBX will call the user who activated the service (their phone display will show the label c2c followed by the number of the extension they originally called), and once they answer, they will make the call to the other user.

If the user wishes to cancel the service, they must dial the CCBS cancel code (disabled by default). KalliopePBX will confirm by playing the "Saved" audio file.

This service is valid for one hour. If the service is activated again before the previous activation expires, the old one will be automatically canceled (only one instance can be active per extension).

Configuration

The service can be enabled/disabled in the PBX -> Numbering plan page.

The service code can be changed in the PBX -> Numbering plan page.

Interoperability

When using the CCBS service, it can be useful to have one key available to make the reservation and one to cancel it. This is normally done by configuring two Speed Dial type function keys with values equal to the reservation code and reservation cancellation, respectively.

Boss/secretary filter

Description

This service lets one or more users (secretaries) intercept calls to another extension (boss). Only the secretaries (and, optionally, other bosses in a customizable group) will be able to directly contact a boss on their extension. The secretaries have the task of answering calls, checking whether the boss is available, and, if necessary, forwarding them to the boss.

The secretaries have the task of answering calls, checking whether the boss is available, and, if necessary, forwarding them to the boss.

The service can be enabled or disabled on a group level (for all secretaries) or just for one. In practice, a boss enabling the service will usually enable it for all secretaries, while individual secretaries can choose to enter or exit the service. Even if the boss has not enabled the service, a secretary can do so for their own extension.

The boss/secretary service can be enabled by dialing the corresponding codes. The service can be enabled on a group level with the activation code (by default *521) followed by the extension number of the boss. KalliopePBX will confirm the activation by playing the "Saved" audio file. The service can be enabled for a specific secretary by adding to the global code * followed by the extension number of the secretary. KalliopePBX will confirm the activation by playing the "Saved" audio file.

Similarly, the service can be disabled by dialing the deactivation code (by default *520) followed by the extension number of the boss. KalliopePBX will confirm the activation by playing the "Saved" audio file. The service can be disabled for a specific secretary by adding to the global code * followed by the extension number of the secretary. KalliopePBX will confirm the activation by playing the "Saved" audio file.

There is also a code to invert the state of the service (by default *52). Similarly to activation/deactivation, the code must be followed by the extension number of the boss to act on a group level, while the state of the service can be inverted for an individual secretary by adding to the global code * followed by the extension number of the secretary. In both cases, KalliopePBX will confirm the activation by playing the "Saved" audio file.

These codes can only be used from a phone linked to an extension that belongs to the boss/secretary group.

Configuration

The service can be globally enabled and the service codes can be changed in the PBX -> Numbering plan page.

The configuration of the Director-Secretary groups is done in the panel Director-Secretary Groups. The parameters to be configured for the groups are as follows:

Parameter	Description	Value
Enabled	Lets you disable a boss/secretary filter without losing its configuration.	Yes / No
Name	Boss/secretary group identifier.	Alphanumeric
Allow calls from other bosses	If this option is enabled, extensions labeled as bosses in other groups will be able to call the boss directly without passing through the filter.	Yes / No
Select boss	The extension whose calls will be intercepted by the secretaries.	Extension
Add secretary	The list of extensions that will intercept the calls to the boss.	Extension

Failovers

Here failover actions are defined in case one of the overflow causes occurs for calls diverted to secretaries. The additional failover action "Forward to Director" is provided for this service

Parameter	Description	Value
Extension	Failover action on calls from extensions (including remote extensions).	
External	Failover action on calls from external numbers.	
Transfer	Failover action on call transfers	
Timeout (sec.)	Interval of time at the end of which the configured failover action will be executed if the call is not answered (no secretary picks up the call in time).	Numeric
No answer	The call is considered unanswered after the specified time has passed.	Forward to the boss / Hang up / Custom selection / Ask for selection / External number / Extension / Group / Queue / Checktime / IVR / Voicemail / MeetMe room
Busy	All secretary extensions are busy. The extension is considered busy if the configured Busy Level has been reached, or if the terminal sends a 486 Busy Here SIP Response.	Forward to the boss / Hang up / Custom selection / Ask for selection / External number / Extension / Group / Queue / Checktime / IVR / Voicemail / MeetMe room
Not available	All secretary extensions are unavailable. The extension is considered busy if the terminal is not registered or not reachable at an IP level, or if the terminal sends a 480 Temporarily Unavailable SIP Response.	Forward to the boss / Hang up / Custom selection / Ask for selection / External number / Extension / Group / Queue / Checktime / IVR / Voicemail / MeetMe room

Interoperability

When enabling/disabling the service from the phone, it can be useful to have a key (with Busy Lamp Field) that lets you verify the state of the service.

For monitoring, KalliopePBX sends SIP NOTIFY messages to communicate changes of state. The phone must send a SIP SUBSCRIBE message to request this information.

This operation is normally executed by configuring a BLF-type function key.

The object that needs to be monitored is bs<boss_extension> to monitor the state of the service on a group level and bs<boss_extension>*<secretary_extension> to monitor the state of the service for a secretary in a specific group. The state of the service on a group level will be active when is is enabled for at least one secretary.

Other than monitoring the state the service, it is also possible to invert it by pressing the corresponding function key.

The key to control the service on a group level is usually enabled on the boss' phone, while the keys for specific extension are enabled on the secretary's phone.

The keys can be configured for all members of the group to accommodate specific requirements (e.g. the boss might wish to set the state of specific secretaries or let the secretaries set the state on a group level).

Examples

On SNOM

- Through the web GUI, you can configure function keys with:

```
Account: select the account from the drop-down (if only one account is configured on the␣
↪phone, it will be the first in the list)
Type: BLF
value: bs<boss_extension> / bs<boss_extension>*<secretary_extension>
```

- Or you can directly edit the configuration file or the template:

```
<fkey idx="%%id%%" context="%%line_id%%" label="" perm="">blf sip:bs<boss_extension>@%
↪%KPBX_IP_ADDRESS%%;user=phone</fkey>
```

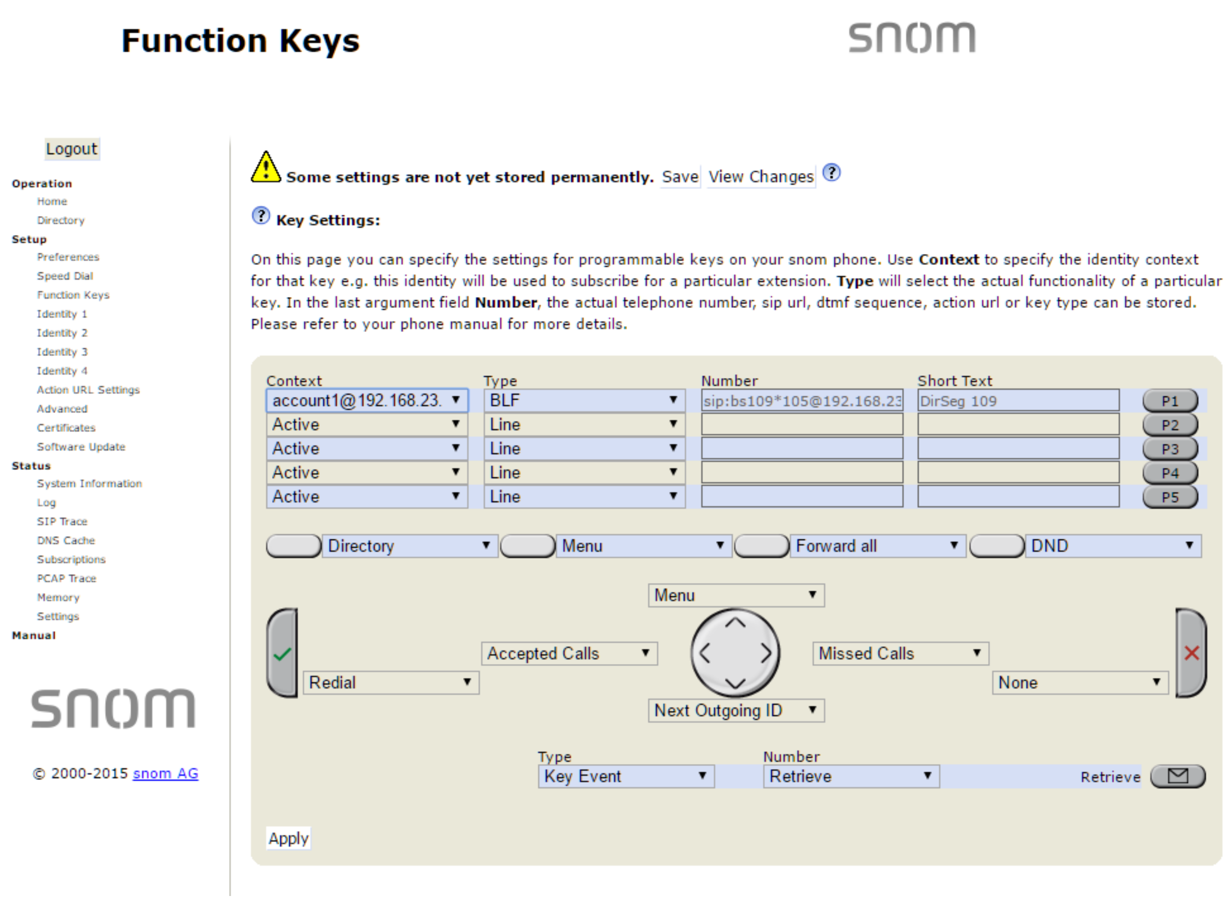
oppure

```
<fkey idx="%%id%%" context="%%line_id%%" label="" perm="">blf sip:bs<boss_extension>*
↪<interno_segretaria>@%%KPBX_IP_ADDRESS%%;user=phone</fkey>
```

where %%id%% is the ID of the key to configure and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
<fkey idx="0" context="1" label="DirSeg 109" perm="">blf sip:bs109*105@192.168.23.112</
↪fkey>
```



On YEALINK

- Through the web GUI, you can configure DSS keys with:

```
Type BLF
Value: bs<boss_extension> / bs<boss_extension>*<interno_segretaria>
Line: The line associated with the account (Line 1 if the account is the only one on the phone)
```

- Or you can directly edit the configuration file or the template:

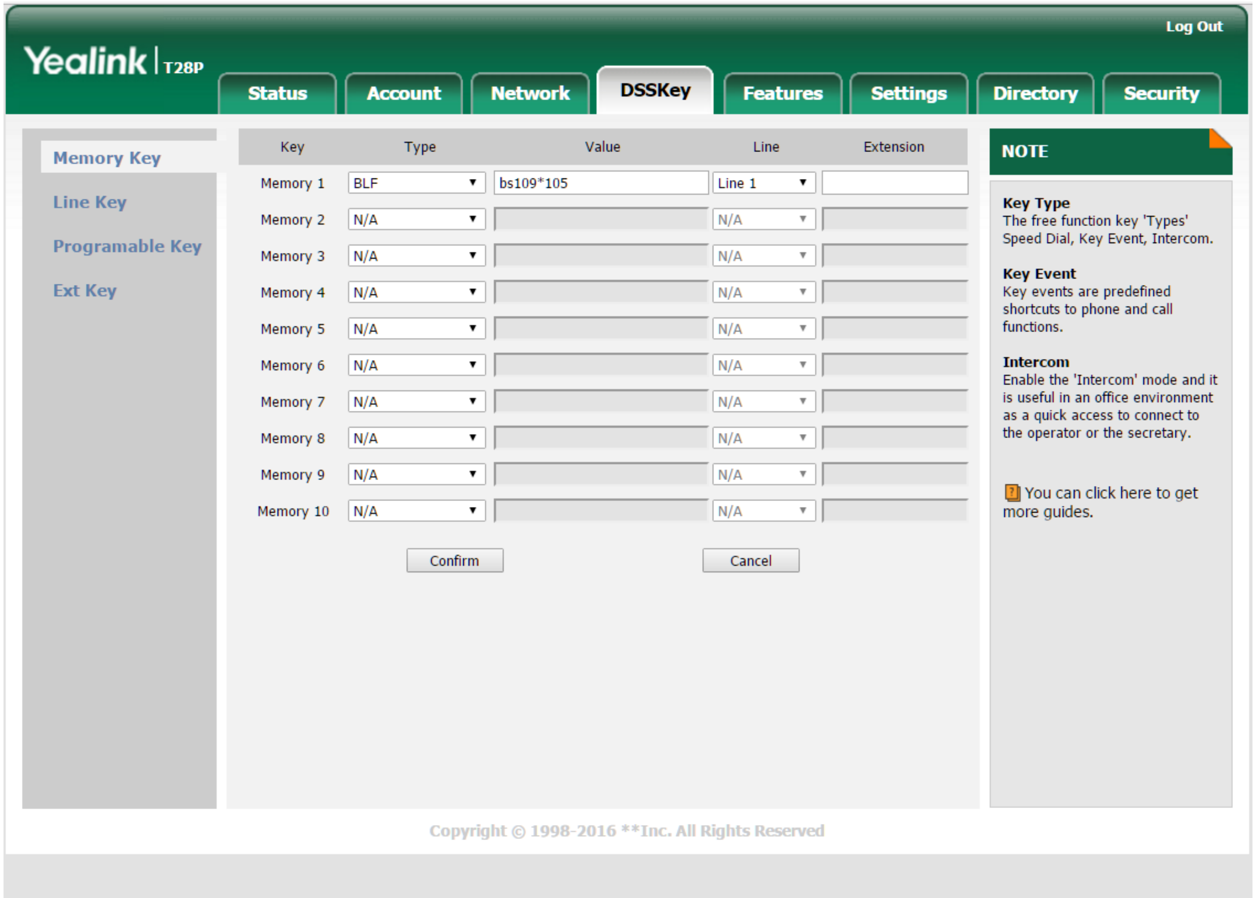
```
memorykey.%%id%%.line=%%line_id%%>
memorykey.%%id%%.value=bs<boss_extension> / bs<boss_extension>*<interno_segretaria>
memorykey.%%id%%.type=16
```

where %%id%% is the ID of the key to configure

and %%line_id%% is the ID of the corresponding account (1 if the account is the only one on the phone).

Example:

```
memorykey.1.line = 1
memorykey.1.value = bs109*105
memorykey.1.type = 16
memorykey.1.pickup_value = %NULL%
```



Paging service

Note: Introduced with firmware version 4.3.1

Description

The paging service lets a user send a live or prerecorded audio message to multiple destinations simultaneously, with the option of having devices answer automatically when they receive the message. This service is used to make informative or emergency announcements.

KalliopePBX lets you define an arbitrary number of "paging groups". Each one is independent from the others and is fully configurable when it comes to permissions, choice of destinations, mode of operation, and messages.

An extension can make a call to a code comprised of a paging service prefix (set when enabling the service in the numbering plan, by default *53) followed by the paging group number (set in the configuration page for the group). If authentication is necessary, KalliopePBX will play the "Password" audio file and the user will need to input a PIN (the one set in the group configuration or their personal one) followed by #. If authentication is successful the paging group will be activated according to its configuration, detailed in the following section.

Configuration

Paging groups can be created and edited in the PBX -> Paging groups page.

The following table describes the configuration parameters of each group are:

Parameter	Description	Value
Name	Identifier assigned to the Paging group	Alphanumeric
Number	Selection of Paging Service	Numeric
Caller ID	Display Name for calls to destination terminals	Alphanumeric
Modality	Mode of operation of the Paging group	Live/Unattended
Audio file	Audio file containing a prerecorded message that is played at the destination terminals of the group. It is used differently in the case of Live or Unattended mode	
Repetitions number	Number of times the prerecorded message must be played before the PBX automatically ends the call (configurable only in "Unattended" mode)	Numeric (0 indicates that the message is to be played in a continuous loop)
Automatic response method	Header to be added to calls made to destination terminals in order to specify the automatic response request	Manual configuration/Call-Info/Header Alert-Info

Account

Parameter	Description	Value
Extension	Paging target terminals	
Account	SIP account associated with the extension	

Controllo di accesso

Parameter	Description	Value
Extension	Caller extension	"Any extension"/Extension
PIN type	Authentication mode required	None/Custom/ Authentication mode required.
Pin value		Alphanumeric

Modality

The operating mode of the paging groups: can be set as either "Live" or "Unattended". In Live mode the extension making the call to the paging group will remain on call as the sole speaker; at the end of the call, they will hang up, and the calls to all destinations will themselves end. In Unattended mode the PBX hangs up the call made from the extension to the paging group and, after a few moments, proceeds to contact all destinations and play the message set under the "Audio file" item.

The audio file will be used differently depending on whether the mode is set to "Live" or "Unattended":

- **Live**: the message is played before the caller begins communication with the group. This can be used to mark the beginning of an announcement.
- **Unattended**: the message is played to all destinations in the paging group. Depending on the following setting (Number of repetitions), the message can be played once or multiple times.

In Unattended mode a second call to the group number will stop message playback regardless of the number of repetitions set.

Note: The second call does not need to be made from the extension that activated the service as long as it is enabled to use the service according to the rules described below.

Access control specifies the permissions necessary to use the service. You can specify one or more access rules of the form, specifying which extensions can use the service (one, more than one, or any) and if an activation PIN is required.

An example of **ACL** is:

Extension	Type of PIN	Value of PIN
101 (Extension 101)	None	
102 (Extension 102)	Extension service PIN	
Any	Custom	123123

In this example, extension 101 can activate the service (or deactivate it if it is set to “Unattended”) without needing to input any PIN. Extension 102 can use the service after dialing their service PIN, while all other extensions need to dial 123123.

Interoperability

The configuration of the automatic answer is tied to the terminals, as different devices might require different ways of signaling requests. The service has been tested with SNOM (firmware 8.7.5.13) and Yealink (firmware v80) terminals and requires the phone to be configured to accept requests. Other models are currently being tested and new headers will be added in the future if necessary.

Specifically:

- **If the paging group is configured not to send additional headers (and therefore uses terminals dedicated to making announcements), automatic answer must be enabled on all phones. This can be done either through provisioning or through the web GUI of each phone. The relevant settings for each tested phone are:**
  - SNOM: “Auto Answer” setting in the “Account -> SIP” page; (http://wiki.snom.com/wiki/index.php/Settings/user_auto_connect)
  - Yealink: “Auto Answer” setting in the “Account -> Basic” page.
- **For non-dedicated terminals, it is necessary to send a header to request automatic answer. The terminals must be configured to accept the directions given by the header, otherwise it will be ignored and the user will need to answer manually.**
  - SNOM: edit the “Intercom Policy” setting in the “Advanced -> Behavior” page (disabled by default). See http://wiki.snom.com/Settings/answer_after_policy for the configuration. Both headers were tested with positive outcome.

Note: upon automatic answer, the phone will, by default, emit an announcement tone; this can be disabled by editing the “Auto Connect Indication” setting (http://wiki.snom.com/Settings/auto_connect_indication).

- Yealink: automatic answer headers (and announcement tones) can be enabled or disabled in the “Features -> Intercom” page.
- Grandstream: automatic answer headers can be enabled in the “Accounts -> AccountX -> CallSettings -> Allow Auto Answer by Call-Info” page. Header Call-Info requests are required.

Other services

High Availability

Description

High availability aims to increase the performance of the VoIP system by bringing together two physical nodes or two virtual machines that work together in active/passive mode. Only one of the two PBXs is active in a specific moment, while the other is on standby and is ready to take over call service when the passive node fails. There is no retention of calls in progress during transferring resources from one node to the other By design, the failure that a high-availability system intends to handle is a complete shutdown of the active node's functionality at that time:

- natively there is no case of monitoring service if it is not actiive, as instead happens in other redundancy mechanisms where the node is considered degraded, even if one of the services is not active
- unpleasant episodes can happen if the two nodes that are part of the cluster, lose their sight of each other, because each of them is convinced to be the only one and therefore activates, causing a "split brain" situation

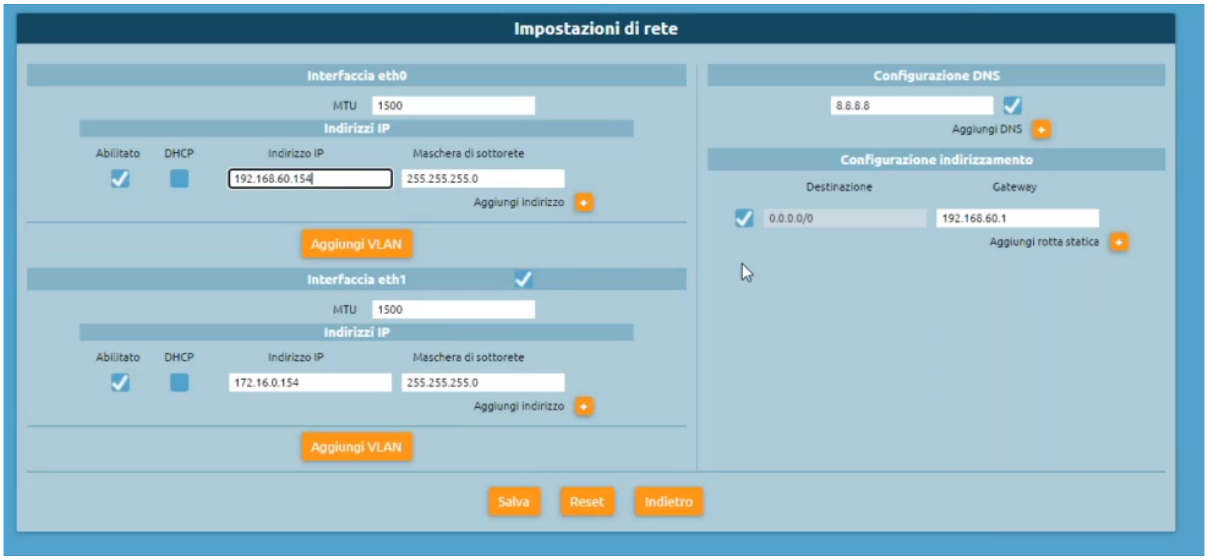
The High availability panel is accesible only to the admin or in case of Multitenant machine, to the PBX Admin.

Configuration

We have to go to Settings > High Availability Status This takes us to the **High Availability Configuration** page.

You can change system's status as "Enabled as Coordinator" or "Enabled". For example, we can choose "Enabled as Coordinator" for a node to keep as coordinator and select (from another browser tab) another node as "Enabled", which will act as a secondary and receive the configuration from the other node.

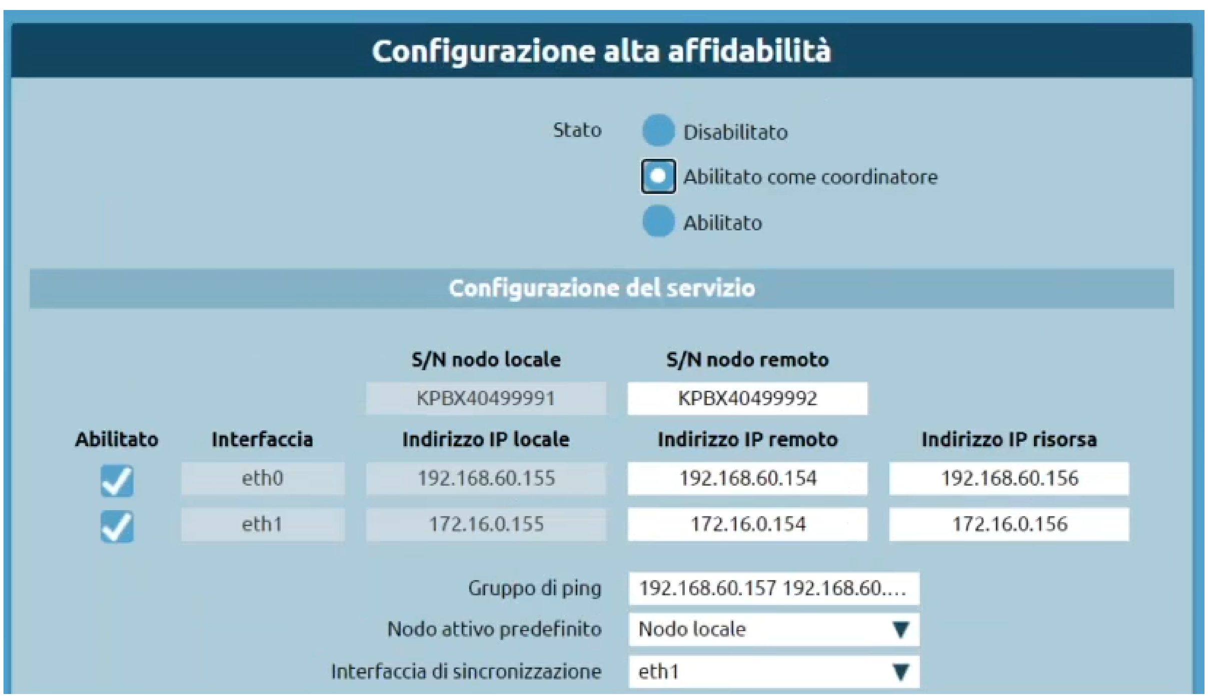
Network settings



For a complete configuration, you need to go to Settings > Network settings and check that two network interfaces are configured for high availability, checking their own IP address. The two interfaces are connected to the same virtual switch that has no public interfacing. It has a single DNS configured and a route.

High Availability Configuration

Service Configuration (Abilitato come coordinatore)



- **S/N remote node**: it's the serial number, the first communication between the two exchanges (coordinator and secondary node) works with an interactive SSH session, in which you have to insert user and password for autentication. The password is calculated from the serial number of the machine. So the coordinator node needs to know the serial number of the other node to get the password to connect with and provide the necessary commands

- **Enabled**: you can choose which service to configure on one or both of them
- **Interface**: it is possible to see all the network interfaces present on the machine
- **Local IP Address**: you can see which is the local IP address
- **Remote IP Address**: you can view the remote IP address.
- **Resource IP Address**: it's the IP address that will be acquired and shared on a given interface
- **Ping group**: you can insert the IP addresses separated by spaces and they will be the nodes that each of the two will ping periodically to check which one sees more of them
- **Synchronization interface**: you can insert a chosen interface that will be used for synchronization of data and logs between the two nodes (e.g. eth1)
- **Default active node**: you can choose which of the two nodes should acquire the default resource in case of indecision.

Advanced settings

- **Timeout keepalive - warning threshold (sec.)**: this is a threshold after which a warning is generated (at the moment only in the logs)
- **Timeout keepalive - out of service (sec.)**: this value defines after how long the node is not communicating with the other node, it must be assumed that it is inactive, and therefore if the other node may acquire resources
- **Initialization timeout (sec.)**: this value guarantees a specific number of seconds before deciding to acquire the resources (if there is already an active node). The higher the value, the more time the machine will take at startup to become operational because both nodes wait for this time

By pressing "Save and start service" you can save the setting to the database and the high availability can start.

High Availability Status

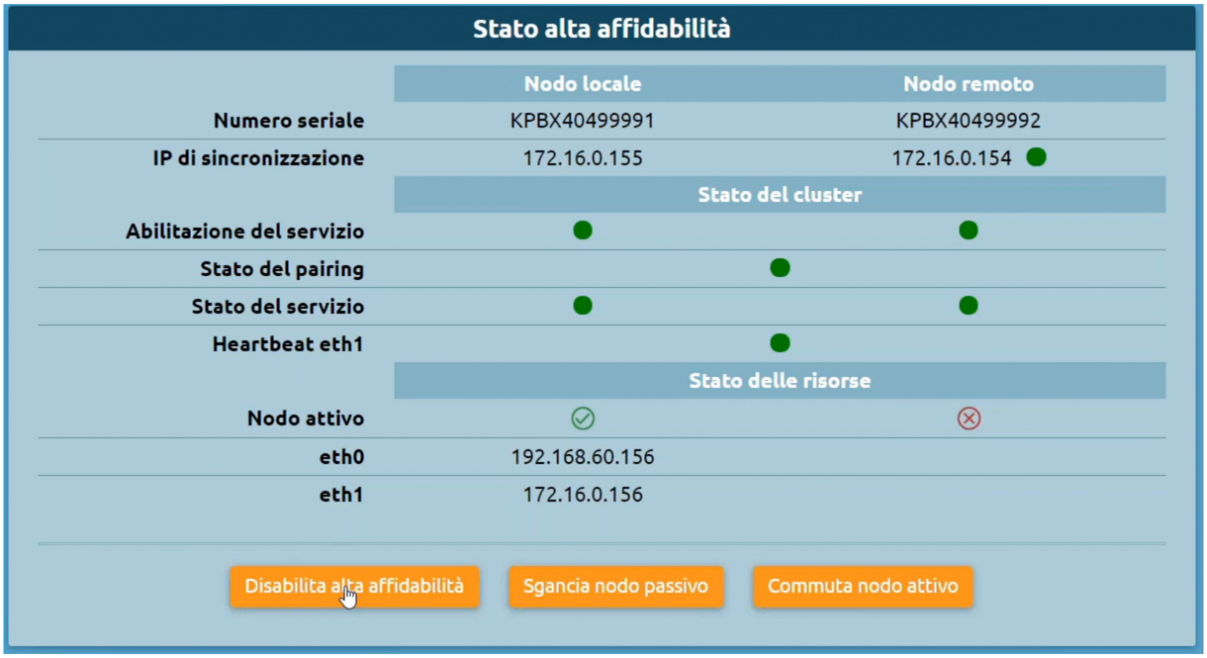
After saving the configuration, we are on the High Availability Status page where we can view:

- The nodes with their respective serial numbers and synchronization IPs
- The cluster status with the service enable, the pairing status, the service status, the heartbeat
- The status of the resources with the active node
- The cluster status - "pair" option: we will hook the secondary node and we will be able to see the status panel in passive mode:

High availability passive node status

- **Unhook node**: disable high availability, thus leaving the cluster degraded and the resource running on the active node
- **Switch active node**: force resource acquisition

High availability active node status



- **Disable High Reliability**: turns it off on both nodes
- **Unhook passive node**
- **Switch active node**: force resource switching on the other node

Scheduled Tasks

Scheduled task management

With this service (introduced in version 4.5.8) you can schedule tasks to be automatically carried out by KalliopePBX following a user-defined planning policy.

The types of tasks that can be scheduled are:

- Send Call Center CDR
- Send CDR (available with firmware version 4.7.0 of later)

Defining a new scheduled task

You can set scheduled tasks on the Operating menu -> Monitoring -> Scheduled tasks page.

To add a new scheduled task click on "Schedule new task" in the top left after selecting the type of task from the drop-down menu.

How to create a new scheduled "send Call Center CDR" task

Note: This type of scheduled task requires a Call Center license to be present on the PBX.

Select "Send Call Center CDR" from the drop-down menu and click on "New scheduled task". A form, pictured below, will appear.

The following table lists the parameters that need to be configured:

General settings

Parameter	Description	Value
Name	The name assigned to the scheduled task. This name will appear in the subject line and the body of the emails sent upon completion	Alphanumeric
Enabled	Enable or disable the scheduled task. Disabled tasks will only be carried out manually	Yes / No
Type of schedule	Select how often the task will be carried out	Daily / Weekly / Monthly
Scheduled time	Select the time at which the task will be carried out and and the result sent via email to the configured addresses.	Hours and minutes
Day of the week	Only available if the schedule is set to "Weekly". Lets you set the day of the week on which the task will be carried out (at the time selected above) and the result sent via email to the configured addresses.	Day of the week
Day of the month	Only available if the schedule is set to "Monthly". Lets you set the day of the month on which the task will be carried out (at the time selected above) and the result sent via email to the configured addresses.	Date

Settings for sending the Call Center CDR

Parameter	Description	Value
Time span	Select the time span based on which you wish to export the Call Center CDR.	Current day / Previous day / Current week / Previous week / Current month / Previous month
File format	Select the format of the exported Call Center CDR.	XLSX / XLSX (detailed) / CSV / CSV (detailed) / JSON / XML
Export operator events	If selected, this will add all operator events to the exported Call Center CDR. This setting can only be enabled if a "detailed" file type has been selected.	Yes / No
Compress if file size exceeds this value in MB (0 = always compress)	Indicate a maximum acceptable file size (in MB); if the file exceeds this, it will be compressed to zip format before being sent as an attachment. If this value is set to 0, the report will always be compressed regardless of its size.	Numeric
Notify all supervisors	Select whether or not the exported Call Center CDR will be sent via email to all users with a supervisor role who have a configured email address.	Yes / No
Recipients	In conjunction with or as an alternative to notifying the supervisors, you can specify an arbitrary number of recipients to which the exported Call Center CDR will be sent.	Alphanumeric

Note: Either one or more recipients must be set or notifications to supervisors must be enabled, otherwise the form will display an error message and the scheduled task settings will not be saved.

How to create a new scheduled "send CDR" task

Select "Send CDR" from the drop-down menu and click on "New scheduled task". A form, pictured below, will appear.

The following table lists the parameters that need to be configured:

General settings

Settings for sending the CDR

Parameter	Description	Value
Time span	Select the time span based on which you wish to export the CDR.	Current day / Previous day / Current week / Previous week /Current month / Previous month
File format	Select the format of the exported CDR.	XLSX / XLSX (detailed) / CSV / CSV (detailed) / JSON / XML
Compress if file size exceeds this value in MB (0 = always compress)	Indicate a maximum acceptable file size (in MB); if the file exceeds this, it will be compressed to zip format before being sent as an attachment. If this value is set to 0, the report will always be compressed regardless of its size.	Numeric
Recipients	In conjunction with or as an alternative to notifying the supervisors, you can specify an arbitrary number of recipients to which the exported Call Center CDR will be sent.	Alphanumeric

Carrying out tasks manually

Among the available actions for scheduled tasks there is the option to carry them out on demand without waiting for the set schedule. To start the task in the background you can click on the play icon.

At the end of the request, a message will notify you that task has started.

Audit Log

Description

The Audit Log contains all configuration changes made, tagged with the user who made the change.

Changes are not immediately irreversible, but it is possible to view what has been changed and, if necessary, go to restore it.

Knowing the user who made the changes is very important; the admin user is not the only one who can make configuration changes, but you can create customizable roles and assign users to these specific roles. In this way, you delegate part of the PBX configuration to customer staff. Then all changes made by the client will be marked with the user name assigned to them. Click on the link below for a more in-depth look at the users and roles *users and roles*

To reach the service, follow the menu path "Logs > Audit Log."

The Audit Log contains change logs sorted by month, and it is possible to export the log in various formats: XLSX, CSV, JSON, XML There is a "Choose Columns" filter to target the search to specific sections present.

In fact, the Audit Log allows us to display:

- **Transaction Id**
- **Day of the month**: a specific date can be selected.
- **Timestamp**
- **Username**: name of the user who made the change.
- **IP address**
- **Action**
- **Entity type**
- **Description**

Auto-Provisioning

Description

The Auto Provisioning service generates the configuration file necessary for the correct operation of a device and transfers them to your phones. This file also contains information on the account and extension linked to the phone.

Configuration

This section collects all the configurations needed to perform auto-provisioning of a telephone device. You can also consult the list built-in devices for auto-provisioning:

Elenco dispositivi built-in per l'auto-provisioning

I dispositivi riportati di seguito sono pre-definiti su KalliopePBX. Se il modello richiesto non è presente in questa lista, ma il meccanismo di generazione è analogo ad un modello esistente, è possibile creare un file di provisioning personalizzando il processo di generazione del nome. Per ulteriori informazioni sulla generazione di un file di provisioning con nome arbitrario si faccia riferimento all'apposito paragrafo.

AudioCodes

AudioCodes 405

AudioCodes 405HD

AudioCodes 420HD

AudioCodes 430HD

AudioCodes 440HD

AudioCodes 445HD

AudioCodes 450HD

Avaya

Avaya 1120E

Avaya 1140E

Avaya 1220

Avaya 1230

Cisco Unified IP Phone

Cisco Unified IP Phone 794x

Cisco Unified IP Phone 796x

Cisco Unified IP Phone797x

Escene

Escene ES220-N

Gigaset

Gigaset DE310 IP PRO

Gigaset DE410 IP PRO

Gigaset DE700 IP PRO

Gigaset DE900 IP PRO

Gigaset Maxwell 2

Gigaset Maxwell 3

Gigaset Maxwell 10

Gigaset Maxwell Basic

Gigaset N720 IP PRO

Htek

Htek UC924

Linksys

Linksys SPA921

Linksys SPA922

Linksys SPA941

Linksys SPA942

Snom

Snom 300

Snom 320

Snom 360

Snom 370

Snom 710

Snom 715

Snom 720

Snom 725

Snom 760

Snom 820

Snom 821

Snom 870

Snom D120

Snom D305

Snom D315

Snom D345

Snom D375

Snom D710

Snom D712

Snom D715

Snom D717

Snom D725

Snom D765

Snom D785

Yealink

Yealink SIP-T19P

Yealink SIP-T19P E2

Yealink SIP-T20P

Yealink SIP-T21P

Yealink SIP-T21P E2

Yealink SIP-T22P

Yealink SIP-T23G

Yealink SIP-T23P

Yealink SIP-T26P

Yealink SIP-T27G

Yealink SIP-T27P

Yealink SIP-T28P

Yealink SIP-T29G

Yealink SIP-T32G

Yealink SIP-T38G

Yealink SIP-T40G

Yealink SIP-T40P

Yealink SIP-T41P

Yealink SIP-T41S

Yealink SIP-T42G

Yealink SIP-T42S

Yealink SIP-T46G

Yealink SIP-T48G

Yealink SIP-T48S

Yealink SIP-T52S

Yealink SIP-T54S

Yealink SIP-T56A

Yealink SIP-T58A

Yealink SIP-T58V

Yealink SIP-VP-T49G

Yealink VP530

Patton

Patton SN4522

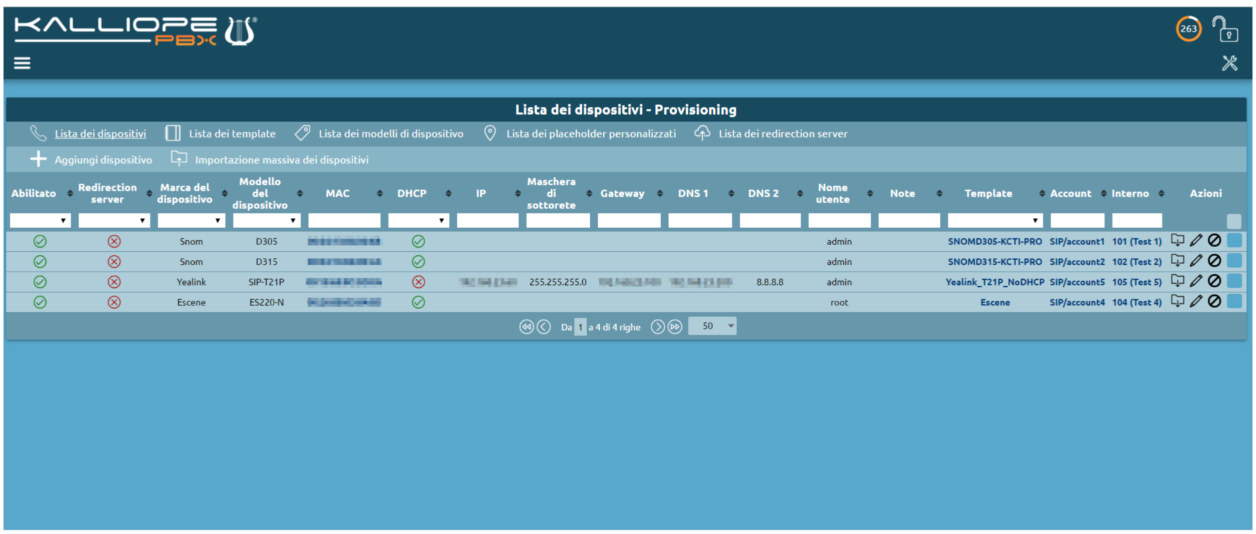
Patton SN4524

Patton SN4526

Patton SN4528

Device list

This page contains the list of all devices for which provisioning file generation has been configured.



The following table lists the configurable parameters for each device.

Parameter	Description	Value
Enabled	Allows you to disable the generation of the provisioning file associated with the device	Yes / No

Device model

Parameter	Description	Value
Brand	List of manufacturers for which at least one device model has been defined	Brand
Model	List of devices associated with the selected manufacturer	Model
Template	List of templates associated with the selected template	Template

Redirection Server

Parameter	Description	Value
Redirection Server for provisioning	List of redirection servers defined for the selected manufacturer	Redirection Server
Provisioned on the redirection server	Read-only field indicates whether the provisiong on the redirection server was successful	Yes / No

Device configuration

Parameter	Description	Value
MAC Address	MAC address of the device (formats that are accepted AABBCCDDEEFF, AA:BB:CC:DD:EE:FF, AA-BB-CC-DD-EE-FF)	MAC Address
Notes	Free field containing annotations about the device	String
Enable DHCP	Set the placeholder value %%IPADDRMODE%% to on/off for this device. The placeholder may be used in the template to generate the configuration file with the with the required network settings.	Yes / No
IP Address	Set the placeholder value %%IPADDR%% for this device. The placeholder may be used in the template to generate the configuration file with the required network settings.	IP Address
Subnetwork mask	Set the placeholder value %%IPNETMASK%% for this device. The placeholder may be used in the template to generate the configuration file with the required network settings.	Subnet Mask
Gateway	Set the placeholder value %%IPGATEWAY%% for this device. The placeholder may be used in the template to generate the configuration file with the required network settings.	IP Address
DNS1	Set the placeholder value %%IPDNS1%% for this device. The placeholder may be used in the template to generate the configuration file with the required network settings.	IP Address
DNS2	Set the placeholder value %%IPDNS2%% for this device. he placeholder may be used in the template to generate the configuration file with the required network settings.	IP Address
Username	Set the placeholder value %%PHONEUSERNAME%% for this device. The placeholder may be used in the template to set the phone login credentials. This same value will be used by KalliopePBX to drive the phone when a KalliopeCTI PRO application is associated with the device.	String
Password	Set the placeholder value %%PHONEPASSWORD%% for this device. The placeholder may be used in the template to set the phone login credentials. This same value will be used by KalliopePBX to drive the phone when a KalliopeCTI PRO application is associated with the device.	String

Remote control

If empty, the PBX will use the IP address from which the account associated with this device is registered at the SIP level

- IP Address
- - Port
  - Same as the previous field, but related to the port on which the phone web interface is visible by the PBX.

– Integer (range 1-65535)

Device user

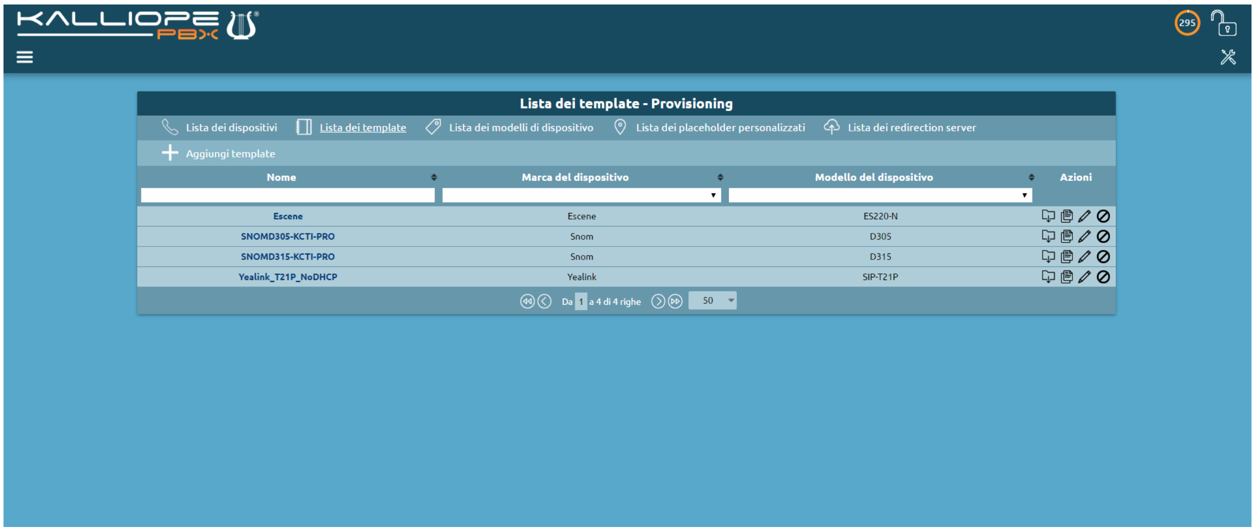
If the account is not associated the configuration file is not generated and the device is entered as Disabled.

- Account

Template list

This page shows the list of templates on the KalliopePBX.

It is necessary to select a template for each phone model for which you wish to generate a provisioning file.



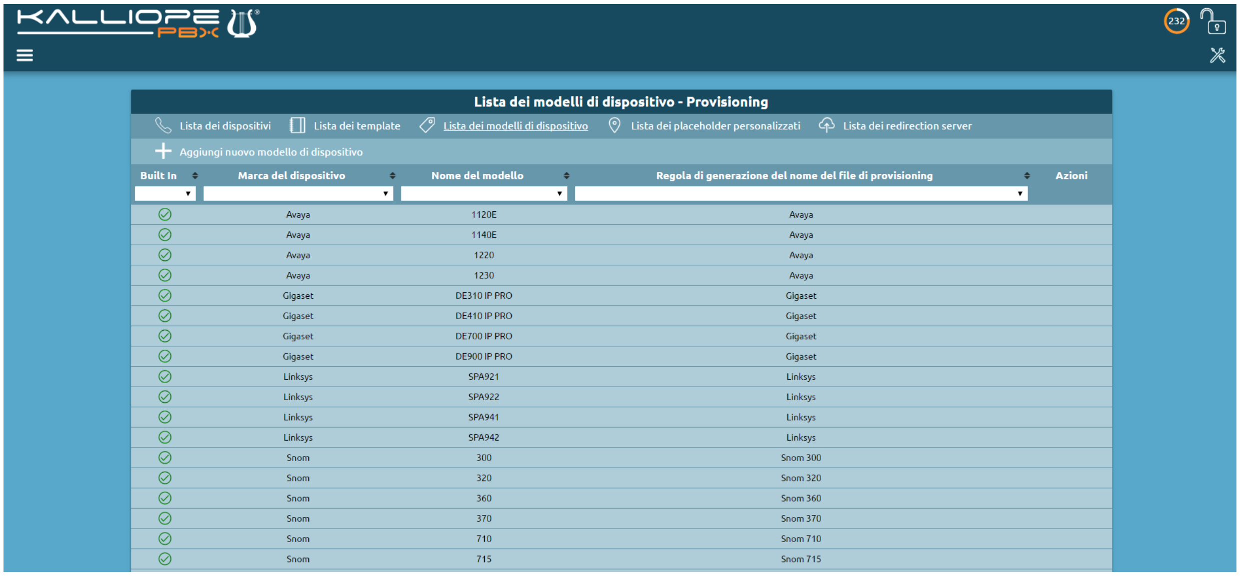
Parameter	Description	Value
Name	Template name	String
Device brand	List of manufacturers for which at least one device model has been defined	Brand
Device model	List of devices associated with the selected manufacturer	Model

Template content

Parameter	Description	Value
Template	This free field should contain the template to be used for generating the provisioning file	Text

List of device models

This page contains the list of all models defined on KalliopePBX. Some models are distributed with the KalliopePBX firmware, while more can be added in order to generate configuration files for models that are not explicitly supported.



You can also specify rules to create provisioning files with arbitrary names. The filename can be composed of a prefix, a MAC address (in several formats) and a suffix.

The following table lists the configurable parameters for each model.

Parameter	Description	Value
Name	Device model name	String

Device brand

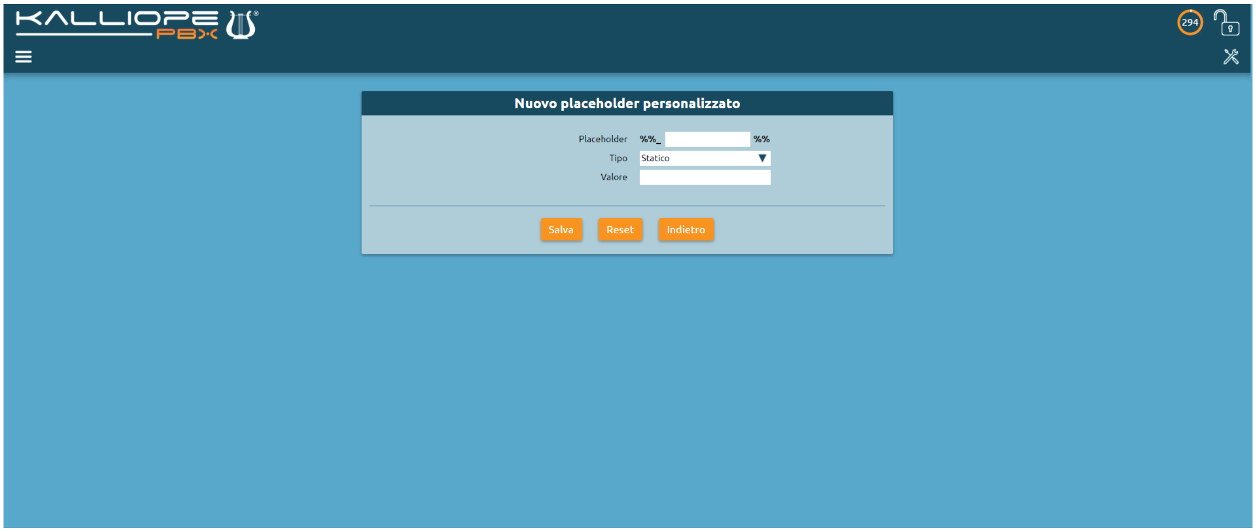
Parameter	Description	Value
Choose Brand	Allows you to select an existing brand or create a new one by selecting New Brand	Brand
Brand name	In the case of New Brand contains the name to be associated with it	String
Name	Device model name	String

Provisioning file name generation rule

Parameter	Description	Value
Choose rule	Allows you to select an existing rule or create a new one by selecting the New Rule item	Brand
Rule name	In the case of New Rule contains the name to be associated	String
Prefix	Prefix to add to file name	String
MAC address format	Allows you to select the format of the MAC address to be included in the filename from a list	MAC Address Format
Suffix	Suffix to be added to file name	String

Custom placeholder list

This page shows the list of all placeholders specified by the user along with the default ones. Custom placeholders are formatted as %%_PLACEHOLDER%% and can be used inside a template.



You can create two types of placeholder:

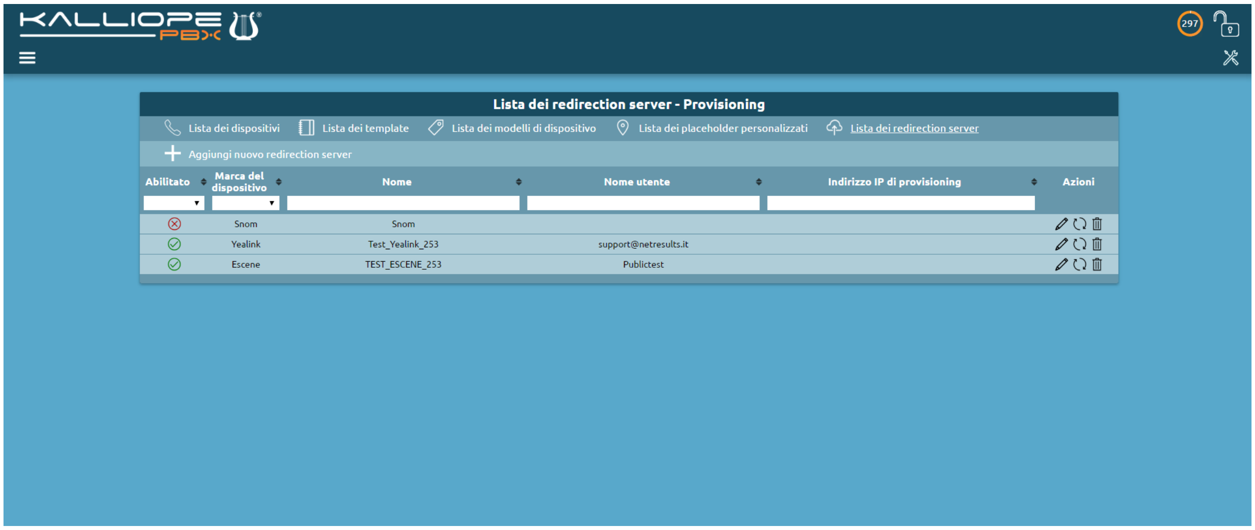
- Static: used in order not to have to edit all templates which contain a specific value.
- Dynamic: used to dynamically update certain values associated to the KalliopePBX and not the specific user. For the time being, the only available dynamic placeholders are the ones relating to the IP addresses associated to the different network interfaces/VLANs.

Parameter	Description	Value
Placeholder	Placeholder identifier. The placeholder to be used is %%_PLACEHOLDER%%	String
Type	Allows you to define the type of placeholder	Static/dynamic
Value	In the case of static placeholder the value to be replaced in generation is entered, in the case of dynamic placeholder the attribute of the KPBX to be used for replacement.	String / KPBX Attribute

List of redirection servers

This panel contains the list of all user-configured redirection servers on the KPBX. Currently, integration with redirection servers from the following manufacturers is supported:

- SNOM (https://sraps.snom.com/)
- Yealink (https://ymcs.yealink.com/)



Note: Due to limitations of the APIs provided by Yealink/Escene, the configuration procedure is different when using the SNOM redirection server. Specifically, for Yealink/Escene, it is necessary to preemptively specify a server by accessing the RPS service web GUI. Its name will be referred to during the redirection server configuration phase.

Parameter	Description	Value
Device brand	Brand of the device for which the redirection server is being defined	Snom / Yealink / Escene

Credentials

Parameter	Description	Value
Username	User for authentication on the manufacturer's RPS server	String
Password	Password for authentication on the manufacturer's RPS server	String

Settings

Parameter	Description	Value
Enabled	Allows you to disable the redirection server without losing its configuration	Yes / No
Name	Name of the redirection server to be created (if SNOM) or to be used (in the case of Yealink / Escene)	String
Provisioning address	In the case SNOM URL to which the redirection is made. (example: https://192.168.0.100/provisioning/)	String

Template

The configuration file for a specific device is generated from a template for the brand and model of the phone.

The template format depends on the brand and model of the phone as well as the version of its firmware.

When defining a template it is possible to use placeholders that Kalliope will automatically replace when generating the file.

These placeholders include:

- KalliopePBX attributes (e.g. SIP UDP/TCP port of the PBX);
- attributes of the extension/account linked to the phone (e.g. SIP credentials, first name, last name, etc.);
- phone attributes (e.g. network parameters, access credentials, etc.).

Once the template has been defined, you must specify the MAC address of the device and the account/extensions you wish to link it to.

Once a file has been generated, it must be transferred to the phone. KalliopePBX provides the following file transfer protocols:

- **TFTP**: files are available directly in the root of the TFTP server for single-tenant KalliopePBX. For multitenant KalliopePBX, you must add the Tenant UUID to the path (e.g. <tenant_uuid>/snom370-0004167898B1.htm).
- **HTTP / HTTPS**: the files are published on http(s)://<ip_address>/provisioning/ for single-tenant KalliopePBX. For multitenant KalliopePBX, you must add the Tenant UUID to the path (e.g. http(s)://<ip_address>/provisioning/<tenant_uuid>/).

All generated files are also visible in the File Manager.

To tell the phone which protocol should be used to download the configuration file in addition to the IP address (and possible path) of the provisioning server, there are several modes whose configuration and execution order depend on the phone model used. The commonly available methods are as follows:

- **SIP PnP**: the phone at startup sends a SIP SUBSCRIBE message to a multicast address. If on the KalliopePBX the SIP PnP service is enabled, the PBX responds with a SIP NOTIFY containing the TFTP server's IP address to be used. This mode cannot be used in the case of multitenant KalliopePBX.
- **Redirection Server****: the phone at startup tries to contact the manufacturer's Redirection Server. If the phone's MAC Address is entered, the phone is redirected to the indicated server to download the configuration file. In this mode, you can use any available protocols (depending on the parameter configured on the redirection server).
- **DHCP OPTION 66**: in the case when assigning the IP address, the DHCP Server also communicates to the phone the DHCP Option 66 containing the URL to be contacted (including the protocol to be used), the phone will use this information to make the download of the configuration file.
- **Manual**: you can also start the configuration manually from the phone or the WEB GUI by entering the protocol to be used and the IP address (as well as the path, if necessary).

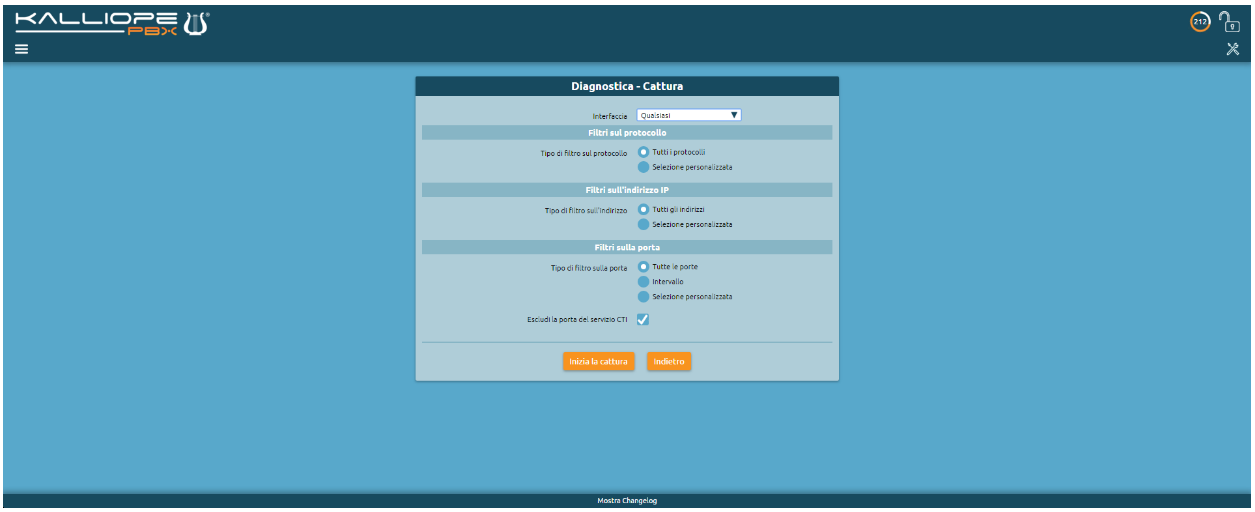
Packet Capture

he capture tool lets you acquire traces of packets to or from the KalliopePBX network interface, applying if necessary a selection filter based on:

- Type of protocol (all or a combination of ICMP/UDP/TCP)
- IP address (source or destination)
- Source or destination port

Once you have set the desired filters, you simply need to click on Begin capture. You can end the procedure by clicking on Finish capture; otherwise the capture will end once the file size has reached 10 MB.

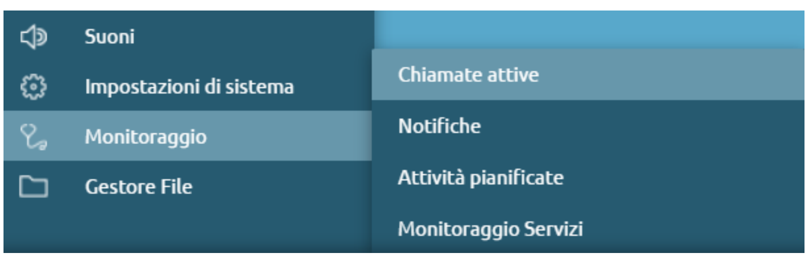
At the end of the capture, you can download the pcap file to your computer by clicking on Download or delete it by clicking Delete. Starting a new capture will overwrite the previous one. Rebooting the PBX will erase the pcap file.



Active Calls

Description

The "active calls" submenu of the Monitoring service can be reached by clicking on "Monitoring > Active Calls," as shown in the image to the right.



The page shows the panel where the list of active calls is available, filtered by:

Source	Destination
Linked ID	
Channel	Channel
Status	Status
Caller ID	Called ID
Caller name	Called name
Called number	
Creation Time	Creation Time
Uptime	Uptime
Connection Time	Connection Time
Uptime since connection	Uptime since connection

PBX Events

Description

The PBX Events page displays all notifications about events that occur when the central unit is accessed.

To reach the service, follow the path “Logs > PBX Events”.

- ID
- Event type
- Day of the month
- Timestamp
- Severity
- Parameters

You can also export data in the following formats:

- XLSX
- CSV
- JSON
- XML

Remote Filesystem Management

Description

To reach the service, follow the path "System Settings > Remote Filesystem Management".

The following page contains the list of remote filesystem on any remote computer.

Configuration

To add a filesystem, press on "Add new remote filesystem" The configuration of the new filesystem allows you to enter:

- Protocol (CIFS/NFS)
- Name
- Server address
- Sharing
- Username
- Password

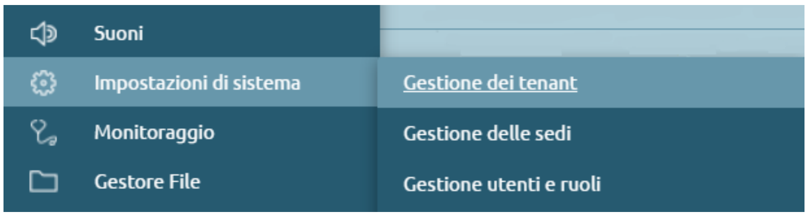
Tenant Management

Description

On the tenants management page, you can view the list of tenants and tenant groups.

Configuration

To reach the "Tenants Management" service, click on "System Settings > Tenants Management".



Unlocking the padlock in the upper right to enter edit mode, you can create a new tenant.

Tenants list

To create a new tenant, it is necessary to click on "Create New Tenant." There are the following fields to fill in and/or select:

- **Operating mode**: can be full, limited with outgoing call blocking (except calls to whitelist numbers) and disabled (operation of that particular tenant is inhibited)
- **Name**
- **Domain**: domain name that works as the username component after the "@" (e.g., admin@domain)
- **Password**: of the tenant admin that can be initialized at tenant creation
- **Custom logo url**: a tenant user that enters the central web interface can see a custom logo in the top left corner
- **Admin email**: email of the referrer for sending the tenant creation notification
- **Group (default group / new group)**: is useful for grouping multiple administratively distinct tenants (each with its admin) into a confederation

The configuration of tenant groups is explained in the next section.

- **SIP account prefix**: ensures uniqueness of SIP accounts, i.e., all SIP accounts of a given tenant will have the same prefix

To prevent a collision (two tenant admins creating the same SIP account), each tenant is assigned a prefix (a 6-character alphanumeric string) so that all SIP accounts of a particular tenant share the same prefix. The prefix characters are randomly generated, but it is possible to customize them.

Note: This is the main difference between the default tenant (the one that is prefixed on the central machine) and the others. The former, which comes from a monotenant device, has accounts that lack prefixes.

For example: if you have account x, defined in the default tenant, to continue using it, you either reconfigure your phone to assign a prefix to the default tenant or risk keeping the account without a prefix.

- **Accountlimit** (-1 is unlimited)
- **Exten limit** (-1 is unlimited)
- **Call limit** (-1 is unlimited): affects the limit of external calls
- **Local storage quota** (in MB): internal machine storage quota that the tenant can use to record custom audio files or voicemail messages
- **Provisioning fallback enabled**

Once configured, you can access your tenant of which you are an administrator.

Tenant groups list

In this section, a new tenant group can be created or modified. The tenant group is then used to group multiple tenants from multiple virtual pbxs – administratively distinct – into one group. Then it is possible, after putting them in the same group, to share the numbering plan and have the tenants communicate directly between extensions.

- **Name**: name of the group of tenants.
- **Tenants in group share selections**: checkable or uncheckable option.
- **Tenants belonging to this group**: list of tenants belonging to that particular group.

Remote extensions

In the numbering plan, the set selections that are served by the selected tenant are defined under the “remote extensions” section.

- **Selection type** (exact selection, selection interval, prefix selection).
- **Selection value**
- **Destination type**
- **Destination value**

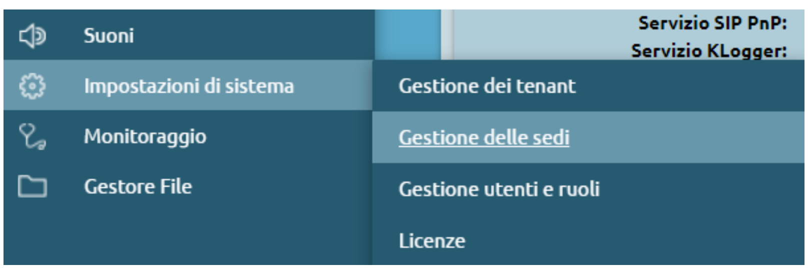
Site Management

Description

Sites can be configured at the PBX admin level. A site is defined by the IP addresses of the phone that belong to a particular site.

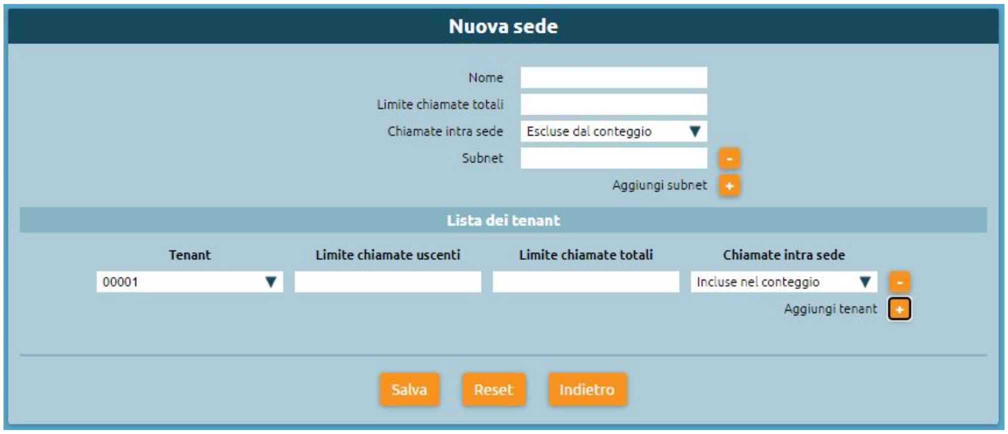
Configuration

To configure the site management service, you can follow the path "System Settings > Site Management"



To create a new location click on "Create New Site" The form must be filled with the following information:

- **Name**: name of the venue
- **Total call limit**: maximum number of total calls that are allowed on this venue
- **Intra site calls**: can be excluded from count/included in the count
- **Subnets**: registration ip of the phone that belongs to a particular venue



In the total call count, there are calls from telephones that go outside or to services at the central office, or to a responder, etc. It is then necessary to inform the PBX office whether any call between two extensions at this location should be charged in the count of calls that occupy the stream or not. This depends on the communication between the telephones, whether they communicate in direct media or not. If a call between two phones in the same location is not in direct media, it occupies two streams within connectivity.

As for intra site calls, if you are not sure whether direct media is applied on all internal calls, you will have to include them in the count. On the other hand, if the intra site calls work in direct media, you can exclude them from the count.

Note: Intra-office calls are different from inter-office calls. With "Excluded from count" intra site calls will go into direct media.

Example of a specific case: In the case where multiple tenants physically insist behind the same access connectivity, the telephones of the various tenants will show up at the exchange with the same IP. One can then partition this capacity to divide it among the various tenants insisting on a given location. In the simple case there will be a single tenant for which you assign:

- **Tenant**
- **Outgoing call limit**: a number that must be less than the total call limit and represents the maximum number that will be engaged by external calls
- **Total call limit** (insisting on the premises)
- **Intra-tenant calls**

Audiofile management

Description

The Audio File Management service includes the uploading and customization of audio files to be performed during the delivery of specific services.

Audio File Configuration

Following the path Sounds > Audio Files, we'll be on the page that lists the audio files present on the control unit and for each you can customize:

- **Play:**
  - Play in browser: the file is played directly in the browser
  - Download: the file is downloaded locally
  - Play on a device: you can choose which extension and which account to play it on. Pressing "Play" the selected extension will ring and you can listen to the audio file through the handset.
- **Replay:**
  - Load a new audio file: you can select it locally
  - Record a new audio file: you can record a new audio file using a telephone terminal. You can choose the extension and the account on which you will be called by the central unit that will provide you with instructions for the correct recording of the file
- Delete audio files we don't need anymore

The first three files we see are builtin, i.e. files made available to the control unit:

1. builtin/rec-start: warning file of the start of recording, in case of audio recording of phone calls is enabled
2. builtin/rec-stop: notifies the end of the recording of phone call audio
3. builtin/spu-start: audio message that notifies a call center operator (if the license is active) that a supervisor is starting to listen passively to that call

Upload new audio file

You can load a new audio file

- Available destination path: the path is a folder that allows you to identify file types more easily
- New destination path: you can insert the new path
- File: you can choose an audio file (.wav and .mp3)

The files will be available in the "Audio file list".

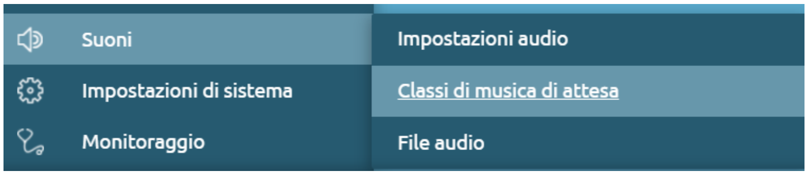
Record new sound file

- File name
- Available destination path
- New destination path
- Recording device: internal and account to be used for recording the audio file.

File size limit cannot exceed 5MB in size.

Music On Hold Classes Configuration

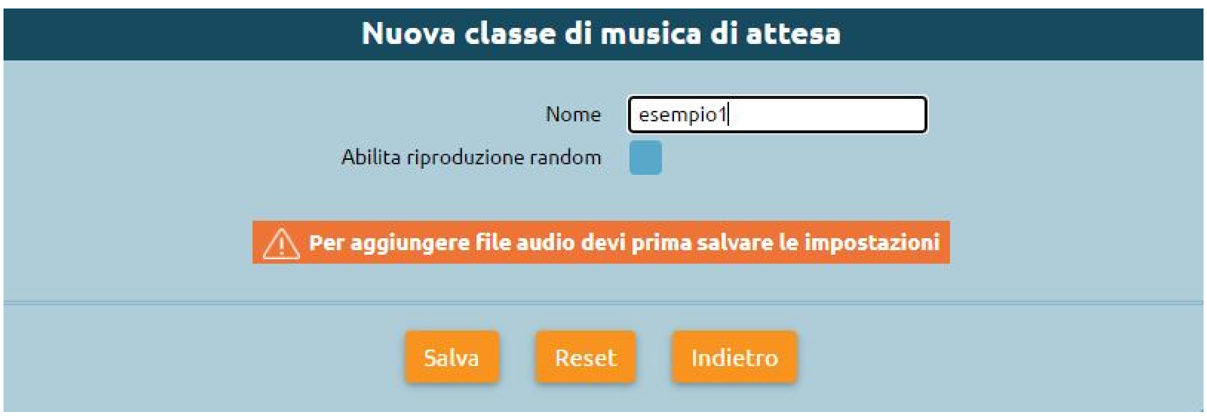
To configure the MOH classes we proceed by going to Sounds > Music on Hold Classes.



Hold music is the playback of audio files used for many services, including queues or during call transfers from one extension to another. The central unit has five pre-configured hold music classes available, but you can create an arbitrary number of new hold music classes.

New MOH class

- Name
- Random Enabled: option to randomly play the audio files in the playlist, which would otherwise be played from top to bottom.

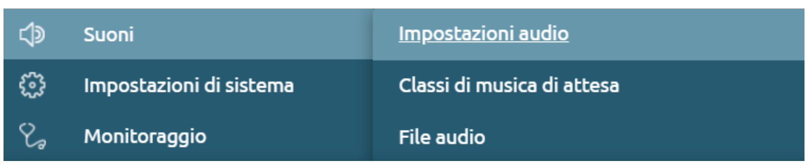


Edit MOH class

- Upload new audio files: you can load a series of audio files

Audio Settings Configuration

To configure the Audio settings we proceed by going to Sounds > Audio Settings.



- System Audio File Lang: you can change the language
- Default MOH Class: you can view the list of preloaded playlists or that we will add later
- Spy service Sound File: indicates the choice of the audio file to be used by default for the passive listening service

Custom Languages Configuratione

The "Custom Languages" function is available for the Singletenant Admin and Multitenant PBX Admin.

To get to the service, press "Sounds > Custom Languages".

In this panel, variants can be defined from a front audio base in a given language.

In the customization tab you can enter:

- Name
- Basic language pack
- Variant

At the audio settings level, you can choose a custom language that inherits all the audio prompts of the base language you have chosen. Clicking on the list of the custom language created shows all the audio files that make up the language pack of the selected base language. You can change the speaker or alter the content of the files by uploading an audio file to replace the standard one. Anything that is not returned is inherited from the chosen primary language.

Load archive with audio files

You can upload a zipper containing the audio files, they are automatically reallocated over the previous ones.

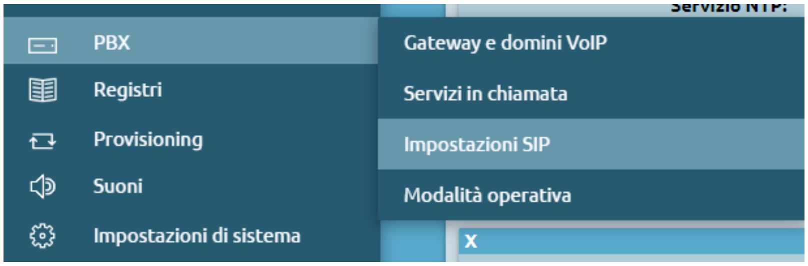
SIP settings

Description

The SIP settings panel contains the specification of telephone engine settings. The SIP protocol governs the communication between the exchange and the telephones and between the exchange and other gateways.

Configuration

The panel can be reached from the menu "PBX > SIP settings". In the case of a multitenant system, the panel is used by the PBX admin because even in a multitenant node, the SIP stack running on the machine is unique and shares settings among all tenants. The tenants, however, can independently perform the configuration of their own extensions/groups/queues/accounts.

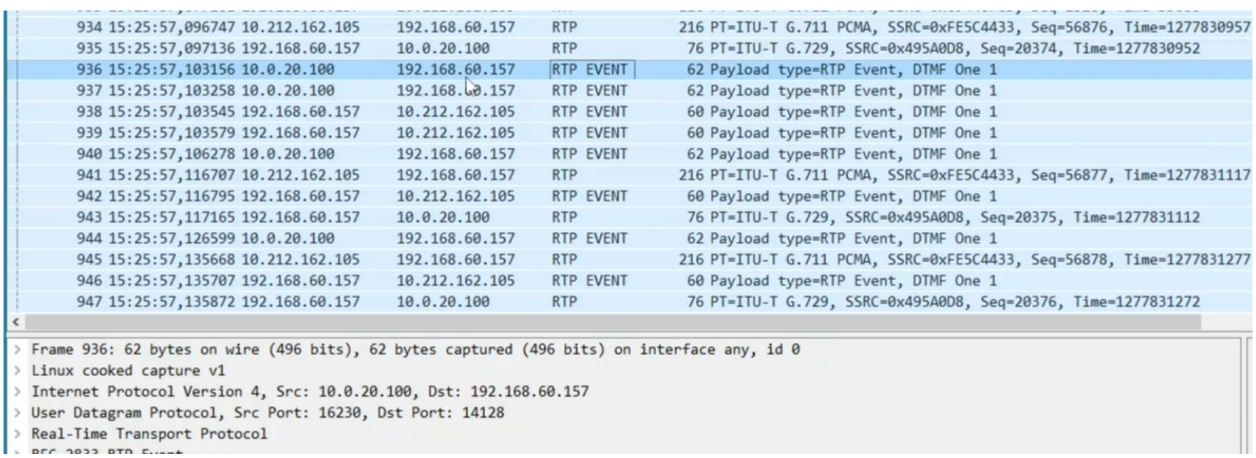


- **Enable pedantic check of SIP messages**: this causes a stricter check of the formal correctness of SIP messages sent to the pbx to be carried out, and if this legal check indicates malformations within the SIP packet, it discards it. This is a security measure to prevent malformed SIP packets from arriving at the exchange, creating malfunctions or attacks on the machine itself. In cases where one interfaces with providers (gateways, telephones) that create SIP messages that are not formatted entirely correctly, it may be helpful to disable the flag. In this case, message parsing is relaxed and made acceptable.
- **DTMF mode: tones that can be sent during a telephone call to send commands or information to the other party. There are three modes provided in the SIP standard by which DTMFs sent by the other party or sent to the other party can be sent and acknowledged.**
  - **RFC 2833**: can be considered side-by-side with 4733, which recalls and extends 2833.

This mode provides that DTMF tones are sent as RTP event packets within the RTP stream of a call. DTMF tones take the same path as the audio stream, but are not sent as tones, but as RTP event packets. These RTP event messages are sent by sending an initial packet (beginning of the tone). Then, with periodicity typical of VoIP, another packet is sent that lengthens the duration of the tone. These RTP event' messages contain the number of keys pressed and the symbols *#.

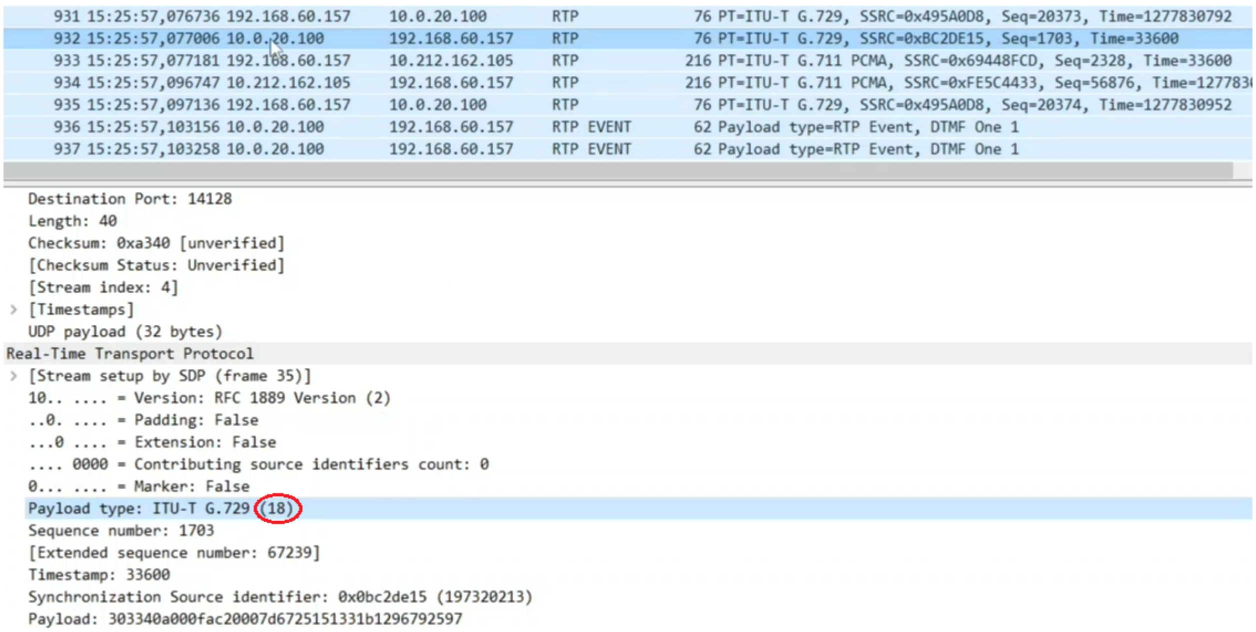
Example:

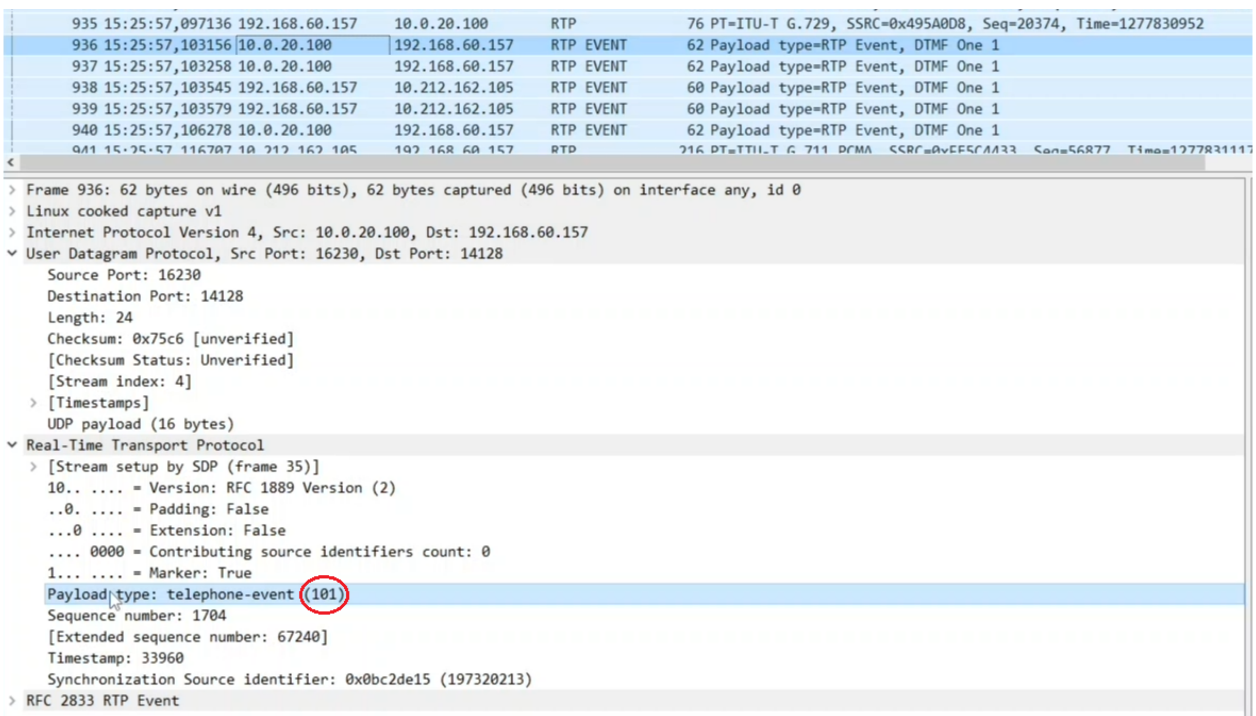
The DTMF tone sent in RFC 2833 mode by the 10.0.20.100 central unit to the PBX can be displayed:



This incoming RTP is passed back from the pbx to the interlocutor.

The audio packet is sent to a stream that has a source and destination port that contains the RTP stream. When the station sends a DTMF in RTP event mode within the same stream, the port extremes do not change (the same as in the audio packet), but the payload type changes.

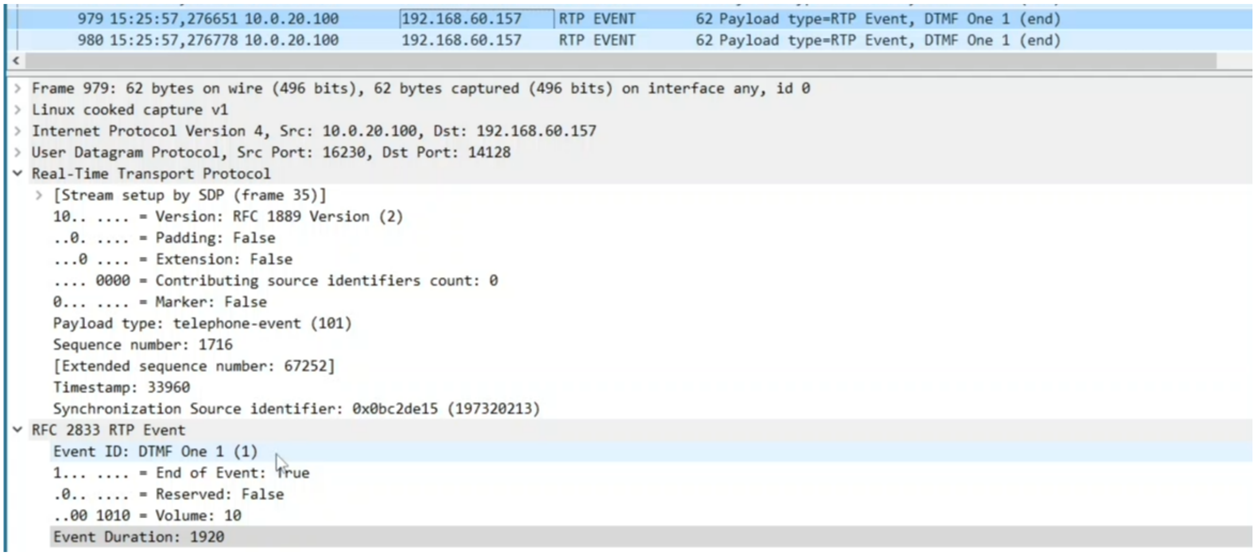




In the second case the payload type is 101 because the RTP event changes the payload type. Kalliope uses 101 by default for sending DTMFs.

The payload becomes explicit information of the key is pressed, the volume, and duration in samples.

UDP, in the case of a PBX with more than one network interface, causes VoIP service to be activated on all the interfaces in the PBX. A second network interface can be added, or VLAN tags can be added.



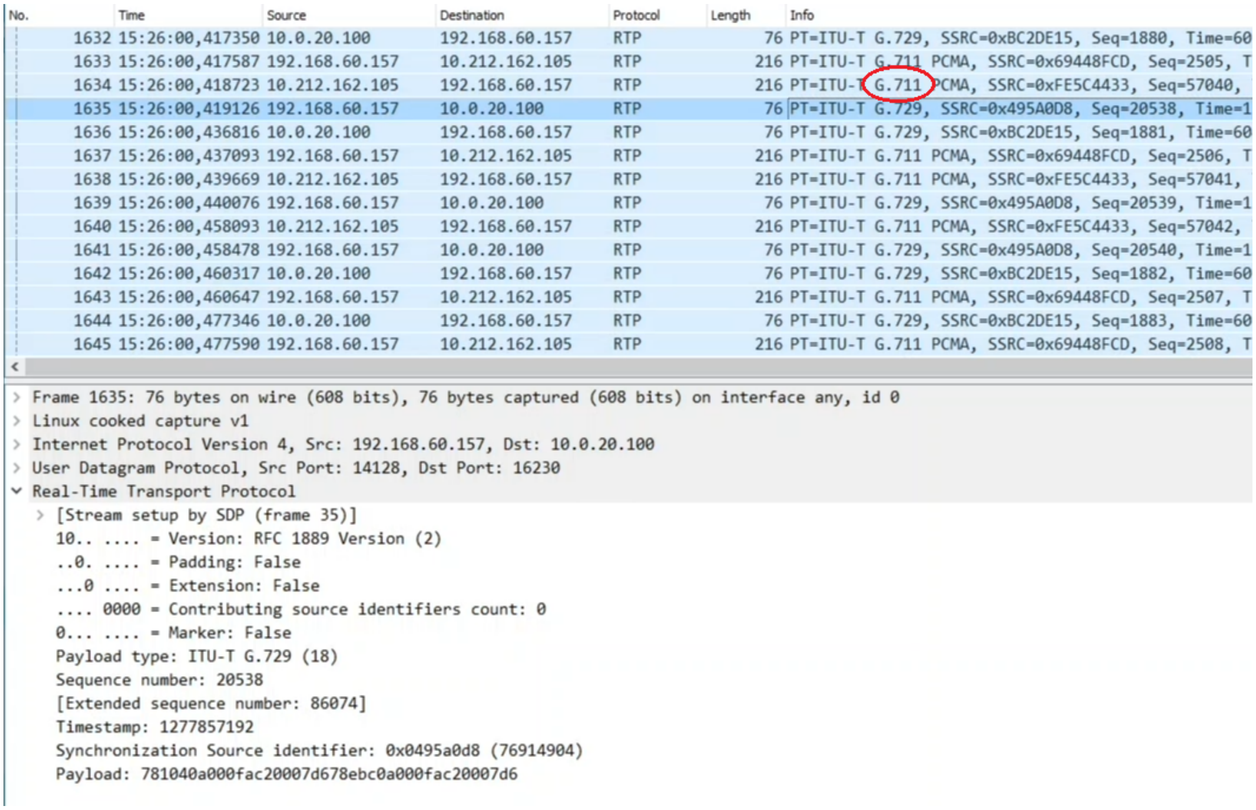
The duration is useful because if the central unit has to convert this DTMF, when sending it to the interlocutor in a format other than RFC 233, it should play it back for a corresponding duration. The setting in this panel applies to all SIP accounts defined in the "Internal and Accounts" panel and is used as the default setting for all incoming and outgoing lines.

Unlike accounts, in the panel, there is no ability to go and change DTMFs: when you send a DTMF to an account, you send it with SIP settings, and the same occurs when a SIP account sends a DTMF. Instead, you can change the DTMF mode in the Inbound lines panel or leave it as default.

This mode is often referred to as AVB (Attribute Value Pair) on other equipment.

- SIP INFO: is a mode that does not send RTP or RTP event messages, but an info type SIP message. The info is a type of SIP message that contains a payload containing the key and duration, is transmitted only once, and has a network path that follows that of the SIP signaling which is different from the path that the RTP flow takes.
- In-band: typed DTMF tones are sent as audio tones within the RTP stream.

If you analyze the tracks containing DTMF tones, you can find RTP only. So you cannot distinguish whether they are tones or not. The problem with DTMF transmission under audio tones is that if codecs other than G.711 are used, they introduce compression into the audio and distortion of the tones. While a conversation is understandable, recognition of these tones may fail.



Thus, the use of in-band DTMF mode is discouraged because if the RTP stream suffers degradation due to losses or congestion, the tones, even using G.711 are distorted and may not be recognized. Thus, the use of DTMF in audio is recommended only in conjunction with a compressed PCM a-low or PCM u-low codec, but it may still be error-prone.

The most commonly used mode is RFC 2833.

The following two fields are attributes that impact both SIP accounts and input and output lines; as in the case of DTMF mode, the next two are also modifiable from the default value in the input and output lines.

- **Enable RPID (Remote Party ID) acceptance**: enables COLP acceptance (named thus in the inbound and outbound lines).

Enabling causes it to extract the caller's identity from the FROM it sends and the PAI, RPID, etc. headers. This is a SIP header that carries additional information that identifies the caller. During a call, there are instances when the identifier of the person being talked to may change over time. The COLP feature causes the exchange to notify that the caller has changed; there are some cases where it is necessary to provide this information to the caller, and others where it is not. This flag causes it to accept from telephones connected to the exchange, the caller ID present in the RPID or PAI (P-Asserted-Identity) header.

Note: At the output level, it is recommended that the entry be disabled since you do not want the pbx to change the identifier of the connected line.

- **Send rpid mode**: Send COLP mode (named so in the input and output lines).

It has three options: disabled / remote party ID / P-Asserted-Identity. In this case, the setting that is used for telephones and that is recommended is sending PAI. This flag causes the pbx to warn that the interlocutor being communicated with has changed, it sends a message that updates the information via the PAI header. The interlocutor change information is useful toward telephones, but generally, on an outgoing line it should be disabled.

Note: If you go to make a trunk to another exchange where you configure remote extensions, in this case the line is not an outgoing trunk to an operator, but it is the interconnection to another exchange on which it is helpful to activate the COLP update mode.

The following three fields set the quality-of-service parameters for the service leaving the exchange. There are TOS hexadecimal values and priority codes in the network to decide which packets to make travel with priority over others in case of congestion. Thus, it ensures that audio and video streams receive priority over sending the mail.

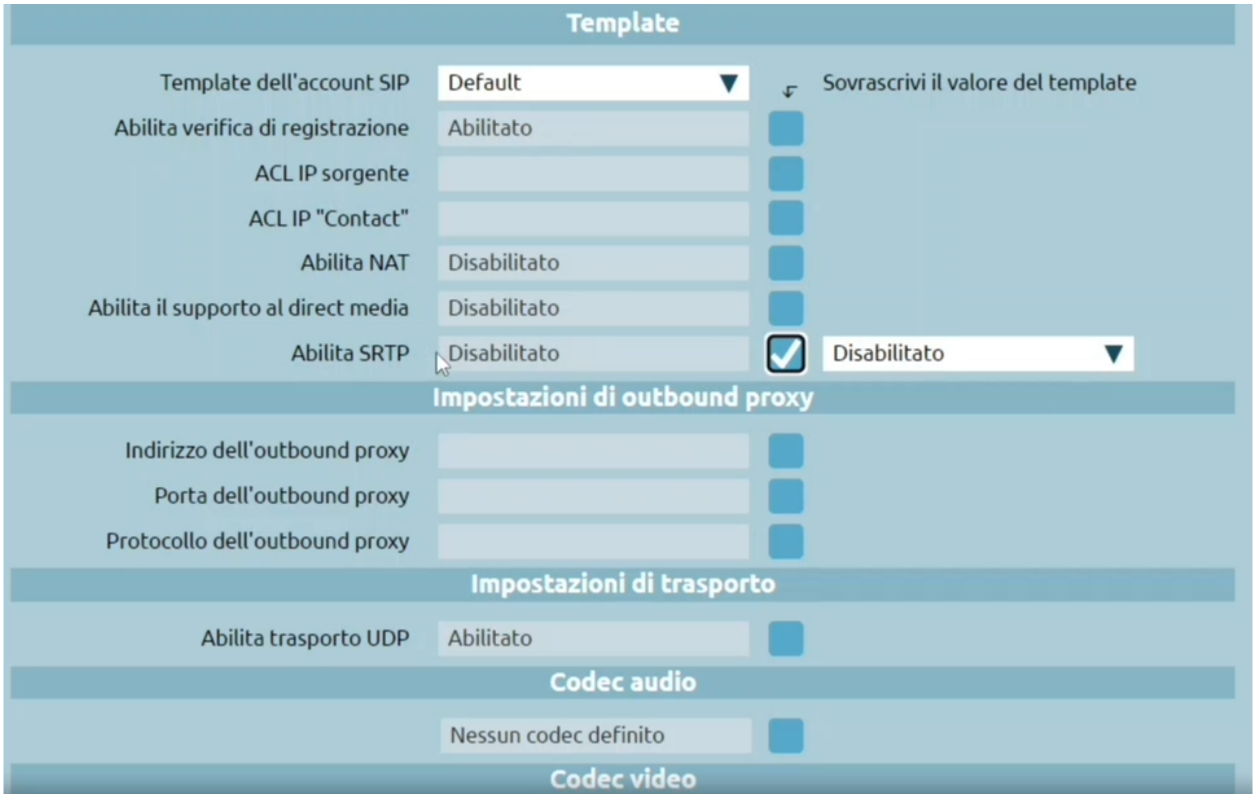
- TOS SIP
- TOS Audio
- TOS Video

These values can be changed if the network in which the control panel is operating requires that different TOS values be used for audio, signaling, and/or video.

- **Enable Video**: to globally enable support for video calls.
- **Maximum bitrate for video calls**: attribute that applies only to video calls and determines the maximum bitrate that the exchange accepts and offers for video calls, by default it is 384 Kbps so as not to take up too much bandwidth. Often values can be raised to increase quality.
- **Enable RTP encryption support (SRTP)**: enables at the pbx level the ability to allow encryption of RTP streams that generally travel in the clear (anyone who can intercept the RTP stream of the call can listen to its contents).

SRPT is a protocol that relies on the prior exchange of keys between the caller and called party; the flow is encrypted with a symmetric key cipher. The key is exchanged during call setup, and if a secure signaling transport protocol such as TLS is not used and TCP or UDP is used instead, it can be read in the cleartext on the SIP message. Enabling SRPT makes sense if coupled with a secure signaling protocol such as TLS.

Warning: Enabling SRTP in the SIP settings does not automatically enable the actual use of SRTP, but it does allow the module from SRPT support. You need to enable it in the Extensions and Accounts panel. Here, there is a checkmark "Enable SRPT": this way it can be inherited by all accounts that use that particular template.



The following two fields are not necessary to configure, except when you want to expose the service via WebRtc (via Web Socket) and specify a STUN server to acquire the information needed to shut down signaling properly and media.

Note: STUN is a protocol used to distribute the public IP information a client uses behind a NAT. It is used by rtc (web phone) web clients to ensure that the central station knows what IP address to send media streams. It goes hand in hand with enabling transport (protocol for SIP signaling) via Web Socket. Once Secure Web Socket is allowed, there is a need to specify a STUN server, which the central office uses to learn which ports it will use for signaling and media.

- STUN server address
- STUN server mail

NAT Helper

This section is especially important for making the station available in access from the public.

- External host: public IP address.
- External host update period (sec.)
- External UDP port
- External TCP port
- External TLS port
- Local networks

All messages sent to networks marked as local use their private address as routed, while what is outside local networks uses the IP address you specify in the external host.

Predefined video codecs

To enable video call support, check the above “Enable video” box and select which video codecs are supported by the central unit.

Codec audio

Also, for audio, you can select which codecs are supported by the power plants.

SSL Settings

Description

The SSL (Secure Sockets Layer) settings section contains the management of trusted CAs (Certification Authorities).

Configuration

To reach the SSL settings section you can follow the path “System settings > SSL Settings” as shown in the right figure.

Trusted CA Management

This section contains the list of Certification Authorities of phone vendors that the pbx considers valid to authenticate a client certificate.

The server certificate is rarely issued directly from one of the CAs on the phone, since they are no-root CAs. Often certificates are issued by intermediate CAs.

Note: It is important to remember the correct sequence of the chain: root CA > intermediate CA > server certificates

The root CA issues a certificate to an intermediate CA and the intermediate CA issues the server certificates. You have to upload the server certificate consisting of the actual certificate and private key and the intermediate CAs that are used to build the chain of trust up to the root CA.

The certificates must be put together inside a single .pem file. The phone then provides the client certificate, the certificate is validated by the pbx using the trusted CAs in the panel.

If the certificate is deemed valid and both the CN (common name) and MAC address match the file it is requesting, then everything matches, the session closes, and the download of the provisioning file can start.

While on browsers intermediate CAs are often preloaded, on phones, for reasons of memory occupancy there are only root CAs. If we load the server certificate signed by an intermediate (signed by a root CA) on the machine, but the server passes only its own certificate and not the intermediate one to the phone, it is not considered valid by the phone.

It is then necessary to upload both the intermediate CA and the server certificate.

Server certificate management

In this section you can upload the server certificate in a single .pem file.

By default, on all Kalliope, there is a self-assigned certificate that is issued by a self-generated, in-house CA.

You can create a new CSR certificate (Certificate Signing Request) by clicking on "Create new CSR" and entering:

the details of the new certificate:

- Country
- State
- Locality
- Organization
- Organizational unit
- Common name
- E-mail

Local CA Management

In this section you can observe the Root Certificate Details and the Certificates List.

It is also possible to:

- Emit new certificate, insert new certificate details and Subject Alternative Names:
  - nstall as server certificate
  - Country
  - State
  - Locality
  - Organization
  - Organizational unit
  - Common name
  - E-mail
- Download root certificate (.pem)
- Download root certificate (.der)
- Delete local CA

Forward On All Unreachable

Note: Information

- Firmware: 4.5.9 or later
- Available in single-tenant and multi-tenant (tenant admin level) systems

Description

This service lets you specify a destination to which inbound calls to the PBX (or the tenant in multi-tenant systems) should be forwarded when all SIP accound linked to the extensions are unreachable.

This service is particularly useful when the PBX is remotely installed with respect to the terminals (for example, when it is installed in a remote deta center while the telephones are located in the client's offices); should the client's offices be cut off from the from the PBX, all inbound calls can be forwarded to a backup destination, such as a mobile number or a courtesy message.

In multi-tenant systems, the service and its destination can be activated independently for each tenant by the tenant admin.

Configuration

The service can be configured in the PBX -> General Settings page, under the "Behavior when all accounts are not registered" section.

Other than the checkbox for activating the system, there is a form in which you can select the forward action, as well as specify an audio file to play to the caller if desired and the action to perform on the inbound call (selectable from a drop-down menu). If you wich to forward calls to an extrernal number (to reach during an emergency), you will nedd to specify the destination number, without any external line selection prefix, the identity (which will determine the calling number), and the class that KalliopePBX will use to make the call.

Interfacing with third party software via AMI

Descrizione del servizio

The Asterisk Manager Interface lets KalliopepBX interface with third party software. This panel lets you define the authentication credentials (username and password) along with an ACL comprised of one or more IP addresses or subnets. The configured user has read permissions for "call" and write permissions for "call,originate".

Configuration

By enabling the AMI interface from the Kalliope GUI, you can interface with external systems to carry out click-to-call operations.

The standard syntax to carry out c2c via AMI (from the extension %extension% towards the destination %toNum%, including the outbound prefix for external numbers) on KalliopePBXv4 is the following, in which some channel variables are set:

```
Action: Originate
Async: true
Channel: Local/%interno%@c2c
Context: from_c2c
```

(continues on next page)

(continued from previous page)

```
Exten: %toNum%
CallerId: %callerId%
Timeout: %timeout%
Priority: 1
Variable: C2C_SRC=%interno%
Variable: C2C_DST=%toNum%
Variable: __TENANT_UUID=%tenantUid%
```

Where:

- **%callerId%** = in "%displayname%" <%number%> format (we set "c2c: %toNum%" <%toNum%>)
- **%timeout%** = the number of milliseconds in which to accept the call on the caller's terminal (we set 10000)
- **%tenantUid%** = the tenant UUID. It must be indicated even in silge tenant systems; it can be found in the the AMI settings page (with firware version 4.2.x) or the dashboard widget (with firmware version 4.3.x)

With TSP Xtelsio Tapi for asterisk (frequently used to integrate the Estos ProCall application with Asterisk systems), it is not currently possilbe to set these variables in the AMI call, so a mechanism has been developed based on context wrappers to set the required variables.

The AMI message to send is therefore (both modes are supported):

```
Action: Originate
Async: true
Channel: Local/%interno%@c2c_%tenantUid%
Context: from_c2c_%tenantUid%
Exten: %toNum%
CallerId: "c2c: %toNum%" <%interno%>
Timeout: %timeout%
Priority: 1
```

Restricted operating mode

Description

This service lets the KalliopePBX administrator specify operating restriction in order to limit certain types of calls while leaving the configuration (extensions, accounts, routing, outbound lines) unchanged.

For the moment, there are three available operating modes:

- Full: KalliopePBX is fully operational and there are no restrictions on calls. This is the default mode.
- Limited: block outbound calls: allows calls between extensions, KalliopePBX service numbers, and whitelisted numbers. All other outbound calls are blocked.
- Disabled: all phone functionality is disabled (account registration, calls between extensions, etc.).

In multitenant PBXs, the operating mode is configurable for each tenant, but only by the pbxadmin.

Configuration

The operating mode can be configured in the PBX –> Operating mode page.

Whitelist

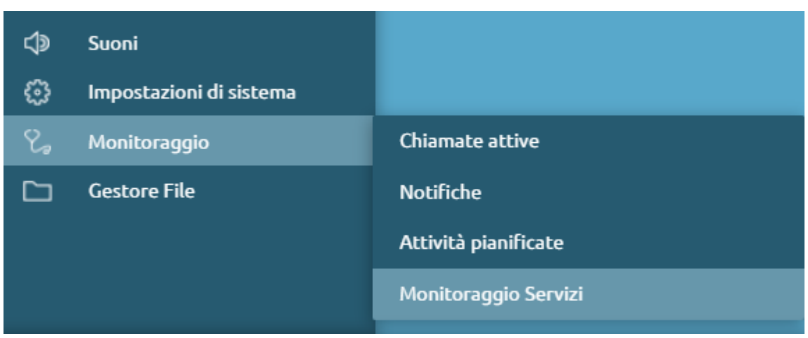
For limited mode, any numbers you wish to allow outbound calls to (e.g. emergency calls) must be added to the whitelist.

This can be done in the PBX –> Whitelist page.

Service Status

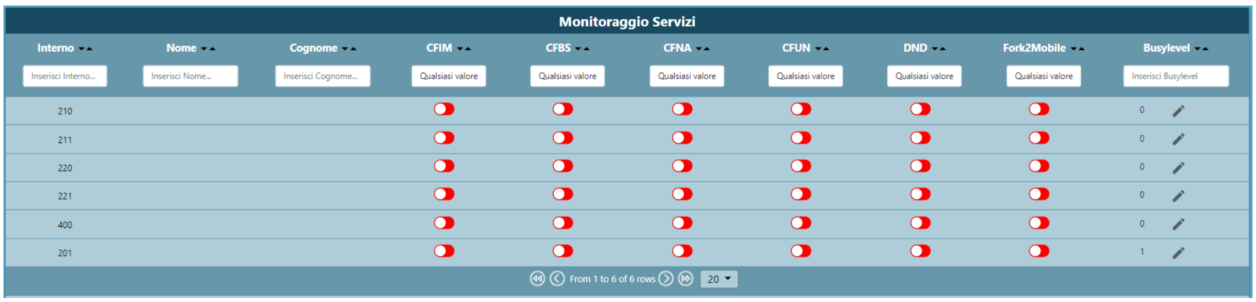
Description

Per accedere al servizio di Monitoraggio Servizi basta seguire il percorso "Monitoraggio > Monitoraggio Servizi".



On the page, you can view and filter the following items to refine your search:

Parameter	Value
Extension	Alphanumeric
Name	Alphanumeric
Surname	Alphanumeric
CFIM	Any value/OFF/ON
CFBS	Any value/OFF/ON
CFNA	Any value/OFF/ON
CFUN	Any value/OFF/ON
DND	Any value/OFF/ON
Fork2Mobile	Any value/OFF/ON
Busylevel	Alphanumeric



By the edit icon, it is possible to “Enable Busylevel” for a selected extension:

Events Notification

Description

This service lets you monitor the selected events by receiving notifications.

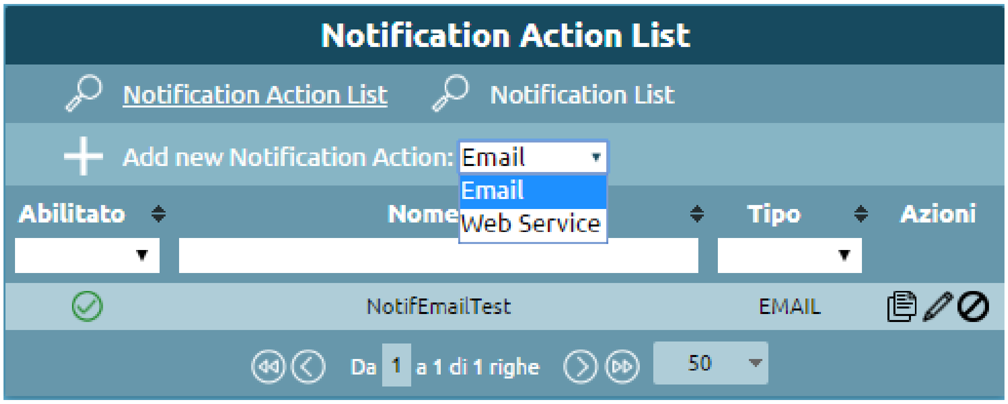
You can associate each selected event to a notification action such as an email or a call to a web service.

Configuration

In the Monitoring → Notifications section, you can manage the notification funcionality

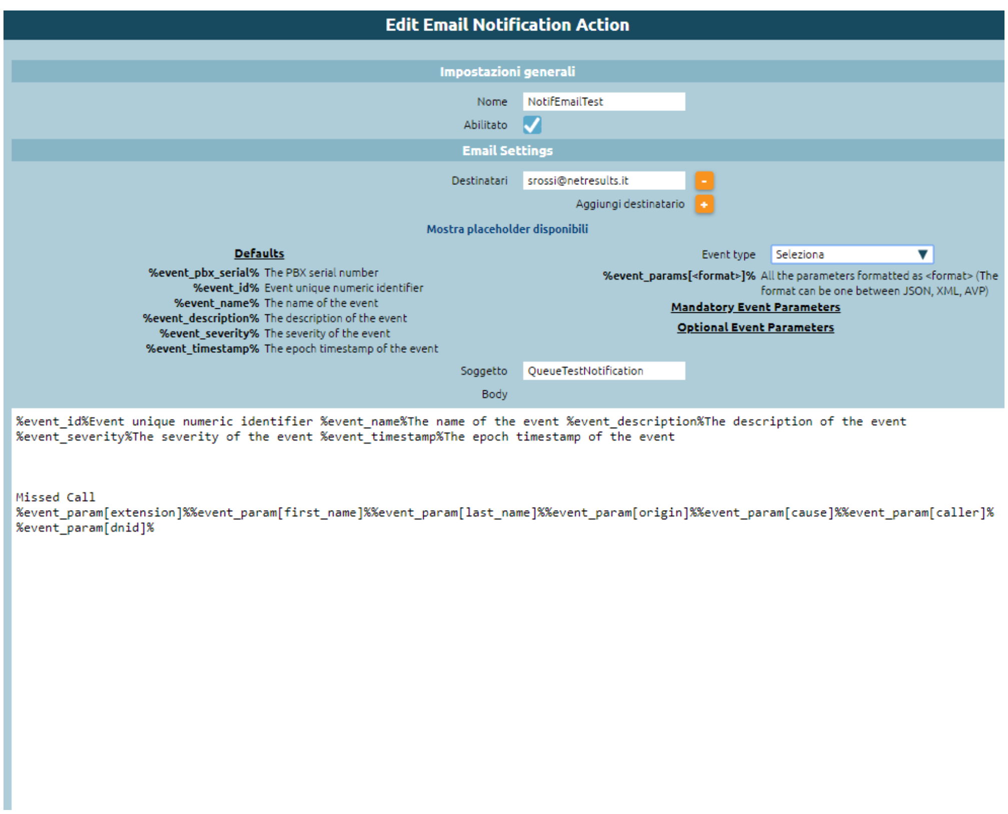
Notification Action List

In Notification Action List you can add a new notification by selecting Email or WebService action.



Email

By selecting Add New Email Notification Action, it is possible to define the recipient of the event notification email and the information we want to convey.



The following table shows the parameters you can define for the Email Notification.

Parameter	Description	Value
Enabled	Allows you to disable Email Notification	Yes / No
Name	Notification ID	Alphanumeric

Email Settings

Parameter	Description	Value
Recipients	Email address of the recipient of the notification	Alphanumeric
Subject	Objects of the notification email	Alphanumeric
Body	Email text containing both default and event-specific placeholders	Alphanumeric

The following table lists the generic parameters:

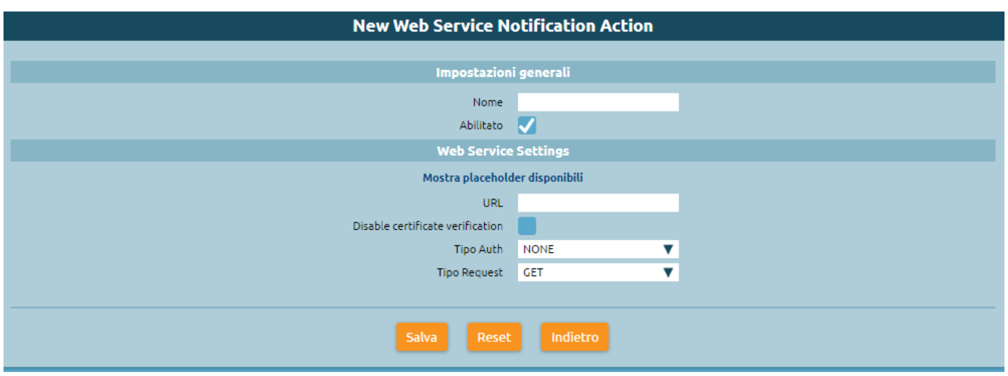
Parameter	Description
%event_id%	Event id
%event_name%	Event name
%event_description%	Event description
%event_severity%	Event severity (numeric, from 4 to 0 corresponding to DEBUG|INFO|WARNING|CRITICAL|FATAL)
%event_timestamp%	Epoch in which the event occurred

Event-specific parameters are listed in the Notification page; JSON, XML, and AVP formats are supported.

The full set of parameters related to an event can be obtained with the placeholder: %call_params[<format>]%

Web Service

By selecting Add New WebService you can add a new web service notification. You will need to insert the name of the notification in the general settings.



The following table lists the configurable parameters for web service notifications.

Parameter	Description	Value
Enabled	Allows you to disable WebService Notification	Yes / No
Name	Notification ID	Alphanumeric

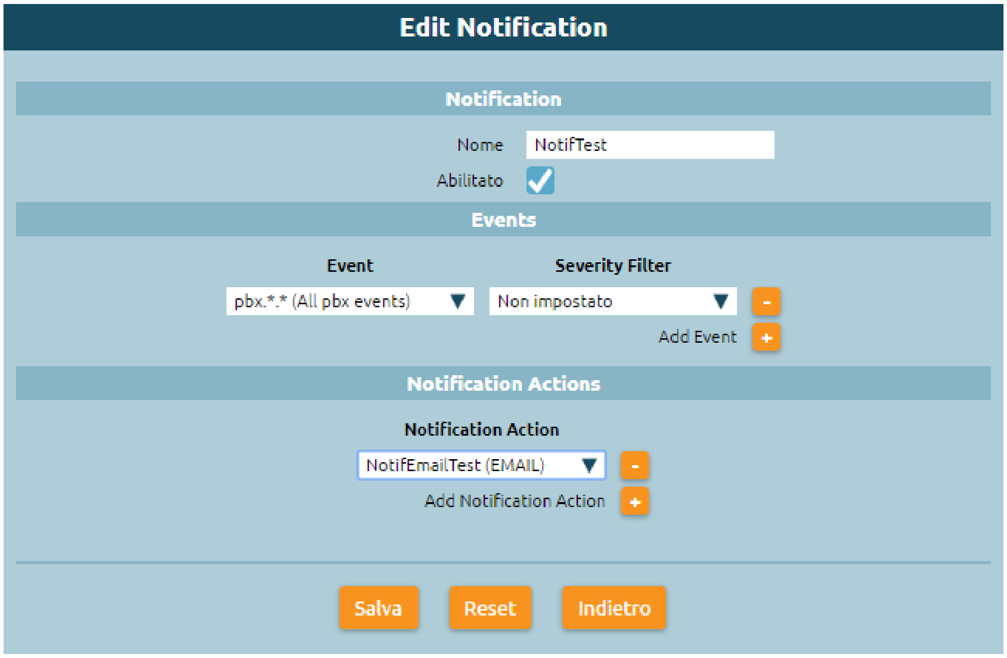
WebService Settings

Parameter	Description	Value
URL	Notification URL	Alphanumeric
Auth Type	Authentication Type	None/ Basic
Auth username	Authentication Username (only in Auth case)	Alphanumeric
Auth password	Authentication Password (only in Auth case)	Alphanumeric
Request type	Type of request	Get/Post
Request content	Content of request (just Post)	Placeholder

When the event occurs, a notification will be sent to the external web service, which will handle the information received.

Notification List

In the Notification List section you can select Add New Notification to select the event for which you wish to be receive notifications.



The following table lists the events that can be monitored and with which a notification can be associated.

Event	Description
ademco.*.* / alarmreceiver.*.*	Specific events used by the optional KalliopeLift module to interface with elevator dialers
cti.client.background	A CTI (mobile operating system) client was put in the backrground
cti.client.login	A CTI client logged in
cti.client.login-failed	A CTI client failed a login
cti.client.logoff	A CTI client logged out
mobile-app.call.incoming	Incoming call to mobile app account
mobile-app.call.timeout	The call to the mobile app account has expired
mobile-app.status.not-logged	The mobile application is not registered
mobile-app.wake-up.registered	The mobile app has been activated
mobile-app.wake-up.sent	Alarm notification sent to mobile app account

Table 3 – continued from previous page

Event	Description
mobile-app.wake-up.timeout	The mobile application does not activate within 5 seconds of sending the notification
pbx.account.incomingcall	A call for an extension forwarded to the account
pbx.account.startcall	Attempted call to account initiated
pbx.account.unavailable	Attempt to call the account not initiated because the account is unavailable
pbx.call.end	Call ends
pbx.call.start	Call starts
pbx.dynamic-routing.enter	A call has entered the Dynamic Routing service
pbx.dynamic-routing.input	A new parameter was entered by the caller in the dynamic routing
pbx.extension.answercall	Extension call answered by one of the associated accounts
pbx.extension.failedcall	Call to extension failed
pbx.extension.incomingcall	Incoming extension call
pbx.extension.missedcall	An extension missed a call; the event is triggered only if “generate event” is checked in the overflow
pbx.queue.enqueue	Call on hold
pbx.queue.enter	A call comes in to the queue service
pbx.queue.ringmember	A call is presented to a queue operator
pbx.queue.ringnoanswer	A selected operator did not handle the call; the call is still in queue and will go to other operators, if
pbx.queue.servedcall	A call in the queue was served, i.e., answered by an operator
pbx.queue.unservedcall	A call in the queue was not served globally; therefore, it represents the final outcome of the call that
pbx.queuemember.added	Queue member added
pbx.queue.enqueue	A call enters the queue service; the queue is open
pbx.queuemember.pause	A queue member entered a pause
pbx.queuemember.unpause	A queue member came unpause
pbx.spy.start	Supervisor spy started
pbx.spy.stop	Supervisor spy stop
pbx.queuemember.removed	A queue member is removed
pbx.wake-up.unanswered	The wake-up service had no response from the room
pbx.user.create	A new Kalliope user has been created
pbx.user.password-change	A Kalliope user’s password has been changed
storage.quota.exceeded	The storage quota reserved for a specific tenant has been exceeded
storage.quota.restored	The storage occupancy of a specific tenant is restored

The following table lists the configurable parameters for notifications.

Parameter	Description	Value
Enabled	Allows you to disable the notification	Yes / No
Name	Notification ID	Alphanumeric

Events

Parameter	Description	Value
Event	Type of events for which you want to be notified	From list
Severity	Event Severity	Fatal/Critical/Warning/Info/Debug

Notification Action

Parameter	Description	Value
Notification Action	Association with a NotificationAction	From list

When the selected event occurs, you will receive a notification with the desired information.

Practical example

For the “Unserved Call” event, if 103 calls 201, who is a member of the queue QueueTest, and after 5 seconds the caller leaves the service, we can request an email containing information on:

- the id of the event
- the name of the event
- the name of the queue and waiting time
- the reason the call was not served

by inserting the corresponding placeholders in the body of the email.

In the Notification List, we associate the “pbx.queue.unservedcall” event to the previously created notification action.

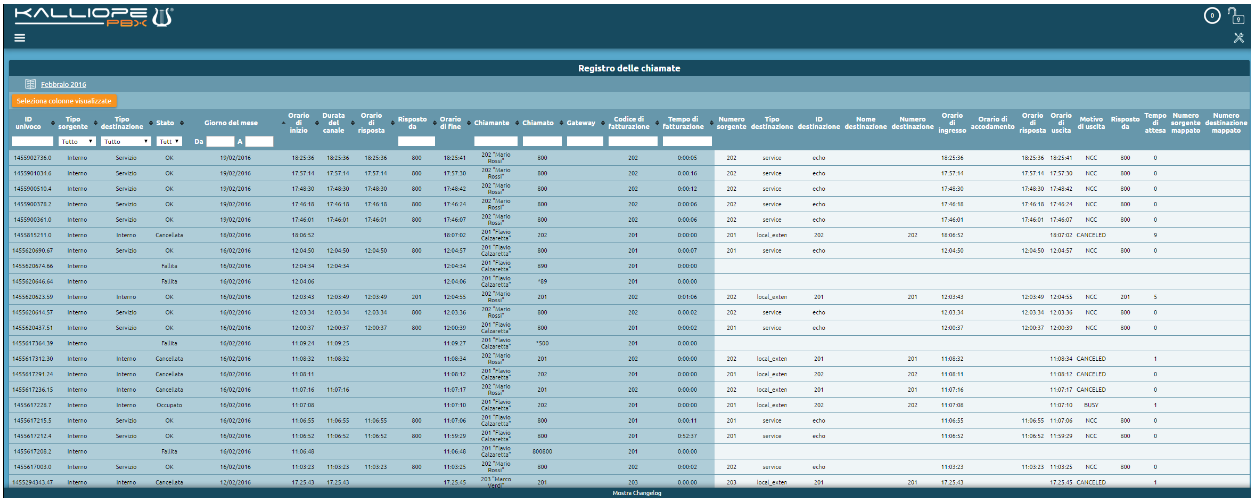
We will then receive an email with the following information:

```
Unserved
1511212918.0
1
Default
103
201
5
CANCELLED
```

Or we can receive the following response by inserting the placeholders:

```
%call_params[<JASON>]% :
 {"reason":"CANCELED","queue_id":"1","uniqueid":"1511212918.0","called_num":"201",
↪"caller_num":"103","queue_name":"QueueTest","waiting_time":"5"}

%call_params[<XML>]%

> <?xml version="1.0"?>

> <response><reason>CANCELED</reason><queue_id>1</queue_id><uniqueid>1511212918.0</
↪uniqueid><called_num>201</called_num><caller_num>103</caller_num><queue_name>QueueTest
↪</queue_name><waiting_time>5</waiting_time></response>

> %call_params[AVP]%:

> reason=CANCELED&queue_id=1&uniqueid=1511212918.0&called_num=201&caller_num=103&queue_
↪name=QueueTest&waiting_time=5
```

Call Detail Record (CDR)



Description

This page lets you view the call detail record through KalliopePBX. A new tab will be automatically created for ease of viewing every month. All the following information are exportable through API REST or in other formats (Excel, XML, JSON, CSV) by clicking on the proper button "Export in format" present in the panel header. All the information contained in the CDR can also be sent periodically as better described in the proper section.

Calls are shown from most to least recent, and display the following information:

Parameter	Description	Value
UniqueID	a unique call identifier	unique_id
Source Type	The type of call source (extension, inbound line, remote extension)	source_type, click *here* for possibile value
Destination Type	The type of call destination (extensions, remote extension, queue, ring group, IVR, outbound line, service)	destination_type, Click *here* for possible value
Status	The outcome of the call (failed, busy, canceled, not answered, OK, forbidden)	status, Click *here* for possible value
Day	Date of the call (day/month/year)	the date is included in subsequent export timestamps
Start Time	Call start time	start_datetime
Channel Up Time	Time of opening of the media channel of the call, i.e. when the PBX answers. In the case of outgoing calls this coincides with the “answer time”.	channel_up_datetime
Answer Time	Call answering time. For incoming calls it indicates the answering time of an extension. For outgoing calls it coincides with the “Answer channel” time.	answer_datetime
Answered by	ID of who answered	answered_by
End Time	Call end time.	end_datetime
Caller	Number and ID (if it is present in the phonebook) of the caller	caller e caller_name
Anonymous	Flag that identifies if the calling number is anonymized (yes/no)	anonymous (0/1)
Called	Number and ID (if it is present in the phonebook) of the called number. In the case of incoming calls it identifies the public number dialed by the caller	caller
Gateway Name	Indicates the gateway or termination used (in the case of outgoing or incoming calls)	gateway_name
Account Code	Indicates the code (or “tag”) assigned to the call. It is possible to assign a “tag” to a particular call using the phone code configurable in the Numbering Plan (available only with the Call Center license).	account_code
Duration	Total duration of the call expressed in hours, minutes and seconds (hh:mm:ss)	duration
Bill Time	Actual call duration once established (following SIP 200 OK message) expressed in hours, minutes and seconds (hh:mm:ss)	bill_secs (secondi.millisecondi)
Source Peer Name	For outgoing calls or calls between extensions, it indicates the SIP source account of the call.	src_peer_name
Source IP/Port	For outgoing calls or calls between extensions, it indicates the IP address and the source port of the call.	src_ip_port

Call path

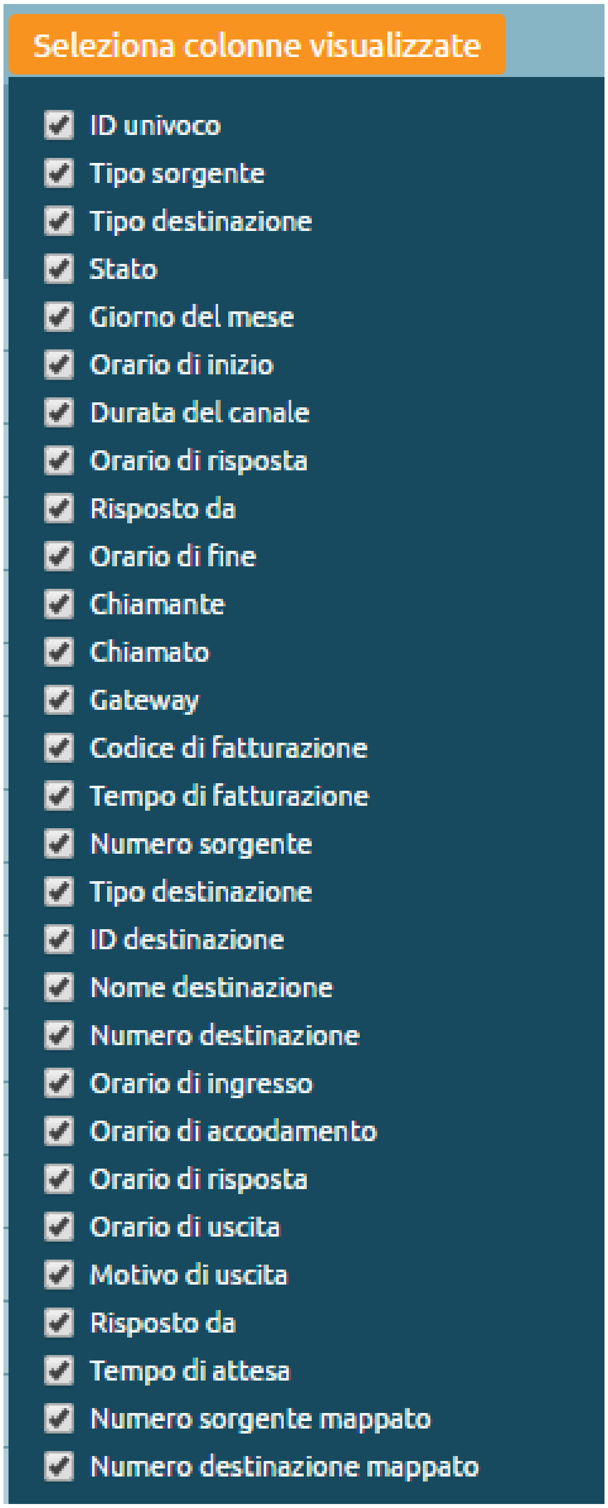
For every call, it is also possible to reconstruct the entire path of the call in the numbering plan, reporting for each step a detail line containing the following fields:

Parameter	Description	Exported field
Source Number	Call source number	detail_source_num
Source Number	Flag that identifies if the calling number is anonymized (yes/no)	detail_anonymous (0/1)
Destination Type	The type of call destination (extension, remote extension, queue, call group, IVR, outgoing line, service)	detail_destination_type
Destination ID	Identifier of the type of destination; in the case of destination “service” indicates the particular service involved, in the case of destination “local_exten” or “voicemail” indicates the recipient extension, in other cases the identifier of the particular destination (e.g. the id of the IVR menu or destination queue)	detail_destination_id
Destination Name	In the case of destination “local_exten” (extension) it shows the name associated with the recipient extension, in the case of destination “queue”, “FAX” or “callg” it shows the name of the recipient entity, in the case of destination “obl” (outgoing calls) it shows the name of the outgoing line used, in the case of destination “service” it shows further details on the service	detail_destination_name
Destination Number	Indicates the destination number in case the “Destination type” is “local_exten” or “obl” (outgoing calls)	detail_destination_num
Enter Time	Time (hh:mm:ss) of call entry in the corresponding detail line	detail_enter_datetime
Enqueue Time	Time (hh:mm:ss) of when the call enters the queue (only if the detail line includes a destination of type “queue”)	detail_enqueue_datetime
Answer Time	Time (hh:mm:ss) the call was answered (only if the call was answered or destined in a voice mailbox)	detail_answer_datetime
Exit Time	Time (hh:mm:ss) of call exit from the corresponding detail line	detail_exit_datetime
Exit Cause	Time (hh:mm:ss) of call exit from the corresponding detail line	detail_exit_cause
Answered By	The number that answered the corresponding detail line	detail_answered_by
Waiting Time	Indicates in seconds the time elapsed between the input time and the response time	detail_waiting_time (secondi)
Mapped Source Number	For outgoing calls it indicates the source number resulting from any manipulation of outgoing calls	detail_mapped_source_num
Mapped Destination Number	For outgoing calls it indicates the destination number resulting from the possible manipulation of outgoing calls	detail_mapped_dst_num
Account Code	Indicates the code (or “tag”) assigned to the call. It is possible to assign a “tag” to a particular call using the phone code configurable in the Numbering Plan (available only with the Call Center license).	detail_account_code

You can order the calls according to each parameter and invert the order (increasing or decreasing) by clicking on the header of the corresponding column.

You can filter the CDR by each of these fields (e.g. by viewing only calls from a specific extension or to a specific number, or during a specific time range) by clicking on the box corresponding to the column.

You can also click on Select columns to display to choose which items to show.



List of SOURCE/DESTINATION TYPE and DETAIL SOURCE/DESTINATION TYPE codes

Source/Destination Type:

- **local_exten** –> PBX extension
- **ibl** –> nbound line
- **obl** –> outbound line
- **service** –> PBX local service
- **callg** –> call group
- **queue** –> queue
- **ivr** –> IVR menu
- **conference** –> dialout conference room
- **fax** –> FAX server istance

Detail Source/Destination Type:

- **local_exten** –> PBX extension
- **ibl** –> inbound line
- **obl** –> outbound line
- **service** –> PBX local service
- **callg** –> call group
- **queue** –> queue
- **ivr** –> IVR menu
- **conference** –> dialout conference room
- **fax** –> FAX server istance
- **dre** –> dynamic routing
- **checktime** –> checktime
- **voicemail** –> voicemail

List of EXIT CAUSE and DETAIL EXIT CAUSE codes

Exit cause:

- **OK** –> ended call after being answered from a service/extension/external number
- **CANCELED** –> ended call because because it was cancelled by the caller before it was answered from a service/extension/external number
- **NOANSWER** –> call ended unanswered from a service/extension/external number
- **BUSY** –> ended call because the called number is busy
- **FAILED** –>ended call because there is no rule to route the call (no destination)
- **UNAVAILABLE** –> ended call because destination is not available (e.g. destination phone not registered)
- **FORBIDDEN** –> ended call because it came from an unknown incoming line
- **??** –> it was not possible to trace the reason why the call ended

Detail exit cause:

- **CANCELED** –> the caller ended the call before it was answered
- **NOANSWER** –> destination did not answer
- **BUSY** –> destination is busy
- **NCC** –> call to destination terminated after being answered (Normal Clearing Code)
- **ANSWNOACC** –> call answered by mobile but not accepted (key 1 not pressed)
- **PICKUP** –> incoming call picked up from another extension
- **PARKED** –> call parked in one of the parking slots
- **UFWD** –> incoming call redirected due to unconditional forwarding
- **CFWD** –> incoming call redirected to another destination
- **CFWD2MOBILE** –> incoming call redirected to the mobile associated to the called extension
- **FORK2MOBILE** –> incoming call was forked to both the extension and the associated mobile number
- **FASTXFER2MOBILE** –> call in progress transferred from extension to associated mobile number
- **FASTXFER2EXTEN** –> call in progress transferred from mobile number to associated extension
- **BLINDXFER** –> call transferred without offer
- **ATXFER_START** –> start a transfer with offer
- **ATXFER_REFUSED** –> the transfer with offer to the destination is terminated because the destination has refused the transfer
- **ATXFER_BUSY** –> the transfer with offer is terminated because the destination is busy
- **ATXFER_UNAVAILABLE** –> the transfer with offer is terminated because the destination is not available
- **ATXFER_NOANSWER –>** the transfer with offer is terminated because the destination has not answered
- **ATXFER** –> call transferred with offer
- **UNAVAILABLE** –> the call is terminated because the destination is unavailable
- **CONGESTION** –> the call to the destination is terminated due to congestion
- **DECLINED** –> the call to the destination has been rejected due to a declined rule on the numbering plan
- **BLOCKED** –> the call to the destination has been blocked by the LCR because there are no lines to route the cal
- **FORBIDDEN_NOCLASS** –> the outgoing call was blocked by the LCR because a class to route the call is not defined
- **FORBIDDEN_NORULE** –> the outgoing call has been blocked by the LCR because no rule is defined to route the call
- **QUEUE_CALLBACK** –> a callback has been requested on a queue
- **CLOSED** –> the destination queue is closed due to time control

Provisioning Requests

Description

To view provisioning requests, you must follow the path "Logs > Provisioning Requests."

Within this section, provisioning requests that come from the phones are recorded.

You can see:

- Day of the month of the request
- Timestamp
- IP address
- Protocol
- Certificate identity
- User Agent
- Requested path
- Requesting MAC address
- Local path
- HTTP status code

Phonebook

Description

The phonebook allows the display of contacts that are in the "Local phonebook" and the "Shared phonebook". The phonebook can be used via the web interface or the CTI client, in this way you can see phonebook contacts and extensions. By logging in with a non-admin user, but associated with an extension, you can have management of a user phonebook, accessible only via the web interface and via the CTI client. Through CTI desktop, you can only make editing and add contacts to the personal phonebook, to add contacts to the shared phonebook, you have to go to the web interface.

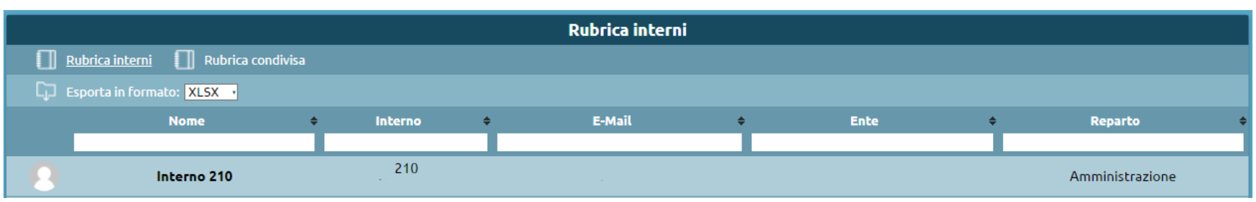
Configuration

Local phonebook

The local phonebook is populated by extensions that have a particular configuration: the "Show in local phonebook" parameter is active. In the extension settings there is a way to make the presence of the extension visible in the LDAP phonebook. "Organization" and "Organizational unit" can also be entered in the configuration to have these fields populated within the phonebook. In addition, the extension can have associated e-mail address and mobile number, see the Extensions configuration

Note: A non-administrator user does not see the mobile number of other extensions because the mobile number is meant to be a reachability number through the Fork2Mobile service and may be personal and not exclusively business.

The non-administrator user displays the phonebook list of extensions in the same way as the administrator user, but has an extra feature: when the mouse hovers over the extension number, "Click2Call" appears. When you click on the contact, the central unit starts a call to the extension and you have 10 seconds to answer. When you answer, the central unit starts a call to the destination number. You can export the phonebook in the following formats: XLSX, CSV, JSON, and XML.



Shared phonebook

The shared phonebook contains the contacts visible to all extensions in the pbx.

The shared phonebook is populated by the admin or other users - if delegated - and contains several tabs that can be added manually or imported from EXCEL files.

Through Add New Contact, a contact can be added manually by entering:

- First name
- Last Name
- Organization
- Organizational unit

One or more contact information can be associated with the card: the extension, home, work, cell phone, work cell phone, fax, or e-mail. Each contact number, except e-mail, can be associated with a Speed-Dial, a speed-dial code that the user can type in to initiate a call to a particular number without having to remember it and/or without knowing it.

Thus, the phonebook contains the contact numbers entered, which are not directly callable by the central unit.

- If you want to make the call to an external number, you must prefix the external line commitment prefix ("0").

The user can remove the "0" for external calls or decide to use a different prefix, this is changed in the "general settings" panel, via the dl box "Outgoing call prefix".

- If you want to make a call to an extension, by running the Click2Call - feature for the non-administrator user - the central unit calls the user and then triggers a call to the selected extension. For non-internal numbers, the system calls the user and, when answered, starts a call to the selected number by prefixing "0".

On the CTI client, in the phonebook, the contact details are marked with different icons. Adding a new card - thus a new contact - is not immediately visible within the CTI client, because the phonebook is updated periodically every 10 minutes. If you want to force the update due to urgencies, you can log out and then log back in, so the phonebook import is done automatically when the client starts up.

Export and import

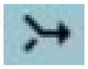
The export of the shared phonebook can be done via XLSX, CSV, JSON and XML. Exporting it in EXCEL (XLSX) format is helpful because the downloaded file will be the same one you can use to do a bulk import. On excel, the contact information for a single card is represented on different rows.

Note: When populating the contactValue column, enter a formation of type "Text" and not "General," otherwise the "0" is not included if you enter a number with that initial value.

The option Phonebook allows you to:

- Replace the current phonebook with the imported one: if this option is not checked, the existing phonebook is integrated with the new one
- Choose whether speed dials in the file include the prefix (of the numbering plan).

If you perform the import by not replacing the current phonebook with the imported one, the warning field highlights the operation with the icon:



Note: If there are contacts in the phonebook that have Speed-Dial, when adding new contacts via import, you must put any Speed-Dial without prefix.

Depending on the export format, the fields vary in nomenclature:

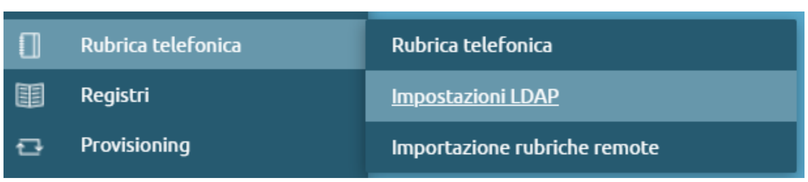
XLSX e CSV	JSON e XML
FirstName	firstName
lastName	lastName
organization	organization
organizationalUnit	organizationalUnit
contactType	typeString
contactValue	
speedDial	speedDial

Delegate the ability to manage in "write" the Shared phonebook

It may be necessary for some non-administrator users to be able to edit the shared phonebook. It is possible through the role mechanism to assign a user permission to make changes to contacts in the shared phonbook. Via the "User and Role Management" panel, in the Role Management panel, press on "Add new role." Priority and description are entered, and in the "Manage Shared phonebook" action, "Write" is checked.

LDAP settings

To reach the LDAP settings panel, click on "Phonebook > LDAP Settings"



You can make use of the phonebook via LDAP clients. You can make the phonebook available to external LDAP clients: from the phone, via keypad, you can search the central office phonebook. You can also access remote phonebooks exposed by other LDAP servers to make them available to Kalliope users. LDAP (Lightweight Directory Access Protocol) is a protocol that organizes data in the form of a tree hierarchy. The leaves (object class) of the tree are the objects that contain the attributes (first name, last name, email, phone numbers), so each contact record can be considered a leaf of the LDAP tree. A lot of information can be held in LDAP, so when defining the leaf type, you have to provide the object class representing what attributes can be obtained from these objects.

The organization of information in the LDAP tree is as follows:

root: dc=root

Subtrees: dc=default (where default is the domain name of the default mono tenant)

Instead, in a multi tenant node, every time a new tenant is created, a new subtree comes up with dc=domain name For each tenant, there are two subtrees:

- dc=users
- dc=phonebook

The latter contains other subtrees

- dc=extension (contains extension phonebook contacts)
- dc=system (contains the phonebook of system contacts)

When the user logs on to the phonebook, they get visibility of everything under the phonebook subtree of their domain.

In the panel, there are enablers for the two subtrees. In the central unit, there is an LDAP server on which you can enable the publication of contacts separately between the internal and shared phonebooks. On LDAP, exposing contacts from the personal phonebook isn't possible.

System contacts

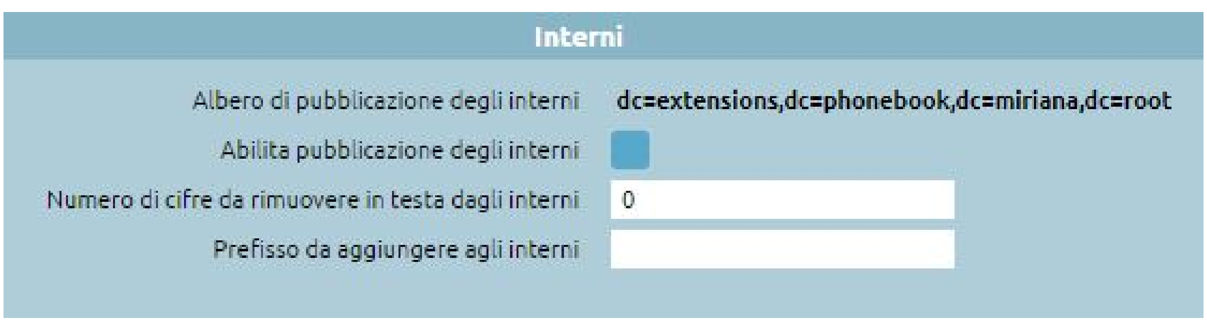
Configure system contacts unlocks the publication on the dc=system subtree of contacts in the shared phonebook:

- System contacts publication tree: the tree is displayed.
- Enable publication of system contacts: publication of contacts in the shared phonebook is enabled.
- Add prefix for outgoing calls: when the central unit publishes contacts to LDAP, it adds a prefix "0" to all numbers, except those marked as extensions, to make outgoing calls. In this way, the client accessing the phonebook, finds the numbers already ready (including the "0") to be called

Extensions

You can enable the export of the extension directory to LDAP as a global function.

- Extensions publication tree: the tree is displayed.
- Extensions publish enabled: the phonebook of extensions can be found searchable via LDAP.
- Extension prefix strip: standard extension publishing rule that is used in case you want to display extension numbers not with the extension number, but with the full geographic number
- Extension prefix prepend

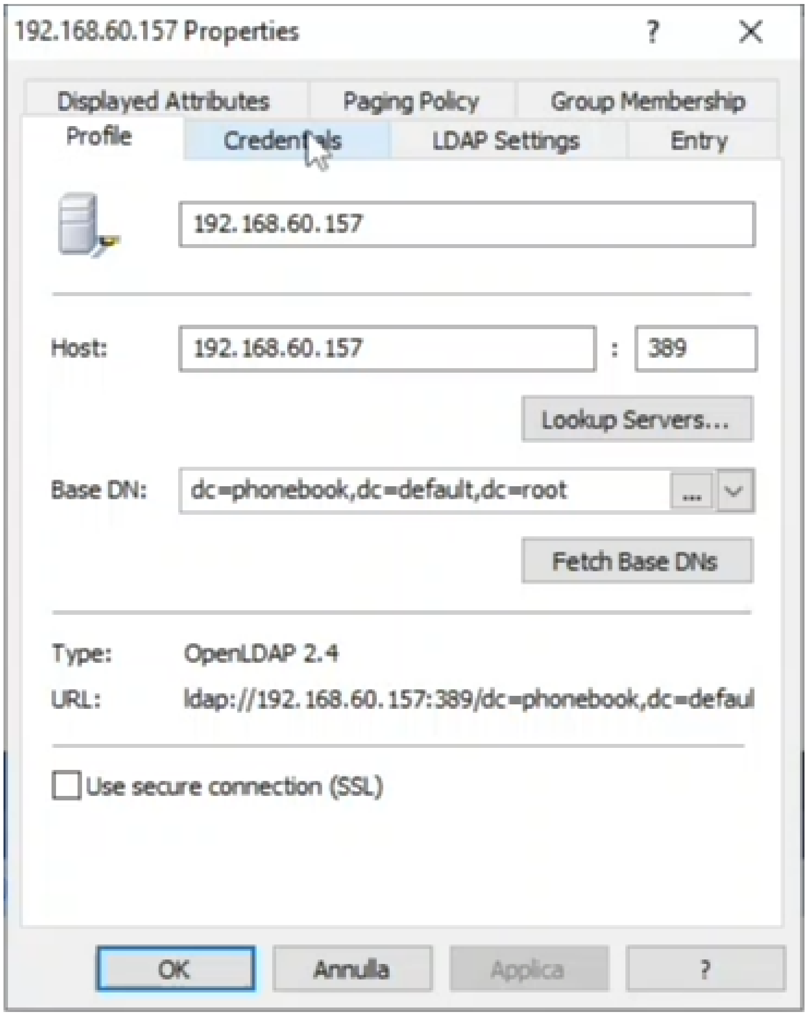


Saving the configuration populates the LDAP tree with the contents of the system phonebook and extensions. You must authenticate with a user who has the right to see the phonebook to access the LDAP phonebook. An LDAP client can be used to consult it.

LDAP client configuration

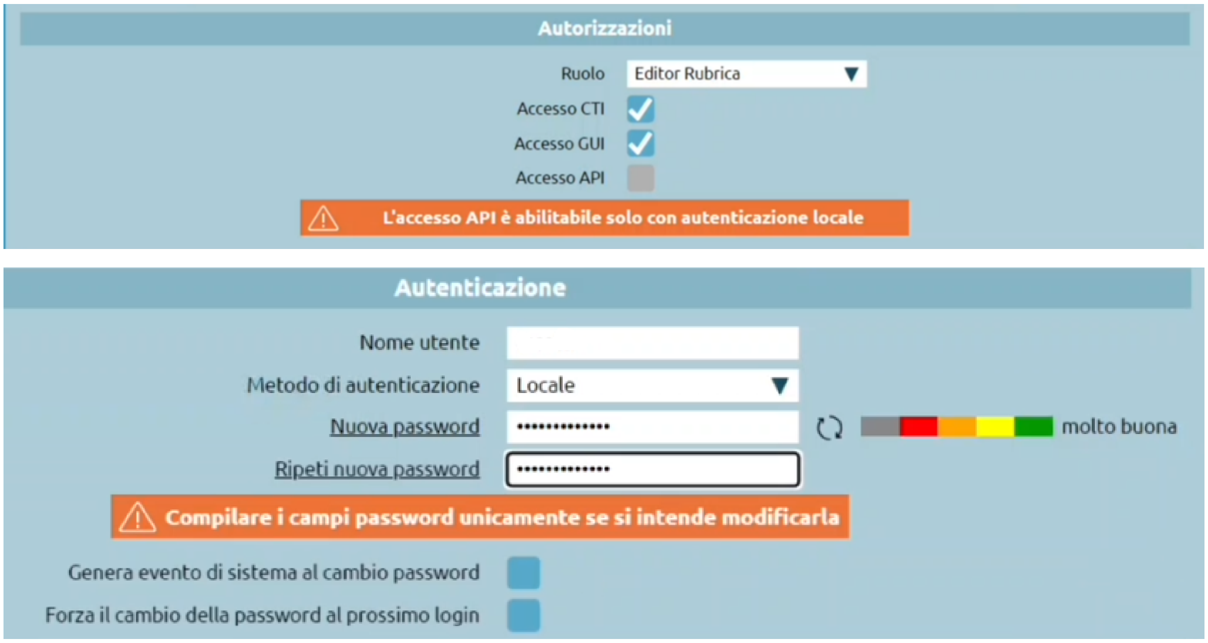
- IP address of the server and port
- Base DN: root of the tree that you want to consult
- Login credentials to be specified: username and password of user authorized to read the LDAP tree

Kalliope users, defined within the "User and Role Management" panel, are by default authorized to access LDAP. So in the credentials to be specified you enter the user via a specific syntax: cn=user@domain, dc=users, dc=domain, dc=root



Then, automatically, the client will make the connection to LDAP.

Note: **Authentication of users**: To gain access via LDAP, users must have a local authentication method; users with domain authentication cannot access LDAP.



Note: The PBX, not knowing the user passwords, could not write the passwords of each user to the phones. Therefore, using personal credentials to access the phonebook is never performed. Within the PBX, there is a default user, phonebook, who has the right to access and edit the shared phonebook; it is a disabled entity by default, but it can be enabled once a password is assigned. Enabling GUI access (to consult the phonebook and make changes from the web) is required; API access is unnecessary.

Once the connection is made, if you look at the dc=extension subtree, you will see that it is empty. It is a domain, but it has no leaves. It is empty because it is not enough to apply the extension publication in the LDAP settings. Still, each extension has a flag that checks whether it should be published to the internal phonebook and/or LDAP phonebook. In the template default extension, the "LDAP publishing mode" must be enabled. There are also ways to make the extension's presence visible on the LDAP phonebook.

- **According to LDAP publishing rule**: exposes the extension number with a manipulation to convert it to a geographic number.
- **Presenting the telephone number below**: exposes the extension number with a specified number
- **According to the LDAP publication rule applied to the extension below**: assigns a dummy extension and the masking specified in the LDAP panel is applied

If the *dc=system* subtree has leaves, the system contact directory has been correctly populated with all the attributes of the tab.

- givenName
- sn: surname
- cn: common name, is constructed by doing concatenation of first name (givenName) and last name (surname)
- o: organization
- ou: organizion unit

Objects collected and described under an LDAP tree are characterized by a chain of object classes. Each object class has a set of mandatory or optional attributes. If it activates a filter on an object class, an LDAP client that wants to read the phonebook can decide how to perform a search.

Mapping of contact types in the shared phonebook in LDAP

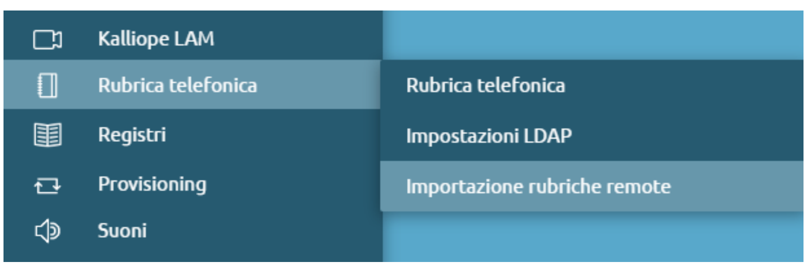
XML attribute	GUI attribute	LDAP attribute	Presence
firstName	Name	givenName	optional
lastName	Surname	sn	required
		cn	required
organization	Organization	o	optional
organizationalUnit	Organizationalunit	ou	optional
Home	Home phone	homePhone	optional
Work	Work phone	telephoneNumber	optional
Mobile	Mobile phone	mobile	optional
Work Mobile	Work mobile	mobile	optional
Fax	Fax	facsimileTelephoneNumber	optional
Extension	Extension	telephoneNumber	optional
Email	E-mail	mail	optional

Warning: All contacts to be published in LDAP must have at least the last name.

Warning: LDAP clients on phones do not always support all object class attributes. Each phone has a set of attributes it can search and return to the client, via display, only the attributes it supports.

Importing remote phonebooks

The import of remote phonebooks can be reached by clicking on "Phonebook > Import Remote Phonebooks."

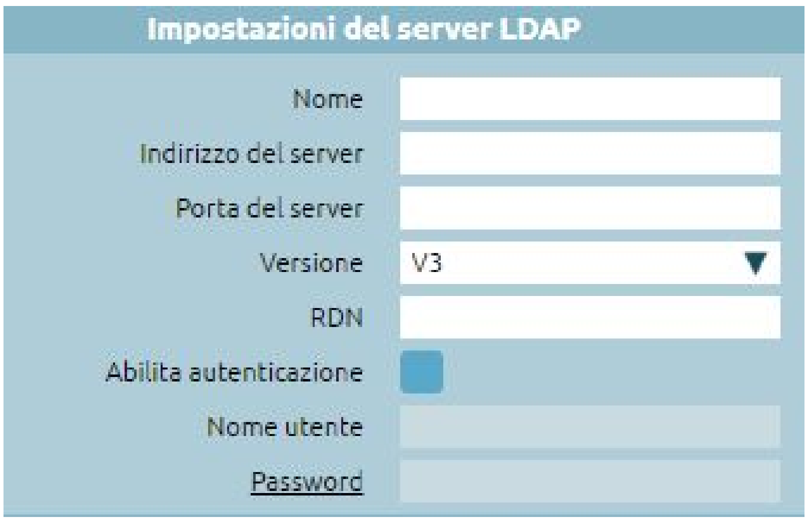


This section offers the possibility of importing contacts in LDAP phonebooks into the Kalliope central unit. A real-time query is not done every time there is a need to consult the phonebook, but a complete import of the contacts in the phonebook is done and entered inside the Kalliope internal database. This import is done periodically. It is possible to define more than one.

Pressing on "Add a new LDAP remote phonebook" will take you to the remote phonebook creation page.

LDAP server settings

- Name
- Host address
- Host port
- Version: V3 / V2 (obsolete)
- RDN: root of the subtree you want to query.
- Authentication enabled
- Username
- Password

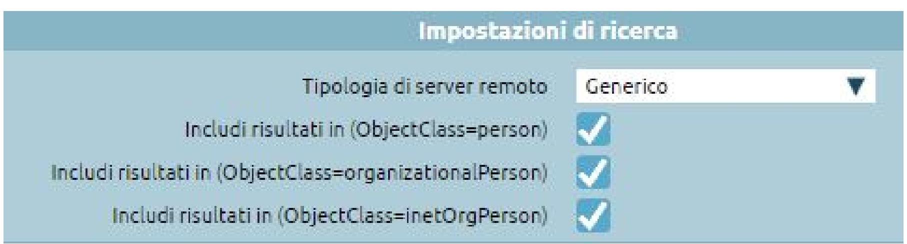


Search settings

You can filter only contacts that have as ObjectClass person, organizationalPerson, inetOrgPerson

- Provider type: Generic / estos MetaDirectory
- Include results in (ObjectClass=person)
- Include results in (ObjectClass=organizationalPerson)
- Include results in (ObjectClass=inetOrgPerson)

Access via SSL is not provided. Metadirectory is software that acts as a phonebook aggregator: it takes data from different sources (LDAP servers, text files, databases), reads it and republishes it in LDAP form to be read by LDAP clients..



Import Settings

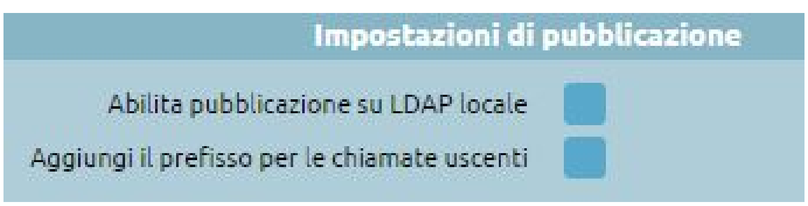
- Import enabled
- Import periodicity: repeated every (tot hours)/repeated every day at the
- Incoming call phonebook loopup enabled: contacts in this phonebook are used to set first and last name based on the calling number
- Export to cti enabled: CTI clients receive the contacts in the LDAP phonebook from the central office when importing
- Show in GUI enabled: the client accessing the interface sees the contacts that are on the remote phonebook in the phonebook
- Import contacts as extensions: phone numbers found in the remote phonebook are entered into the local import database as extensions (extensions). When you then go to show them to clients, they will be treated as internal numbers. Any click2call will occur to the exact number, without putting the "0" in.

Import settings

- Enable publishing to local LDAP: if you want to export the contacts that I imported, make them accessible to phones. This function is helpful for aggregating: this way the central unit presents both the contacts in the shared phonebook and the remote phonebooks.
- Add outbound prefix: as in publishing system contacts, you can add the "0" to contact these numbers as if they were external.

If you check the above option "import contacts as internal," the "0" is omitted because they are reached with the short number.

Contacts in the remote phonebook are visible in the shared phonebook to the administrator; non-administrator users can only see them in the shared phonebook if the "Enable display in GUI" flag is present. With each incoming call, the central unit consults whether the calling number is in a phonebook. If so, it goes to edit the call display name (caller name) to insert the value present in the phonebook and corresponding to that caller number. In the case of direct calls to an extension, the personal phonebook is used as a priority, while for calls to groups or queues, the shared phonebook is used.



On Call Services

This section of the Kalliope interface lets you configure codes for on-call services.

The only service for which it is not possible to set a custom code is canceling an attended transfer, which by default is *0.

The codes for the following services are freely customizable.

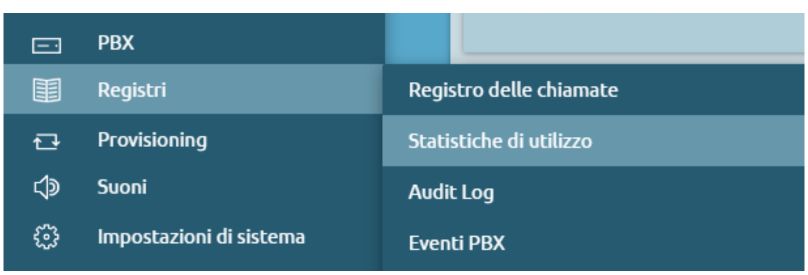
Service	Default
Attended call transfer	*4 Transfer, *0 Cancel transfer, *9 Shuttle, *3 Convert to a three-way conference
Blind call transfer	#4
Call parking	#8
Fast transfer	**
Call recording on demand	*1
Call hang-up code	*0

Usage Statistics

Warning: Usage statistics are available only if the multitenant license is active.

Description

The service can be reached through “Logs > Usage Statistics”.



The usage statistics are a useful tool for viewing the yearly and/or monthly trend of PBX center utilization. The page shows how occupancy is growing at the PBX level.

Tenants defined with the Name, Domain, and UUID are displayed. The first number represents the number of extensions configured on the Tenant, the second is the number of FAX instances, and the third is the number of hotel rooms that have been created.

The display can also be scaled to the current month and the previous month with the display of individual days of the month.

You can also export data in the following formats:

- XLSX
- CSV
- JSON
- XML

SNMP Support

Description

Protocol Operation

SNMP is the standard for network management and monitoring. Thus, it is a protocol used to monitor the state of a machine, specifically, to acquire from a monitoring server external to Kalliope status parameters related to CPU load, memory occupancy, disk space, and concurrent calls. It is a standard protocol available on Kalliope; OIDs (monitored objects) are made available on the central. Those defined in the standard MIB 2 are used, where object identifiers are made available through the agent. The server is the monitoring system that queries, while the SNMP agent is the service used to expose the data to a client that requests it. SNMP involves the client requesting the agent to find out the value of the OID, and the agent returns the value.

The data within the agent is organized in the form of a tree. The data indicate, for example, the version of the primary and secondary firmware, information regarding telephone services, the size of virtual memory occupied by the process, and the total number of failed authentication attempts by SIP clients.

Objects are defined as an index of a leaf under a parent: a tree structure allows you to access by specifying the path address between the subtrees to the leaf, which is the object you want to monitor. SNMP provides:

- Manager
- Agent
- Protocol

The representation of an OIP is done as a sequence of numbers, e.g., 1.3.6.1.2.1.4.6 is the path within the tree, and each point in the path has its textual correspondence.

Configuration

To enable NSMP support, go to “System Settings > SNMP Settings” and hit the “Enabled” checkmark.

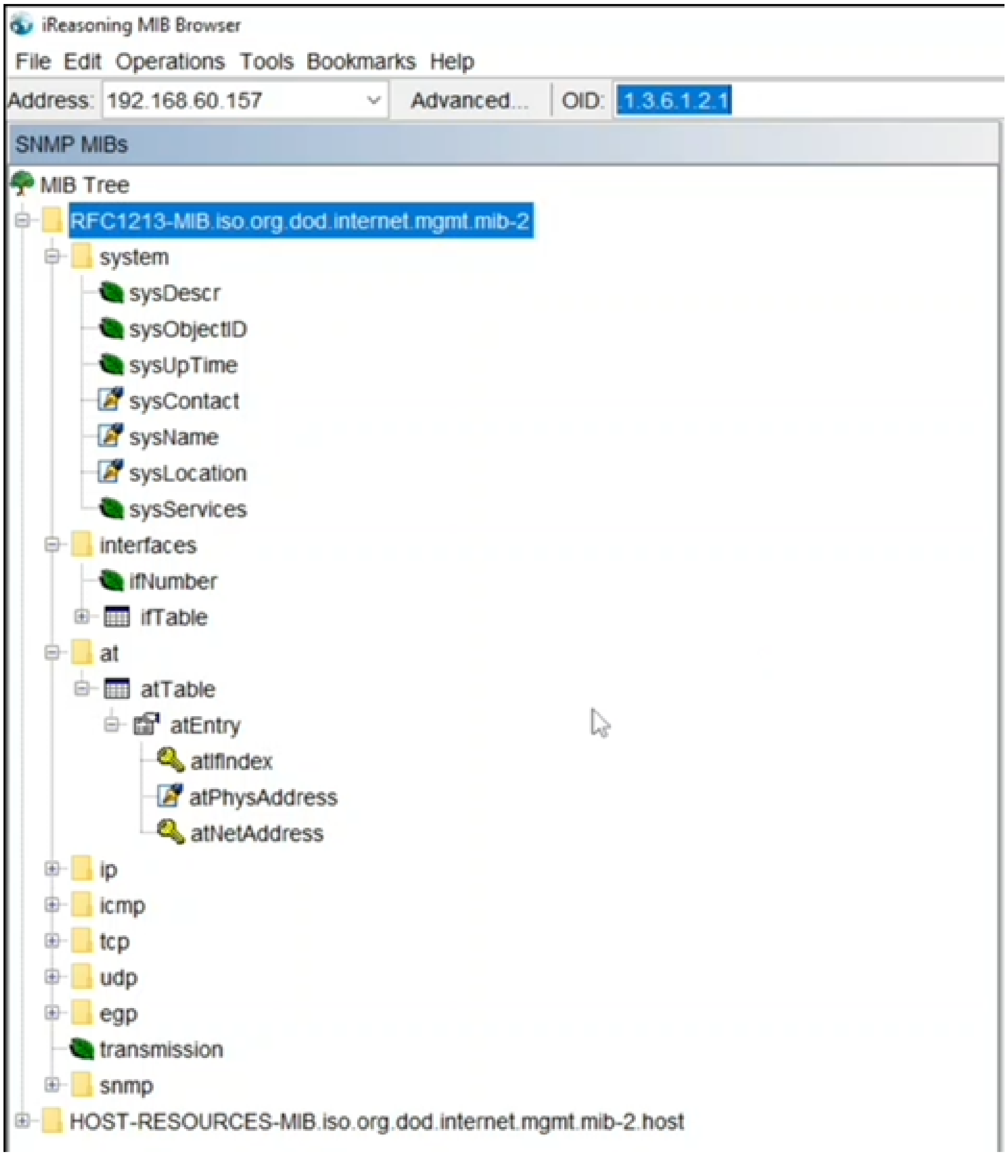
System info

The following information is mandatory; if it is not entered, the SNMP service cannot start.

- **sysName**: is a particular OID under the system tree (which is the first child and is denoted as 1.3.1.2.1.1), sysName is the machine name and is 1.3.6.1.2.1.5.0 (the 0 at the bottom since it is scalar type).
- **sysLocation**: is the physical location, it acts as a kind of inventory
- **sysContact**: email address of the contact person to contact if there is an anomaly in that particular node

In the defined settings, MIB tree 2 and host-resources (1.3.6.1.25) are the two that are exposed on Kalliope. Proprietary MIBs are also exposed and the software can read the whole tree since it scans it entirely, and it is the agent's job to return the values. The MIBs file allows the software monitoring to know the single leaf that only knows by number what it is.

The iReasoning MI Browser client was used for the configuration examples, it can make SNMP queries and shows the subtree organization.



SNMP Access Settings

- **Listening address**: e.g., by default 0.0.0.0 means it listens on all interfaces of the central unit. But, if we have a PBX with multiple network interfaces (some public and some private) and you want the SNMP service to be accessible from only one of these interfaces, you can put the IP address of the interface you want the service to be active. Then you can do a bind of the service on the IP entered; this must be one of the central unit's IPs (either one of its interfaces or that of the High Reliability in the case of a cluster). In the last case, in the case of HA, explicit monitoring should be done not pointing to the resource IP but the individual IPs of the two nodes.
- **Listen port**: 161 is the standard port since SNMP uses the UDP transport protocol.
- **Community v1/v2**: these are the basic versions of SNMP supported by monitoring systems, SNMP v3 support is not yet enabled. The default value that is used is "public"
- **ACL**: Access Control List, is a restriction on which IP address the client must have for querying. If an SNMP request comes in outside the ACL, it is rejected.

Warning: ACL 0.0.0.0/0 is not recommended, it is always good to restrict access only to authorized IPs

TRAP settings

Traps are a reactive-type mechanism that SNMP agents have for notifying the occurrence of events. Kalliope has not added specifications to the system, and basic ones are present:

- 0: ColdStart
- 1: WarmStart
- 2: linkDOwn
- 3: linkUp
- 4: authenticationFailure
- 5: egpNeighbortLoss
- 6: enterpreseSpecific

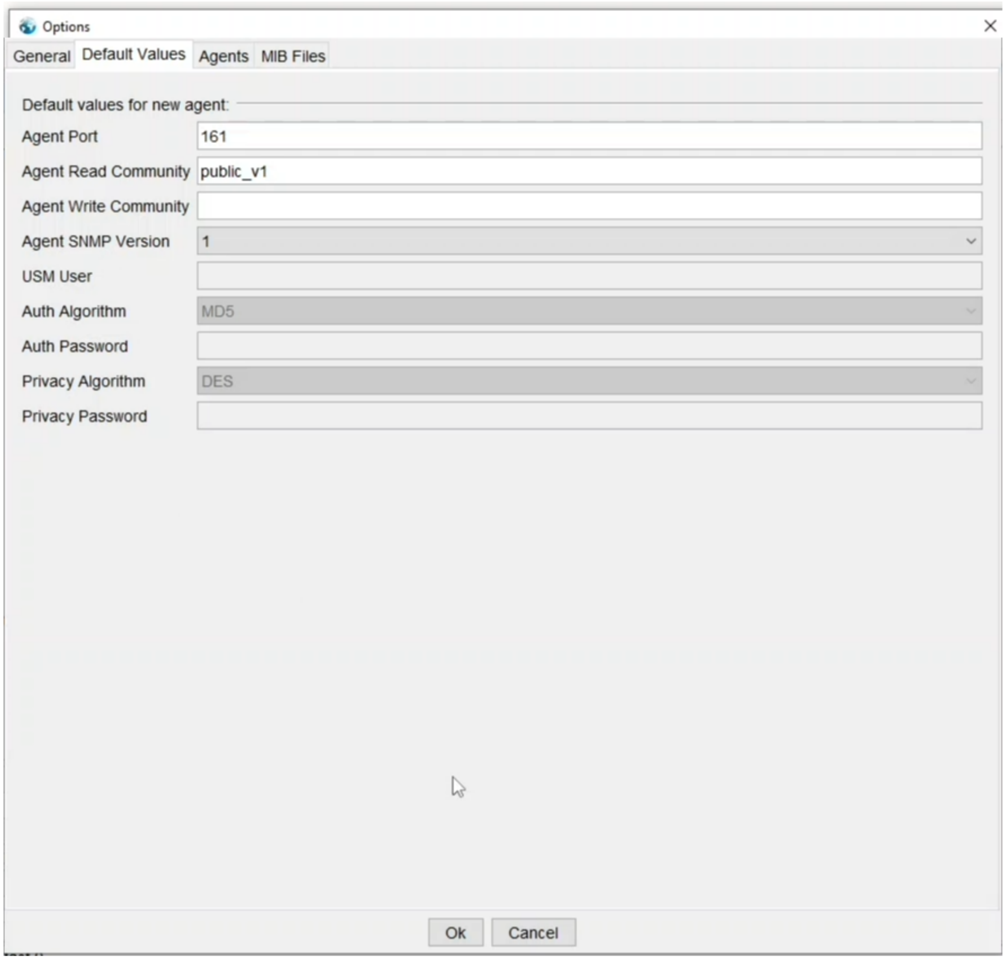
Directions on the configuration of TRAPs:

- TRAP send mode: specify whether you want to send the trap in version 1 or 2
- TRAP destination address: indicate IP address of the monitoring server to send TRAPs to
- TRAP destination port: e.g. 162 is the standard one

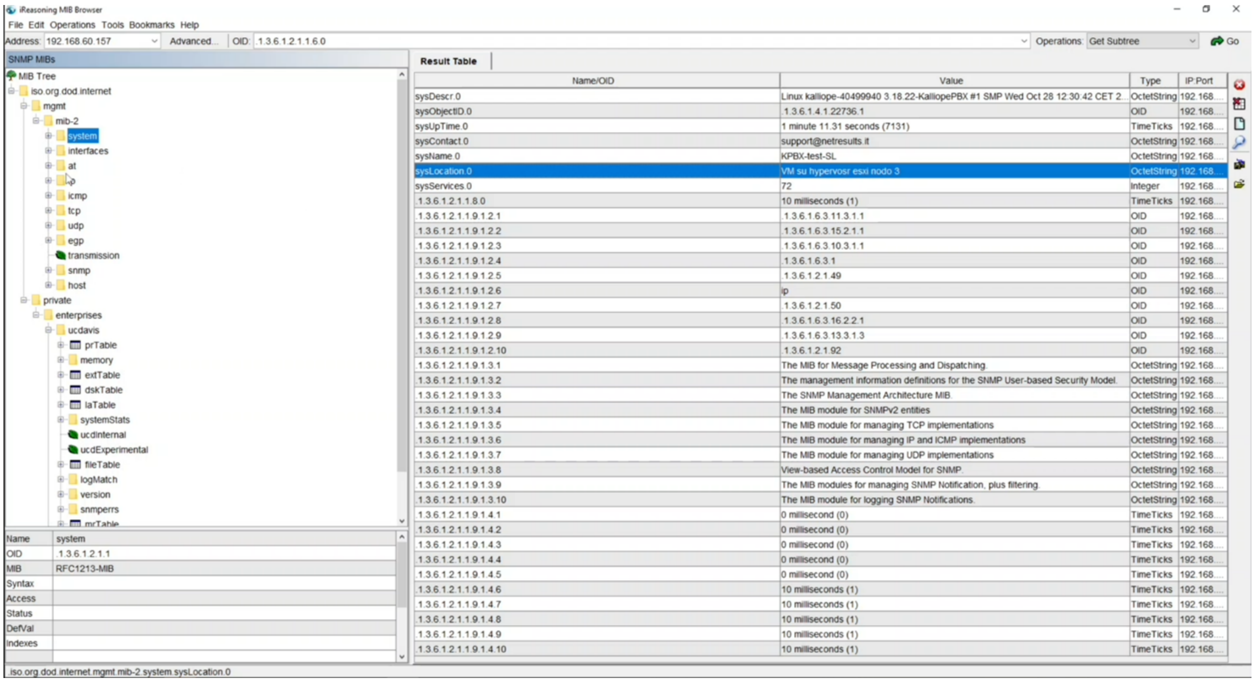
On the dashboard, the execution of the SNMP service is indicated by the green dot and the "Active" status.

You can do an agent query on Kalliope via any SNMP client (in this case you can use the iReasoning MIB Browser). You need to indicate the IP address to connect to and then set up:

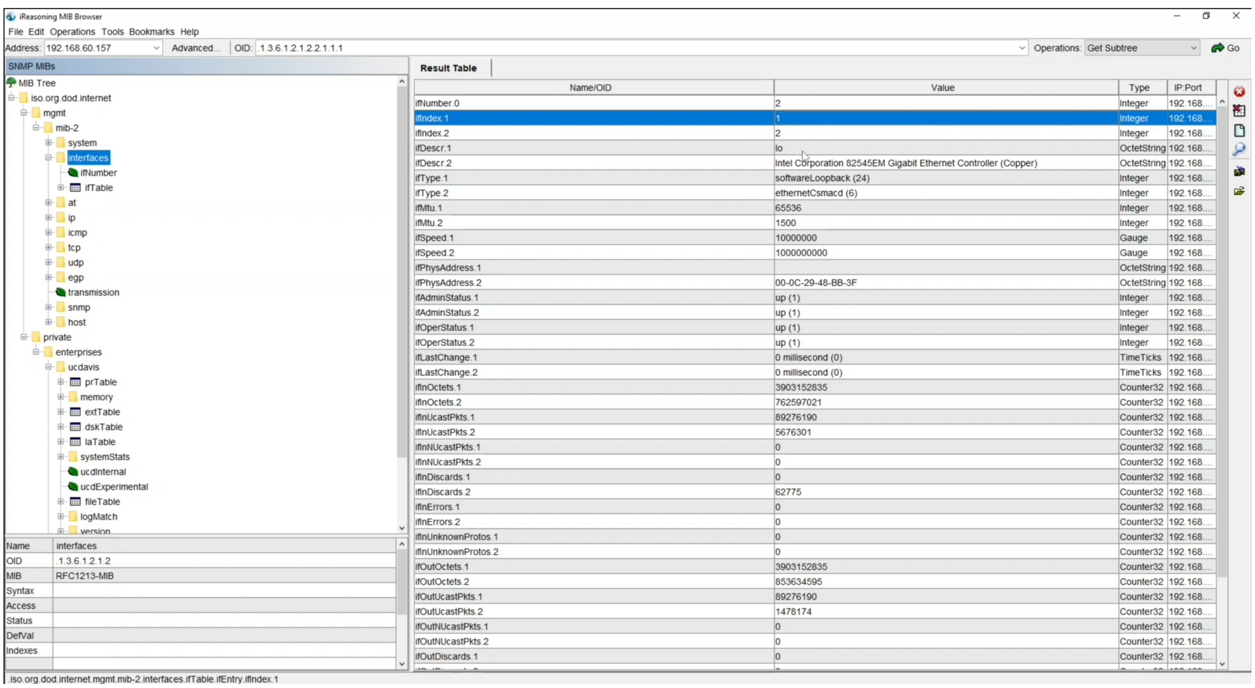
- Agent NSMP Version: the default values with which to make the request, i.e. you can choose whether to make it in version 1 or 2
- Agent Read Community: what is the read community in order to get access to the data
- Agent Port: you can indicate the standard port, 161
- Agent Write Community: the ability to do write via SNMP is not enabled



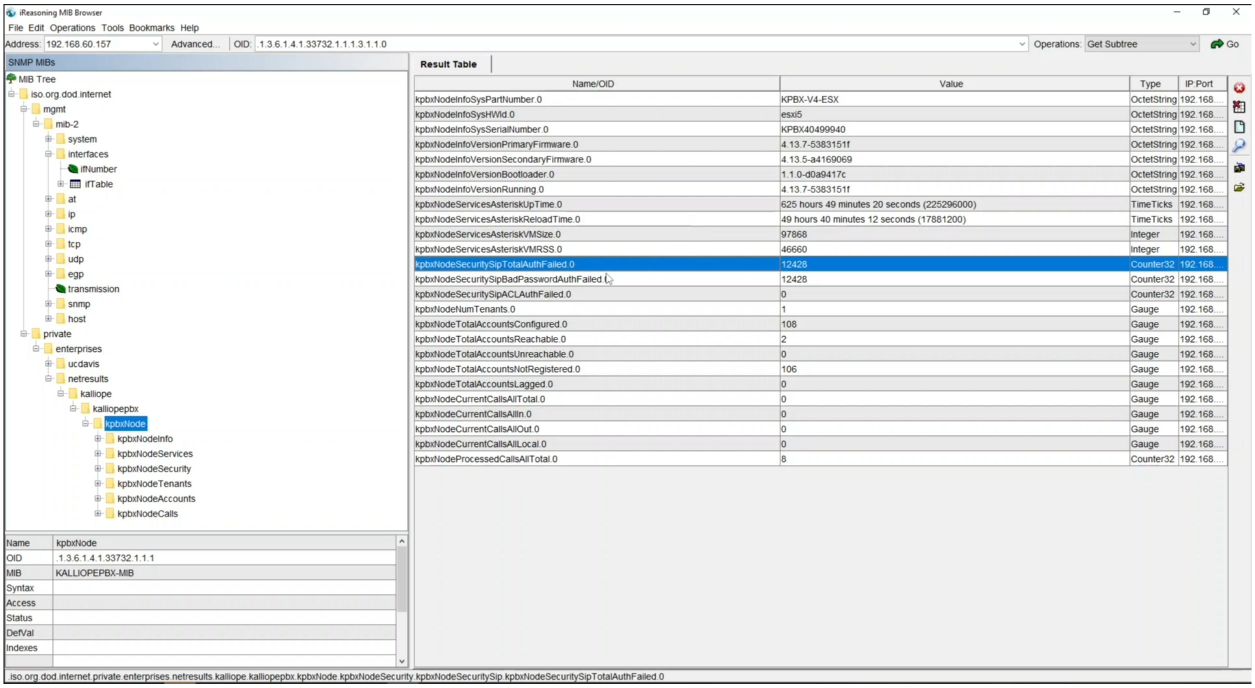
After making the query, it is possible to read the contents of the tree that is present on Kalliope. The reading can be done on an object-by-object basis: The subtree system present the following information including the sysContact, sysName and sysLocation.



The interfaces panel returns the various interfaces present and for each indicates the operational status and bytes exchanged in and out, this allows monitoring systems to show the occupancy graph:



kpbxNode example:



Note: The counter of failed authentications (highlighted in blue) is essential data because if a burst of failed authentications comes in, it is likely to be an attack from outside.

KalliopePBX implements MIBs that allow you to monitor the functioning of the equipment through the SNMP standard communication protocol. The MIBs that can be consulted are:

- RFC1213-MIB - This MIB defines objects for managing and monitoring an entity in a TCP/IP network. On KalliopePBX the following subtrees are implemented: system, interfaces, at, ip, icmp, tcp, udp, transmission, snmp OID: 1.3.6.1.2.1.1/2/3/4/5/6/7/10/11

https://datatracker.ietf.org/doc/rfc1213/

- RFC 2790 Host Resources MIB - This MIB defines a set of objects containing the configuration of hosts (servers/computers) connected to a TCP/IP network independently of the operating system, the network services, and the installed software. OID: 1.3.6.1.2.1.25

https://datatracker.ietf.org/doc/rfc2790/

- UCD-SNMP-MIB - This MIB defines objects for monitoring the performance of a host (e.g. CPU /RAM / disk occupation). OID: 1.3.6.1.4.1.2021

http://www.net-snmp.org/mibs/UCD-SNMP-MIB.txt

- Kalliope MIB - This proprietary MIB provides information on the configuration and the functioning of the services implemented on the Kalliope node, such as number of configured or registered accounts, simultaneous calls, total number of calls. OID:1.3.6.1.4.1.33732

Download the MIB definitions files:

`MIB definitions`

Recovery Mode

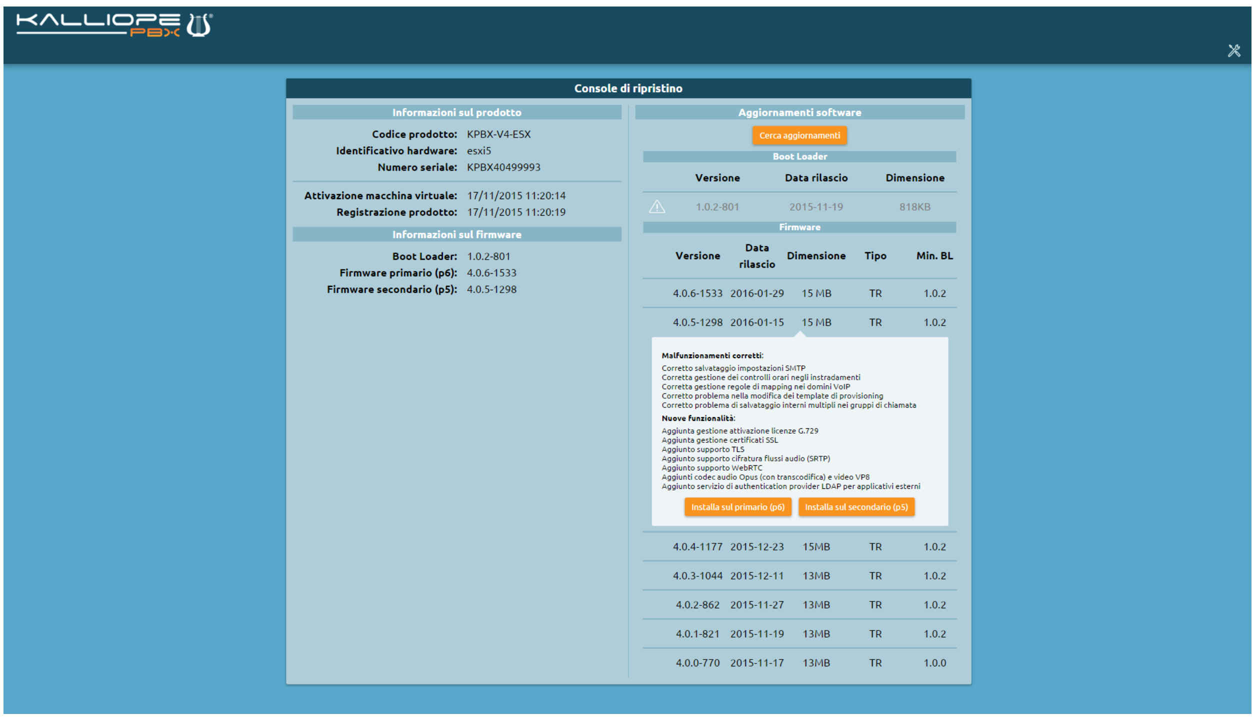
You can access the KalliopePBX recovery console from the resource IP address (192.168.0.100 by default) on port 10080 in HTTP.

The recovery console displays information on the product and the firmware, and lets you:

- register the product (this serves as a timestamp for the beginning of the warranty and free firmware updates);
- activate a virtual machine using an activation key (only for VM);
- update the bootloader;
- reset to factory default by installing firmware on a partition.

Clicking on each release will display new features and bug fixes. You can also choose on which partition to install the selected firmware.

> **Warning:** Installing firmware from the recovery console will erase all information on the partition. Firmware installation should be performed through the Kalliope configuration GUI and not the recovery console.



Firmware Update

From the system menu, you can access the Update management section. Just as with the recovery console, this section displays information on the product and the currently installed firmware.

By clicking on Find updates, you can access the list of available updates, both for the bootloader (which can only be updated from the recovery console) and for the firmware. Clicking on specific releases will display the changelog, which lists new features and bug fixes.

Updates that cannot be installed (because they have already been installed or because they require an extra step) are grayed out and shown next to an alert icon. Updates that are available for installation will instead appear in a black font.

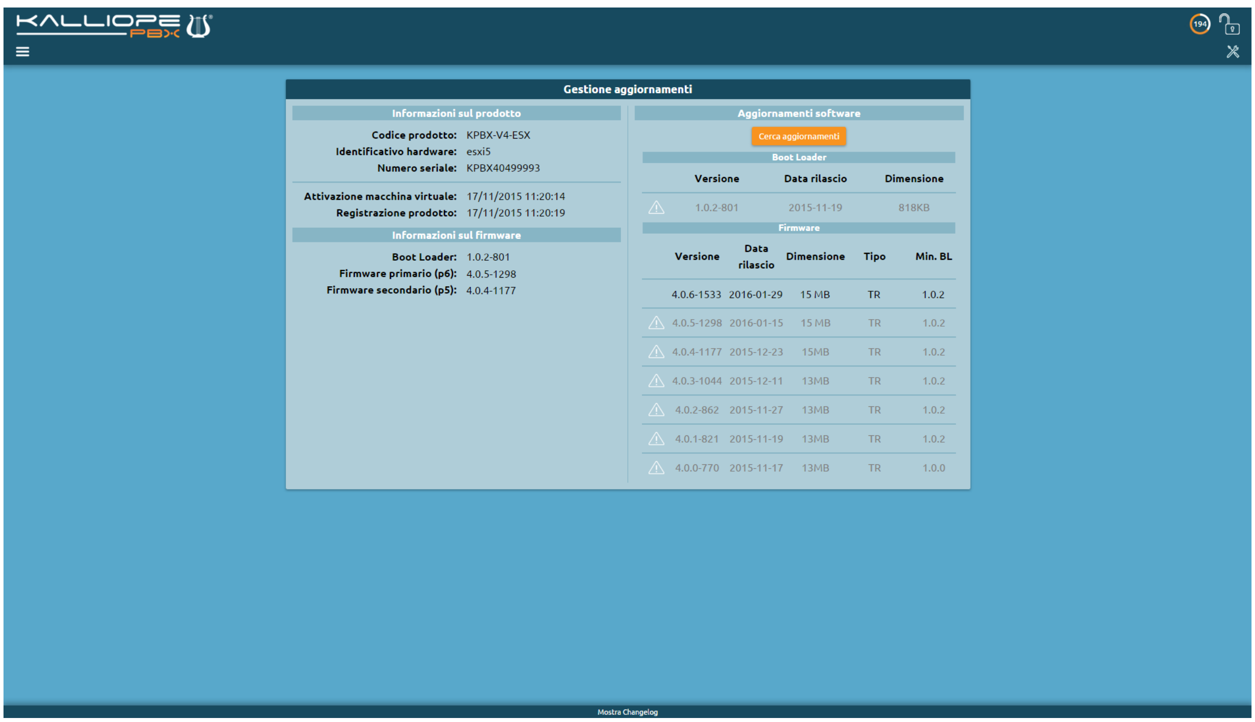
Clicking on an available update will begin the download, extraction, and installation process on the secondary partition.

KalliopePBX has two partitions dedicated to saving firmware: one for the primary firmware, and one for the secondary firmware.

Under normal conditions, the main partition will contain the firmware currently in use. During an update, the new firmware will be installed on the secondary partition. Once the update is finished, you will be presented with the option of rebooting the PBX using the secondary firmware. By logging in as admin, a banner will appear on the configuration GUI informing you that the firmware in use is the secondary one. You can then click on Make primary to promote the firmware version and update the PBX (you will first need to acquire the lock).

You can return to a previously installed firmware version by rebooting on the secondary firmware from the shutdown menu.

Warning: Since the configuration of the PBX is saved in the same partition as the firmware, restoring the secondary firmware will also restore the corresponding settings.



Privacy Admin

Special user privacyadmin

The privacyadmin user has access permissions that are completely independent from those of the system admin in order to completely preserve the privacy of sensitive information contained on KalliopePBX.

When accessing the KalliopePBX web GUI with privacyadmin credentials, the operating menu will only show the following items:

- Phonebook
- Call detail record
- System settings
- Call recording

The call detail record displays all calls made and received by KalliopePBX, with complete details and unobscured numbers. The normal admin cannot view this information on the CDR in any way, as all numbers will have the last 3 digits obscured as a way to safeguard the privacy of the end user.

From the System settings the privacyadmin can delegate their "powers" (i.e. permissions to access private information) to standard PBX users. Only the privacyadmin can delegate other users and edit access permissions to private information.

The call recording service is only available to the privacyadmin and the users delegated by them (and only them). This service is comprised of two pages:

- View log (of recorded calls)
- Edit settings (of the recording service)

Viewing the call log

On the log of recorded calls, the privacyadmin (and any users delegated by them) can view the list of recorded calls, use the advanced search function, listen to recordings directly from their browser(by clicking on Play symbol) or download them to their computer (by clicking on Download symbol).

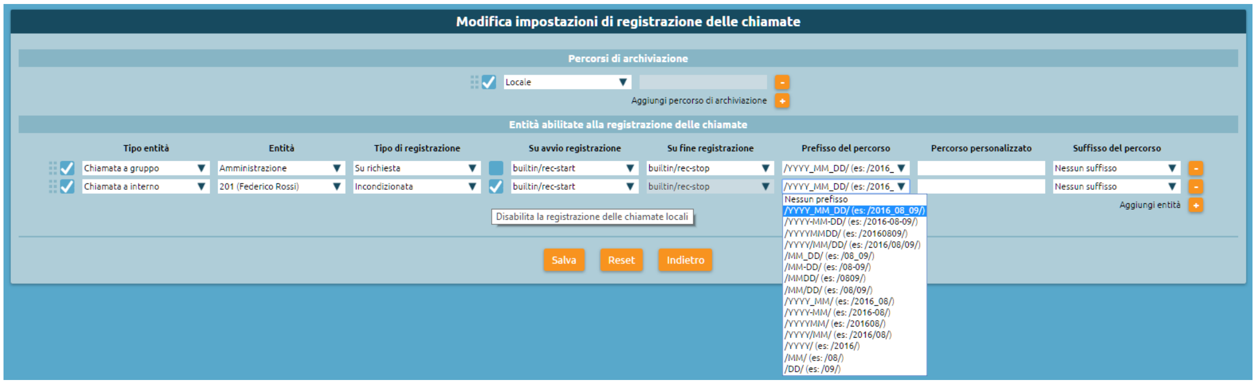
Every month, a new tab will be automatically created for ease of viewing. The calls are shown by default from most to least recent, and list the following information:

- UniqueID, a unique call identifier
- Date and time of the beginning and the end of the recording
- Direction, i.e. which conversation flow was recorded (in / out / both / mix)
- Recording type (unconditional / on demand)
- Archiving volume (local / remote storage)
- Archiving path

- Requester
- Destination type
- Destination name

Call recording service settings

On this page you can configure the entities that regulate the behavior of the call recording service. First of all, you can set one or more paths with which to save recordings. Aside from local storage, you can also save recordings remotely (if appropriately configured). In the Entities enabled to record calls field you can set the desired call recording rules.



The following table lists the configurable parameters for each recording entity (from left to right).

Parameter	Description	Value
Enabled	Enable or disable a recording entity	On/Off
Entity type	Type of call for which to activate recording.	Call to group / Call to queue / Call to extension / Call from extension
Entity	Depends on the previous field. Shows the identity involved in the recording rule.	Group name / Queue name / Extension
Type of recording	Shows whether calls should be recorded automatically or on demand, with the option of starting a recording in the middle of a call. In this case, the recording will include the conversation that occurred between the moments the command to begin and to end recording were given (code defined and editable in PBX menu > On-call services).	Unconditional / On demand
Enable/disable service for local calls	Enable or disable this recording entity for calls between two KalliopePBX extensions.	On/Off
On start of recording	Choose the file to play in order to signal the beginning of the recording (from those available on KalliopePBX through the audio file settings).	Path to the audio file from those available
On end of recording	Choose the file to play in order to signal the end of the recording (from those available on KalliopePBX through the audio file settings).	Path to the audio file from those available
Path prefix	Lets you specify the prefix of the path for saving the recording file among those available (see above image for examples). This can be left empty.	No prefix / One of the values from the dropdown menu
Custom path	Lets you specify a custom path for saving recording files to append to the prefix specified in the previous field (if any) for ease of cataloging.	Custom path text
Path suffix	Lets you specify the suffix of the path for saving the recording file among those available (see above image for examples). This can be left empty. The suffix will be appended to the specified prefix and custom path (if any). The full path will therefore be prefix + custom path +suffix.	No suffix / One of the values from the dropdown menu

You can create an arbitrary number of recording entities, offering you maximum flexibility in defining access criteria to this service.

Enable privacy access to Kalliope users

As previously mentioned, the privacyadmin can delegate other users to access private information and consult and configure recordings.

The privacyadmin can enable access for Kalliope users from the Users management page by selecting the checkbox next to the desired user and clicking on Enable privacy access, as shown in the image below. Similarly, they can disable access for a previously enabled user.

Once privacy access has been enabled, all the user has to do is log into Kalliope with their own credentials and they will be able to see the items relative to privacy directly from the operating menu.

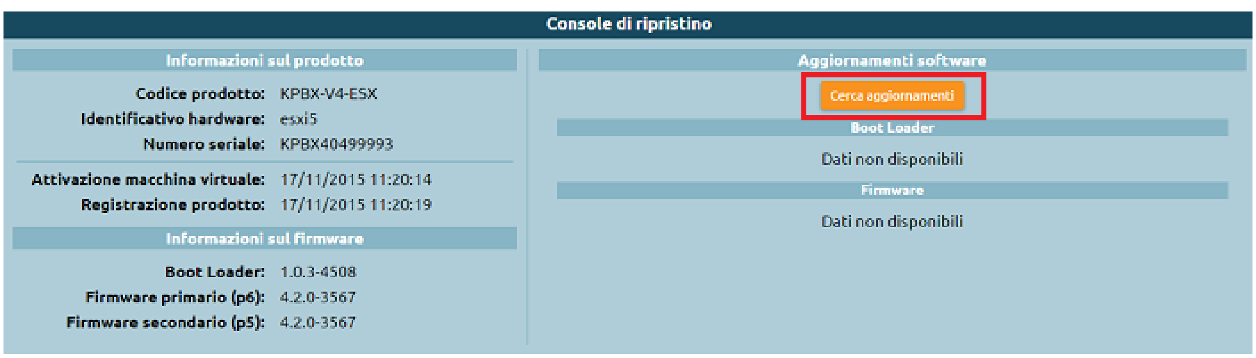
3.1.3 Procedures

How to update the bootloader

To update the bootloader, begin by restarting the PBX from the recovery console.

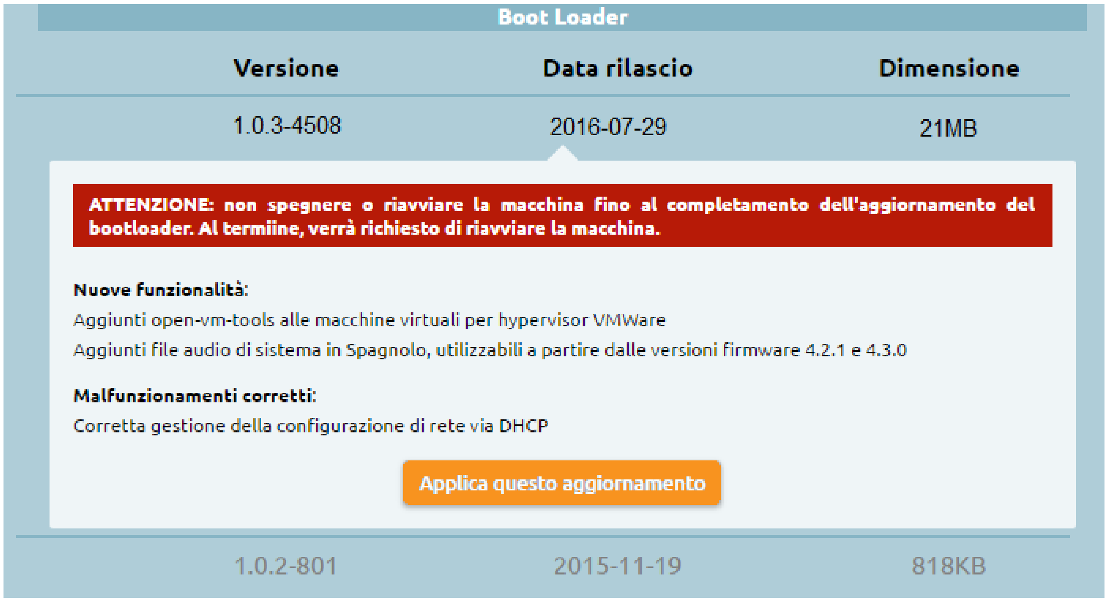
Once the PBX has been booted in recovery mode, you can find available updates.

> **Warning:** Only the bootloader can be updated from the recovery console. In order to update the firmware you must boot KalliopePBX from one of the two partitions.

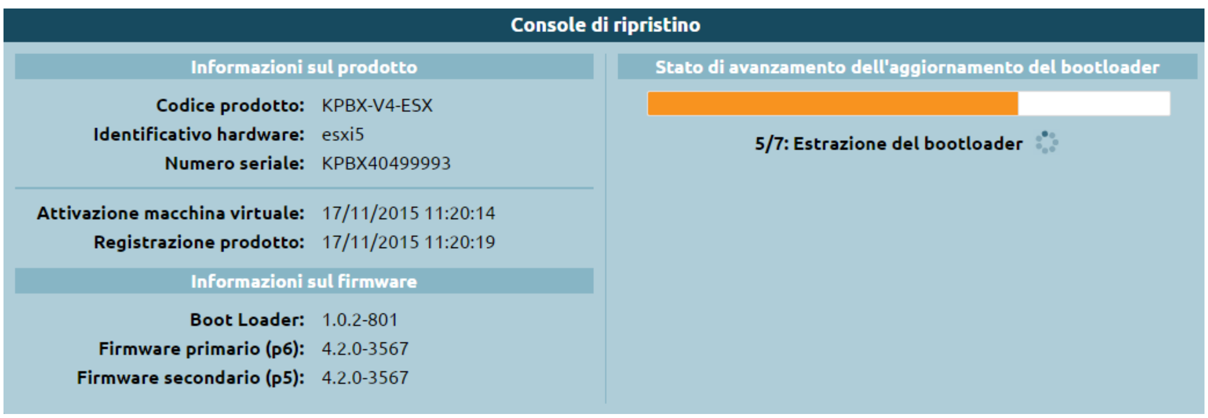


The system will download the list of released versions. The ones that cannot be installed (because they have already been installed or because they require an extra step) will be grayed out. Updates that are available for installation will instead appear in black font.

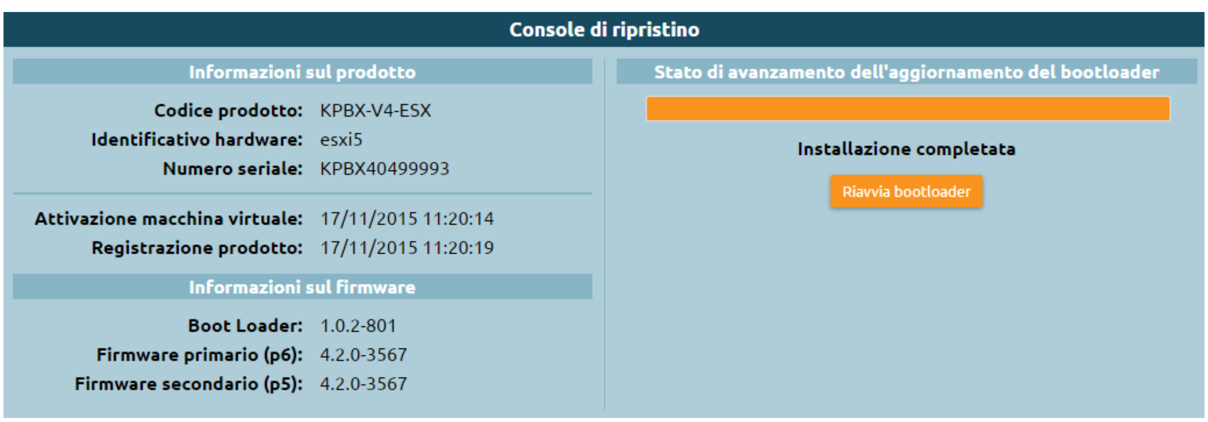
Clicking on a release will display the changelog for that version. If a bootloader update is available, the Apply this update button will be orange and selectable, as shown below.



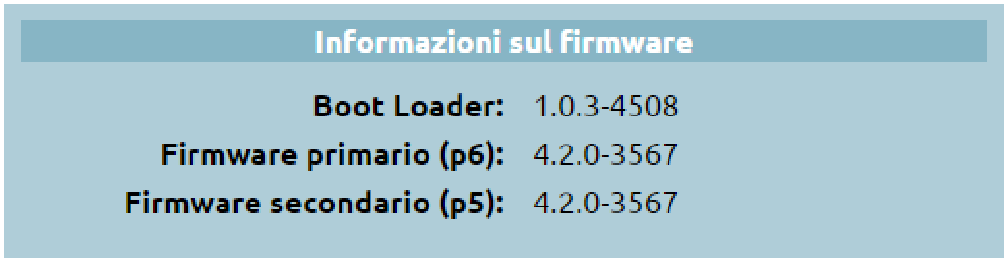
The bootloader will then be downloaded, extracted, and installed.



At the end of installation, you will need to restart the bootloader.



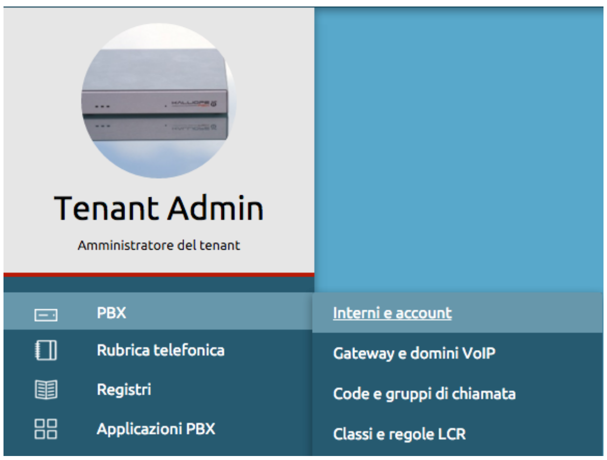
Once the bootloader has restarted the update is complete and you can install the firmware (if this is your first setup) or restart the PBX on the existing firmware.



How to create your first extension

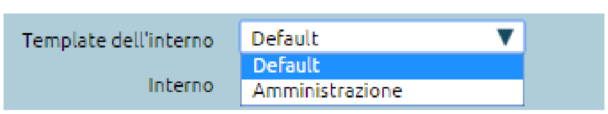
To create or edit an extension you must first acquire the lock.

Go to the PBX -> Extensions and Accounts page.



To create a new extension, click on Add extension. To edit an existing one, click on its number.

To configure an extension, you must first select a template from the drop-down menu. The template can be the default one or one you have previously created.



Each setting has two columns: the one on the left shows values that have been manually edited, and the one on the right shows the values from the template.

To overwrite a value from the template, you can simply select the checkbox and the item on the left will become editable.

Once configuration has been completed, click on Save to finish creating the extension.

By returning to the Extension list page you can edit existing extensions. To edit/delete a single extension you can click on the corresponding pencil or trash icon respectively.

You can select multiple elements and act on them through certain mass actions. You can:

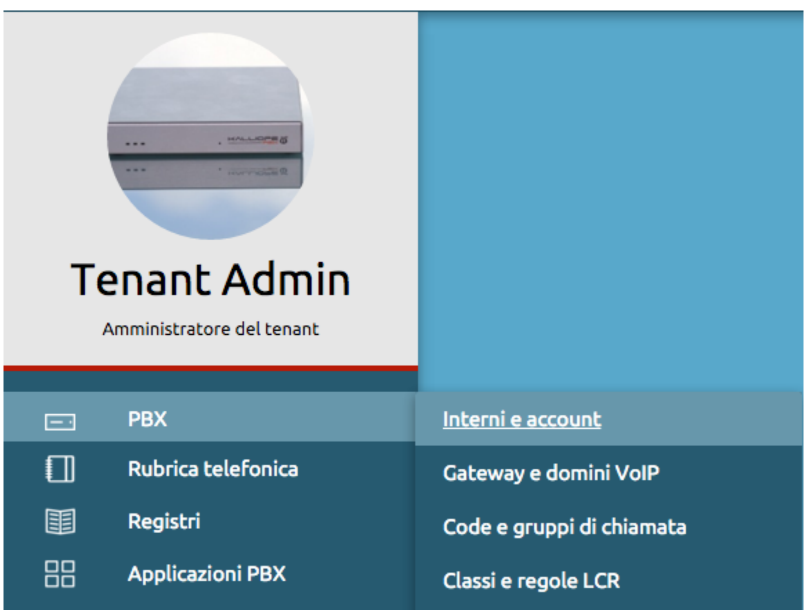
- Delete
- Enable
- Disable
- Change template

When you change the template, the previous configuration will be overwritten with the values from the selected template.

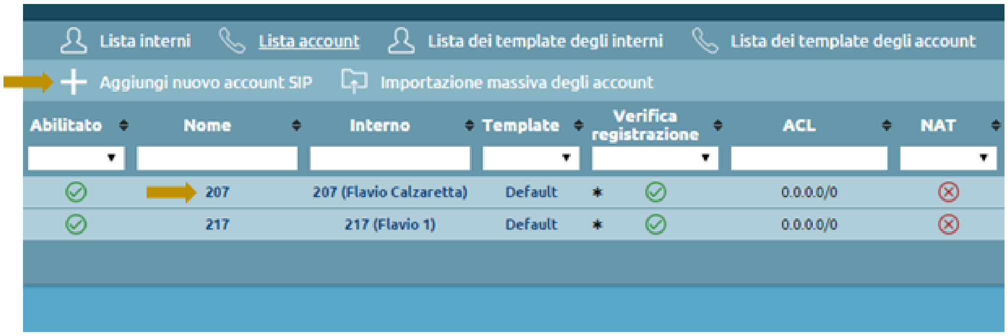
How to create your first account

To create or edit an account you must first acquire the lock.

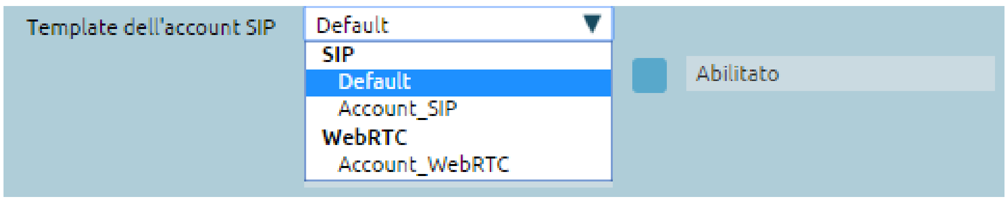
Go to the PBX -> Extensions and accounts page and click on Account list.



To create a new SIP account, click on Add SIP account. To edit an existing one, click on its name.



To configure a SIP account, you must first select a template from the drop-down menu. The template can be the default one or one you have previously created.



Each setting has two columns: the one on the left shows values that have been manually edited, and the one on the right shows the values from the template.

To overwrite a value from the template, you can simply select the checkbox and the item on the left will become editable.

Once configuration has been completed, click on Save to finish creating the account.

By returning to the Account list page you can edit existing account. To edit/delete a single account you can click on the corresponding pencil or trash icon respectively.

You can select multiple elements and act on them through certain mass actions. You can:

- Delete
- Enable
- Disable
- Change template

When you change the template, the previous configuration will be overwritten with the values from the selected template.

How to convert a backup from V3 to V4

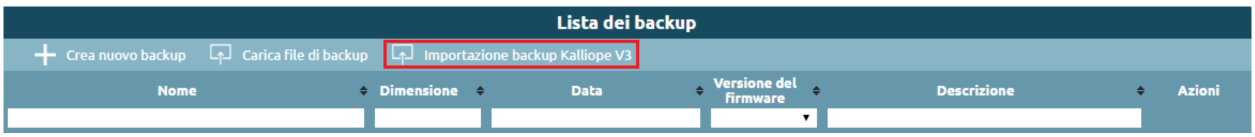
KalliopePBX V4 comes with much improved PBX configuration logic. Because of this, V3 and V4 configurations are not directly compatible.

In order to aid migration, we have developed a built-in tool available for firmware version 4.2.0 or later. This tool converts a backup from V3 to V4, except for a few elements that cannot be remapped and must therefore be handled manually.

Requirements:

- KalliopePBX V4 with firmware version 4.2.0 (or later) with factory settings
- V3 configuration backup made with firmware version 3.12.3

To restore a V3 configuration you must go to the Backup page from the system menu and click on Import Kalliope V3 backup, as shown below.



Before uploading the backup, carefully read all information provided by the system, the click on Choose file and select the V3 configuration backup you wish to convert.

You will now see a summary page showing the details of the selected backup. Once you have made sure it is the correct file, click on Confirm.

The system will bring the configuration to the new PBX and show which parts of the configuration it was unable to convert and will therefore need to be manually edited.

> **Warning:** Take note of these alerts before exiting the page

The configuration has now been imported and applied, and it is possible to view it in the KalliopePBX interface.

3.1.4 Interface description

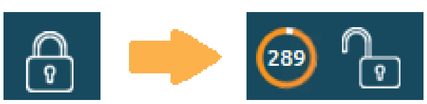
KalliopePBX's configuration interface consists of the top bar, the operating menu, the system menu, the content area in the center, and the bottom bar.

Top bar

By clicking on the Kalliope logo in the top left you can return to the home page from any configuration page, while clicking on the lock in the top left will lock the configuration.

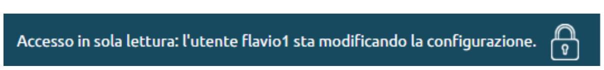
In KalliopePBX V4, configuration is based on lock (with editing permission), which can only be acquired by a single user at a time. Lock management is strictly based on roles.

Once lock is acquired a 5 minute countdown will begin. The countdown will be reset every time the user interacts with the interface.



When a user is making edits, other users will be notified and they will only be able to acquire a lock if they have a higher priority. If they do not, they will have to wait for the lock to expire.

In the image to the left, the user has higher priority and can therefore access the lock; in the image to the right, on the other hand, the user has a lower priority and therefore the lock is hidden.



Operating menu

The operating menu is located on the left side of the Kalliope interface. It includes information on the logged-in user and allows access to the configuration pages for telephone services and UC. The menu has a different layout depending on the type of Kalliope license and the type of user logging in:

Multitenant:

- PBX Admin

Primo livello	Secondo livello
PBX	*Gateways and VoIP domains*
	On-call services
	SIP settings
	Operating mode
Logs	*CDR*
	Usage statistics
	Audit Log
	PBX events
	Provisioning Requests
Provisioning	*Provisioning*
Sounds	*Lingue personalizzate*
System settings	*Tenants management*
	Sites management
	Users and roles management
	Licenses
	SSL settings
	SNMP settings
	AMI settings
	Remote filesystem management
Monitoring	*Packet capture*
	Active calls
	Notifications
	Scheduled tasks
File Browser	File Browser

- Tenant Admin

Primo livello	Secondo livello
PBX	*Extensions and Accounts SIP*
	Gateways and VoIP domains
	Management of assigned lines
	ACD e *Ring groups*
	Outbound routing
	Checktime
	Switches
	Numbering plan
	On-call services
	General settings
PBX applications	*MeetMe rooms*
	Call Campaign
	IVR menu

continues on next page

Table 4 – continued from previous page

Primo livello	Secondo livello
	Dynamic Routing
	Boss/secretary filter
	Paging groups
	Hot Desking
	Call Center Service Accessibility
	Billing module
FAX	FAX entities
	FAX settings
Phonebook	Phonebook
	LDAP settings
	Remote phonebook settings
Kalliope LAM	Kalliope LAM
Logs	*CDR*
	Call center CDR
	Call Center accessibility log
	Audit Log
	PBX events (OLD)
	PBX events
	Provisioning requests
Provisioning	*Provisioning*
Sounds	Audio settings
	Music on hold classes
	Audio files
System settings	*Users and roles management*
	Remote filesystem management
Monitoring	*Active calls*
	Notifications
	Scheduled tasks
	Services status
File browser	File browser

Singletenant:

- Admin

Primo livello	Secondo livello
PBX	*Extensions and Accounts SIP*
	Gateways and VoIP domains
	ACD e *Ring groups*
	Outbound routing
	Checktime
	Switches
	Numbering plan
	On-call services
	SIP settings
	General settings
	Operating mode
PBX applications	*MeetMe rooms*
	CallCampaign

continues on next page

Table 5 – continued from previous page

Primo livello	Secondo livello
	IVR menu
	Dynamic Routing
	Boss/secretary filter
	Paging groups
	Hot Desking
	Alarm receiver
	Call Center Service Accessibility
FAX	Send FAX
	FAX register
	FAX entities
	FAX settings
Kalliope LAM	Kalliope LAM
Phonebook	Phonebook
	LDAP settings
	Remote phonebook settings
Logs	*CDR*
	Call center log
	Call center accessibility log
	Audit Log
	PBX events (OLD)
	PBX events
	Provisioning requests
Provisioning	*Provisioning*
Suonds	Audio settings
	Music on hold classes
	Audio files
	Custom languages
System settings	*Sites Management*
	Users and roles management
	Licenses
	SSL settings
	SNMP settings
	AMI settings
	Remote filesystem management
Monitoring	*Packet capture*
	Active calls
	Notifications
	Scheduled Tasks
	Services Status
File browser	File browser

System menu

The system menu is located on the right side of the Kalliope interface. It allows access to the configuration for the global Kalliope system settings. The menu has a different layout depending on the type of Kalliope license and the type of user logging in:

Multitenant:

- **PBX Admin**
    - Credits
    - Change language
    - Customize GUI
    - Clock Settings
    - Network Settings
    - SMTP Settings
    - Backup
    - Update Management
    - High Reliability Status
    - Change Password
    - Logout
- **Tenant Admin**
    - Credits
    - Change language
    - Configuration Wizard
    - Change password
    - Logout

Singletenant:

- **Admin**
    - Credits
    - Change language
    - Guided configuration
    - Date and time
    - Network settings
    - SMTP settings
    - Backup
    - Firmware update
    - High availability state
    - Change password
    - Log out

- Shutdown/restart menu (also from the recovery console)

Privacy admin menu

Access to sensitive data and call recording management is handled by a special user called privacyadmin. When logged in with privacyadmin credentials, the operating menu will only display the following items:

First level	Second level
Phonebook	Phonebook
Logs	Call detail record
System settings	GUI user management
Call recording	View logs
	Edit settings

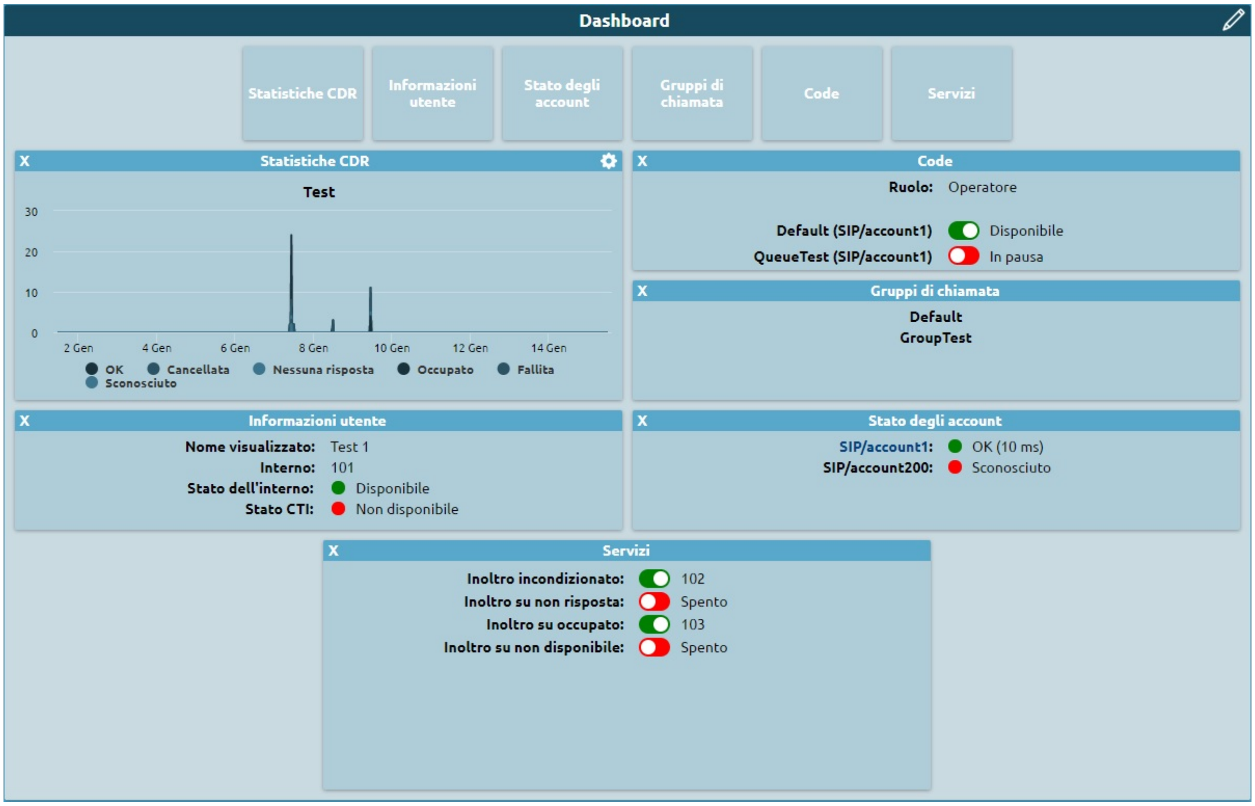
User menu

When accessing Kalliope as a user, you can view in the operating menu, on the left side of the the Kalliope interface, information on the logged-in user and access the configuration pages that the user has permissions to access.

As a tenant user, you can always view the phonebook and CDR pages. If the tenant admin assigns you further permissions, you can also view from this menu the other configuration pages.

You can also customize your dashboard with the information relevant to you.

Select the pencil icon on the top left to view all widgets: CDR statistics, user information, account status, ring groups, queues, services, function keys. By clicking and dragging a widget to the space below you can make them visible, as shown in the picture below.



CDR statistics

This widget displays a graph that summarizes the information in the CDR. You can set which information to display in the statistics.

You can:

- select the type of filter (outcome or direction);
- insert a time span by inserting the number and selecting year, months, weeks, days, hours, or minutes;
- insert the span with which the information will be grouped;
- insert the number of minutes between updates;

Click on “Save” to save and apply the settings.

User information

This widget displays information on the logged-in user:

- displayed name
- extension number
- extension status: green dot is available, red dot if unavailable, gray dot if suspended
- status of the associated CTI: green dot is available, red dot if unavailable

Account status

This widget displays the accounts associated with the extension and their status (green if active, red if not active).

Ring groups

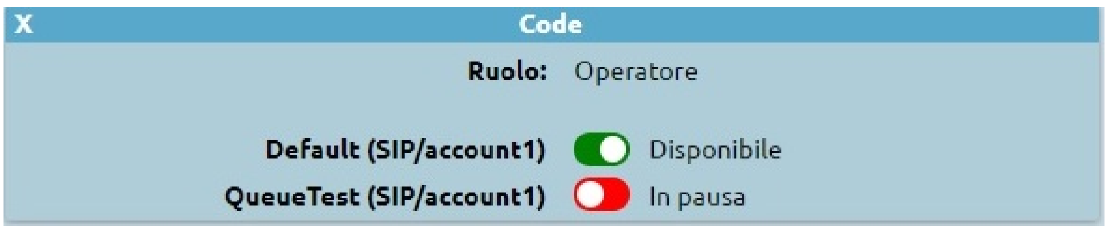
This widget displays the groups the user belongs to. IF the user does not belong to any groups, the widget will remain empty.

Queues

This widget displays the queues the user belong to:

- user role (operator or supervisor)
- status of the user on the queue (available or paused)

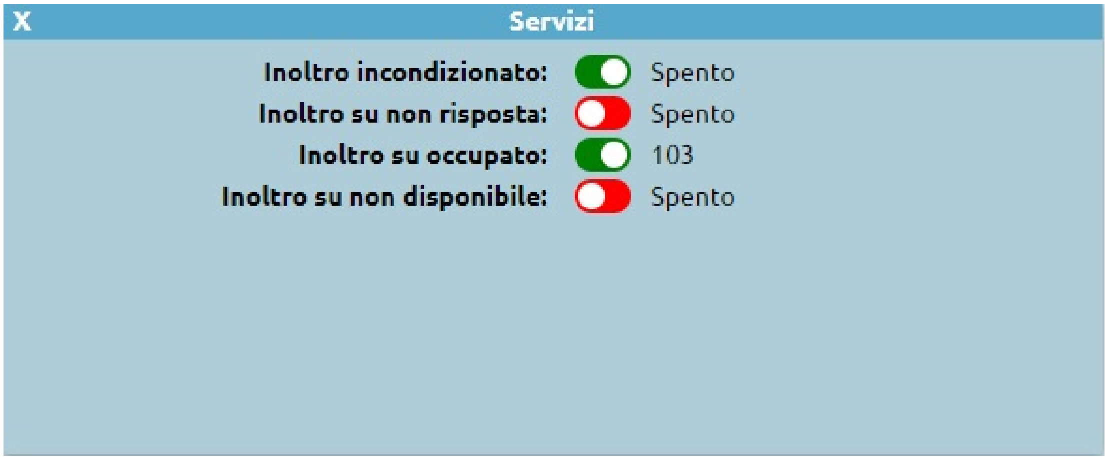
By clicking on the user status, you can change it.



Services

This widget lets you enable or disable certain services:

- Unconditional Forward
- Forward when not answering
- Forward when busy
- Forward when not available
- Fork to Mobile

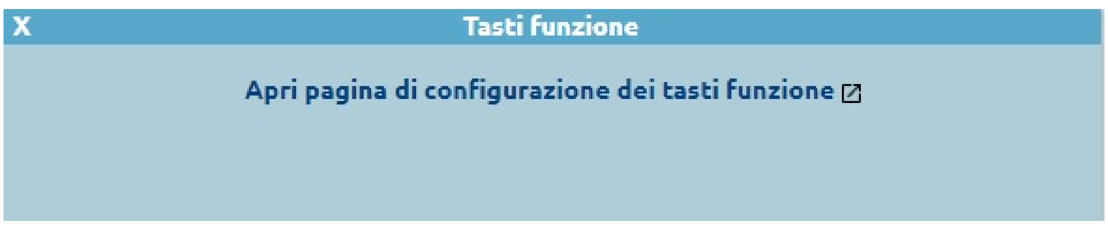


Forwarding services can be enabled by clicking on the red button and inserting the number to which you wish to forward calls and disabled by clicking on the green button.

If you have an associated mobile number, you can view the Fork to Mobile status and enable or disable it by clicking the corresponding button.

Function key configuration

From this widget you can access the function key configuration pages if your extension is associated with a device.



You can set each function key to a service by clicking on the link.

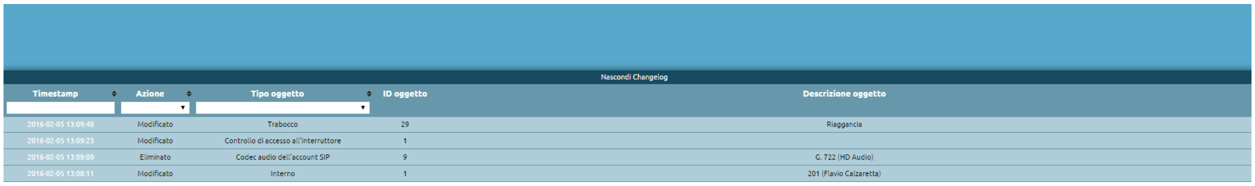
To add a function key, select the "+" button, inset the name of label that will be displayed on you device, select the type of service (BLF, boss/secretary, speed dial, do not disturb, unconditional forward, fork to mobile, switch, or parking slot), insert the corresponding value if required, and save.

After saving the settings, you can enable or disable the service with the phone buttons.

Since unconditional forward requires an extension number to be inserted, you can only disable and not enable it from you device.

Bottom bar

The bottom bar allows access to the changelog containing pending changes (i.e. changes that have been saved but not yet applied).



By clicking on Show changelog you can see a list of pending changes, with each line displaying the following information:

- Date and time
- Type of action
- Object
- Object ID
- Object description

When changes are applied they will automatically be removed from the changelog.

CHAPTER

FOUR

ADD-ON AND KALLIOPE APPLICATIONS

4.1 Kalliope Applications

4.1.1 KalliopeCTI 4

Warning: The KalliopeCTI 4 version 4.5.23 manual for the client installation and configuration is available here. To download the client for your operating system, visit kalliope.com, menu Support –> Downloads and then Applications

Introduction

KalliopeCTI 4 (or KCTI 4) is an accessory application of the KalliopePBX V4 VoIP that helps you use its services and access information.

KCTI 4 (available in Free, Pro, and Phone modes) is a cross-platform application on Windows, Mac OS X, and Ubuntu. A KCTI 4 Pro or Phone license includes a license for the KCTI Mobile app for Android and iOS.

The KalliopeCTI Mobile user manual can be found here.

The following table lists the features available with each mode:

	KCTI 4 Free	KCTI 4 Pro	KCTI 4 Phone	KCTI 4 Mobile
Extension phonebook	✓	✓	✓	✓
Shared phonebook	✓	✓	✓	✓
CDR	✓	✓	✓	✓
Click-to-call	✓	✓	✓	✓
Inbound call notification	✓	✓	✓	✓
Instant Messaging	✓	✓	✓	✓
Presence	✓	✓	✓	✓
Opening custom URLs	✓	✓	✓	
Voicemail access	✓	✓	✓	✓
Synchronization with Outlook contact		✓	✓	
BLF		✓	✓	✓
Number dialing		✓	✓	✓
Blind transfer		✓(2)	✓	✓
Attended transfer		✓(2)	✓	✓
Do not Disturb		✓(2)	✓	
Call recording		✓(2)	✓	
Unconditional forward		✓(2)	✓	
Call parking		✓(2)	✓	
Call pickup		✓(2)	✓	
Forking to Mobile		✓(2)	✓	✓
Queue statistics	✓(2)	✓(2)	✓(2)	
Supervisor mode	✓(2)	✓(2)	✓(2)	
Selective pause	✓(2)	✓(2)	✓(2)	
Touchscreen support				✓
Mobile phonebook integration				✓

1 Available only when paired with a Snom or Yealink phone

2 Available only when paired with Kalliope Call Center

Download and installation

KCTI 4 can be downloaded for free from the Download page of the official KalliopePBX website.

The user can then go to the configuration page to select which version to open each time (as long as the corresponding license has been activated on KalliopePBX V4). KCTI Free, Pro, or Phone are different modes of the same software.

Note: Installation does not prompt for any activation keys. Instead, the application will automatically check for the correct license on KalliopePBX every time it is opened.

Unlike KCTI 4 Free, KCTI 4 Pro and Phone require a specific license to be activated on KalliopePBX V4 for each workstation. The KCTI 4 Phone license also includes KCTI 4 Pro. The user can choose their desired mode from the configuration page.

For instructions on how to activate the KCTI 4 Pro or KCTI 4 Phone licenses, see the licenses page.

System requirements:

Windows:

- Windows 10 or later
- Sound card (for KCTI Phone)

- 2 GB RAM
- 100 MB of free disk space

MAC:

- macOS: Ventura, Monteray and BigSur
- recommended: Apple hardware from 2013 or later

Linux:

- Ubuntu 22.04 and 20.04

To install the application on Windows, simply open the executable file and follow the instructions.

Note: In case the SIP service of the central unit is exposed on a different IP from the CTI connection one, or on a different port from the standard one (5060) it is necessary to set on the KalliopePBX one or both of the following custom placeholders inside the Provisioning panel: **%%_KPHONE_SIP_REGISTRAR_IP%%** and **%%_KPHONE_SIP_REGISTRAR_PORT%%%** . This configuration is normally necessary when the SIP service is exposed to the public through a Session Border Controller that uses a different IP from the one on which the PBX is reachable, while it is not necessary if the PBX and the SBC are NATed on the same public IP, each one for the ports of its competence (TCP/5039 and TCP/5222 for the CTI and Chat components, towards Kalliope, and SIP + RTP for voice, towards the SBC).

User Manual

The KalliopeCTI 4 manual for version 4.5.23 and the installation and configuration of the cient is available here.

Special parameters of the KalliopeCTI INI file

Starting from **4.5.18** of KalliopeCTI the following parameters can be added or modified in **kcti.ini**. The following table shows the editable parameters with the version of KalliopeCTI from which the parameter is present and editable.

Parameter	Example Value	Description	Starting from (version KCTI Desktop)
echoCanceller	on	To enable (on) or disable (off) the echo cancel. Default: on	4.5.23
echoCancellerLmsFilter	off	To enable (on) or disable(off) the echo caneller LMS filter. Default: off	4.5.23
echoSuppressor	on	To enable (on) or disable (off) the echo suppression. Default: on	4.5.23
suppressorLowerThreshold	10	To set the lower threshold (percentage) of echo suppression. Allowed value range: [0, 100]. Default: 10	4.5.23
suppressorUpperThreshold	40	To set the highest threshold (percentage) of echo suppression. Allowed value range: [0, 100]. Default: 40	4.5.23
suppressorDivider	15	To set the echo suppressor divider. Allowed value range: [2, 50]. Default: 15	4.5.23
suppressorConvergenceMillis	4000	To set the echo suppressor convergence (millis). dell'echo suppression. Allowed value range: [1000, 120000]. Default: 4000	4.5.23
chatKeepAliveSendIntervalSecs	120	To XMPP ping messages, default 2 minutes	4.5.19
contactsRefreshTimeoutSecs	600	To the contacts refresh, default 10 minutes	4.5.18
cdrRefreshTimeoutSecs	90	To cdr/voicemail refresh, default 1 minutes and 30 seconds	4.5.18
identitiesRefreshTimeoutSecs	300	To identities refresh, default 5 minutes	4.5.18
queueStatsRefreshTimeoutSecs	300	To queue stats refresh, default 5 minutes	4.5.18
codecOrderRefreshTimeoutSecs	30	To codec order refresh, default 30 seconds	4.5.18
excludedOutlookFolders	Public folders;Cache;Lync;Skype	list of strings contained in the Outlook folders to be excluded. Including the relative subfolders from contacts imports. As a separetor you can use ;	4.5.18
queueNames	Queue1;Queue2	list of the queue names to be displayed in the Supervisor panel. As a sepaetor you can use ;	4.5.15

kcti.ini is located in the same directory of the traces (AppDataRoamingKCTI4 under the user path).

Note: kcti.ini must be edited manually when Kcti is not running.

API

The client provides the user with APIs to perform actions without the use of the graphic interface. The available actions are:

- To call a number
- To close the current call
- To transfer with/without an offert the call in progress
- To pause/unpause on all queues
- To enable/disable DND
- Retrieve the data of the active call

For versions 4.5.0 and up a Postman collection is available to test the API of the KalliopeCTI client. You can download it from this link: `KalliopeCTI Client (vers.4.5.x) postman collection`. You can also download a Kalliope CTI client API operation manual in pdf format from here `link`

4.1.2 KalliopeCTI Mobile

Introduction

For those who need to always be on call, Kalliope has designed the KalliopeCTI mobile app for Android and iOS.

The KalliopeCTI mobile app is an optional product that lets you access telephone and UC services even outside your office through your extension number, using the company lines.

To guarantee stability and improve conversation quality, KalliopeCTI mobile lets you choose how to handle your calls: based on data traffic availability, you can choose whether to use the GSM network or use the app in softphone mode.

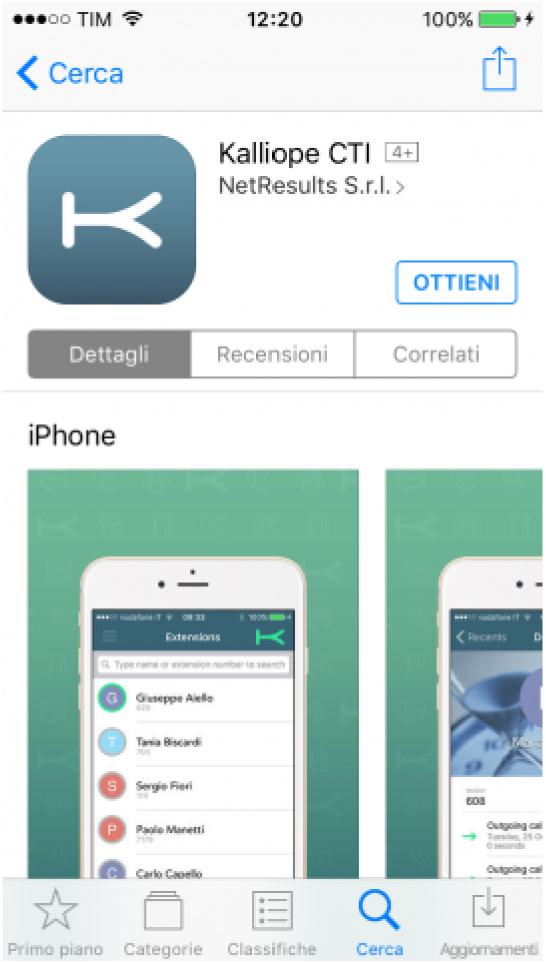
KalliopeCTI mobile lets you call using three different modes:

- Click-to-call: the call will be handled through the desk phone associated with your extension;
- Call-back: call from your smartphone using Kalliope lines;
- Softphone: using an integrated SIP telephone engine, your smartphone will register to KalliopePBX like any other IP client.

KalliopeCTI mobile is available on Playstore and App Store.

View the policy on Processing of users' personal data

Note: Each extension can only be linked to **one** KCTI account.



Initial configuration

In order to use the KalliopeCTI mobile app you must first:

- Enable "KCTI Mobile App" in the account settings

Note: Only one account per extension may be enabled to use the mobile app

- Assign one of the following KalliopeCTI licenses to the user:
  - KalliopeCTI Pro license -> enables the use of the KalliopeCTI mobile app only in callback mode (GSM™)
  - KalliopeCTI Phone license -> enables the use of the KalliopeCTI mobile app in both softphone and callback mode
- If you wish to externally access the company network with the mobile app, you will need to publish the following KalliopePBX ports

Note: **It is recommended** to not expose the central unit's SIP service on the public in a direct way **that can be reached from arbitrary IPs, but to use a Session Border Controller that can properly protect the central unit from intrusion attempts from outside)**

- TCP 5039 port for CTI services

- TCP 5222 port for XMPP chat services
- UDP for SIP service port as configured in the “SIP settings” page (only for softphone mode, default value is ‘5060’)
- UDP 10000-20000 ports interval for RTP streams (only for softphone mode)

Also, it is necessary that the KalliopePBX can send messages to the push notification servers of Apple and Google, so the reachability by the KalliopePBX of the following addresses must be guaranteed:

- iOS: api.push.apple.com (HTTPS, port TCP/443)
- Android: fcm.googleapis.com (HTTPS, port TCP/443)

Since these hosts may resolve different IPs over time, it is recommended that no filters be applied to HTTPS traffic exiting KalliopePBX.

In addition, in order for mobile devices running the KalliopeCTI Mobile APP to receive notifications, they must be connected and reachable by the sending services. For more information about the flows that must be ensured, please refer to the official documentation of:

- Google: https://firebase.google.com/docs/cloud-messaging/concept-options#messaging-ports-and-your-firewall
- Apple: https://support.apple.com/en-ph/HT203609

Note: In order to avoid abnormal behavior in sending incoming/ending call notifications to clients, it is necessary that each user is logged in on a single device, because each SIP account (and therefore associated user) can be associated with a unique Firebase token (used to identify the recipient of the notifications).

Note: In case the central unit SIP service is exposed on a different IP than the CTI connection IP, or on a different port than the standard one (5060), it is necessary to set on the KalliopePBX one or both of the following custom placeholders within the Provisioning panel: %%_KPHONE_SIP_REGISTRAR_IP%% and %%_KPHONE_SIP_REGISTRAR_PORT%%%. This configuration is normally necessary when the SIP service is exposed to the public through a Session Border Controller that uses a different IP from the one on which the PBX is reachable, while it is not necessary if the PBX and the SBC are natted on the same public IP, each one for the ports of its competence (TCP/5039 and TCP/5222 for the CTI and Chat components, towards Kalliope, and SIP + RTP for voice, towards the SBC).

More and specific informations regarding the configuration and use of the KalliopeCTI Mobile app in the two operating systems Android and IOS can be found on the following dedicated pages:

- Manuale KalliopeCTI Mobile app Android
- Manuale KalliopeCTI Mobile app IOS

Kalliope CTI Mobile Android

Changelog vers. Android

KalliopeCTI Mobile App Version 4.8.12

- Fixed an issue where on firmwares prior to 4.13.0 the call transfer (without offer) didn't work properly

KalliopeCTI Mobile App Version 4.8.11

- Fixed an issue where on some phone models following an unanswered outgoing call the media and notification volume was muted

KalliopeCTI Mobile App Version 4.8.10

- Fixed an issue that could prevent receiving calls if the app was not closed for too long

KalliopeCTI Mobile App Version 4.8.9

- Added log with statistics for monitoring audio quality.
- Fixed an issue that could cause muted audio on a GSM call received while a VoIP call was in progress.
- Fixed an issue where the transfer did not work properly if the call was paused.
- Resolved an issue where the hardware button did not properly control the ring volume on some smartphone models.
- Fixed an issue where the Bluetooth audio device could be disconnected when exiting the call screen.
- Fixed an issue where re-entering the call screen from the current call notification could present the incoming call widget.

KalliopeCTI Mobile App Version 4.8.8

- Fixed an issue that could cause startup crashes on some phone models.
- Stability improvements

KalliopeCTI Mobile App Version 4.8.7

- Fixed an issue that could prevent receiving push notifications

KalliopeCTI Mobile App Version 4.8.6

- Increased the clickable area of buttons in the call screen for a better user experience.
- Stability improvements

KalliopeCTI Mobile App Version 4.8.5

Bugfixes

- Fixed an issue that could prevent a transfer with offer from completing (on firmware 4.13.0+).

KalliopeCTI Mobile App Version 4.8.4

Bugfixes

- Stability improvements

KalliopeCTI Mobile App Version 4.8.3

Bugfixes:

- Fixed an issue that could prevent voicemail downloading on some - phone models.
- Fixed an issue that would stop the ringtone before replying
- Stability improvements

KalliopeCTI Mobile App Version 4.8.2

- The app notifies the user that he received a call while he was in VoIP conversation with another caller: this is a different notification than the "missed call" one sent by Kalliope
- Updated SIP engine
- Stability improvements

KalliopeCTI Mobile App Version 4.8.1

- Optimized call notification engine
- Updated SIP engine
- Minor graphical improvements

Bugfix:

- Fixed an issue that prevented call handling in kcti mode.
- Fixed an issue that did not disable the proximity sensor when the app was not in the foreground.
- Fixed an issue where when returning from a GSM call, the Bluetooth headset would not reconnect

KalliopeCTI Mobile App Version 4.8.0

Features:

- Replaced blind transfer feature with offer transfer (only connected to panels with firmware 4.13+). Call transfer can only start when the call has been established and is in progress.

For more details on this new feature *click here*

- Improved echo cancellation algorithm (added a toggle in settings to disable it)

Bugfixes

- Fixed an issue where the Bluetooth music stream would not stop when a call came in.
- Fixed an issue that prevented ringing in KCTI mode

KalliopeCTI App Mobile Version 4.7.3

Features:

- Improved audio quality in case of small packet loss events
- Swipe-to-refresh in the CDR list
- Added bigger popup notification that indicates if you received a call while you are on another VoIP call

Bugfixes:

- Fixed some random crashes and minor graphical retouches.

KalliopeCTI Mobile App Version 4.7.2

Bugfix:

- Library update fix 4.7.1 resulted in a crash when starting the app (on 32bit phones). This version fixes the crash introduced in 4.7.1.

KalliopeCTI Mobile App Version 4.7.1

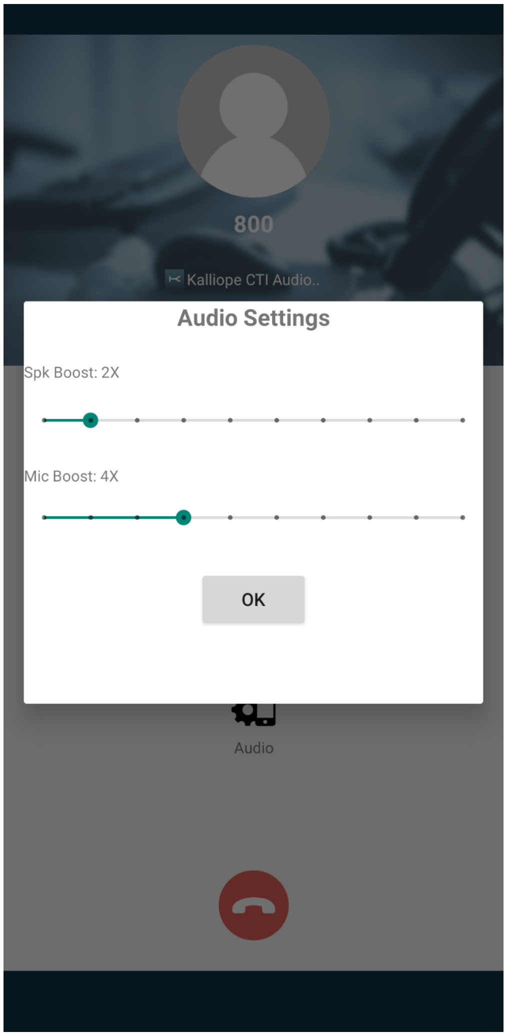
Bugfix:

- Fixed an issue where an outgoing VoIP call would be closed when a second VoIP call (incoming) came in and terminated. The problem was present also in previous versions, but instead of ending the call, the audio was lost
- Call volume linked to phone volume and no longer to multimedia volume also for arm7 (32bit) phones

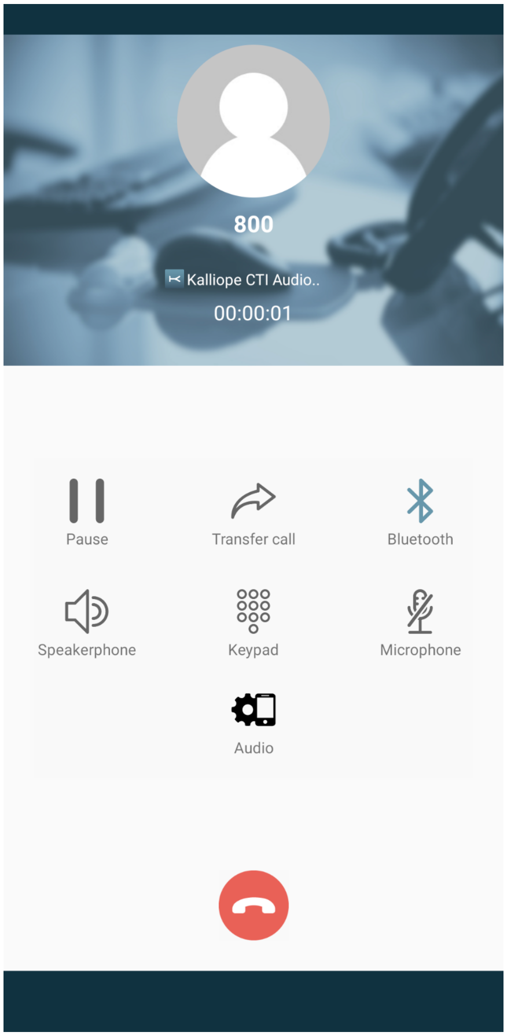
KalliopeCTI Mobile App Version 4.7.0

Features:

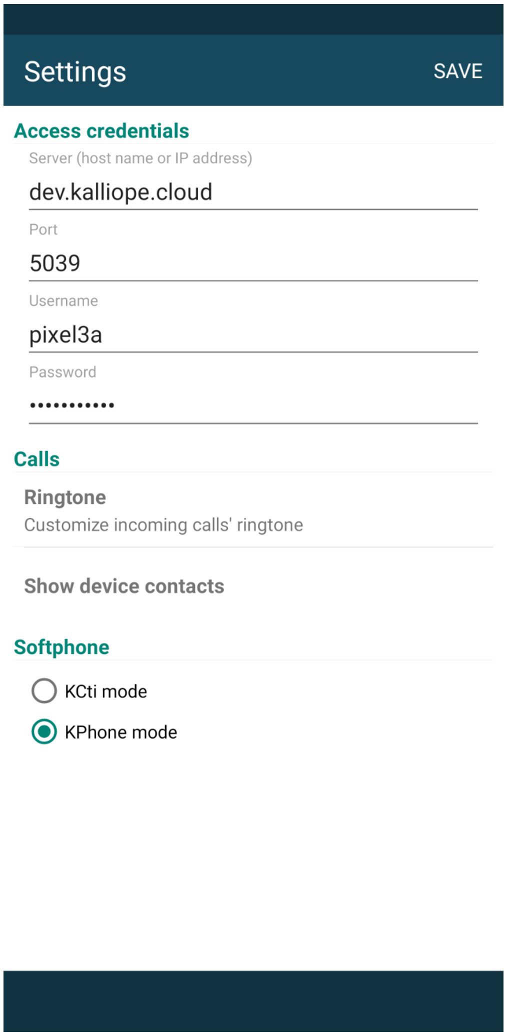
- Mode of operation of the app communicated to the central unit (requires firmware version 4.11.7) that improves communications stability.
- Bluetooth headset, if present and connected at the arrival of a call, are automatically activated
- Call volume connected to the phone volume and no longer to the multimedia volume

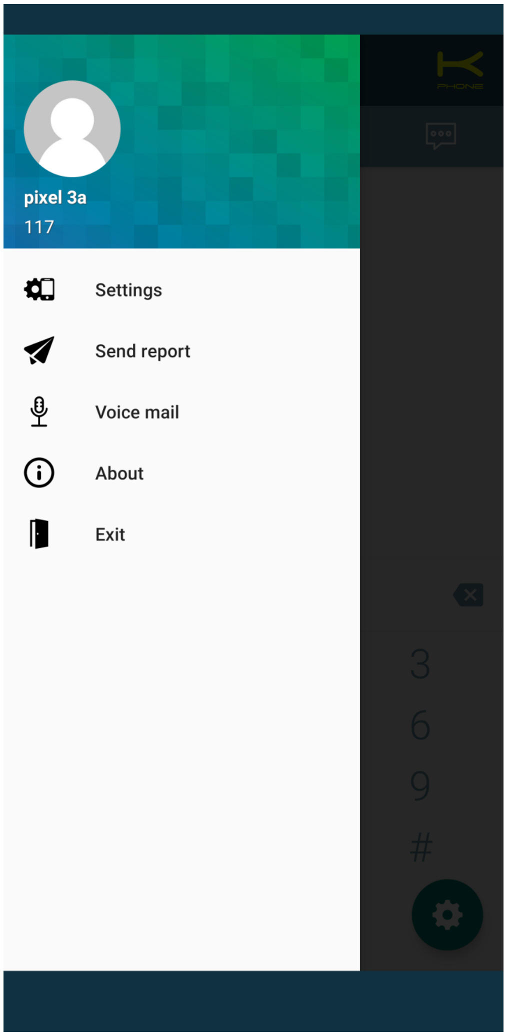


- Added a panel in the call screen to be able to use an audio boost on microphone and speakers



- Removed softphone mode on wifi only, operating mode settings are now in the settings page
- Introduction of androidX libraries


Settings
SAVE
Access credentials
Server (host name or IP address)
dev.kalliope.cloud
Port
5039
Username
pixel3a
Password
Calls
Ringtone
Customize incoming calls' ringtone
Show device contacts
Softphone
KCti mode
KPhone mode



Bugfix:

- Fixed a possible crash when terminating calls.
- Removed the initial tutorial as obsolete
- Fixed an issue that prevented the activation of the handsfree before the call was fully established
- Fixed an issue where the app would remain busy if an end of call message did not arrive.
- Fixed an issue where the app tried to connect to the standard SIP port 5060 instead of the one reported by the central office during login.
- Fixed an issue where the chat could not reconnect to the server.
- Fixed some minor stability issues

KalliopeCTI Mobile App Version 4.6.0

Features:

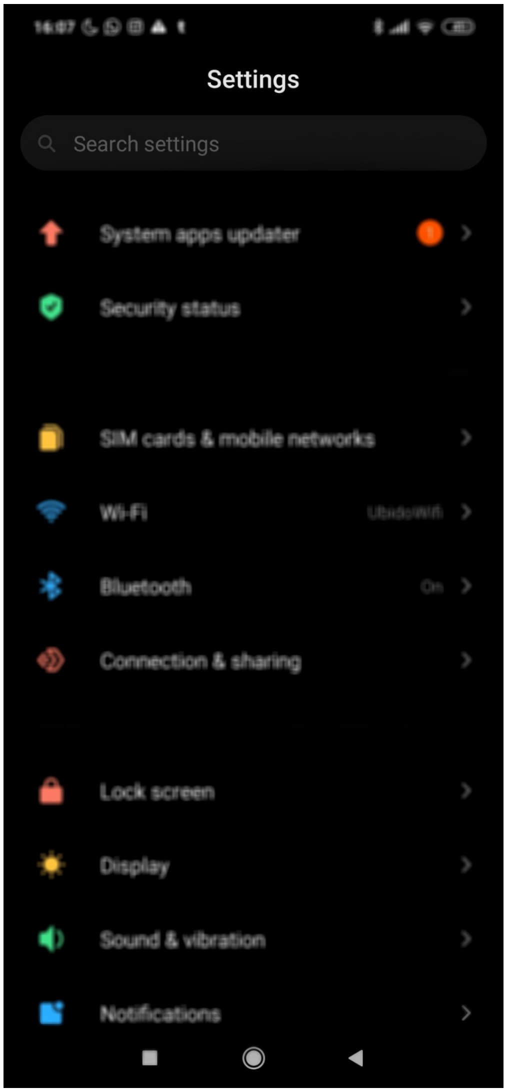
- A new call mode has been introduced: now next to each contact, there is an icon that allows you to call him directly with a SIP call (if in Softphone mode) or to make a call-back on your cell phone (if in CTI mode). If no mobile number is specified on KalliopePBX for the user, then the default action in Cti mode will be the click-to-call on the landline.
- The icon indicating which address book (Kalliope, google, etc.) the contact is from, has been moved to the contact information tab. The contact information tab appears upon clicking on the contact's entry.
- When entering the number for the unconditional forwarding a switch has been added to specify whether the number is an extension or an external, so the number to be entered does not need the line commitment prefix (which will be added automatically or not depending on the choice).

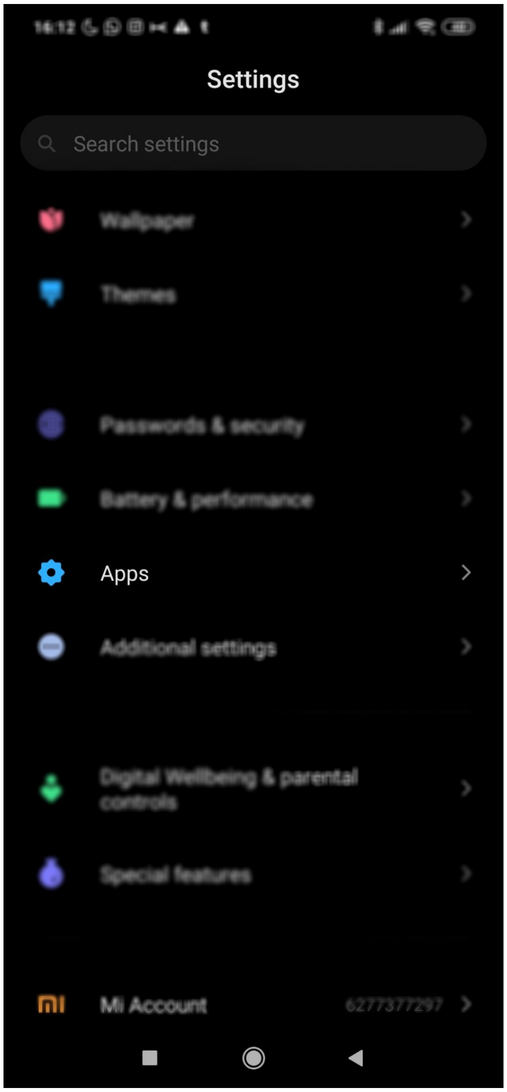
Bugfix:

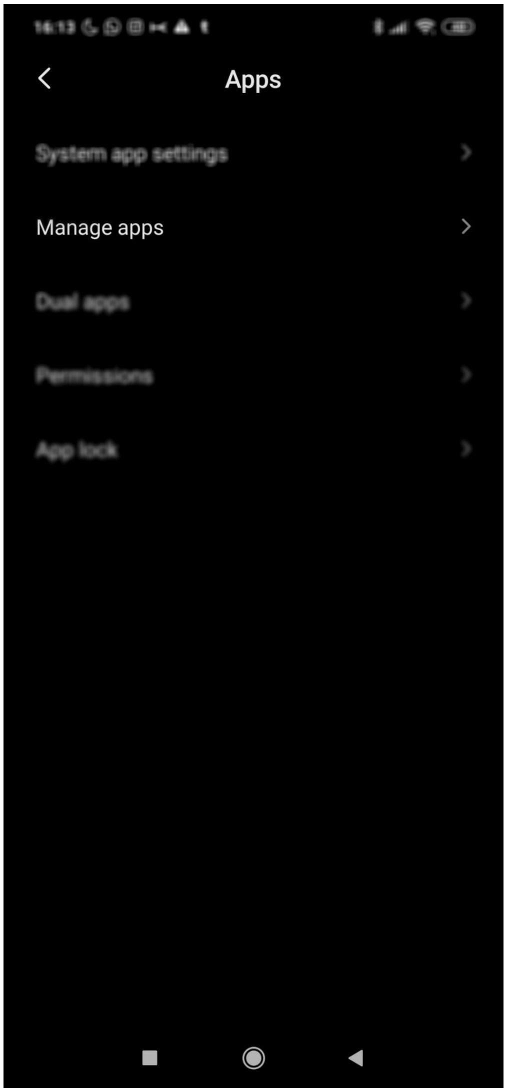
- Fixed a bug that allowed trying to make calls from the keypad even after the connection to Kalliope was lost; this would bring the app into an inconsistent state.
- Fixed a bug that showed as "unknown" calls from numbers not present in Kalliope's address book but in the phone's native address book (this feature is active only if the address book integration is active).
- Fixed an issue where pressing the "back" button from the filtered contacts list did not show the correct page.
- Fixed an issue where two icons overlapped on the password field of the settings on some phones.

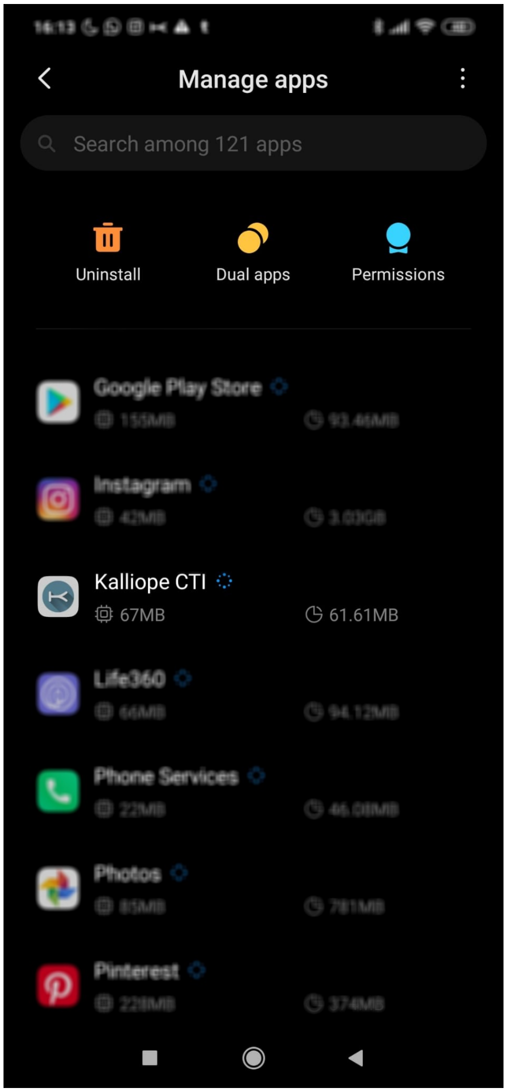
KalliopeCTI Mobile App Version 4.4.6

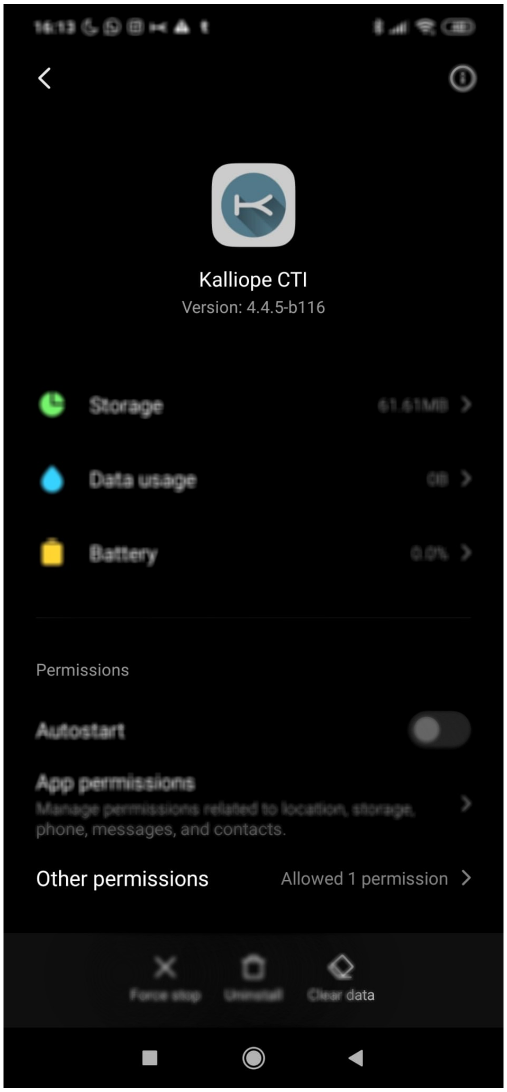
In this release, a new tutorial has been added at startup that will be displayed on Android phones with the Xiaomi Miui version. On these Android phones, it is necessary (shown in the tutorial screens) to manually set some additional permissions to allow you to receive notifications while using other apps or with the screen locked.

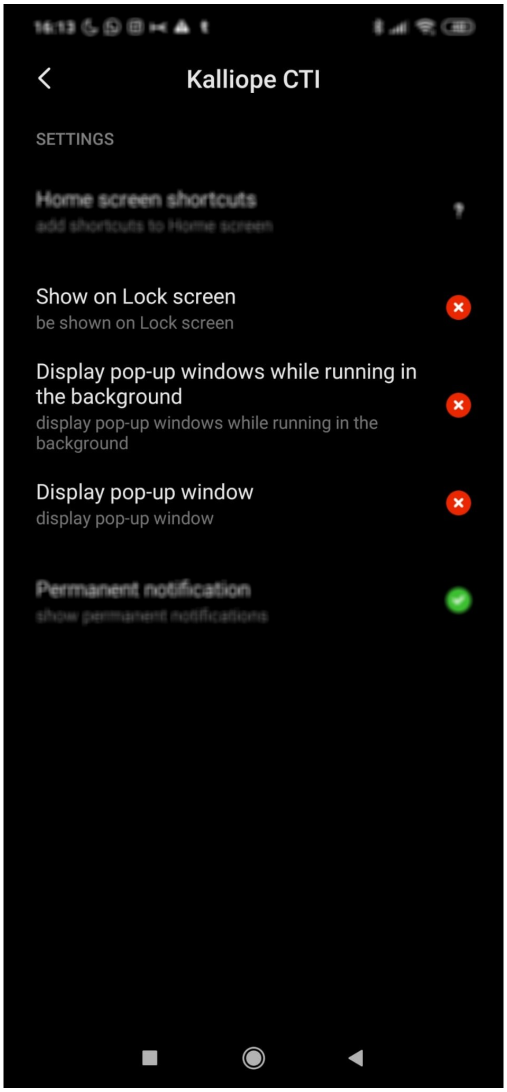

Settings
Search settings
System apps updater
Security status
SIM cards & mobile networks
Wi-Fi
Bluetooth
On
Connection & sharing
Lock screen
Display
Sound & vibration
Notifications

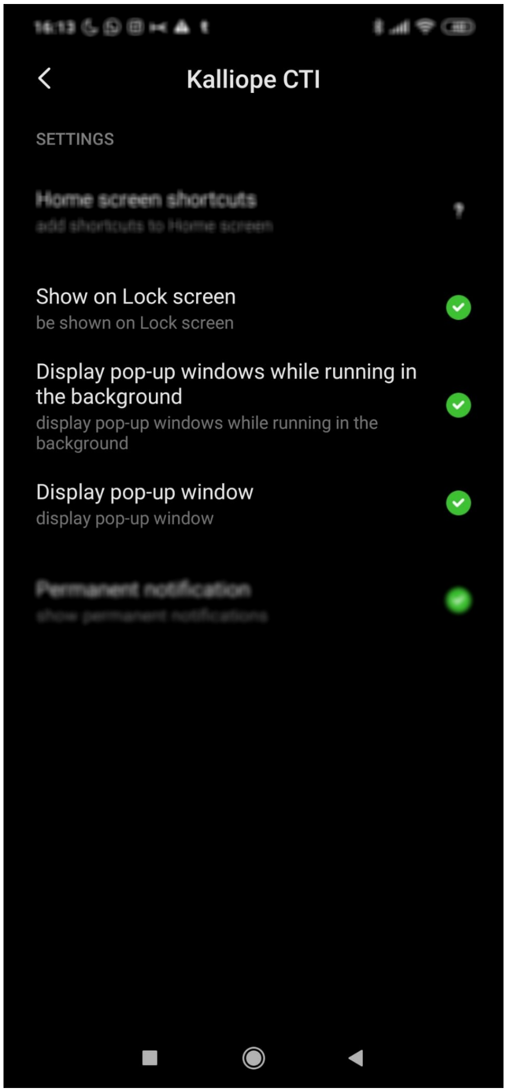

Settings
Search settings
Apps


Apps
Manage apps


Manage apps
Search among 121 apps
Uninstall
Dual apps
Permissions
Kalliope CTI
67MB
61.61MB


Kalliope CTI
Version: 4.4.5-b116
Storage
Data usage
Battery
Permissions
Autostart
App permissions
phone, messages, and contacts.
Other permissions
Allowed 1 permission
Clear data


Kalliope CTI
SETTINGS
Show on Lock screen
be shown on Lock screen
Display pop-up windows while running in the background
display pop-up windows while running in the background
Display pop-up window
display pop-up window



Android Specification:

- Requires Android 6 or higher
- Languages: English, Italian
- Product available only in combination with a KalliopePBX® V4 VoIP PBX.
- Minimum KalliopePBX firmware version: 4.6.0 (does not support softphone mode)
- Suggested KalliopePBX firmware version: 4.7.15+.
- Integration of the native Connection Service libraries to guarantee that the softphone can always be reached even when the terminal is in standby mode.
- A KalliopeCTI Pro license is required to use only the GSM™ mode. A KalliopeCTI Phone license is required to use both modes, GSM™ and Softphone.

Note: Softphone mode is in beta and currently has the limitations listed below. These limitations will not be present in the final version, which is nearing completion.

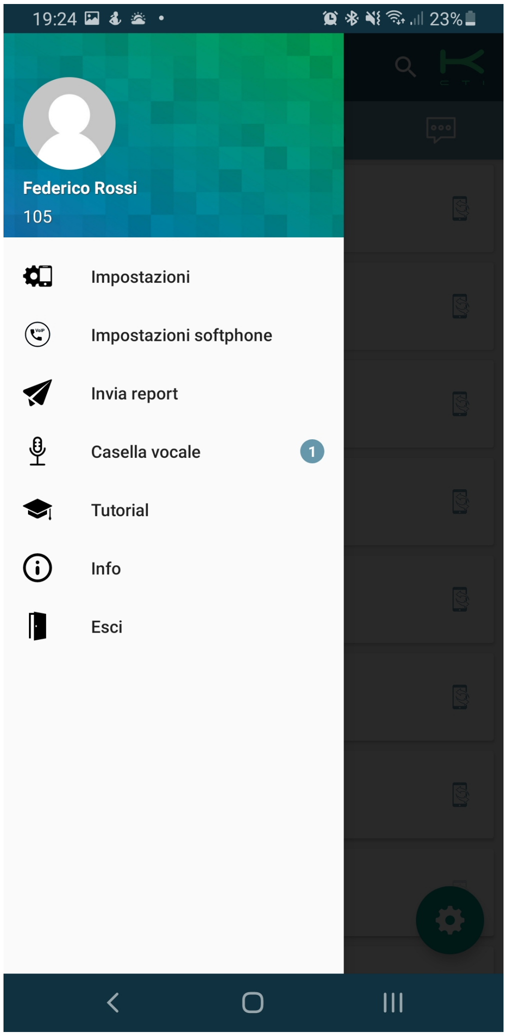
- It cannot handle multiple SIP calls at the same time
- Bluetooth support has yet to be perfected
- SIP/TLS is not yet enabled
- Resuming a paused call may sometimes fail

Organization of the GUI

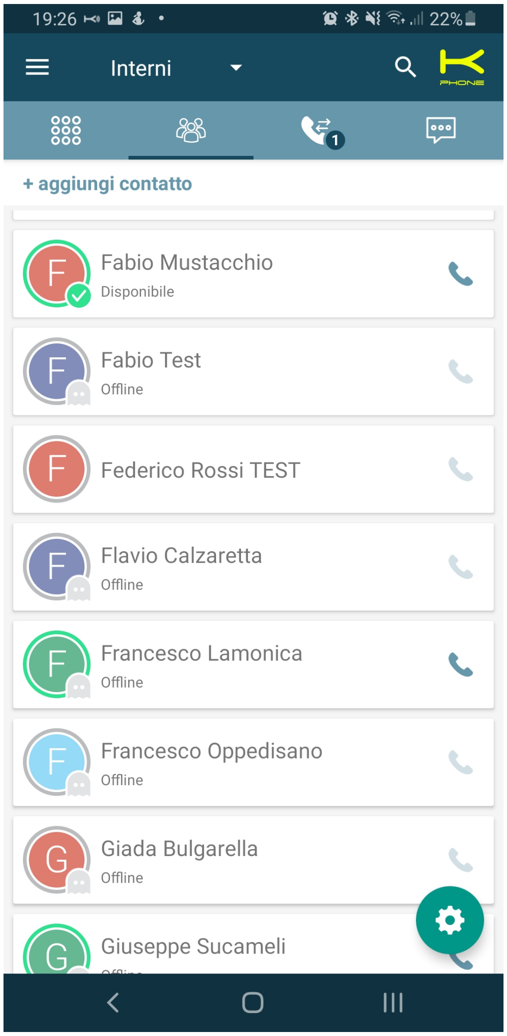
When you first start the KalliopeCTI app, you need to go and set the configuration parameters to allow the app to connect to KalliopePBX.

Tapping the menu icon on the top left lets you access the system menu containing the following items:

- Settings: application configuration panel
- Softphone settings: allows setting the mode of the application (iOS only)
  - Softphone disabled: the app forcibly works in call-back mode. In this case, the KalliopeCTI Pro license is sufficient.
  - Softphone enabled: the app is forced to work in softphone mode. In this case, the KalliopeCTI Phone license is sufficient.
  - Softphone enabled on WiFi only: the app will automatically switch to softphone mode if the connection is established with a WiFi network. Switching to softphone mode will only be possible if the KalliopePBX user is associated with a KalliopeCTI Phone license.
- Voicemail: direct access to voicemail messages
- Info: provides information about the installed version
- Send report: allows you to send a report to the developers in case of abnormal behavior of the app
- Tutorial: restarts the app tutorial (Android only)
- Exit: closes the app and you will not receive any more notifications until the next time you reopen the app.



The main screen of the application is shown here and consists of an upper bar.



The top-level bar shows the name of the tab you are currently viewing (in this case, Address Book), allows you to search within each tab and shows the connection status of the app via the K symbol in the top right corner. Below is shown the top-level bar (header) for Android.

The connection status is indicated by the color of the K symbol in the upper right corner:

- gray = not connected
- green (with the word "cti") = connected in call-back mode.
- yellow (with the word "phone") = connected in softphone mode.

In the Android version, the header also includes the navigation bar between the app's main sections.

In both versions, the navigation bar lets you access the main tabs of the app:

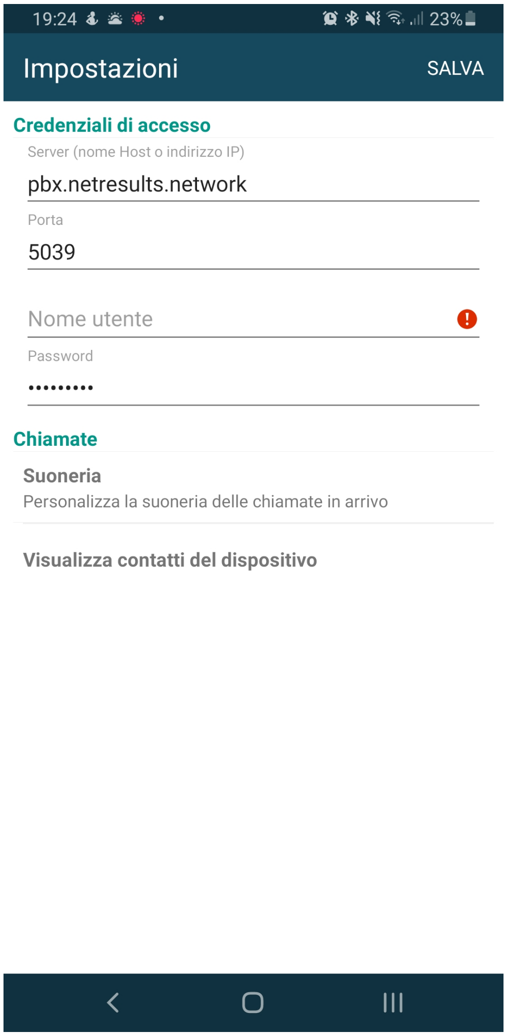
- Keypad: the numeric keypad used to dial numbers;
- Address book: extensions, Kalliope phonebook, favorite contacts, and, if enabled, the personal contacts of the device;

- Call log: list of dialed, received, and missed calls, grouped by contact;
- Chat: conversations with other KalliopeCTI users, both mobile and desktop.

Settings Panel

First you need to configure some simple parameters from the Settings panel which can be reached through the system menu:

- Login credentials
  - Server (Host name or IP address): IP address of the KalliopePBX central unit
  - Port: 5039
  - User name GUI/CTI user name
  - Password of the GUI/CTI user
- Calls
  - Ringtone: useful to differentiate the incoming calls to the mobile number from the incoming calls to the extension
  - Display device contacts: allows to choose the address books from which to display contacts



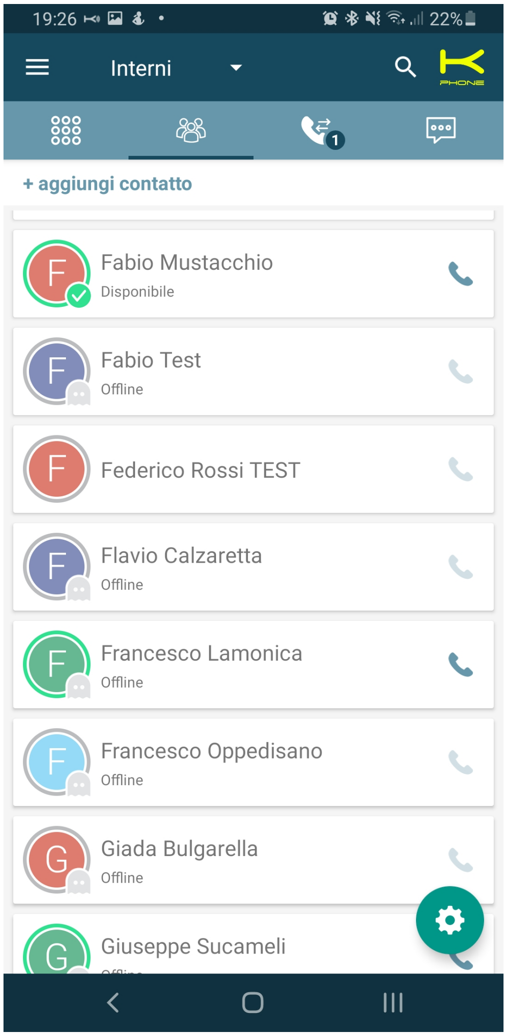
Contacts

The Contacts tab (second icon from the left) shows all KalliopePBX contacts, both extensions and phonebooks, with favorite contacts at the top.

Extensions can be identified by the presence of a BLF field (colored dot) and the chat presence (icon and status).

Clicking on the various contacts will take you to the detail screen containing the following items:

- Contact details
- Contact First and Last Name
- Origin of contact (Kalliope or phonebook)
- List of phone numbers associated with the contact with icons for the various calling modes available



For more information on the various calling modes, see the call management section of this guide.

By tapping the tab title (default: Contacts) you can filter the phonebook by:

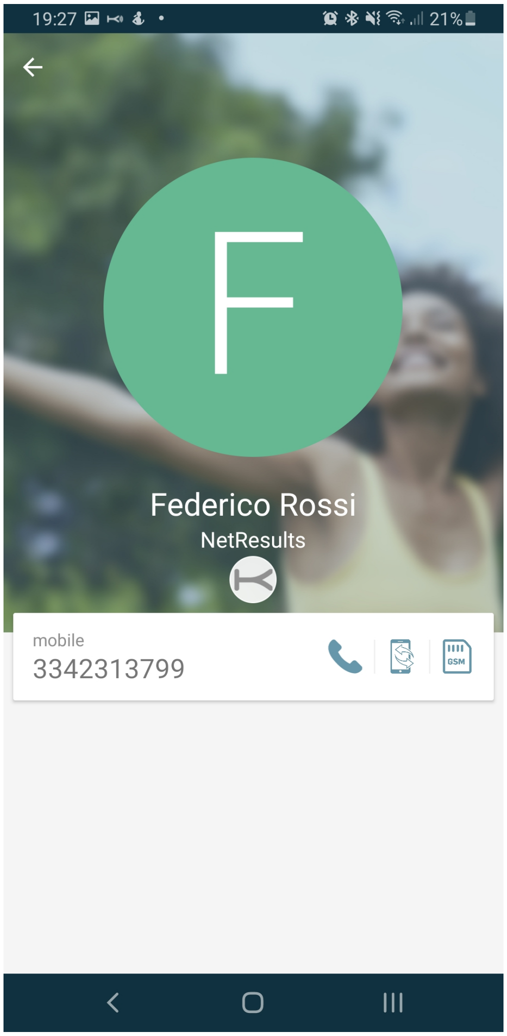
- Contacts: show all contacts
- Extensions: only show Kalliope extensions
- Phonebook: only show Kalliope and device contacts

For extensions, presence and BLF status will also be displayed.

The BLF status can be:

- Green: available
- Yellow: ringing
- Red: busy
- Gray: not registered

From this tab, you can also add the contact to your favorites by tapping the Kcti favourite icon in the top right. Favorite contacts will be displayed at the top of the Contacts list.



Add contacts

By clicking on the button "+ add contact" placed immediately below the top-level bar (header) it is possible to insert a new contact by filling in the form shown in the figure. The created contact will be inserted in the user's personal Kalliope address book.

Call history

This tab displays the call history of the logged-in user.

The calls are divided in:

- Made calls
- Answered calls
- Missed calls

Calls are grouped based on the called or calling contact/number, and the total number of calls is shown in parentheses.

Tapping each row will display all calls to or from the selected contact, as shown in the figure.

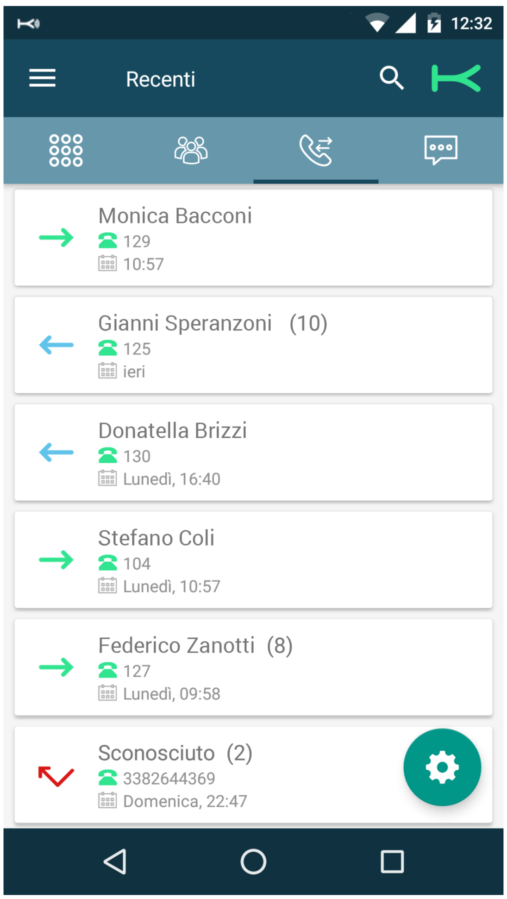
A notification will be displayed if there are any voicemail messages; tapping the notification will open the voicemail box, where you can listen to and manage messages.

The call detail by contact includes all calls to and from the contact in chronological order.

For each call, the following information is displayed:

- Type (made, answered, missed)
- Date
- Time
- Duration

You can also call the contact with the available modes based on the type of contact (extension or external number).



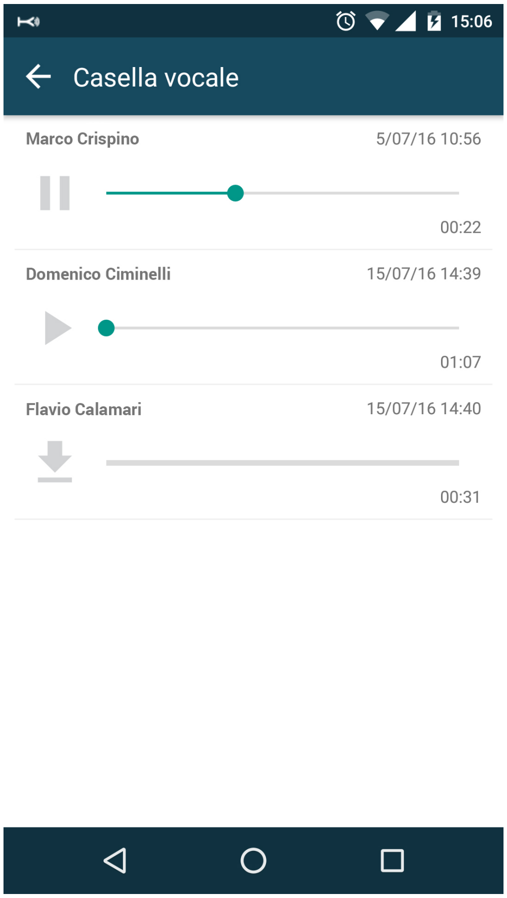
Voicemail

You can access your voicemail from the system menu to quickly and easily access all messages saved on the PBX or remote storage.

For each voicemail message sender, date, time, and length are displayed.

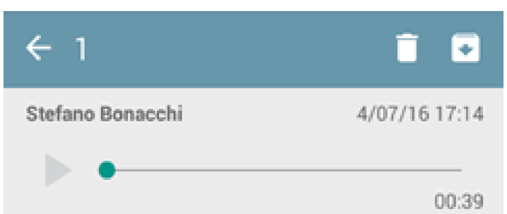
The messages are not automatically saved to the device but can be downloaded by tapping the icon next to each message.

Once downloaded, you can play the message directly on the device.



The app also lets you mark messages as read, unread, or urgent and delete messages from local storage, remote storage, or both, by tapping the trash icon.

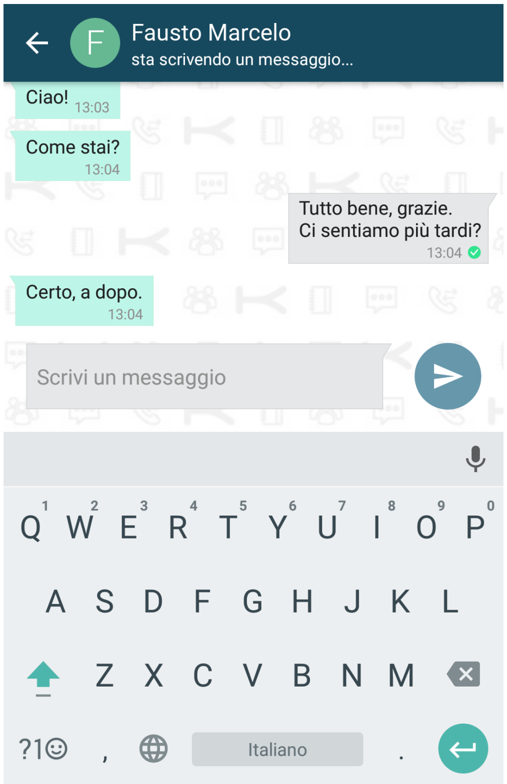
To access the message management menu, tap and hold a message for a few seconds and the management options will appear.



Note: Messages forwarded via email and automatically deleted will not be shown on this page.

Instant Messaging

KalliopeCTI offers a handy chat and presence management service that lets you easily talk to other Kalliope users. To open a new chat, you can tap on the desired contact, then on the chat icon.



Within the chat, KalliopeCTI displays the status of sent messages with the following icons:

- Not sent
- Sent
- Delievered

Note: Chat messages that could not be sent because CTI is offline will be sent upon the next login.

On Android, the presence status can be changed from any tab by tapping the gear icon then the presence service icon (first from the top). On iOS, the status can be changed directly from the Chat tab.

You can also set a custom message to be displayed along with the presence.

The possible statuses are:

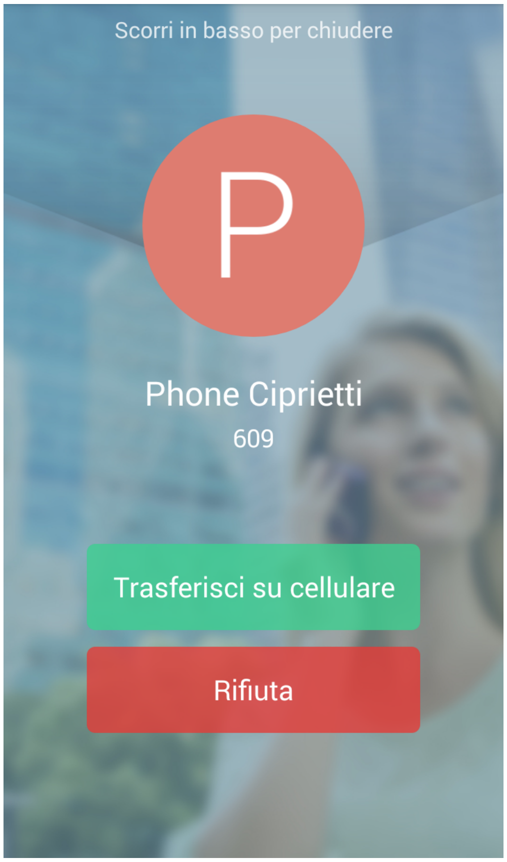
- Available
- Away
- Busy
- Not available
- Offline

Call management

Inbound calls in in CTI mode (call-back)

For inbound calls to the extension, it will send a notification to the mobile app and the screen shown in the picture to the right will be displayed.

As they use the official notification engine of their operating systems, these notifications will be sent even when the app is closed.



This screen allows you to perform three actions:

- **Slide down**: the call is ignored, the notification on the mobile is muted and the user's extension continues ringing
- **Transfer to mobile**: This button is only present if a mobile number has been entered in the user's settings on Kalliope. The user's extension stops ringing and at the same time Kalliope establishes a call to the mobile number associated with the extension by the system administrator in the extensions. configuration panel. The call will then arrive on the mobile's SIM showing the geographic number of the line associated with KalliopePBX as the sender. By answering this call, you are directly connected to the caller. This call forwarding is entirely transparent to the caller, who will hear the ringing tone during the whole call establishment period.
- **Reject**: the call is dropped by the PBX and, consequently, the extension stops ringing.

Incoming calls in Softphone mode

In case of incoming calls to the extension, KalliopePBX will send a push notification that will trigger the incoming call screen on the mobile like the one shown in the figure. These notifications arrive even when the app is turned off, ensuring that the user can be reached even when the smartphone is on standby.

This screen lets you select one of two options:

- Refuse: the PBX will refuse the call and the extension will stop ringing.
- Accept: the KalliopeCTI app will wake up in softphone mode, the “Active call” screen will be displayed, and the SIP call will be established directly via the app. The smartphone will effectively become an extension of KalliopePBX.

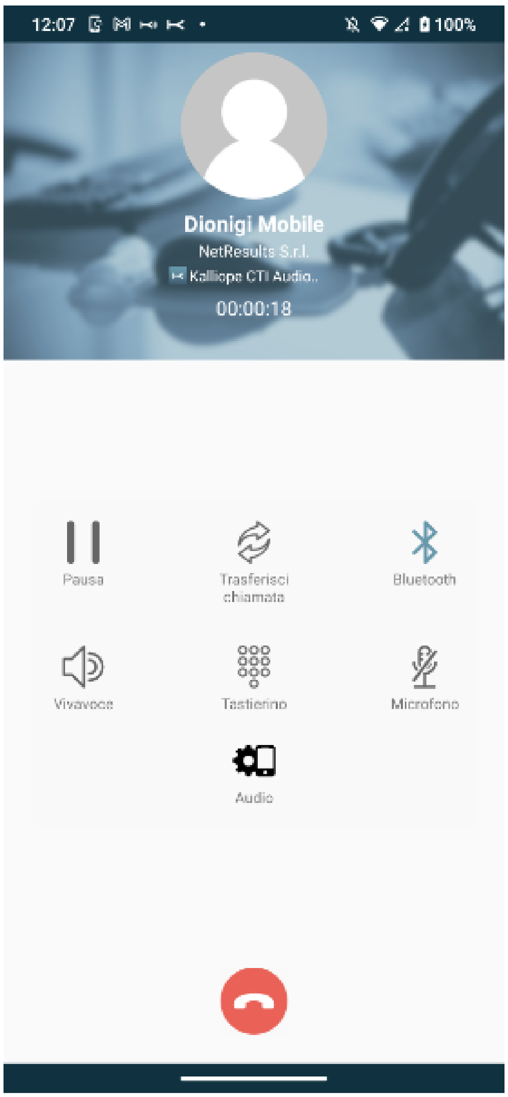
The Call in progress screen shows the name and number of the caller at the top, and at the bottom the buttons for the features available on the current call:

- Pause: the call is put on hold. The button can also be used to resume the call from the pause.
- Call transfer: blind transfer of the call to another number (internal or external).
- Bluetooth: activate/deactivate bluetooth headset.
- Speakerphone: enable/disable the speakerphone of the smartphone.
- Keypad: activates the keypad screen (e.g. to navigate a IVR menu)
- Mute: enable/disable smartphone’s microphone

Call transfer (with offer)

Starting with version 4.8.0, the transfer mode has been replaced from "blind" to "with offer" when connected to exchanges with firmware higher than 4.13.0. Call transfer can only start when the call has been established and is in progress. After clicking on the "call transfer" button, the user will be presented with a popup from which he can choose how to insert the contact to which the call should be transferred

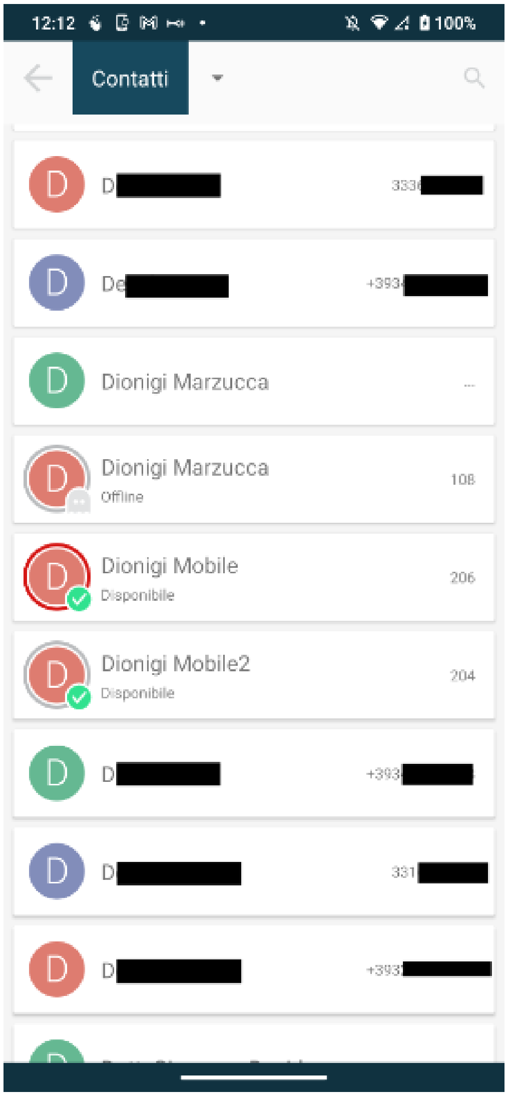
- Address book
- Keypad



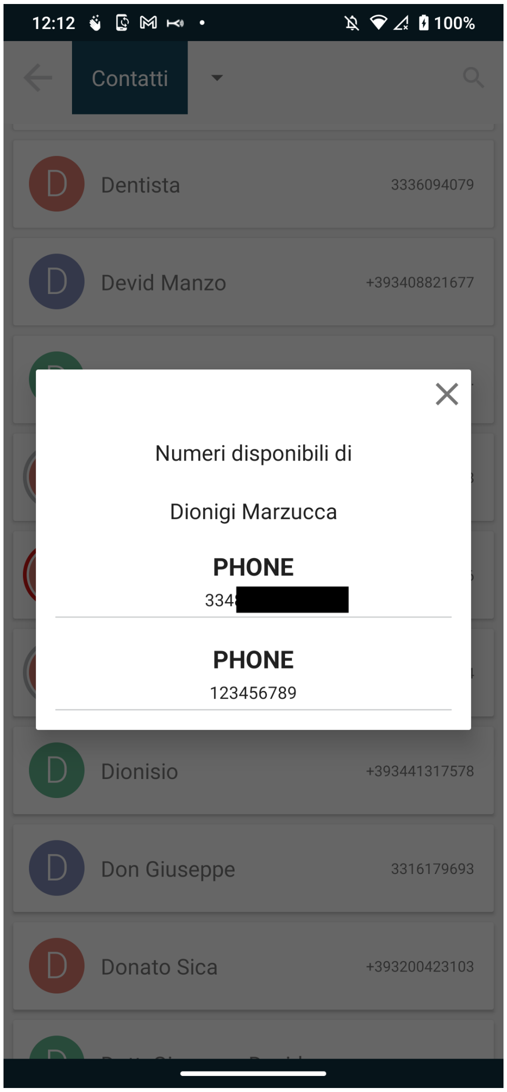
Address Book

In this case, the contact can be selected from the Kalliope address book. This screen is equivalent to the address book already present in the app.

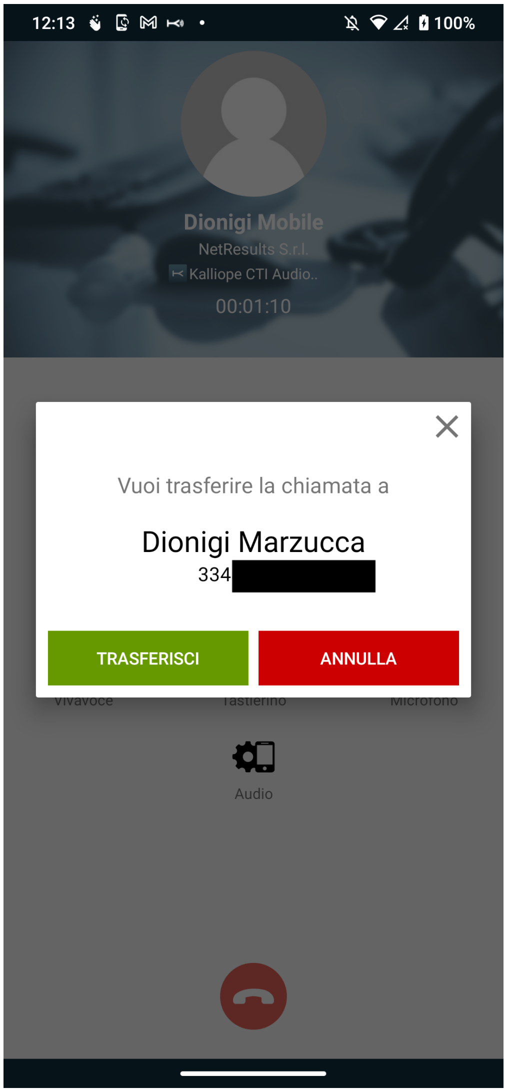
The contact search can also be carried out using the search bar at the top. Compared to the address book search, the main difference of the presented list, is represented by the presence of the contact number on the right of it (if it has only one number) or by three dots (in case of multiple numbers associated to the contact).



If you press on a contact with multiple numbers, a screen opens where you can choose from the available ones. In this screen you have information about the contact you want to transfer the call to (name and number)

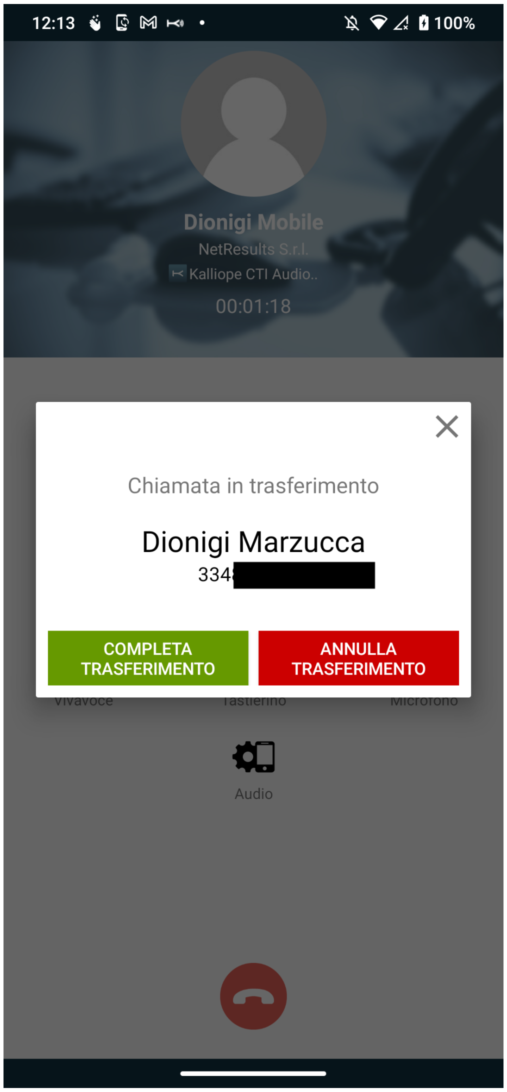


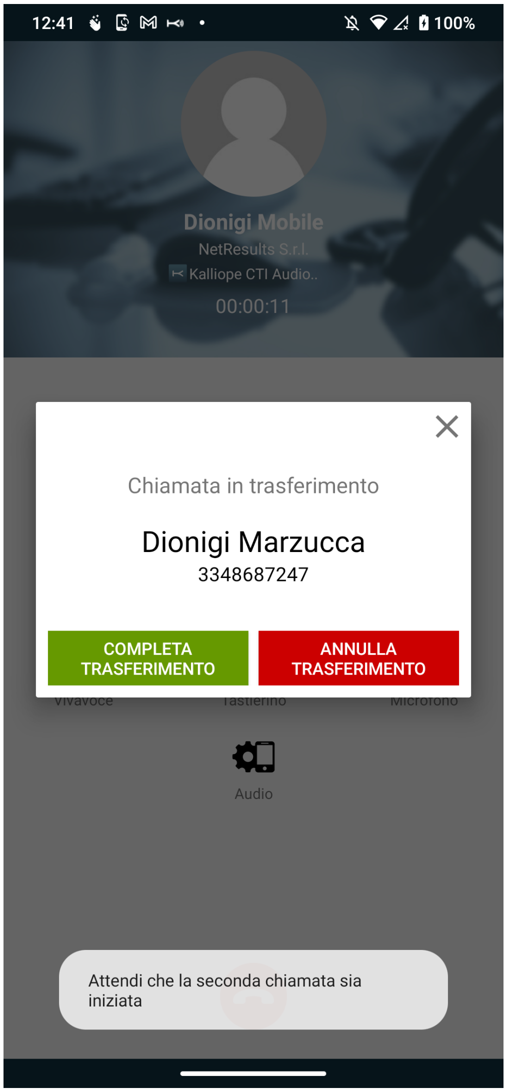
At this stage, you can choose whether to transfer the call (green button with the word Transfer) or cancel the transfer (red button with the word Cancel).



In this in-progress transfer screen, you have the option to complete the transfer (green button) or cancel the transfer and return to the current call (red button).

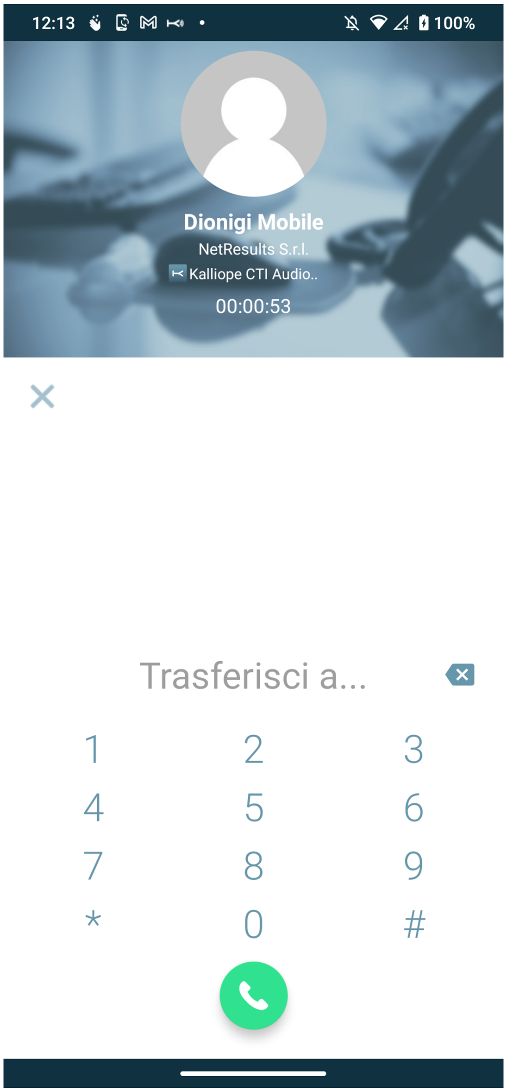
Note: If the green button is pressed before the transfer has started, you will receive a notice to wait for the transfer to start (You must wait for the transfer to start).

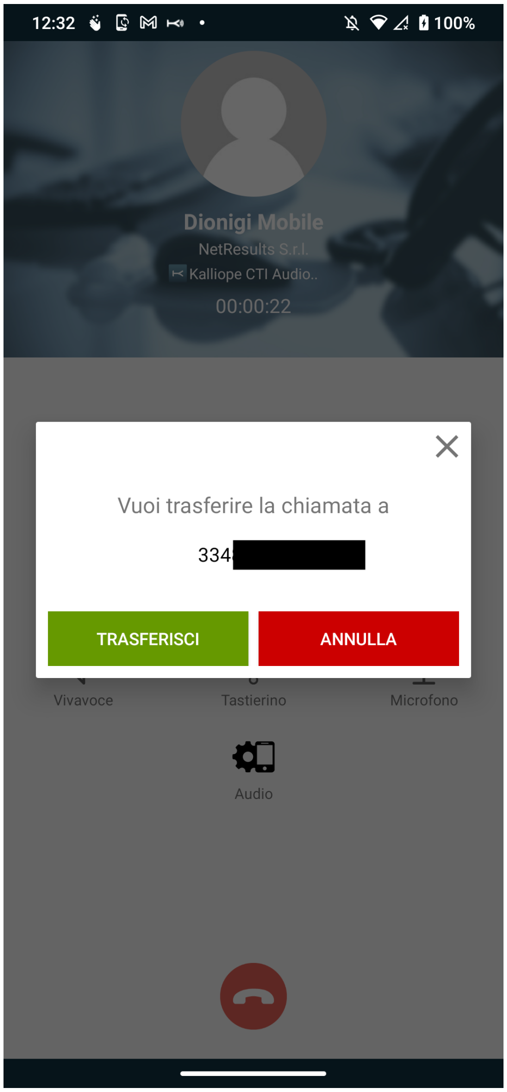

12:13
100%
Dionigi Mobile
NetResults S.r.l.
Kalliope CTI Audio..
00:01:18
Chiamata in trasferimento
Dionigi Marzucca
334[redacted]
COMPLETA TRASFERIMENTO
ANNULLA TRASFERIMENTO
Audio



Numeric Keypad

In this case, the contact number can be entered directly from the keypad and the call can be transferred. After entering the number and pressing the green button, you will get to the Contact Information Screen, but you will see only the number and not the name of the contact (figure below).


12:13
100%
Dionigi Mobile
NetResults S.r.l.
Kalliope CTI Audio..
00:00:53
Trasferisci a...
1 2 3
4 5 6
7 8 9
* 0 #



Outgoing calls in CTI mode

Regarding outgoing calls in CTI mode, KalliopeCTI mobile app offers three different types of call setups.

Call-back

The Call-back service allows you to call from your mobile device using the KalliopePBX's lines. By pressing this key, the KalliopePBX calls the mobile number associated with the extension, answering and pressing key 1 (as requested by the guiding voice), you are communicating with the desired number. The caller will see a call coming from the geographical number assigned to the Kalliope PBX. Questa is the type of outgoing call setup used by default in CTI mode.

Click-to-call

Click-to-call mode: the stationary device associated with the extension receives a call from KalliopePBX, whose caller is c2c: called number. Answering this call will initiate a new call to the desired number.

Using directly the SIM of the device

In this mode, the call is made directly from the SIM of the device on which the KalliopeCTI Mobile app is installed. A simple direct GSM call is then established and then the caller will see the user's mobile number as the calling number.

Outgoing calls in Softphone mode

Regarding outgoing calls in Softphone mode, KalliopeCTI mobile app offers the following three types of call setups.

SIP call

SIP call: outgoing calls leave directly from the app via SIP protocol using the data network provided by the softphone. It is fully equivalent to a call originating from the KalliopePBX extension associated with the user.This is the type of outbound call setup used by default in Softphone mode.

Call-back

The Callback call service allows you to call from your mobile device using the KalliopePBX's lines. By pressing this key, the KalliopePBX calls the mobile number associated with the extension, answering and pressing key 1 (as prompted by the guiding voice), you are actually put in communication with the desired number. The caller will see a call coming from the geographical number assigned to the Kalliope PBX.

Using directly the SIM of the device

In this mode, the call is made directly from the SIM of the device on which the KalliopeCTI Mobile app is installed. A simple direct GSM call is then established and then the caller will see the user's mobile number as the calling number.

Dialing a number directly from the keypad will establish a call via SIP. Once the call is established, the "Active call" screen will automatically be displayed.

Keypad

Aside from searching the phonebook, you can also dial the number you wish to call with Kalliope directly on the KalliopeCTI mobile app keypad. While dialing, you will be shown autocomplete[1] suggestions.

In call-back mode: Tapping the call button lets you choose which mode to use among the ones listed above.

In softphone mode: Tapping the call button will directly establish a SIP call.

Note: The KalliopeCTI keypad does not automatically add the outbound prefix. You will need to dial the number as you would on a landline phone.

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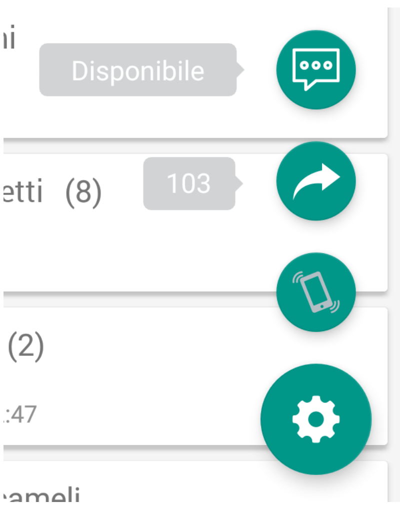
[1] Feature only available on Android.

Services

On Android, you can access the KalliopeCTI app services by tapping the gear icon in the lower right. On iOS, you can access them via the system menu.

Three services are currently available: presence (seen above), unconditional forward, and fork to mobile.



Unconditional Forwarding

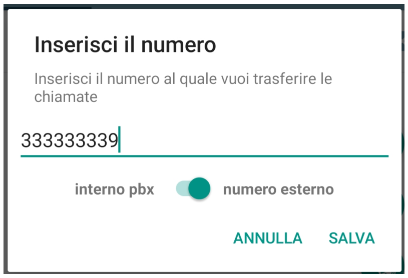
Unconditional forwarding allows you to configure an extension number to which all incoming calls are automatically forwarded (this function is valid only for calls directed to the extension and not for those coming from groups or queues).

This service is helpful when you are not available but wish to redirect inbound calls so they can be dealt with.

To enable the unconditional forward service, tap the arrow icon, input the extension number you wish to forward calls, and save.

When this service is enabled, the icon will change color, and a notification will be shown containing the number of calls are being redirected to.

To disable this service, tap on the arrow icon again.



Fork to mobile

Fork to mobile consists of distributing the direct call to an extension and the associated mobile number. This function can be set on Kalliope only by the administrator and not by the user.

Answering the call on the mobile phone will cause the extension to stop ringing, and vice versa.

To enable the service, tap its icon. As with unconditional forward, the color of the icon indicates the status of the service.

Privacy Policy

Secure connection

To offer its services (e.g. telephony, real-time notifications, chat), KalliopeCTI must be able to communicate with KalliopePBX. This communication is made through an encrypted connection, guaranteeing the security of transmitted personal data and messages.

Contacts

You can give KalliopeCTI permission to access the contacts on your device. The saved contact data will only be used within the app and will never be sent to KalliopePBX or exchanged with other users, except for the information required to make calls via the PBX.

Files

KalliopeCTI will access the phone's memory to send diagnostic data to the KalliopePBX developers via email. This data will only be sent when the user selects the "Send report" item in the KalliopeCTI menu.

Kalliope CTI Mobile IOS

Changelog iOS

KalliopeCTI Mobile App Version 4.9.3

Bugfix:

- Fixed bug where it was not possible to transfer/pause a received call with the device in standby.
- Fixed bug that caused the echo cancel software to be activated even in non-handsfree mode.

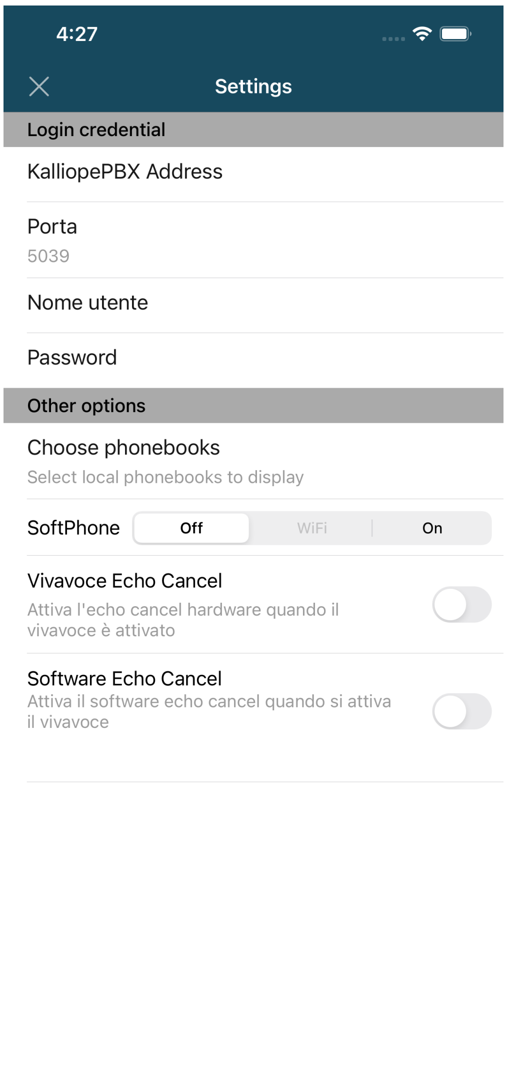
- Fixed graphics on audio boost in call

KalliopeCTI Mobile App Version 4.9.2

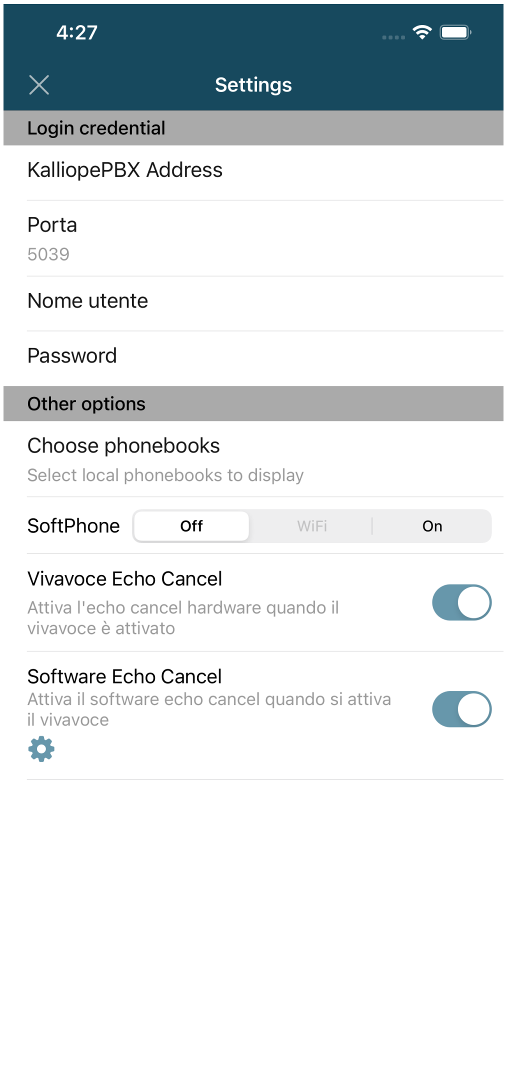
Features:

- Added panel for configuring echo cancellation algorithm parameters.

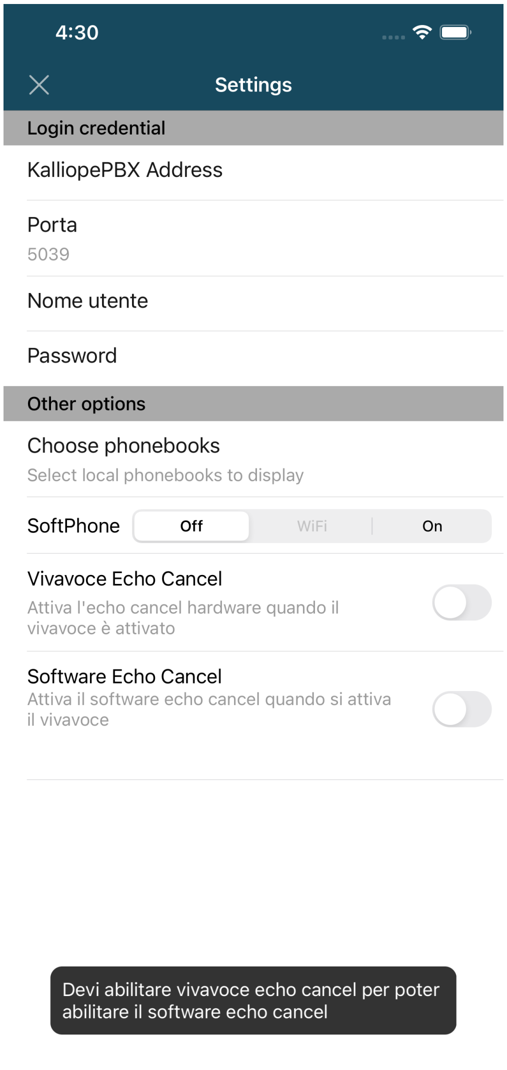
Starting from version 4.9.0 it is possible to enable/disable the hardware echo cancellation when the hands-free mode is active. This feature can be enabled in the Settings panel of the application.



Starting from version 4.9.1 it is also possible to enable, in addition to the hardware echo cancel, a software echo cancel algorithm implemented in the app. Also this feature is enabled in the Settings panel of the application and the algorithm parameters are configurable by clicking on the gear under the label of the item to be enabled.


4:27
Settings
Login credential
KalliopePBX Address
Porta
5039
Nome utente
Password
Other options
Choose phonebooks
Select local phonebooks to display
SoftPhone
Off
WiFi
On
Vivavoce Echo Cancel
Attiva l'echo cancel hardware quando il vivavoce è attivato
Software Echo Cancel
Attiva il software echo cancel quando si attiva il vivavoce

The software echo cancellation feature requires that hardware echo cancellation is also enabled.



Bugfix:

- Fixed call widget display issue on IOS 15.1.
- Minor graphic fixes on IOS 15.1

KalliopeCTI Mobile App Version 4.9.1

Features:

- Added echo cancellation functionality on hands-free speaker.

KalliopeCTI Mobile App Version 4.9.0

Features:

- Transfer mode has been replaced from “blind” to “with offer” when connected to exchanges with firmware greater than 4.13.0. Call transfer can only start when the call has been established and is in progress. For more details on this new feature Click here
- Speakerphone can now be activated when starting a call and not only when it is in progress

Bugfix:

- Fixed an issue where the app kept ringing if a group call was answered elsewhere.

KalliopeCTI Mobile App Version 4.8.2

Features:

- Added an action to refresh the cdr list manually. Swipe-to-refresh on CDR list that will download the updated call list from PBX

Bugfix:

- The app notifies the PBX of the rejected call when the app is forcibly closed.
- In cti mode, the speed dial icon calls click.to.call to the deskphone instead of the cell phone
- Speaker has the same low volume when activated from the default iOS panel, instead of the kcti screen
- When a second VoIP call ends, the current one is terminated
- Try changing the numbering index of iCloud accounts to resolve duplicate accounts

KalliopeCTI Mobile App Version 4.8.1

Features:

- Improved PLC algorithm that increases call quality in case of limited packet loss events.

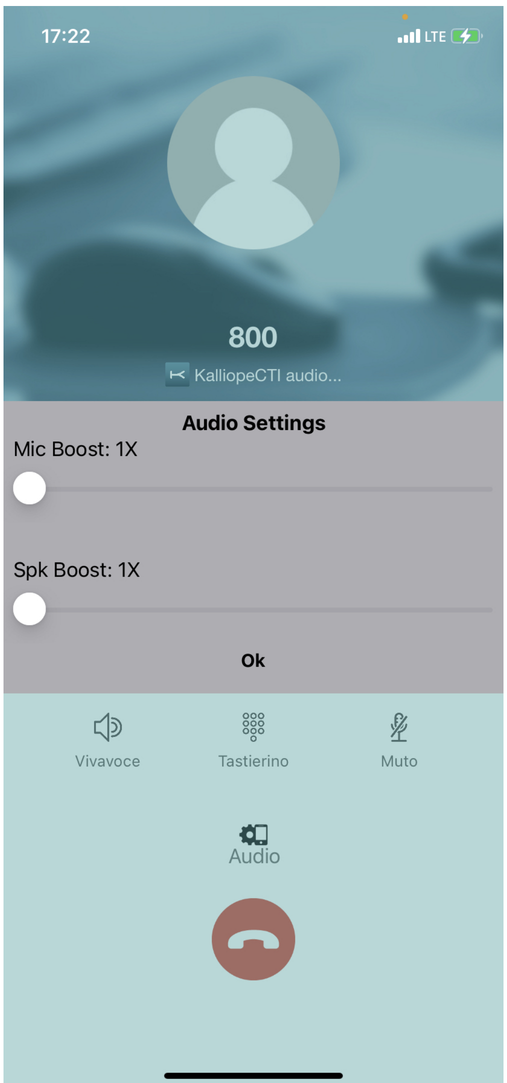
Bugfix:

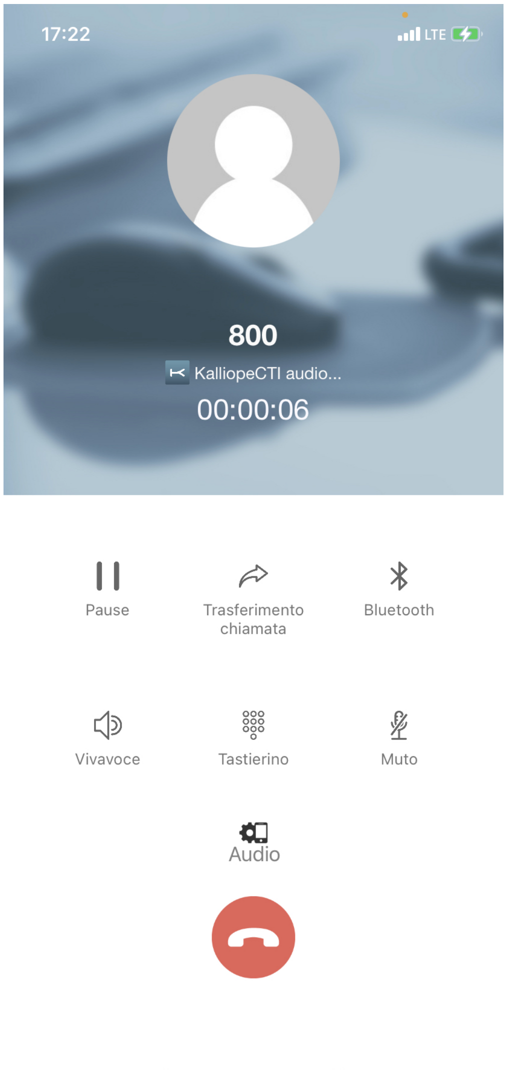
- Increased the speakerphone volume.
- Fixed an issue where the Bluetooth headset (if connected) would not automatically activate during a call

KalliopeCTI App Mobile Version 4.8.0

Features:

- Introduced Siri / Carplay support, now you can use the app by saying:
  - Hey Siri call using Kalliope / KalliopeCittii / KalliopeSittiai
  - Hey Siri call the number using Kalliope / KalliopeCittii / KalliopeSittiai
  - Hey Siri use Kalliope / KalliopeCittii / KalliopeSittiai to call
  - Hey Siri use Kalliope / KalliopeCittii / KalliopeSittiai to call the number
- Added a popup panel to boost the audio signal of microphone and speaker





Bugfix:

- Fixed an issue where it was impossible to use central addresses with very long domain names.
- Fixed an issue where in the cdr display the calls were always identified as incoming.
- Fixed an issue where the lookup of the address book did not recognize the number if it was presented without country-code and was stored with CC instead (and vice versa).

For iOS operating systems

It is no longer possible to select the mode to run the app while not connected to Kalliope (due to a restriction by apple on sending notifications). For the same reason the option to use softphone mode only if the phone is connected to wifi has been disabled, KalliopePBX needs to know how to send notifications, but can't know if the phone is in wifi coverage or not.

We also added the possibility (starting from KalliopePbx firmware 4.7.12 or higher) to add, modify, delete a contact from the KalliopePBX personal address book directly from the app: pressing the label "+ Add Contact" opens a new screen. Once the fields are filled, the added/modified contact will appear in the contact list with a loading symbol. This symbol will automatically disappear as soon as Kalliope notifies the app that the changes have been accepted and registered on the pbx.

iOS Specifications:

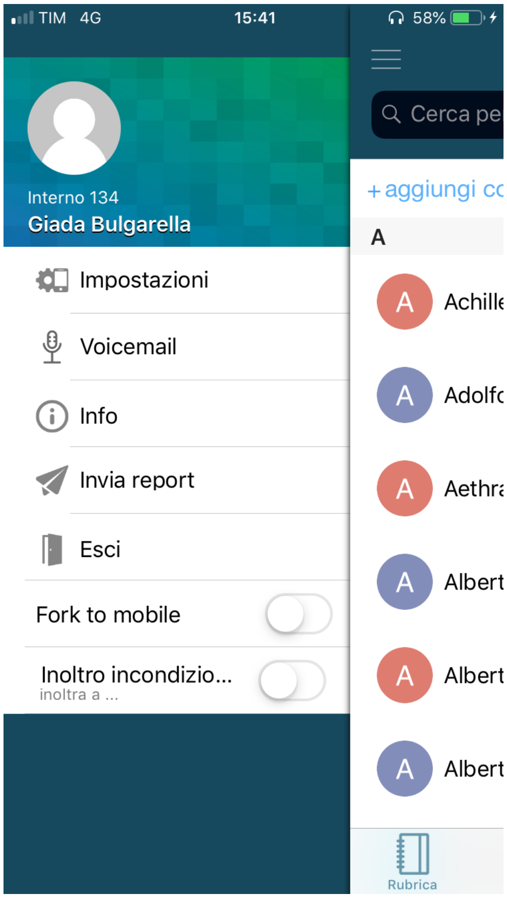
- Requires iOS 10.3 or later. Compatible with iPhone, iPad, and iPod touch." -> app does not support iPad.
- Devices using iOS 13 and higher must connect to kalliope 4.10.x or higher for proper operation.
- Languages: Italian, English
- Product available only if associated with a KalliopePBX® V4 VoIP PBX
- Minimum KalliopePBX firmware version: 4.10.x
- Suggested KalliopePBX firmware version: 4.11.x
- Integration of the native iOS CallKit libraries to ensure that the softphone can always be reached even when the terminal is on standby.
- A KalliopeCTI Pro license is required for the use of GSM™ mode only. A KalliopeCTI Phone license is required to use both modes, GSM™ and Softphone.

Note: Softphone mode is in beta version and currently has the following limitations that will not be present in the final, upcoming version:

- It is not possible to handle more than one SIP call at a time
- Bluetooth support needs to be refined
- SIP/TLS has not been enabled yet
- Resuming a paused call may sometimes fail

Organization of the GUI

When you first start the KalliopeCTI app, you need to go and set the configuration parameters to allow the app to connect to KalliopePBX.

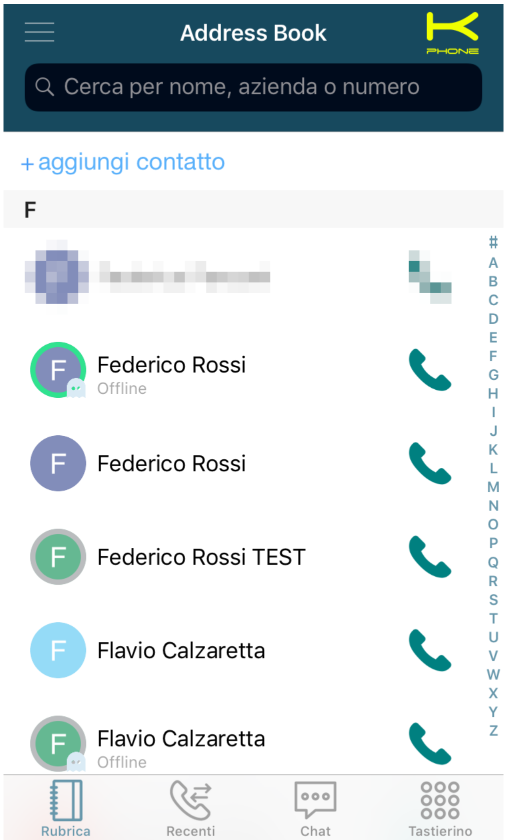


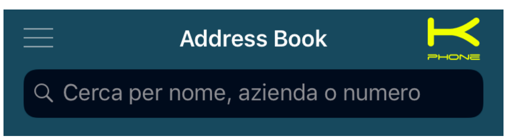
By clicking on the menu button symbol at the top left you can access the system menu containing the following items:

- Settings: application configuration panel
- Softphone settings: allows to set the mode of the application (iOS only)
  - Softphone disabled: the app forcibly works in call-back mode. In this case the KalliopeCTI Pro license is sufficient.
  - Softphone enabled: the app is forced to work in softphone mode. In this case, the KalliopeCTI Phone license is sufficient.
  - Softphone enabled on WiFi only: the app will automatically switch to softphone mode if the connection is established with a WiFi network. Switching to softphone mode will only be possible if the KalliopePBX user is associated with a KalliopeCTI Phone license.
- Voicemail: direct access to voicemail messages
- Info: provides information about the installed version
- Send report: allows you to send a report to the developers in case of abnormal behavior of the app
- Exit: closes the app and you will not receive any more notifications until the next time you open again the app.

The main screen of the application is shown here and consists of a top bar.

The top level bar shows the name of the tab you are currently viewing (in this case Address Book). Allows you to search within each tab and shows the connection status of the app via the K symbol in the top right corner. The top level bar (header) for iOS is shown below.

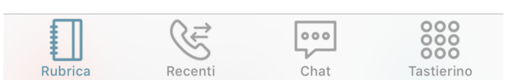




The connection status is indicated by the color of the K symbol in the upper right corner:

- gray = not connected
- green (with the word "cti") = connected in call-back mode.
- yellow (with the word "phone") = connected in softphone mode.

The footer includes the navigation bar between the main sections of the app.



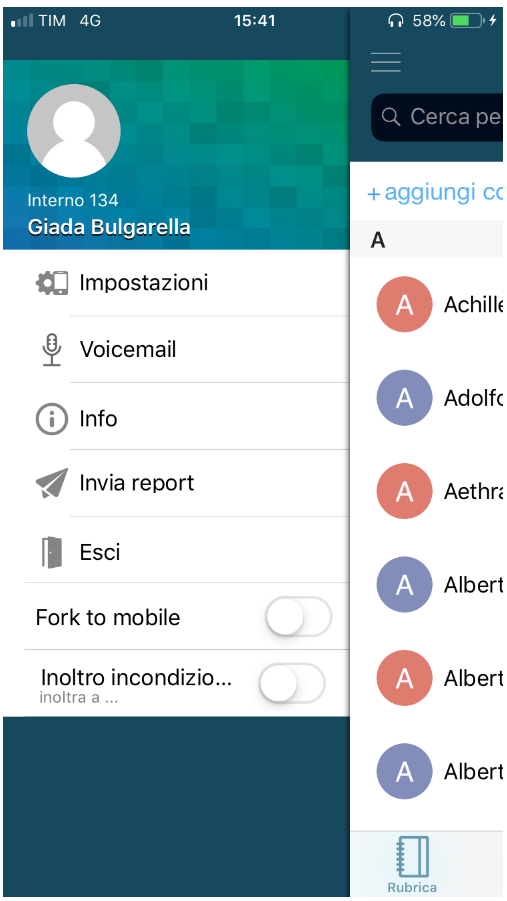
In both cases, the navigation bar allows you to explore the main tabs of the app:

- Keypad: the numeric keypad for dialing numbers
- Phonebook: extensions, KalliopePBX phonebook, favorite contacts, and, if enabled, personal contacts of the device in use
- Call History: list of dialed, received and missed calls grouped by contacts
- Chat: conversations started with other KalliopeCTI users, both mobile and desktop.

Settings Panel

First of all you need to configure some simple parameters from the Settings panel reachable through the system menu:

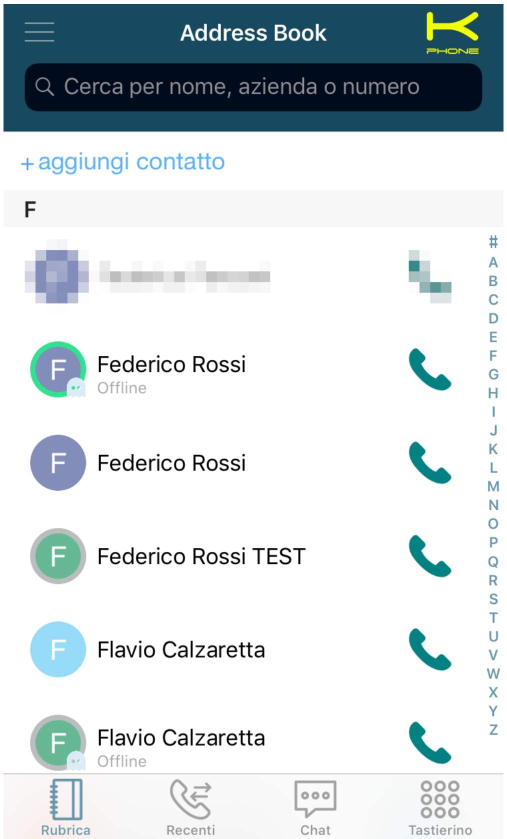
- Login credentials
  - Server (Host name or IP address): IP address of the KalliopePBX central unit
  - Port: 5039
  - User name GUI/CTI user name
  - Password of the GUI/CTI user
- Calls
  - Ringtone: useful to differentiate the incoming calls to the mobile number from the incoming calls to the extension
  - Display device contacts: allows to choose the address books from which to display contacts.



Contacts

The Contacts screen, second icon from the left, shows all KalliopePBX contacts, both extensions and address books, placing favorite contacts at the top.

The extensions are recognizable by the presence of the BLF field (colored circle) and the chat presence (icon and status).



Clicking on the various contacts will take you to the detail screen containing the following items:

- Name and Surname of the contact
- Origin of the contact (Kalliope or phonebook)
- List of phone numbers associated with the contact with icons for the different calling modes available

Please refer to the Call management section of this manual for more information about the different call modes.

By pressing on the title of the screen (default Contacts) you can filter the address book as follows:

- Contacts: shows all contacts
- Extensions: shows only the extensions of Kalliope
- Address Book: shows only the contacts of Kalliope address book and of the device

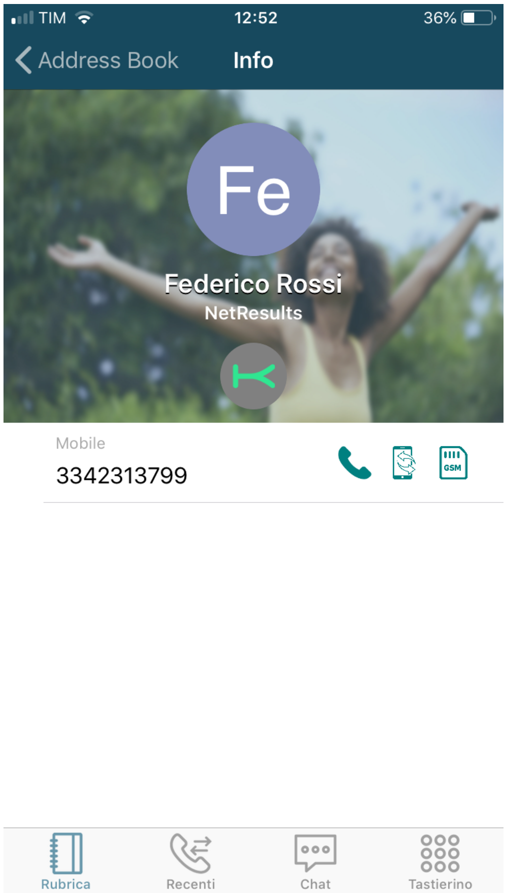
For extensions only, presence and BLF status are also displayed.

The BLF status can be:

- Green: available
- Yellow: ringing
- Red: busy

- Gray: not registered

From this tab you can also add the contact to you favorites by tapping the Kcti favourite button.jpg icon in the top right. Favorite contacts will be displayed at the top of the Contacts list.



Add contacts

By clicking on the button "+ add contact" placed immediately below the top level bar (header) it is possible to insert a new contact by filling in the form shown in the figure. The created contact will be inserted in the user's personal Kalliope address book.

Call history

This tab displays the call history of the logged-in user.

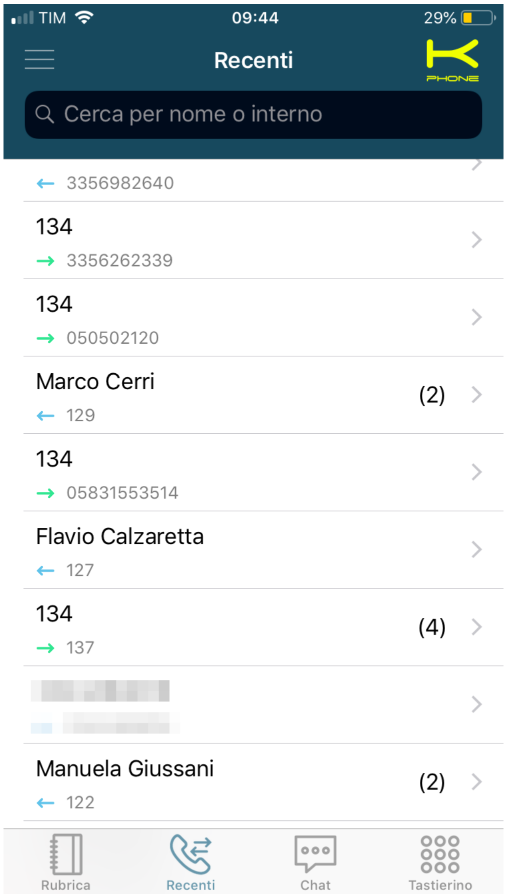
The calls are divided in:

- Made calls
- Answered calls
- Missed calls

Calls are grouped based on the called or calling contact/number, and the total number of calls is shown in parentheses.

Tapping each row will display all calls to or from the selected contact, as shown in the figure.

A notification will be displayed if there are any voicemail messages; tapping the notification will open the voicemail box, where you can listen to and manage messages.



The call detail by contact includes all calls to and from the contact in chronological order.

For each call, the following information is displayed:

- Type (made, answered, missed)
- Date
- Time
- Duration

You can also call the contact with the available modes based on type of contact (extension or external number).

Voicemail

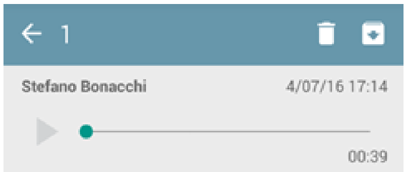
You can access your voicemail from the system menu to quickly and easily access all messages saved on the PBX or a remote storage. For each voicemail message sender, date, time, and length are displayed.

The messages are not automatically saved to the device but can be downloaded by tapping the icon next to each message.

Once downloaded, you can play the message directly on the device.

The app also allows you to manage the status (read, unread, urgent) and the deletion of messages. Messages can be deleted from local storage, remote storage or both directly from the app by clicking on the trash symbol.

To access the message management menu, simply tap and hold a message for a few seconds and the management options will appear, as shown in figure.



Note: Messages forwarded via email and automatically deleted will not be shown on this page.

Instant Messaging

KalliopeCTI offers a handy chat and presence management service that lets you easily talk to to other Kalliope users. To open a new chat, you can simply tap on the desired contact, then on the chat icon.

Within the chat, KalliopeCTI displays the status of sent messages with the following icons:

- Not sent
- Sent
- Delievered

Note: Chat messages that could not be sent because CTI is offline, will be sent upon the next login.

The presence status can be changed directly in the Chat tab.

In addition to the status, the user can also enter a custom message that will be displayed along with the presence.

The possible statuses are:

- Available
- Away
- Busy
- Not available
- Offline

Call management

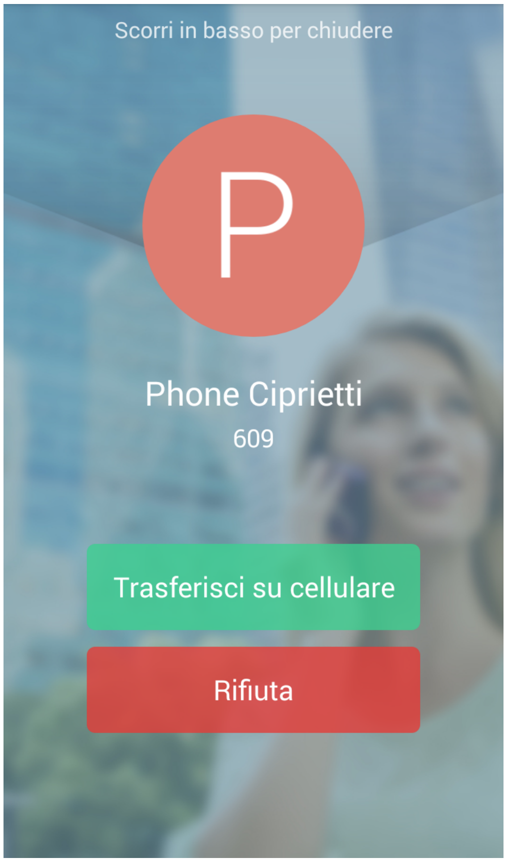
Inbound calls in CTI mode

For inbound calls to the KalliopePBX extension, will send a notification to the mobile app and the screen shown in the picture to the right will be displayed.

As they use the official notification engine of their operating systems, these notifications will be sent even when the app is closed.

This screen allows you to perform three actions:

- Slide down: the call is ignored, the notification on the mobile is muted and the user's extension keeps ringing
- Transfer to mobile: This button is only present if a mobile number has been entered in the user's settings on Kalliope. The user's extension stops ringing and at the same time Kalliope establishes a call to the mobile number associated with the extension by the system administrator in the Extensions configuration panel. The call will then arrive on the mobile's SIM showing the geographic number of the line associated with KalliopePBX as the sender. By answering this call you are directly connected to the caller. This call forwarding is completely transparent to the caller, who will hear the ringing tone during the whole call establishment period.
- Reject: the call is dropped by the PBX and, consequently, the extension stops ringing.

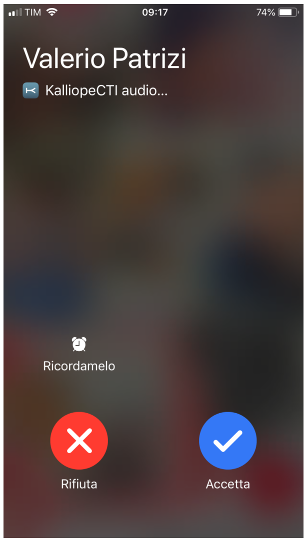


Incoming calls in Softphone mode

In case of incoming calls to the extension KalliopePBX will send a push notification that will trigger on the mobile the incoming call screen like the one shown in the figure. These notifications arrive even when the app is turned off, ensuring that the user can be reached even when the smartphone is in standby.

This screen lets you select one of three options:

- Refuse: the PBX will refuse the call and the extension will stop ringing.
- Accept: the KalliopeCTI app will wake up in softphone mode, the “Active call” screen will be displayed, and the SIP call will be established directly via the app. The smartphone will effectively become an extension of KalliopePBX.



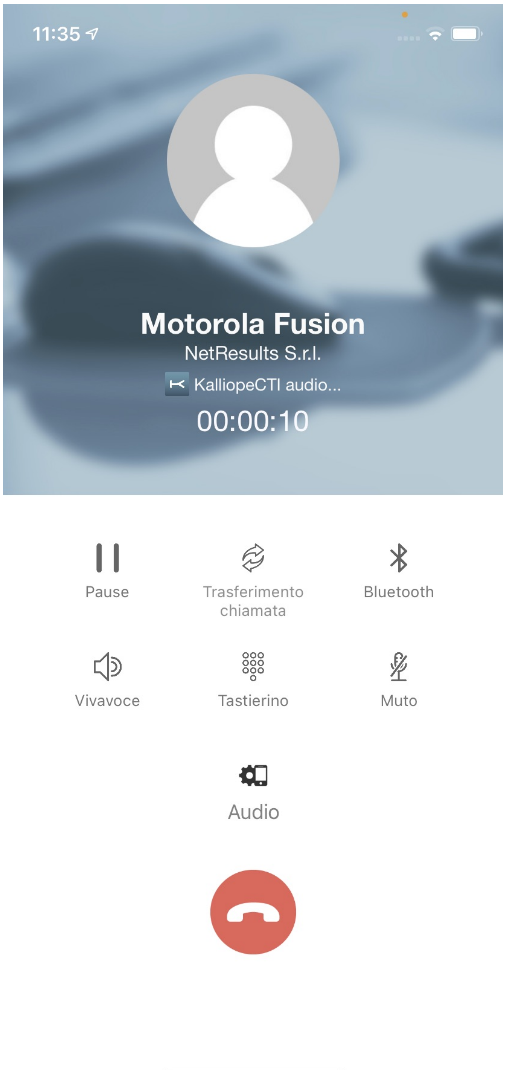
The Call in progress screen shows the name and number of the caller at the top, and the buttons for the functions available on the current call at the bottom:

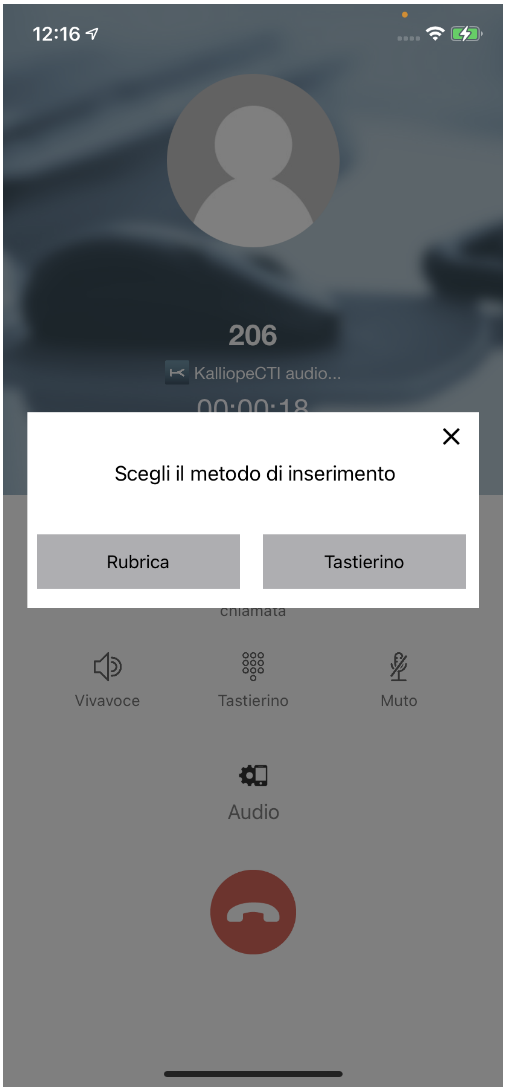
- Pause: the call is put on hold. The button can also be used to resume the call from the pause.
- Call transfer: blind transfer of the call to another number (internal or external).
- Bluetooth: activate/deactivate bluetooth headset.
- Speakerphone: enable/disable the handsfree of the smartphone.
- Keypad: activates the keypad screen (e.g. to navigate a IVR menu)
- Mute: enable/disable smartphone's microphone

Call transfer (with offer)

Starting with version 4.9.0, the transfer mode has been replaced from “blind” to “with offer” when connected to exchanges with firmware higher than 4.13.0. Call transfer can only start when the call has been established and is in progress. After clicking on the “call transfer” button, the user will be presented with a popup from which he can choose how to insert the contact to which the call should be transferred

- Address book
- Keypad

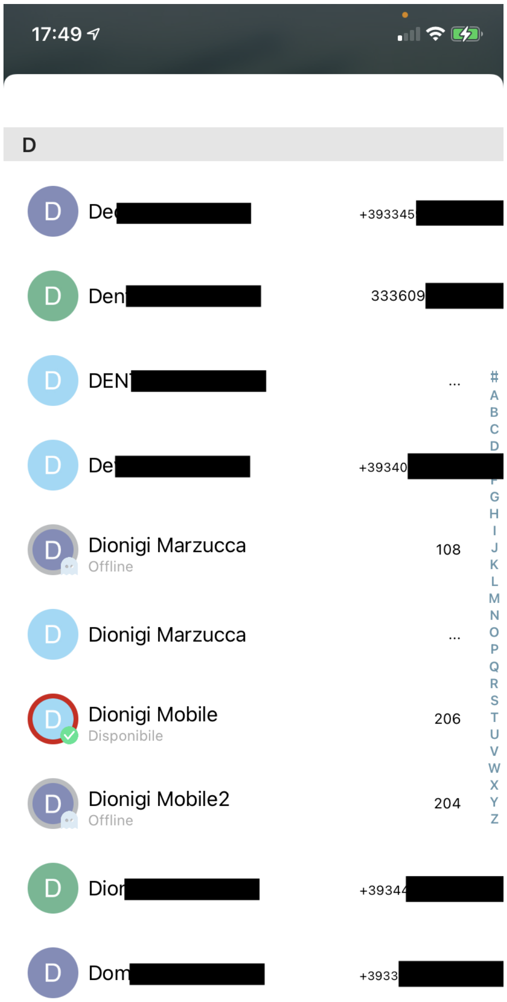

11:35
Motorola Fusion
NetResults S.r.l.
KalliopeCTI audio...
00:00:10
Pause
Trasferimento chiamata
Bluetooth
Vivavoce
Tastierino
Muto
Audio



Address Book

In this case the contact can be selected from the Kalliope address book. This screen is equivalent to that of the address book already present in the app.

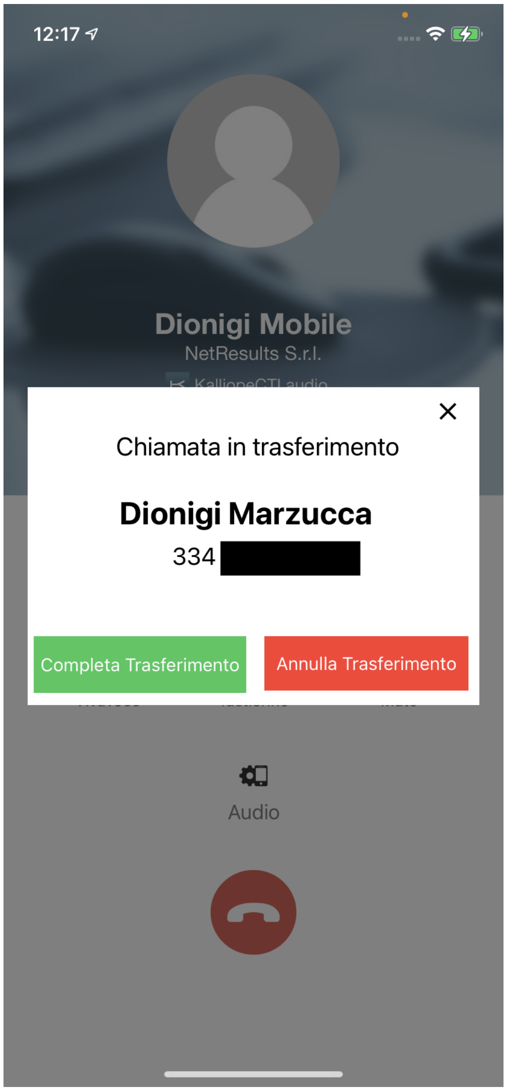
The contact search can also be carried out using the search bar at the top. The main difference between this list and the address book search is the presence of the contact number to the right of the contact (if it has only one number) or three dots (if there are multiple numbers associated with the contact).



If you press on a contact that has multiple numbers, a screen opens where you can choose from the available ones. In this screen you have information about the contact to whom you want to transfer the call (name and number). At this stage you can choose whether to transfer the call (green button with the word Transfer) or cancel the transfer (red button with the word Cancel).

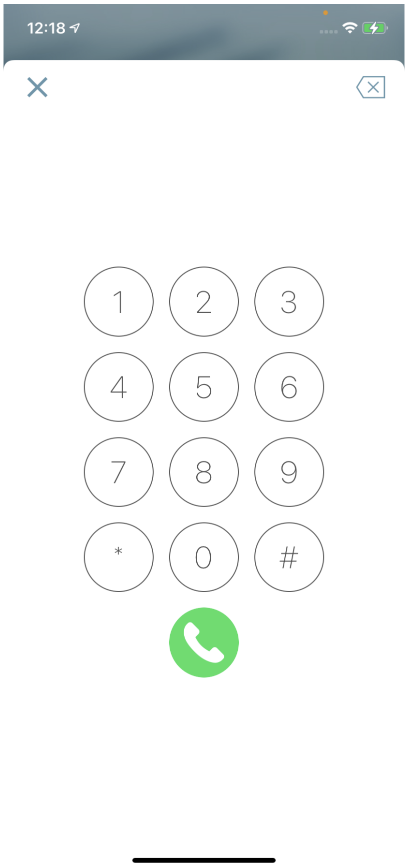
In this in-progress transfer screen you have the option to complete the transfer (green button) or cancel the transfer and return to the current call (red button).

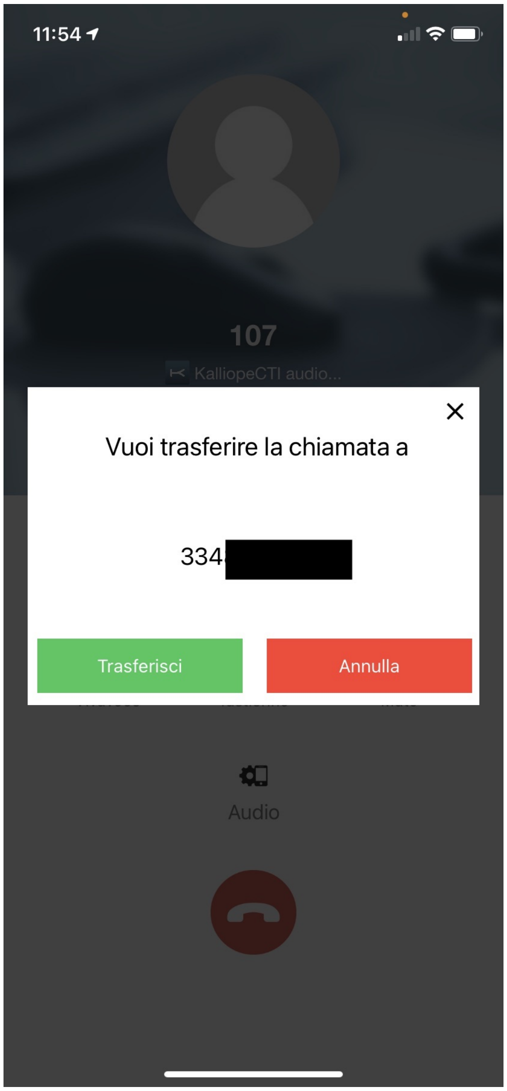
Note: In case the green button is pressed before the transfer has started you will receive a notice to wait for transfer to start (You must wait for the transfer to start).



Numeric Keypad

In this case the number of the contact can be entered directly from the keypad and the call can be transferred. After entering the number and pressing the green button you will get to the Contact Information Screen), but in this case you will see only the number and not the name of the contact (picture below)


12:18
1
2
3
4
5
6
7
8
9
*
0
#



Outgoing calls in CTI mode

Regarding outgoing calls in CTI mode, KalliopeCTI mobile app offers three different types of call setups.

Call-back

The Call-back service allows you to call from your mobile device using the KalliopePBX's lines. By pressing this key, the KalliopePBX makes a call to the mobile number associated with the extension, answering and pressing key 1 (as requested by the guide voice) you are actually put in communication with the desired number. The caller will see a call coming from the geographical number assigned to the Kalliope PBX. Questa is the type of outgoing call setup used by default in CTI mode.

Click-to-call

Click-to-call mode: the stationary device associated with the extension receives a call from KalliopePBX whose caller is c2c: called number. Answering this call will initiate a new call to the desired number.

Using directly the SIM of the device

In this mode, the call is made directly from the SIM of the device on which the KalliopeCTI Mobile app is installed. A simple direct GSM call is then established and then the caller will see the user's mobile number as the calling number.

Outgoing calls in Softphone mode

Regarding outgoing calls in Softphone mode, KalliopeCTI mobile app offers the following three types of call setups.

Chiamata SIP

SIP call: outgoing calls leave directly from the app via SIP protocol using the data network provided by the softphone. It is fully equivalent to a call originating from the KalliopePBX extension associated with the user. This is the type of outbound call setup used by default in Softphone mode.

Call-back

The Callback call service allows you to call from your mobile device using the KalliopePBX's lines. By pressing this key, the KalliopePBX makes a call to the mobile number associated with the extension, answering and pressing key 1 (as prompted by the guide voice) you are actually put in communication with the desired number. The caller will see a call coming from the geographical number assigned to the Kalliope PBX.

Using directly the SIM of the device

In this mode, the call is made directly from the SIM of the device on which the KalliopeCTI Mobile app is installed. A simple direct GSM call is then established and then the caller will see the user's mobile number as the calling number.

Dialing a number directly from the keypad will establish a call via SIP. Once the call is established, the "Active call" screen will automatically be displayed.

Keypad

Aside from searching the phonebook, you can also dial the number you wish to call with Kalliope directly on the KalliopeCTI mobile app keypad. While dialing, you will be shown autocomplete[1] suggestions.

In call-back mode: Tapping the call button lets you choose which mode to use among the ones listed above.

In softphone mode: Tapping the call button will directly establish a SIP call.

Note: The KalliopeCTI keypad does not automatically add the outbound prefix. You will need to dial the number as you would on a landline phone.

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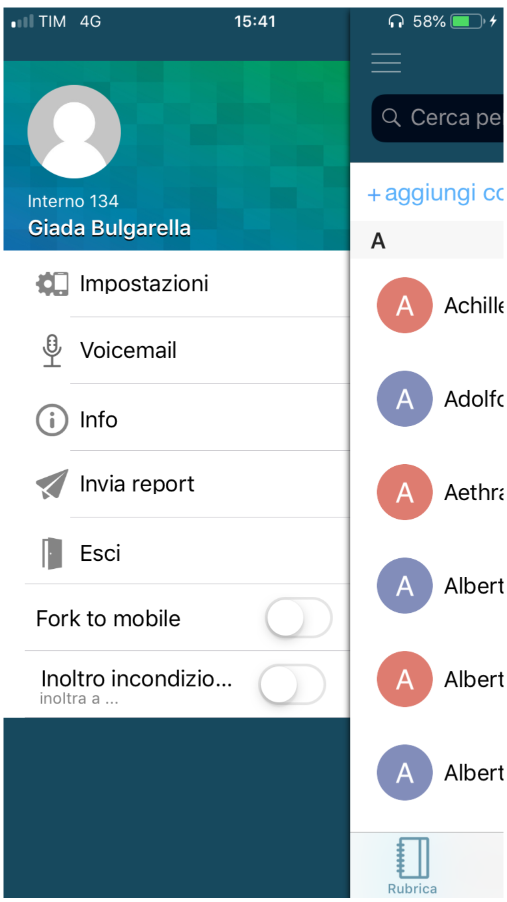
[1] Feature only available on Android.

Services

Services related to the KalliopeCTI mobile app can be accessed on iOS via the System Menu.

Three services are currently available: presence (seen above), unconditional forward, and fork to mobile.



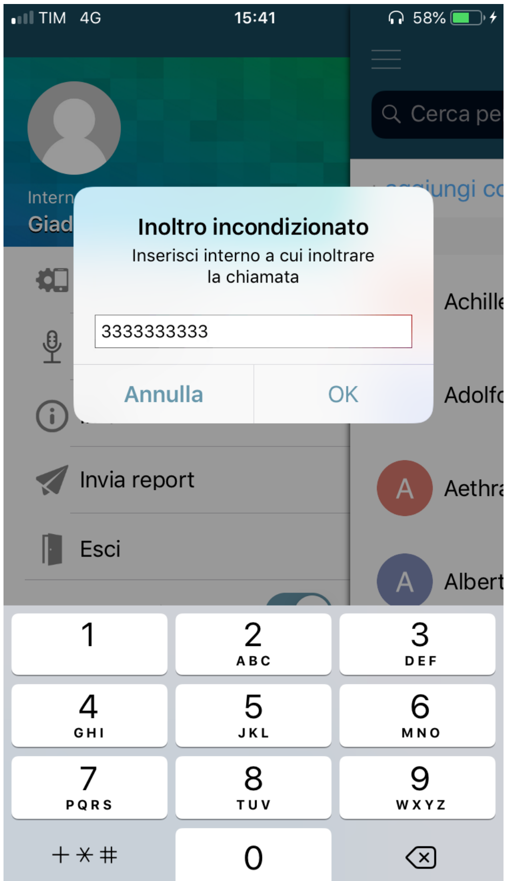
Unconditional Forwarding

Unconditional forwarding allows you to configure an extension number to which all incoming calls are automatically forwarded (this function is valid only for calls directed to the extension and not for those coming from groups or queues).

This service is useful when you are not available but wish to redirect inbound calls so they can be dealt with.

To enable the unconditional forward service, simply tap the arrow icon, input the extension number to which you wish to forward calls, and save.

To disable the service, simply click on the same switch again.



Fork to mobile

Fork to mobile consists in distributing the direct call to an extension also to the associated mobile number. This function can be set on Kalliope only by the administrator and not by the user.

Answering the call on the mobile phone will cause the extension to stop ringing, and vice versa.

To activate the service, simply click on the appropriate switch on the System Menu. Again, the status of the switch denotes the activation status of the service.

Privacy Policy

Secure connection

In order to offer its services (e.g. telephony, real-time notifications, chat), KalliopeCTI must be able to communicate with KalliopePBX. This communication is made through an encrypted connection, guaranteeing the security of transmitted personal data and messages.

Contacts

You can give KalliopeCTI permission to access the contacts on your device. The saved contact data will only be used within the app and will never be sent to KalliopePBX or exchanged with other users, except for the information required to make calls via the PBX.

Files

KalliopeCTI will access the phone's memory in order to send diagnostic data to the KalliopePBX developers via email. This data will only be sent when the user selects the "Send report" item in the KalliopeCTI menu.

4.1.3 Kalliope Attendant Console

Introduction

KalliopeATC (or KATC) is an accessory application of the KalliopePBX VoIP that incorporates features specially designed for phone operators into KalliopePhone's usual features.

KATC is available for Windows.

The following table lists the main features of KATC.

Kalliope Attendant Console features
Extension phonebook
Shared phonebook
CDR
Inbound call notification
Chat
Presence
Opening a custom URL for inbound calls
Voicemail access
Drop-to-Call
BLF
Blind transfer
Attended transfer
Queue statistics
Inbound call classification
Do Not Disturb
Call recording
Unconditional forward
Call parking
Call pickup
Pickup with invite
Forking to Mobile
Touchscreen support
Drag & Drop call transfer

Installation

KalliopeATC can be bought separately from KalliopePBX and can be activated by purchasing a license.

On the configuration page you can insert the credentials of an extension and the system will automatically check for the required license.

Note: Installation does not prompt for any activation keys. Instead, the application will automatically check for the correct license on KalliopePBX every time it is opened.

For instructions on activating the KATC license on KalliopePBX V4, see the licenses page.

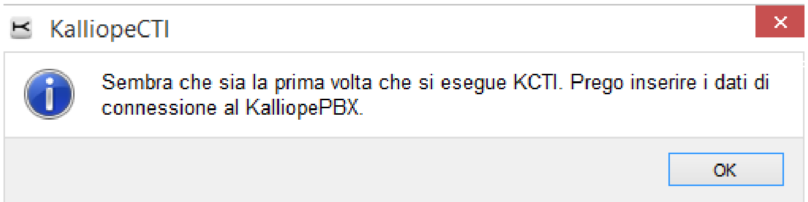
System requirements:

- Windows 7 or later
- Sound card (for KATC Phone)
- 2 GB of RAM
- 40 MB of free disk space

To install the application on Windows, simply open the executable file and follow the instructions.

Configuration

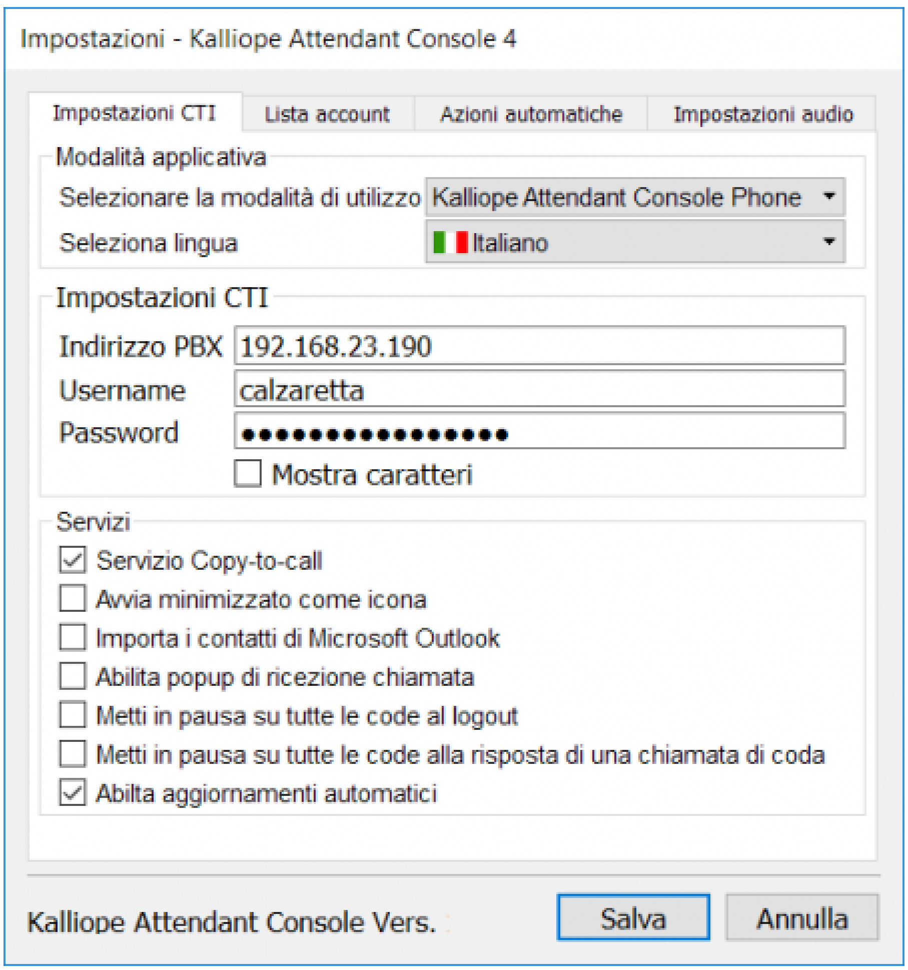
When you first open the application, you will see a message prompting you to insert the configuration parameters.



You will then be taken to the configuration page. The settings are grouped by category.

Application mode

In this tab you can select which mode to use: Kalliope Attendant Console CTI or Kalliope Attendant Console Phone. You can also choose the language (English, Italian, Spanish).



KATC settings

In this tab you can configure the following settings:

- KalliopePBX IP address: the IP address of the KalliopePBX the application should connect to
- KCTI username: the name assigned to the user
- KCTI password: the password assigned to the user

Note: The username and password are the same as the credentials on the System Settings -> Users management page on KalliopePBX V4.

Services

In this tab you can configure the following settings:

- Open minimized as system icon: if selected, the application will open in minimized mode.
- Import Microsoft Outlook contacts: if selected, KCTI will automatically sync with Microsoft Outlook contacts when opened. The imported contacts will appear in the phonebook marked by an icon.
- Enable call reception popup: enables or disabled the call reception popup.
- Pause on all queues upon logout
- Pause on all queues upon answering a call from the queue
- Enable automatic updates: automatically download available Kalliope Attendant Console updates

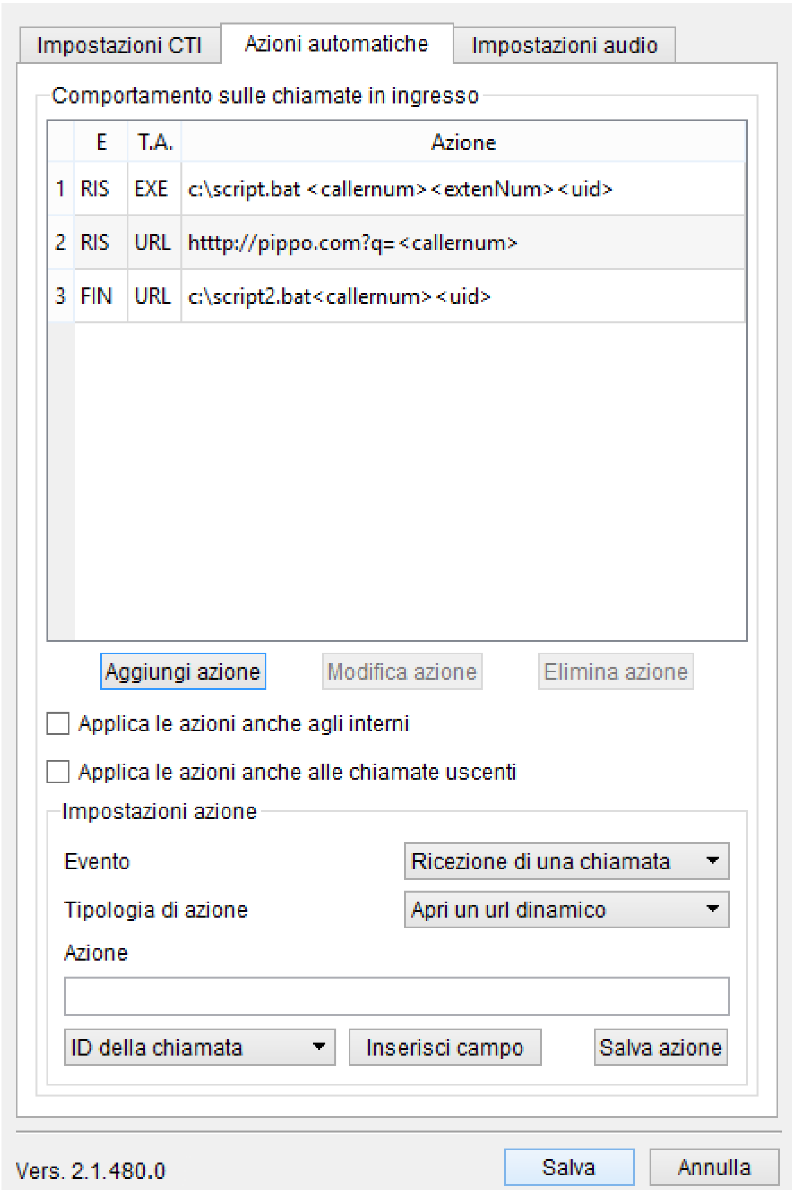
Phone integration

In KCTI 4 Pro mode, you can remotely control a linked phone through native APIs. This feature is currently only available on Snom or Yealink phones.

Account list

This tab shows all accounts linked to the user, listing the brands, models, firmware, and MAC addresses of each one. It is up to the user to select which accounts to link to KATC.

Automatic actions



In this tab you can manage the behavior of the application during inbound calls by configuring actions triggered by the following events:

- Arrival of call: the action will be executed when the phone begins to ring
- Answering call: the action will be executed when the user answers a call
- End of call: the action will be executed when the call is terminated, either by the local user or by the remote one

For each of these events, you can choose one or more actions of two types:

- Open a custom URL
- Execute an external process

In both cases you can use the following dynamic parameters:

- <callername>: this parameter will be replaced by the name assigned to the caller if present in the phonebook
- <callernum>: this parameter will be replaced by the number of the caller
- <extenNum>: this parameter will be replaced by the extension number of the local user
- <uid>: this parameter will be replaced by a unique call ID.

The figure to the side shows the automatic event configuration tab. The table on the top part of the tab contains the list of configured actions, showing the type of event ("IN" = arrival of call, "RIS" = answering call, "FIN" = end of call), the action type ("EXE" = execute an external process, "URL" = open a dynamic URL), and the action details.

Once the action has been saved, you can always edit or delete it by selecting it and clicking on Edit action/Delete action. For each action you can indicate whether it must be executed only on calls from external users or on calls between extensions as well, and whether or not it must also be executed on outbound calls.

Audio settings

This tab is only available in KATC Phone mode and lets you select the audio devices to use for audio input (microphone), audio output (to listen to the call), and for notifications (audio output for the ringtone and other notifications).

Manuale Utente

The KATC v4 user manual is available in pdf format at this link.

4.1.4 Kalliope Call Center

Introduction

Kalliope Call Center is an optional KalliopePBX module that adds advanced features specially designed for call centers.

The main features included in Kalliope Call Center are:

- Kalliope Supervisor Panel
- Call Center CDR
- Call Tagging
- Spy/Whisper/Barge services
- Queue position reservation and automatic callback

This page describes the Kalliope Supervisor Panel. For further information on the other features included in Kalliope Call Center, see the user manual.

Kalliope Supervisor Panel

The Kalliope Supervisor Panel software lets a user access queue statistics based on their role, monitor and change the state of the operators, and manage dynamic operators.

When first opening the application, you will need to go to the Settings page and insert the credentials of the extension. The system will automatically check for the required Call Center license and the role of the user (supervisor or operator).

Note: Installation does not prompt for any activation keys. Instead, the application will automatically check for the correct license on KalliopePBX every time it is opened.

Menu bar

The menu bar lets you select the Settings sub-menu.

The Settings menu contains the following items:

- Connect to KalliopePBX: start the connection with KalliopePBX
- Disconnect from KalliopePBX: interrupt the connection with KalliopePBX
- Fullscreen: view the application in fullscreen mode (shortcut: F11)
- Settings: opens the Settings page
- Quit: close the application (shortcut: CTRL+Q)

Settings page

From the menu bar you can open the configuration page, which consists of two parts: Credentials and Customize logo.

Credentials

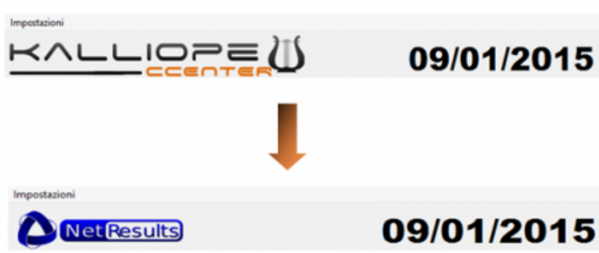
This section lets you configure the following parameters:

- KalliopePBX IP address: the IP address of the KalliopePBX the application should connect to
- Username: the name assigned to the user
- Password: the password assigned to the user

Note: The username and password are the same as the credentials on the System -> KCTI/Web users page on KalliopePBX V4.

Customize logo

This section lets you upload an image file containing the custom logo that will be displayed in the main screen of the application instead of the KalliopeCC logo.



Main screen

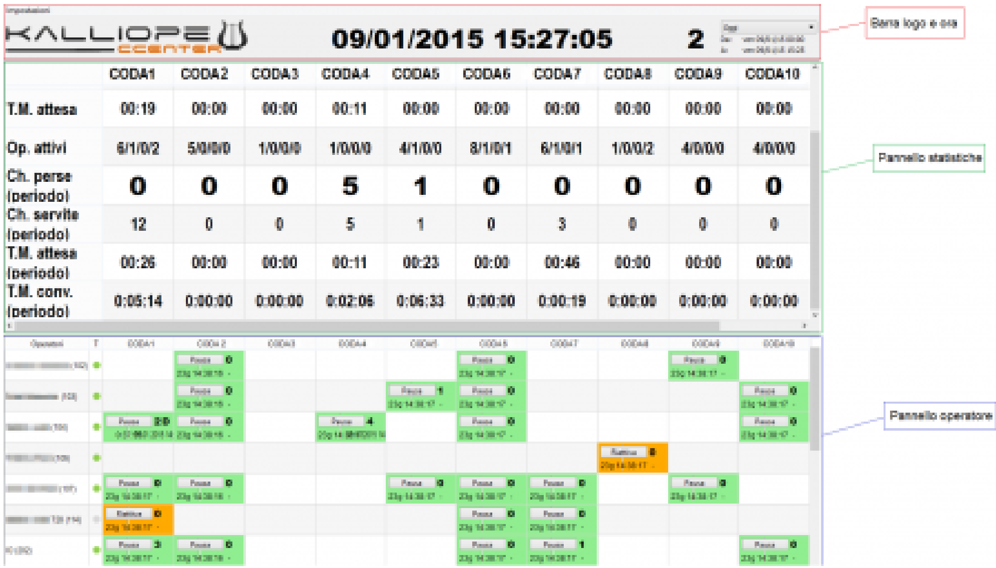
The main Kalliope Supervisor Panel screen is divided in thee sections:

- Logo bar
- Statistics panel
- Operator panel

The first has a set size and cannot be hidden, while the other two can be expanded or reduced and can be set to completely fill the main screen. This can be useful if, for example, you wish to only view aggregated data without seeing the names of every operator.

The main screen changes depending on whether the logged-in user is a supervisor or an operator. A supervisor will see information on all active queues on the PBX and will have permissions to pause and assign queues to every operator. An operator will only see the queues and statistics that concern them.

The three sections are detailed below.



Vediamo, di seguito, le singole le sezioni in dettaglio.

Time and logo bar

As explained above, this section can be partly customized by changing the logo from the Settings page.

The time and logo bar displays the following information:

- The KalliopeCCenter logo or the custom logo uploaded by the user
- Date and time
- The total number of calls in every queue
- Time range selection: from the drop-down menu you can choose the time range for which to display aggregated data in the section below

Statistics panel

This columns of this table display information on each active queue on the PBX.

The rows display data on:

- Users in the queue: dynamically updates when a user enters or exits the queue
- Oldest call in the queue: the time, in seconds, that the oldest call has been in the queue;
- Average waiting time: always refers to the last hour regardless of the selected time range and updates every 5 minutes by automatically downloading and analyzing the month's CDR;
- Active operators: reports the state of the operators in real-time; the first number is the number of active registered operators (green BLF), the second the number of busy registered operators (red BLF), the third the number of paused registered operators (orange BLF), and the fourth the number of operators whose state is unavailable (gray BLF);

- Missed calls (in the selected range): varies depending on the time range selected in the drop-down menu;
- Answered calls (in the selected range): varies depending on the range of time selected in the drop-down menu;
- Average waiting time (in the selected range): varies depending on the range of time selected in the drop-down menu;
- Average conversation time (in the selected range): varies depending on the range of time selected in the drop-down menu;

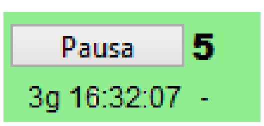
	CODA1	CODA2	CODA3	CODA4	CODA5	CODA6	CODA7	CODA8	CODA9	CODA10
Utenti in coda	0	0	0	0	0	0	1	0	0	0
Ch. più antica							00:02			
T.M. attesa	00:00	00:00	00:00	00:00	00:07	00:00	00:19	00:00	00:00	00:00
Op. attivi	6/1/0/2	5/0/0/0	1/0/0/0	1/0/0/0	4/1/0/0	8/1/0/1	6/1/0/1	1/0/0/2	4/0/0/0	4/0/0/0
Ch. perse (periodo)	0	0	0	5	1	0	0	0	0	0
Ch. servite (periodo)	12	0	0	5	4	0	5	0	0	0
T.M. attesa (periodo)	00:26	00:00	00:00	00:11	00:11	00:00	00:35	00:00	00:00	00:00
T.M. conv. (periodo)	0:05:14	0:00:00	0:00:00	0:02:06	0:03:10	0:00:00	0:00:18	0:00:00	0:00:00	0:00:00

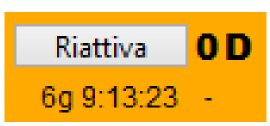
Operator panel

The first column shows the registered operators on the PBX, the second the corresponding BLF state, and the remaining ones correspond to each queue, as with the statistics panel.

Each row corresponds to a single operator and their data for each queue they are statically or dynamically assigned to. This table gives an overview of the situation across all queues.

Each cell corresponding to a specific operator and a specific queue gives the supervisor important information.





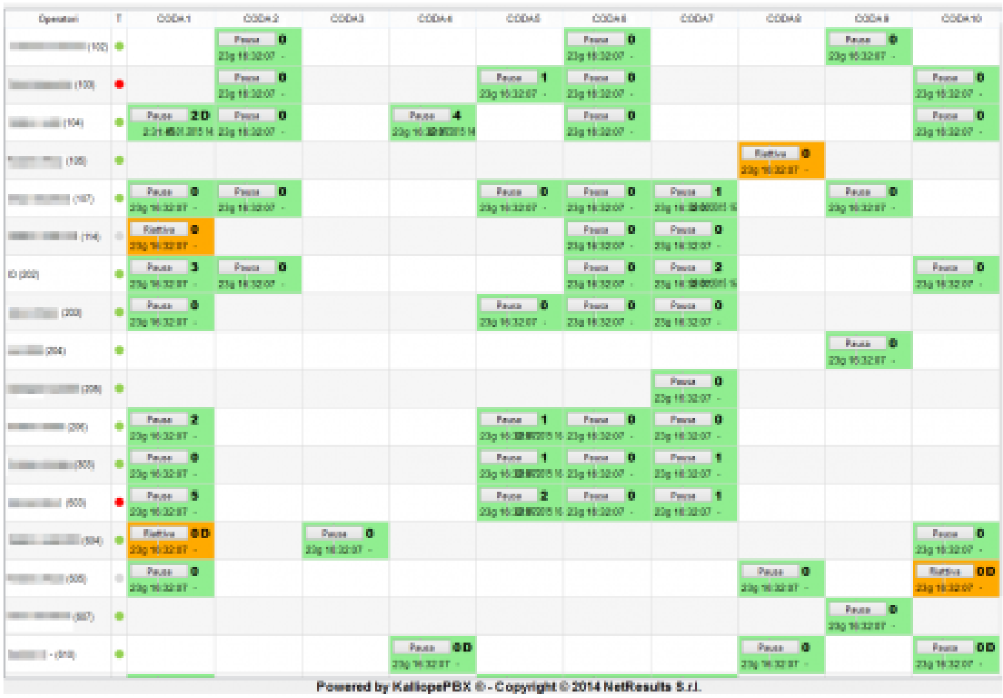
The color of the cell indicates the state of the operator. If the operator is active, the cell is green; if the operator is paused, the cell is orange. Similarly, the button will change between Pause and Reactivate.

Note: Operators are only able to change their own status, while supervisors can act on all operators.

Each cell also displays the following information:

- The number of answered calls in the selected time range, on the top right;
- Whether the operator is static or dynamic on that queue (in the latter case, a "D" will be displayed next to the number of answered calls);

- Time operating on the queue: the amount of time since the status was last changed from active to pause or vice versa;
- Time of the last served call.

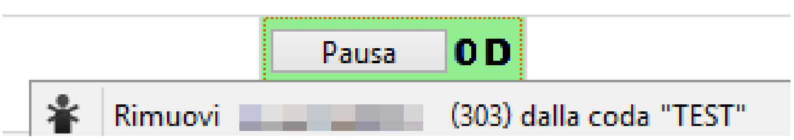


Adding dynamic operators

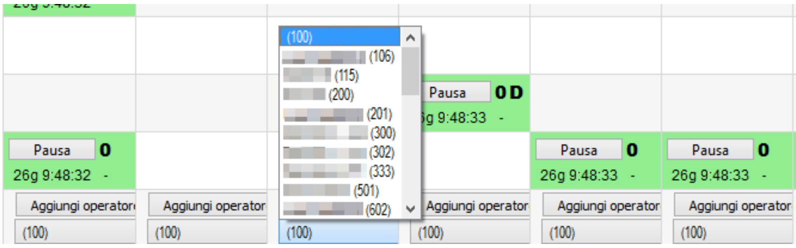
A dynamic operator is a user that has been temporarily assigned to one or more queues by a supervisor. Unlike static operators, dynamic operators are assigned to the queues on an application level rather than a PBX level.

Any extension on the PBX can be set as a dynamic operator, not just those already assigned to a queue.

To add an extension that is already an operator on at least one queue, the supervisor can simply right-click on the empty cell that corresponds to the desired queue and selecting Add from the context menu. Similarly, they can remove the operator by right-clicking on the relevant c



If the extension is not an operator on any queue, the supervisor must use the drop-down menu at the end of the column that corresponds to the desired queue. From there they can select the desired extension then click on Add to assign it to the queue as a dynamic operator. A new row will appear displaying information on the new operator.



As with other dynamic operators, the supervisor can simply right-click on the relevant cell to remove the extension from the queue.

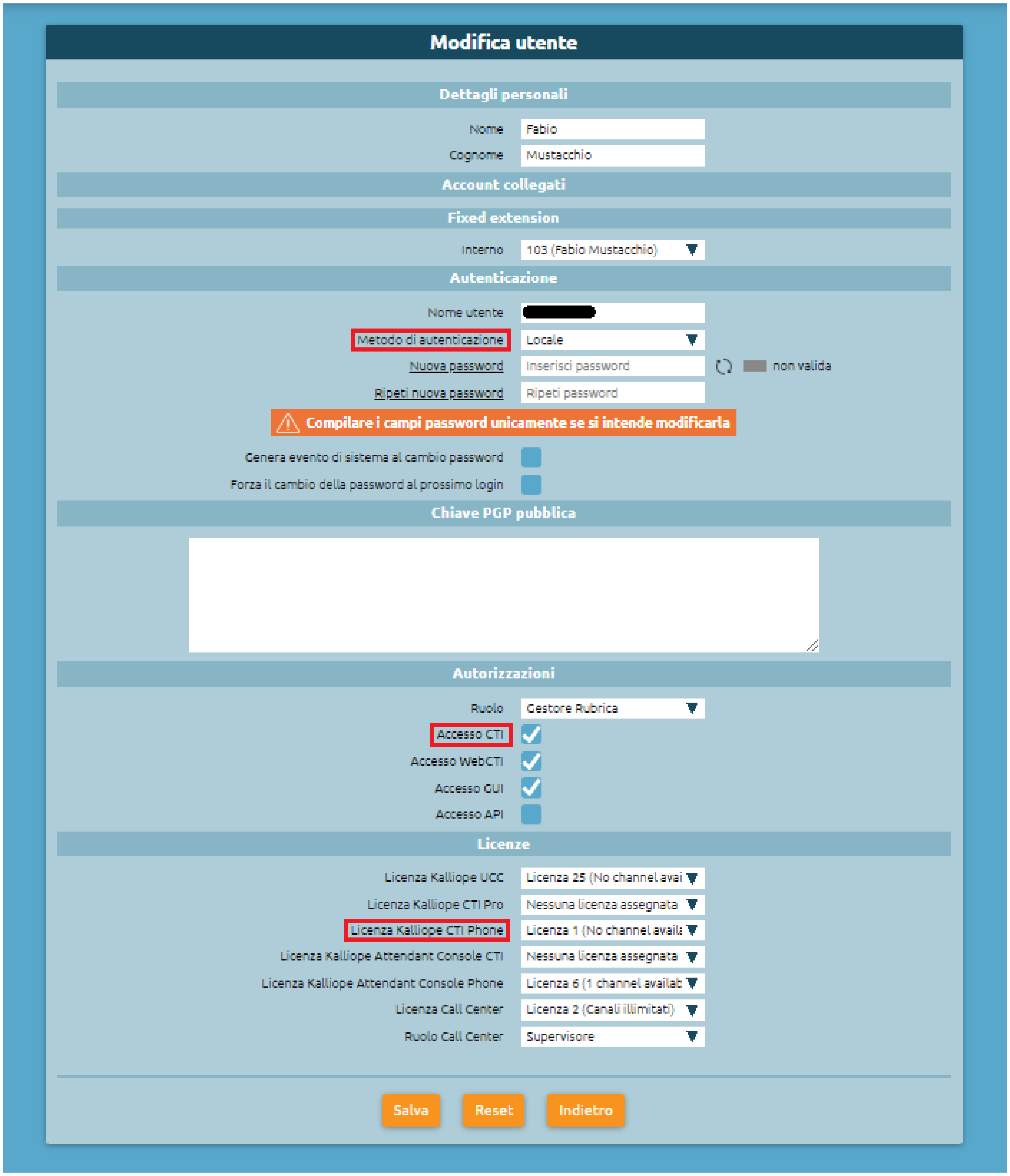
4.1.5 Kalliope Phone

Enabling Kalliope Phone on KalliopePBX

Guide to enabling Kalliope Phone on KalliopePBX.

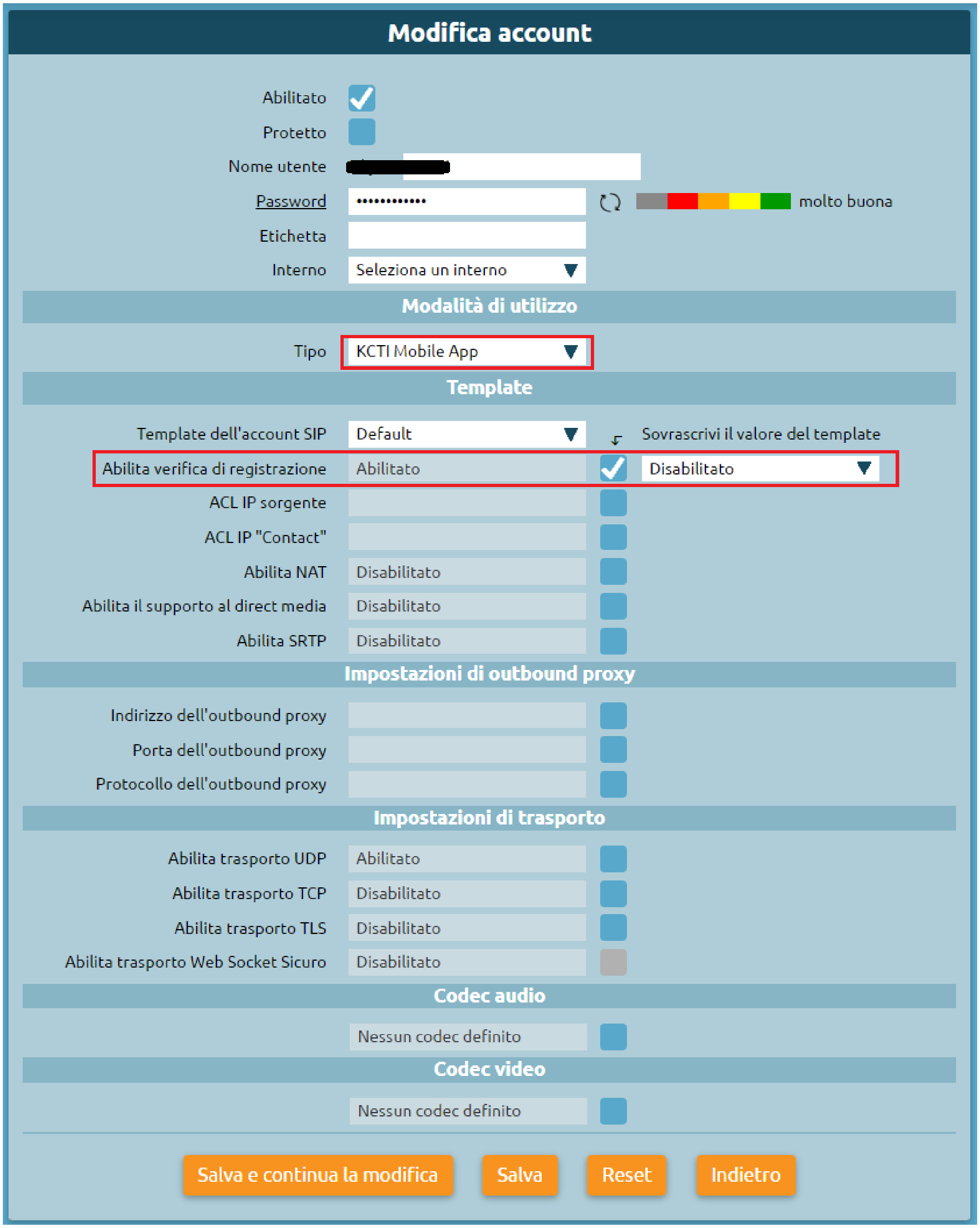
User Setting

- Set the “Authentication method” to “Local”
- Enable the “CTI Access” option
- Enter the KalliopeCTI Phone license in the “Licenses” section.


Modifica utente
Dettagli personali
Nome
Fabio
Cognome
Mustacchio
Account collegati
Fixed extension
Interno
103 (Fabio Mustacchio)
Autenticazione
Nome utente
[redacted]
Metodo di autenticazione
Locale
Nuova password
Inserisci password
non valida
Ripeti nuova password
Ripeti password
Compilare i campi password unicamente se si intende modificarla
Genera evento di sistema al cambio password
Forza il cambio della password al prossimo login
Chiave PGP pubblica
Autorizzazioni
Ruolo
Gestore Rubrica
Accesso CTI
Accesso WebCTI
Accesso GUI
Accesso API
Licenze
Licenza Kalliope UCC
Licenza 25 (No channel avai
Licenza Kalliope CTI Pro
Nessuna licenza assegnata
Licenza Kalliope CTI Phone
Licenza 1 (No channel avail
Licenza Kalliope Attendant Console CTI
Nessuna licenza assegnata
Licenza Kalliope Attendant Console Phone
Licenza 6 (1 channel availab
Licenza Call Center
Licenza 2 (Canali illimitati)
Ruolo Call Center
Supervisore
Salva
Reset
Indietro

Account settings

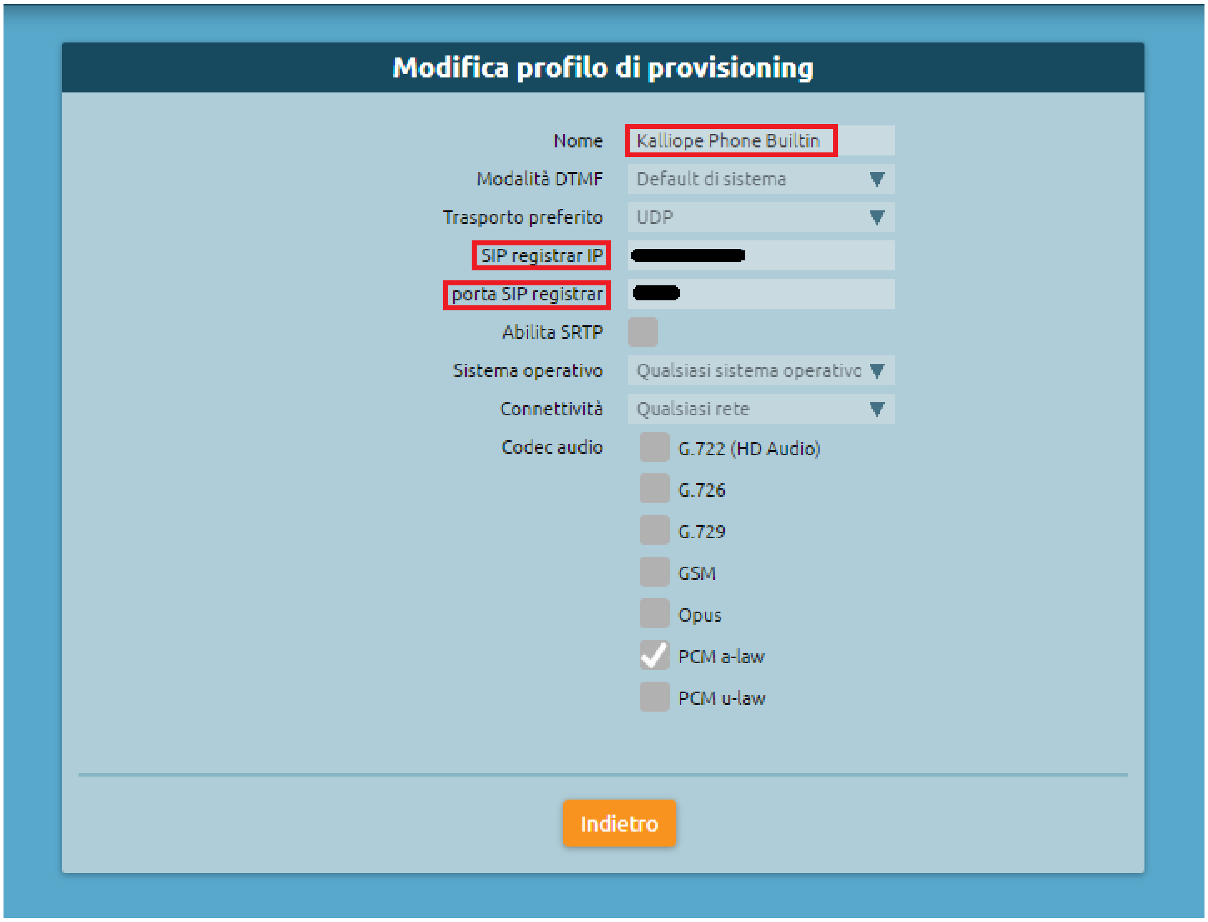
As shown in the template below, an account of type **KCTI Mobile App** must be created and **registration verification must be disabled**.



Applications provisioning

To access the provisioning profiles, follow the path "System Settings > Application Provisioning > Provisioning Profiles." In case there is an SBC or the connection IP of the application is not the same as the one to which the phone component is to register, you need to go and edit the built-in provisioning profile **"Kalliope Phone Builtin"** by entering **the IP** and the **port of the SIP Registrar**.

You can also specify other options in the provisioning profile, such as the **codec** to be used and the use of **TLS and/or SRTP**.



User's Manual

Introduction

The Kalliope Phone application enables business voice calls via Internet connection, allowing you to stay connected to Kalliope from a mobile device and improving the efficiency of business communications.

… warning::

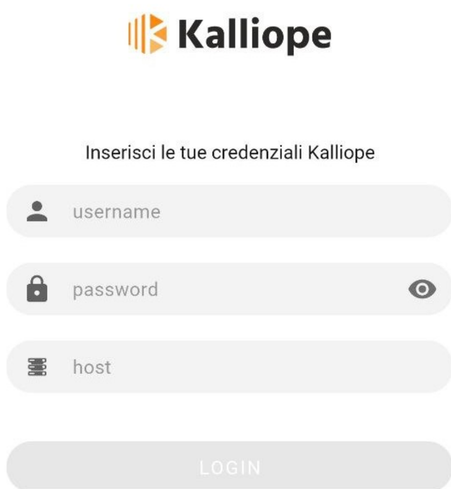
The product is only available when paired with a KalliopePBX® V4 VoIP PBX. Minimum firmware version: 4.15.6 KalliopePhone license required.

Structure

You can download the application on **Android** devices directly from Playstore and on **iOS** from the App Store.

Once downloaded, you must register and enter:

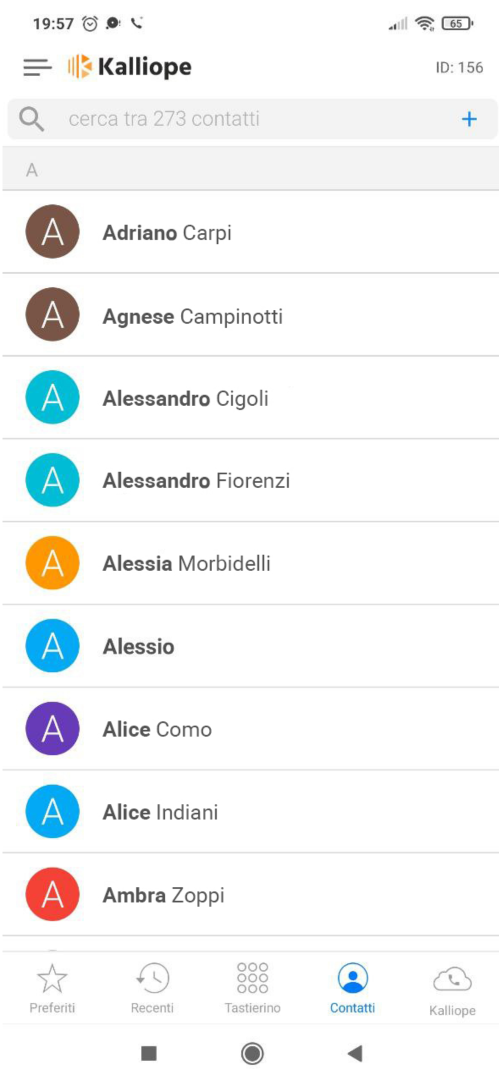
- **Username**
- **Password**
- **Host**: address of the KalliopePBX central unit you have.

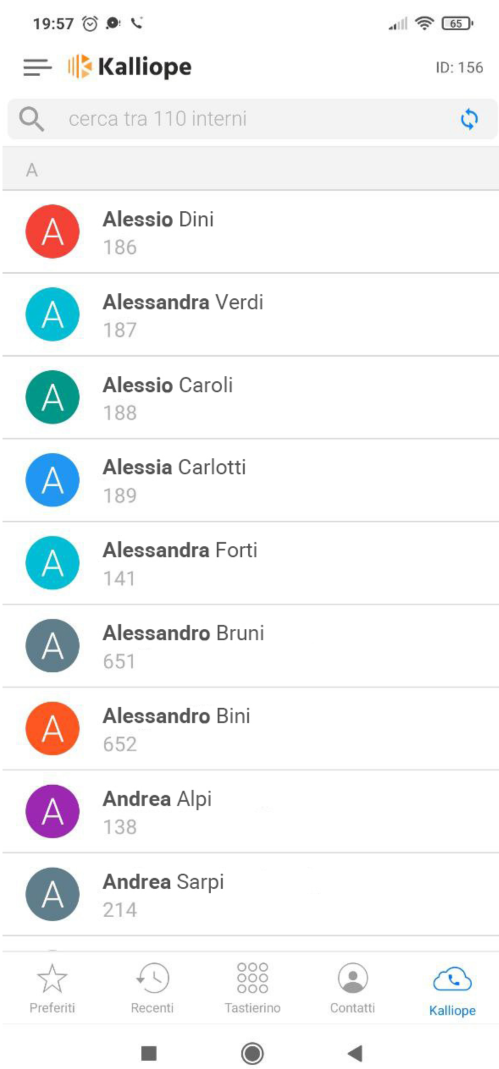


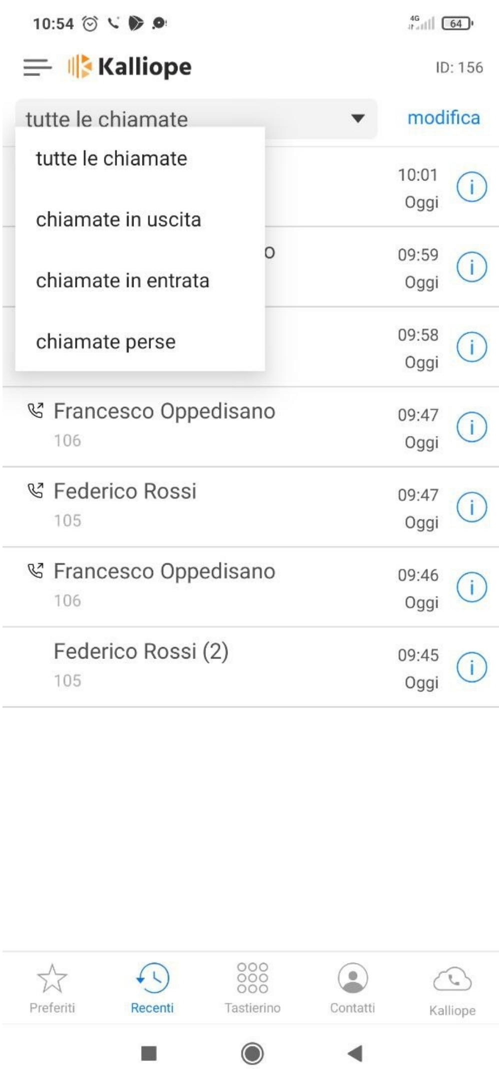
The app has a series of icons arranged at the bottom that identify specific sections described below and depicted in the corresponding images:

- **Numeric keypad** (1): allows typing in the phone number.
- **Contacts** (2): contains the list of contacts on the smartphone. Pressing on a specific contact displays its details and it is possible to bookmark it via the star icon
- **Kalliope** (3): list of Kalliope extensions.
- **Recent** (4): list of recent calls (also called CDR), filterable into "all calls," "outgoing calls," "incoming calls," "missed calls"
- **Favorites** (5): list of contacts entered as favorites


8:42
Kalliope
ID: 156
1
2 ABC
3 DEF
4 GHI
5 JKL
6 MNO
7 PQRS
8 TUV
9 WXYZ
*
0 +
#
Preferiti
Recenti
Tastierino
Contatti
Kalliope

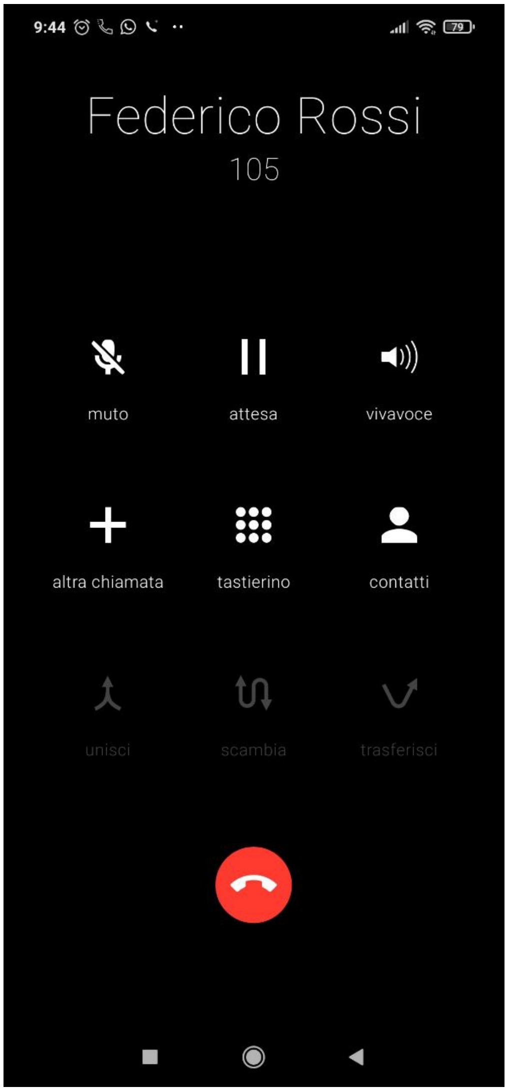

19:57
Kalliope
ID: 156
cerca tra 273 contatti
A
Adriano Carpi
Agnese Campinotti
Alessandro Cigoli
Alessandro Fiorenzi
Alessia Morbidelli
Alessio
Alice Como
Alice Indiani
Ambra Zoppi
Preferiti
Recenti
Tastierino
Contatti
Kalliope


19:57
Kalliope
ID: 156
cerca tra 110 interni
A
Alessio Dini
186
Alessandra Verdi
187
Alessio Caroli
188
Alessia Carlotti
189
Alessandra Forti
141
Alessandro Bruni
651
Alessandro Bini
652
Andrea Alpi
138
Andrea Sarpi
214
Preferiti
Recenti
Tastierino
Contatti
Kalliope


10:54
Kalliope
ID: 156
tutte le chiamate
modifica
tutte le chiamate
chiamate in uscita
chiamate in entrata
chiamate perse
10:01
Oggi
09:59
Oggi
09:58
Oggi
Francesco Oppedisano
106
09:47
Oggi
Federico Rossi
105
09:47
Oggi
Francesco Oppedisano
106
09:46
Oggi
Federico Rossi (2)
105
09:45
Oggi
Preferiti
Recenti
Tastierino
Contatti
Kalliope

Outgoing call

To make a call, you can enter the phone number via keypad or, by pressing on a specific contact in the **"Contacts"** list, click on the relevant phone number. During the current call, the following screen will be displayed:



The actions that can be performed are:

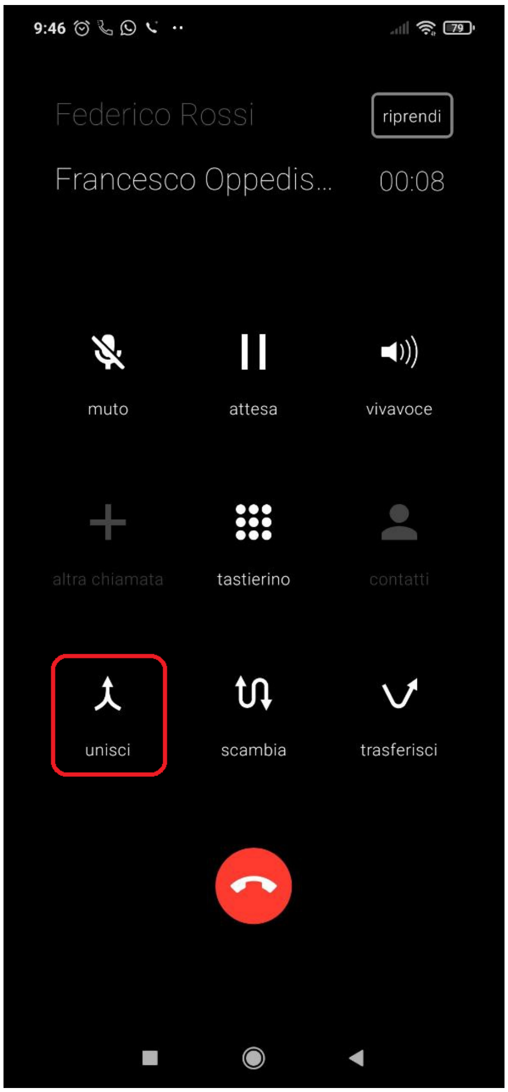
- **Mute call**
- **Put on hold**
- **Change the audio device** you are using (phone, speakerphone or bluethooth)
- **Make, during the active call, a second call** using the "other call" button, *see the next section*.
- **Open the numeric keypad** to send DTMF tones.
- **Open the contact section**

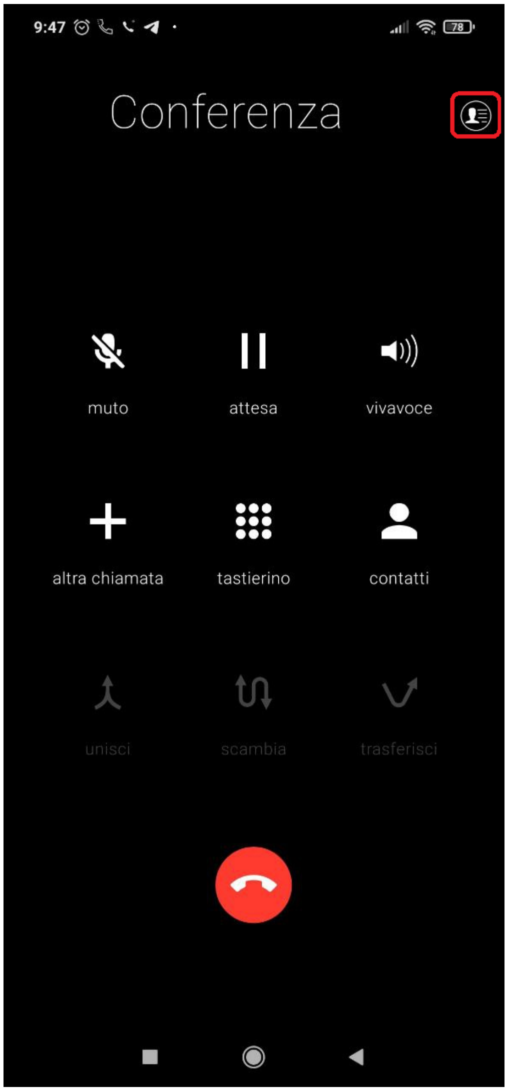
Adding a call

While a call is in progress, you can call another number by pressing on the **+"other call "** key and entering the number using the numeric keypad, searching it among contacts or, if present, among recent calls.

Once the desired number is added, the main call is put on hold and you can:

- **Unite** the two calls, so you get a **Conference**, where the calls are in communication with each other. To *view the participants** of the conference just press on the icon in the upper left corner, in this screen you can remove individual participants via the icon or disengage them from the conference

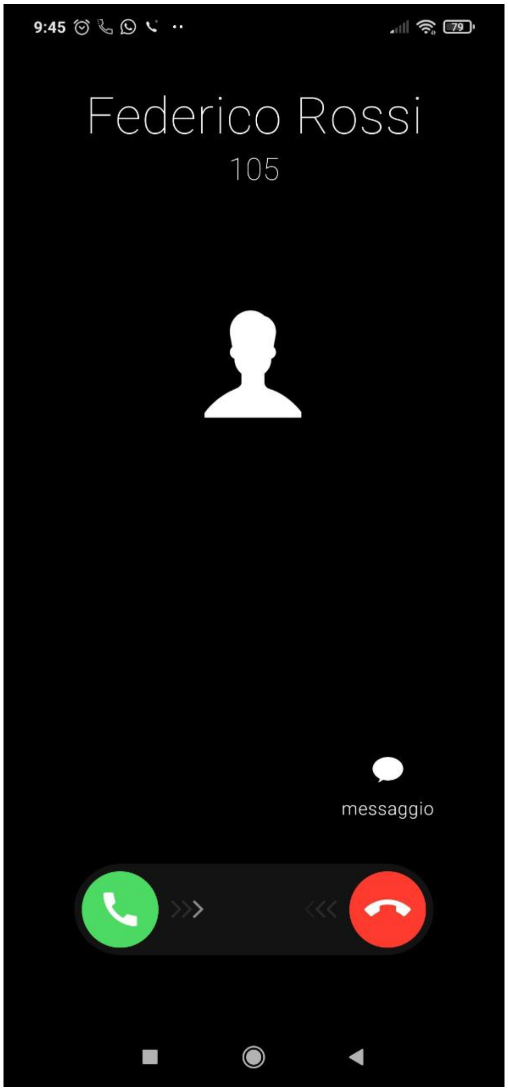



9:47
78
Conferenza
muto
attesa
vivavoce
altra chiamata
tastierino
contatti
unisci
scambia
trasferisci

- **Switch** the call: switch between the two calls in progress, automatically pausing one of them (shuttle service)
- **Transfer** the call: the two calls in progress are put into communication, while one's own line becomes free again

Incoming call

The incoming call is displayed as follows and you can answer or reject it.



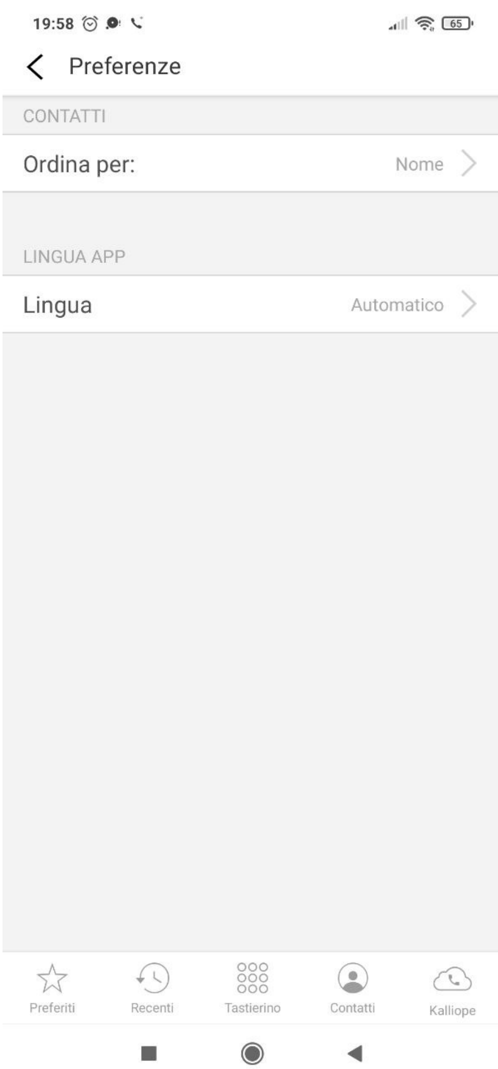
Settings menu

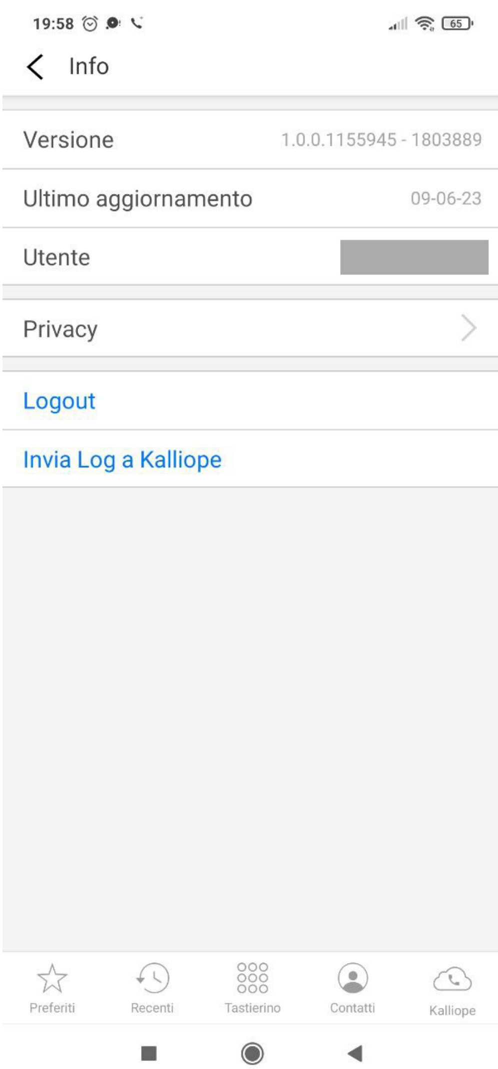
The settings menu can be reached by pressing on the icon in the upper left corner, next to the "Kalliope" logo, as shown in the following figure:

In this section you can set the **contact sorting preference**, sortable by First Name or Last Name, and the **language setting** (English, Italian, Spanish).

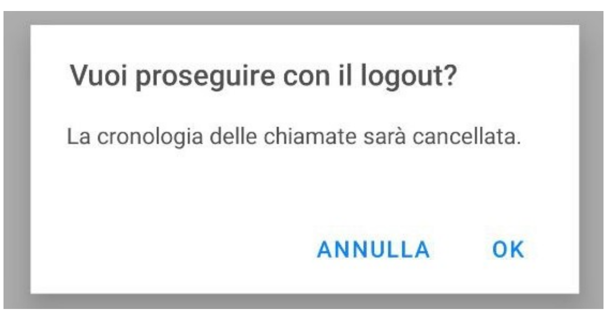
Also present are the **info** about the current version of the application, the last update, reference to the user using the application, and the privacy policy.

In the info it is also possible to **logout** and **send Log to Kalliope** (send call history).


19:58
Preferenze
CONTATTI
Ordina per:
Nome
LINGUA APP
Lingua
Automatico
Preferiti
Recenti
Tastierino
Contatti
Kalliope



When logging out, the user will no longer receive calls on their smartphone. In addition, when logging out, the application warns that information about calls in the log (the "Recent" section) will be lost.



FAQ

(Android) After starting the Smartphone, I do not receive incoming calls on Kalliope Phone. What can I do?

The behavior is introduced by Android security settings from version 10 onwards. The terminal is really operational only after unlocking by PIN, sign or biometric reading, the times vary from a few seconds to a few minutes depending on the hardware characteristics of the terminal.

(Android) I am not receiving incoming calls when Kalliope Phone is in the background. What can I do?

Go to Settings > App > Manage App > Kalliope Phone and check that no background usage or power saving restrictions are set. Go to the Play Store, search for the Operator Services (Google LLC) app and make sure you have the latest version installed. Also check that the ports needed for Google to deliver push notifications are not blocked by routers, firewalls, or antiviruses. Read here for more information.

(Xiaomi) Kalliope Phone does not appear in the foreground when I receive incoming calls. What can I do?

Kalliope Phone may not have the necessary permissions to appear in the foreground on the lock screen. Go to Settings > App > Manage App > Kalliope Phone > Other Permissions and make sure the permissions are all enabled.

(Android) The ringtone is not played when I receive an incoming call. What can I do?

In case the ringtone is saved on external storage, Kalliope Phone may not have the necessary permissions to read the audio file. Therefore, you need to go to your smartphone settings and authorize Kalliope Phone to access the phone's memory. For example, on Samsung terminals, go to Settings > Applications > Kalliope Phone > App Permissions > Memory and select Allow.

I cannot make or receive calls and the message "not registered" is displayed. What can I do?

In case of connecting via mobile data network, make sure you:

- you have 3G/4G/LTE coverage
- you are not in airplane mode
- you have not run out of available Internet traffic

In case of connection via WiFi network, make sure that:

- the ports on SBC or PBX used for signaling and audio streams are reachable (depends on installations). *Es. 5060 UDP and 10000-20000 UDP on Kalliope IP in case the default Kalliope configurations are used and there is no SBC.*
- port 443 TCP on the IP address used during configuration is reachable
- outbound traffic to IPs 78.152.108.54, 78.152.108.55, 80.93.131.132 and 80.93.131.134 (ports 443 TCP, 24998 TCP and 4998 TCP/UDP) is not blocked

If you continue to have problems, please contact your designated Kalliope specialist technician.

(Android) During installation you are prompted to enable permission to display Kalliope Phone in the foreground, but I cannot change this setting. What can I do?

Check Settings > Applications > Kalliope Phone for the Display Featured item and grant permission. The item may have a different name depending on the model (e.g., show above other apps). If your smartphone does not show such an item or the item is not enabled, it is possible that Android Go, a particular version of Android that is not supported by Kalliope Phone, is installed on your smartphone. You can check this in the following ways:

- the version of Android currently installed on your device can be found within the Phone Settings. Remember that the section may have different names depending on the model.
- the screen shown when restarting your device should say go edition in addition to powered by Android.
- standard applications installed on your smartphone should have the suffix Go in addition to their name (e.g., Google Assistant Go, Gmail Go).

- standard app icons may have the word GO in them.

I experience poor audio quality under WiFi network. What can I do?

When connecting over WiFi network, the quality of service (QoS) Voice with Kalliope Phone is dependent on the correct LAN configuration. You can best manage the priority of traffic generated by Kalliope Phone by taking into account that RTP packets are classified and marked (within the TOS, Type Of Service, field) with a DSCP value equal to 46. Make sure that the access-points used allow prioritization of voice packets. If you continue to have problems, contact your Kalliope technical specialist of choice.

4.2 Add-on Kalliope

4.2.1 Kalliope FAX

Description

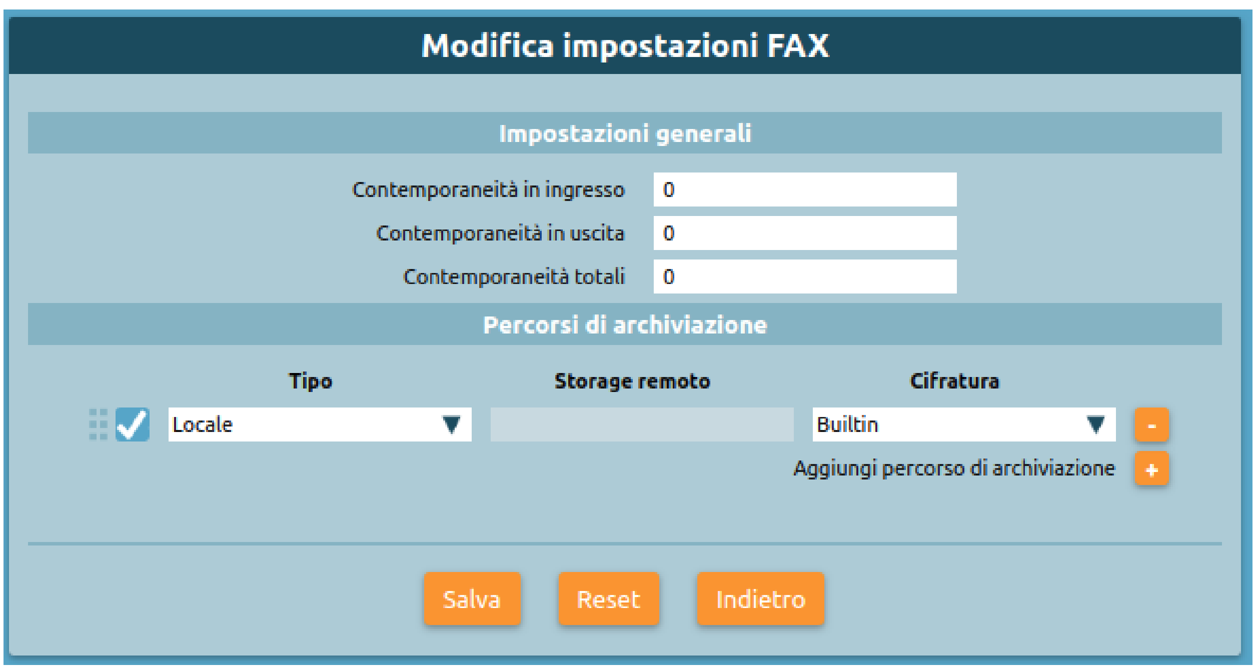
The fax service lets Kalliope users send and receive faxes from their user portal on the web GUI or by email.

The service requires the purchase of a KPBX-V4-FAX license. Each license channel allows the configuration of one fax entity. A fax entity is conceptually equivalent to a physical fax machine that one or more users can access.

Configuration To configure the service, it is necessary to define the general settings for simultaneous use and storage of received/sent files (including reports). This configuration is done in the panel FAX → FAX Settings.

FAX Settings

In the FAX settings edit panel it is possible to set the configurations common to all FAX instances in the system.



Account settings

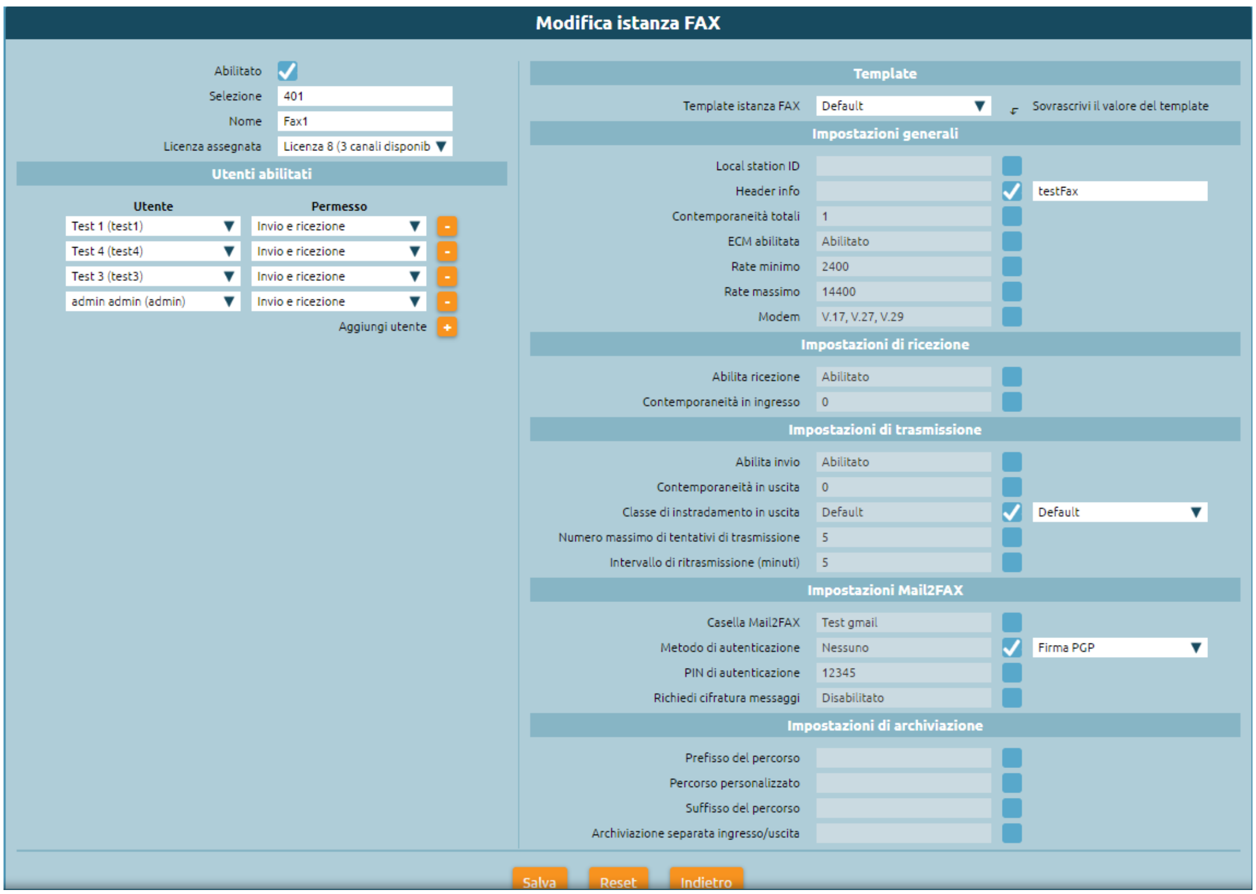
Parameter	Description	Value
Simultaneous inbound FAXes	Maximum number of inbound FAXes allowed	Numeric (0 = infinite)
Simultaneous outbound FAXes	Maximum number of outbound FAXes allowed	Numeric (0 = infinite)
Total simultaneous FAXes	Maximum number of FAXes allowed in both directions	Numeric (0 = infinite)

Archival paths (ordered list)

Parameter	Description	Value
Enabled		Checkbox
Type	Type of storage: local or remote	Local / Remote
Remote storage	Storage name	Dropdown
Encryption	The encryption algorithms to use for the assets on the archival path	None / Built-in

FAX Instances

Next you need to configure the specific FAX instance in the **FAX panel** → **FAX Instances**. The FAX Instance panel defines the attributes associated with a FAX entity. A FAX instance must correspond to a numbering plan selection (such as an extension), have a name, and commit a channel of a Kalliope FAX Module license.



Parameter	Description
Enabled	

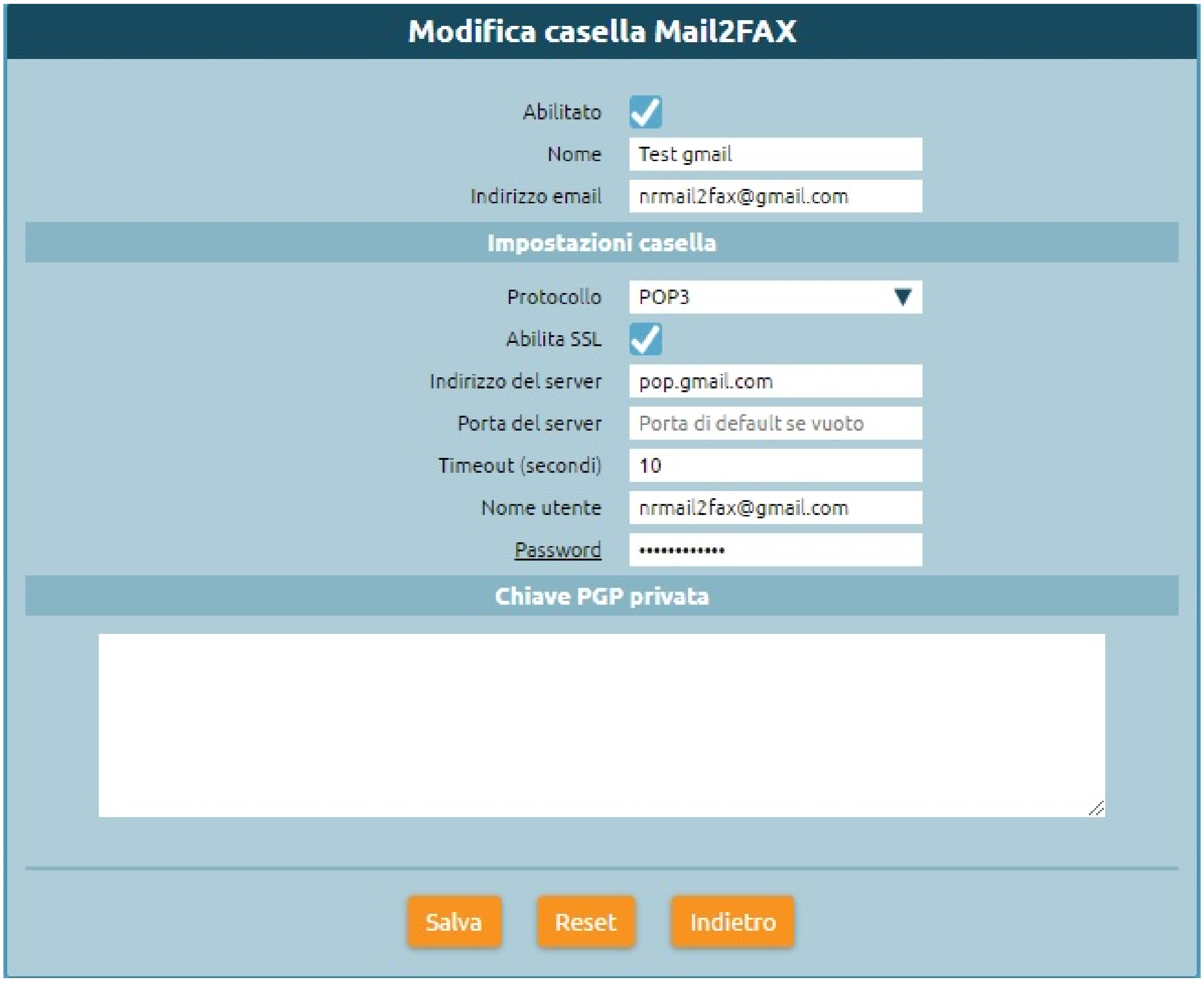
Table 1 – continued from previous page

Parameter	Description
Selection	The numbering plan selection that corresponds to the fax entity
Name	Entity name
Enabled users (list)	
User	The user to which the permission will be assigned
Permission	The permission that will be assigned to the user
Template	
Fax entity template	The template that will be used for the configuration of the instance
General settings	
Local station ID	The fax identifier that will be sent to the remote device
Header info	The text string that will be included in the upper margin of each sent page
Total simultaneous faxes	Maximum number of faxes allowed in both directions
ECM enabled	Enable Error Correction Mode
Minimum rate	Minimum transfer speed
Maximum rate	Maximum transfer speed
Modem	Standard supported modem
Reception settings	
Enable reception	Enable inbound faxes
Simultaneous inbound faxes	Maximum number of inbound faxes allowed
Transmission settings	
Enable sending	Enable outbound faxes
Simultaneous outbound faxes	Maximum number of outbound faxes allowed
Outbound routing class	The routing class used for outbound faxes
Maximum number of transmission attempts	Maximum number of transmission attempts after which the FAX will be considered fai
Re-transmission interval (minutes)	The interval between a transmission attempt and the next
MAIL2FAX settings	
Mail2Fax box	Name of the mail2fax box
Authentication method	Authentication method with which the fax is saved and received
Authentication PIN	PIN with which the fax is authenticated, inserted in the text of the mail
Request message encryption	Enable or disable message nercyption
Archival settings	
Path prefix	The prefix that will precede the custom path of the archived file
Custom path	The custom path to which the archived file will be saved
Path suffix	The suffix that will follow the custom path of the archived file
Separate inbound/outbound archival	If and how to archive inbound and outbound documents separately

Note: Always remember to select the outbound routing class.

Mail2Fax

If you want to configure also the MAIL2FAX service you have to select from the panel FAX → FAX Instances, Mail2Fax boxes list and add a new Mail2Fax box.



Parameter	Description	Value
Enabled	Enable the Mail2FAX box	Checkbox
Name	Name of the Mail2FAX box	Alpha-numeric
Email address	Email address associated with the mailbox	Alpha-numeric
Box settings		
Protocol	Protocol	Drop-down
Enable SSL	Enable SSL	Checkbox
Server address	Address of the mailbox server	Alpha-numeric
Server port	Port number of the mailbox server	Numeric
Timeout		Numeric
Username	Mail address of the user	Alpha-numeric
Password	Password associated with the mail address of the user	Alpha-numeric
Private PGP key		
Insert only if sending encrypted faxes		

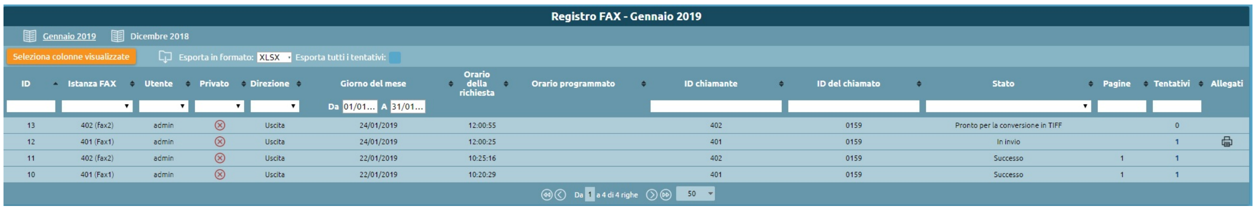
Save the settings and apply the changes.

Fax Register

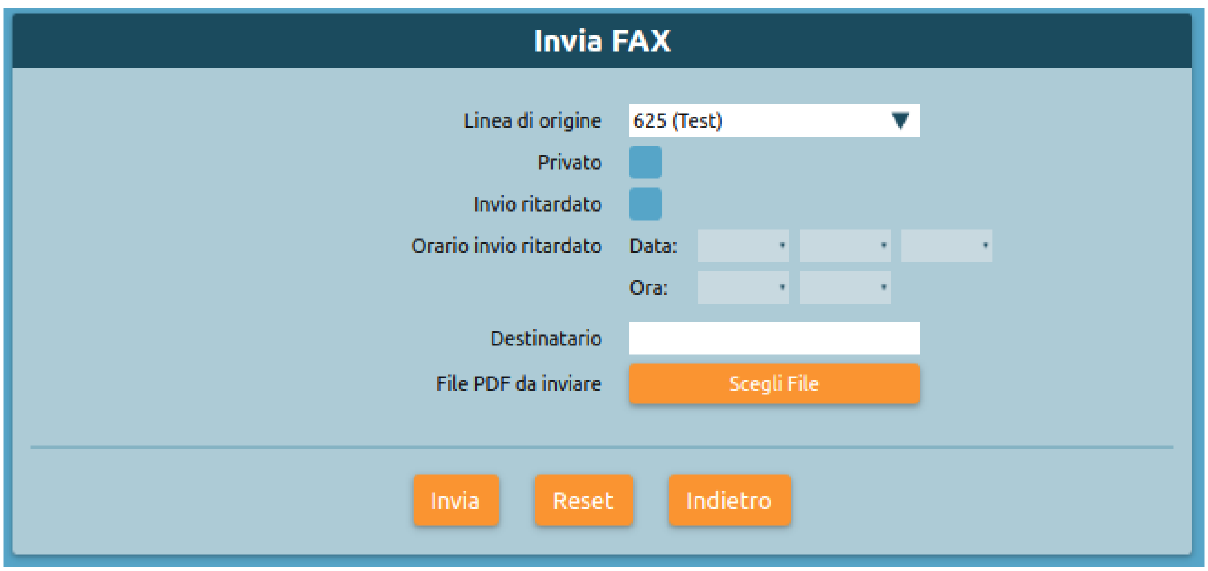
Each user assigned send/receive permission on a FAX instance displays the FAX panel -> FAX Register

In this module, you can:

- view the status of all received faxes and download the received file
- view the status of all sent faxes and download the sent file and the sending report



Sending a fax



Each user with send permissions for one or more fax entities can view the Fax -> Send fax page.

Accedendo a questo pannello è possibile impostare le opzioni di invio del FAX:

From this page, you can set the options for sending faxes:

- origin line (corresponds to the created entity)
- time of sending if required (otherwise the fax will be sent immediately)
- addressee
- pdf file to be sent (pdf, doc, docx, odt are supported)

Selezionando Invia il fax viene inviato.

Click on Send fax to send the fax.

You will be notified via email for each sent fax with the report attached.

Sending MAIL2FAX

Each fax entity must have an associated Mail2Fax box. In the fax entity settings page, select the Mail2Fax settings:

- **Mail2FAX box**: name of the previously created mailbox
- **Authentication method:**
  - None: the fax is sent and received only by checking the sender's email address
  - PIN: the FAX request is also authenticated by a pin that must be inserted in the text of the email
  - PGP signature: the identity of the sender of the email is authenticated using a PGP key
- **Request message encryption**: The email attachment must be encrypted using the sender's PGP key; in this case, it is necessary to load PGP public keys in the settings of each user authorized to use the service.

To send FAXes through mail2fax service, the sender email must be one of the users enabled to use the service (NOTE: the sender email control is case-sensitive, like the one of the mailbox used to collect the fax request). Then, from the "Users and Roles" panel, you have to insert for the user who manages the fax instance, the email address from which the FAX mail is sent. Select "edit user" and enter the email in the proper field. It is important that the same email address is not present for more than one user (even those belonging to different tenants) to avoid that the email is not correctly sent and/or received.

It is now possible to send an email with the fax attached from the email box. The email must have:

- in the "SUBJECT" field, the telephone number where to send the fax. It is also possible to insert the instance to which the fax should be sent by filling in the subject with PhoneNumber@istance
- in the "TO" field, the address of the mail2fax box; the fax line used for sending is (unless it is explicitly specified in the SUBJECT field) the first one to which the mail2FAX box is associated
- **in the "BODY" of the mail, any PIN associated with the FAX instance, if this authentication mode has been chosen.**
  - The body of the mail (in text format only) must contain only the string "FAXPIN:12345" (where 12345 is the PIN assigned in this example).
- The fax file (supported formats are pdf, doc, docx, odt) must be attached.

If the fax is successfully sent, the Fax module page will display the outcome of the transition and an email will be sent to the sender.

Receiving a fax

Each user receiving permission for a fax entity will receive an email notification with the document attached.

When the FAX is received, an email gets sent. It includes a pdf file of the received FAX as an attachment and, as a subject, a string formatted as follows:

Subject: [SERIAL NUMBER] FAX successfully received from 0XXXXXXXX where **0XXXXXXX** is the calling number preceded by 0 of the commitment line.

The following line shows an example of email notification of a FAX sent from the phone number 0501234567.

```
Subject: [KPBX40299999] FAX successfully received from 050123456
```

where 0501234567 is the phone number from which comes the FAX.

We also show an example of the email body (containing more information):

```
FAX received from 0501234567
Date and time: 12/03/2020 11:51:38
Transmission time : 34 seconds
Number of pages: 1
```

The pdf file attached to the email has the following naming pattern:

F<0calling_number>_T<reached_number>_YYYY-MM-DD_HH_MM_SS_FAXID.pdf

The following line shows the potential name of the pdf file attached to the mail of the received fax.

```
F00501234567_T0509655637_2020-03-12_11_51_38_211.pdf
```

They can also view the status of all received faxes and download the document from the Fax module.

4.2.2 Kalliope Hotel

Note: This service is tied to optional licenses and is available from firmware version 4.9.4 onwards.

Description

The Kalliope Hotel module is an add-on of KalliopePBX, which can be activated through a special license. It offers a set of specific functions for hospitality. The functions included are:

- **Receptionist panel**: this panel is used by the hotel receptionists for the ordinary management of the rooms and related services, including the occupancy status, the guest name, any notes and the next set of alarms. For each room, it is possible to access a detailed status panel, in which the room data can be viewed in detail and related operations can be carried out (activation of an alarm clock, check-in/check-out, setting the name of the main guest, etc.).
- **Check-in/check-out service**: this service changes the occupancy status of a specific room from "free" to "occupied" and the other way around. The status change events are time-stamped and are used to generate the debits documentation report for calls made from the room telephone, according to the configured rates (available from firmware 4.9.6). Using the Enabling Classes mechanism, it is also possible to differentiate access to telephone services in each of the "free" and "busy" states of the room.
- **Alarm clock service**: this service allows to generate a call at a predetermined time to the extension associated with a room; the response from the guest can be answering the call or by explicit confirmation before or after listening to the wake-up message; in case of no warranty, the system can repeat the ring several times, according to the configuration made for the service. The programming of the alarm can be done via web through the receptionist panel; for each room can be active more alarms, even at different times for each day.
- **Room cleaning service**: this service allows the room to be marked as clean/dirty and to switch its status either from the Receptionist panel of the GUI or by telephone via a service code from the person who cleans the room. The cleaning status of all occupied rooms is automatically set to the value "dirty" during the night and can then be reset to "clean" in manual mode.

Configuration

The Hotel Module is configured in the "Hotel Module" "Configuration" panel. The visibility of the panel and its tabs is conditional on the activation of the License. The configuration panel has four tabs:

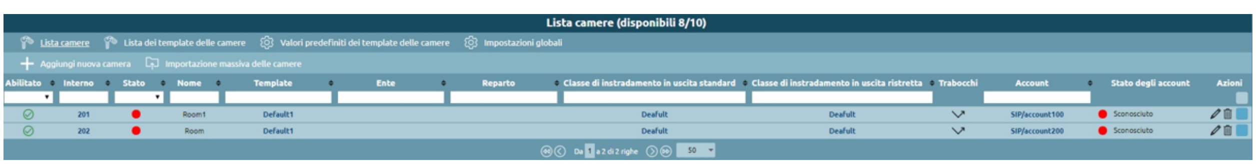
- Room List
- Rooms Template List
- Default values of room templates
- Global settings

Global settings

In this panel, you can set the prefix of the numerical selection that you need to type to set a clean/dirty room by telephone. From inside each room, it's possible to call this prefix followed by the number 0 to mark the room as clean or by the number 1 to mark the room as dirty. For example, if the code *33 is configured as a prefix, it is sufficient to call *330 from the room telephone to mark the room as clean. When the service is enabled, the relative selection will be displayed in the numbering plan in read-only mode. The other option allows to enable or disable the blocking of direct calls between rooms. With the active block, it will not be possible to make direct calls from one room to another; the rooms will still be able to call the other extensions of the central unit (or be called by them) or other selections of the numbering plan.

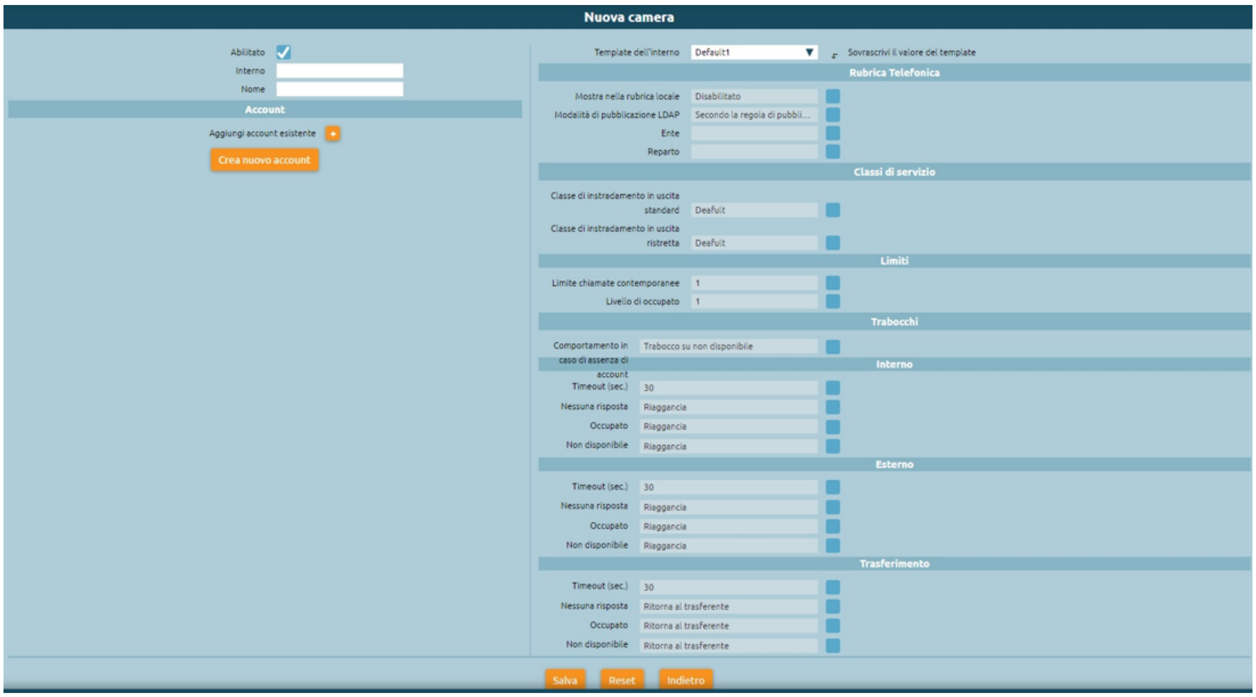
Rooms configuration

The configuration of the rooms is carried out in a similar way to what is done for the configuration of standard extensions. Also, there is a configuration mechanism based on templates to manage the settings common to multiple rooms and a panel in which to specify the default values that will be used each time a new room template is created. The workflow involves first configuring the default values of the templates (creating one or more room templates) and finally making the actual rooms with their extension numbers. As for the standard extensions, it's possible to assign one or more SIP accounts to those of the rooms and it can be used on one or more room extensions; unlike the extensions, there is no particular distinction for the SIP accounts used in the room extensions, so before proceeding to the creation of the rooms it is possible to create the necessary accounts through the usual SIP account management panel, in the "PBX" menu > "Extensions and Accounts". As for the extensions, to facilitate the creation of many rooms, it is possible to use the massive import procedure by XLS/CSV file, using as a template the file available by clicking on the link "Massive import of rooms".



Selecting "Add new room" it is possible to add a room to the list and configure it. As indicated above, rooms' configuration follows the extensions' configuration by replicating the principle of configuration using templates; the parameters available for the configuration of the rooms are a subset of those of the standard interiors. The configuration parameters of a room are:

- **Extension**: the telephone number associated with the room;
- **Name**: The name of the room. Not necessarily the same as the room number, it is listed along with the extension on the receptionist dashboard;
- **Add existing account / create account**: allows you to associate one or more previously created SIP accounts with the extension, or to create one in line with the room configuration;
- **Extension template**: indicates the template containing the default parameters to be used for the selected extension type. All the other attributes present in the panel import the default values but it is possible to overwrite them if necessary.

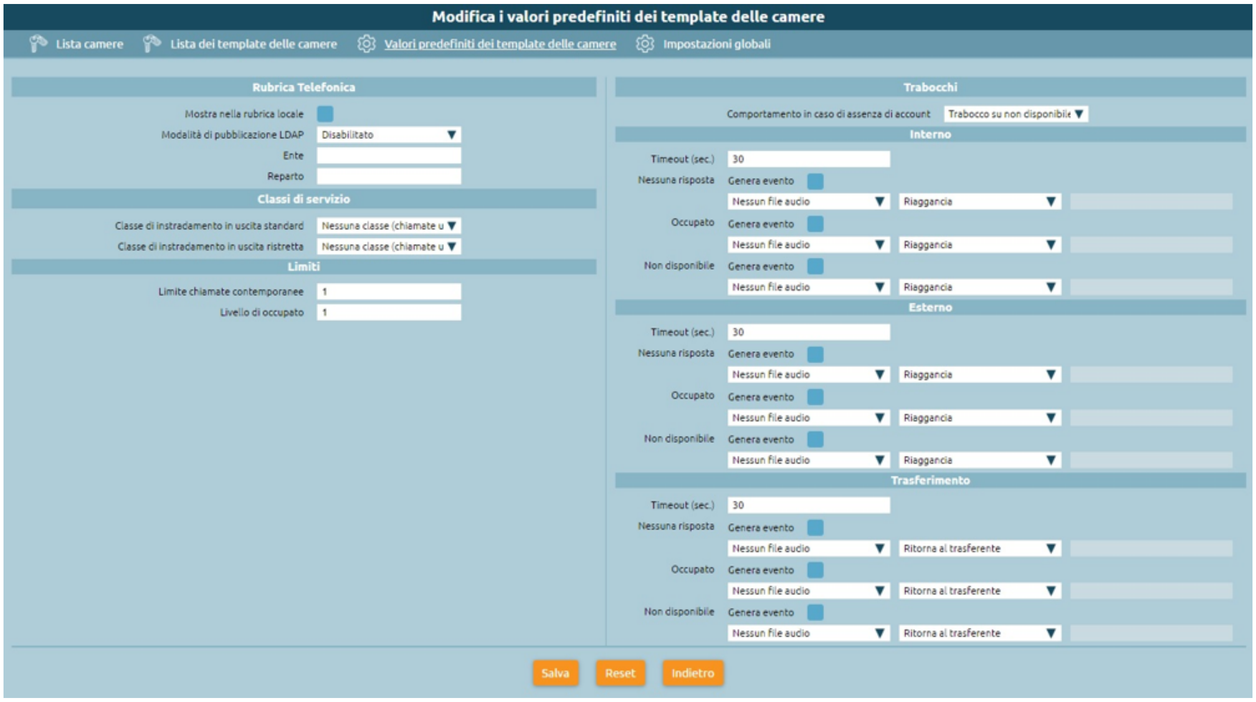


The following settings are a consequence of the assigned template (adjustable from the "List of Room Templates" panel, the same used for standard extensions templates) with the possibility of overriding each setting. When creating a new room template, default values are initialized to those specified in the "Default values of room templates" panel. The room configuration parameters that can be inherited from the template are:

- **Show in local phonebook**: Enables or disables the display of the extension in the extension phonebook
- **Ldap publishing mode**: Indicates how the extension is published in LDAP among the various options available, in a similar way as for extensions
- **Organization/Organizational unit**: these two configuration attributes are used as filtering parameters within the Receptionist dashboard, where they are interpreted with the label "Building" and "Floor"
- **Standard/Restricted outbound routing class**: as for extensions, these two classes define the type (and routing) of external calls allowed. When a room is in the "free" state (i.e. not occupied), the restricted class is assigned to it, while when it is in the "occupied" state (i.e. associated with a guest, after check-in), it is possible to select from the room management widget which of the two classes it uses. In this way, it is possible to prevent outgoing calls to the phones of the rooms, even if they are occupied, and eventually unblock them upon request of the guest
- **Calls limit/Busy level**: identical to the same settings of the extension, they allow to define respectively the maximum number of simultaneous calls possible for the extension and the number of calls in progress on an extension

upon reaching which this must be considered busy (and therefore any further attempt to call this extension will end with this result);

- **Failovers**: as in the case of standard extensions, indicate how a call destined to the extension that ends with one of the three possible outcomes (not answered, busy, not available) should be handled for each of the three possible origins (internal, external or transfer to the extension).



Alarm clock service

The license of the Hotel module includes the enablement of the wake-up service. Before you can use the alarm clock service within the Hotel module, it is necessary to configure it in advance from the "PBX Applications" > "Alarm clock Service" panel. The alarm service allows setting one or more alarms for each room, in the form of date and time. At the scheduled time, KalliopePBX will make one or more calls to the room and optionally collect the confirmation of reception from the guest.

The general configuration of the service is done in the "Alarm Service Settings"; these include:

- The explicit enabling of the service
- The assignment of a Name (which will be used as Display Name for calls made by the service
- The audio file to be played to the guest when they answer the wake-up service call

In addition to these necessary settings, there are other parameters with which to customize the way the service is used:

- Number of call attempts: this is the maximum number of calls that PBX will make to the extension for each set alarm clock, if the guest doesn't acknowledge the answer. By default, answering the call constitutes an answer confirmation, but you can request confirmation by typing a key before or after the alarm sound message is played
- Timeout: is the time for which the destination extension is made to ring before considering the wake-up attempt failed. The system will repeat the call a maximum number of times equal to the value set in the previous step
- Ask for confirmation answer: before playing the audio message, you must press key 1. Failure to press the key prevents the call from continuing; if you hang up, the alarm is considered "not confirmed" and a further call attempt (if there are any) is made by the control unit;

- Ask for confirmation of listening: similar to the previous one, but at the end of the playing of the alarm message (in this case you are asked to press button 9)

At the end of the number of call attempts without confirmation (according to the configured modalities described above), the alarm will be "not answered" and a warning will appear in the dashboard of the receptionist for information purposes. These alerts can be acknowledged (and therefore deleted) by the receptionist after he has carried out the necessary actions (manual call to the room, etc.). The two tabs in the panel "List of wake-up calls" and "List of finished wake-up calls" contain the list of active and finished wake-up calls. The first panel contains those programmed in the future, currently active or finished without confirmation of reception; the second panel contains the list of completed alarms with answers or those not answered but manually taken over by the receptionist (following the cancellation of the alert on the receptionist panel). In both cases, for each alarm it is possible to visualize the complete list of events related to each call attempt with the relative timestamps.

Receptionist panel

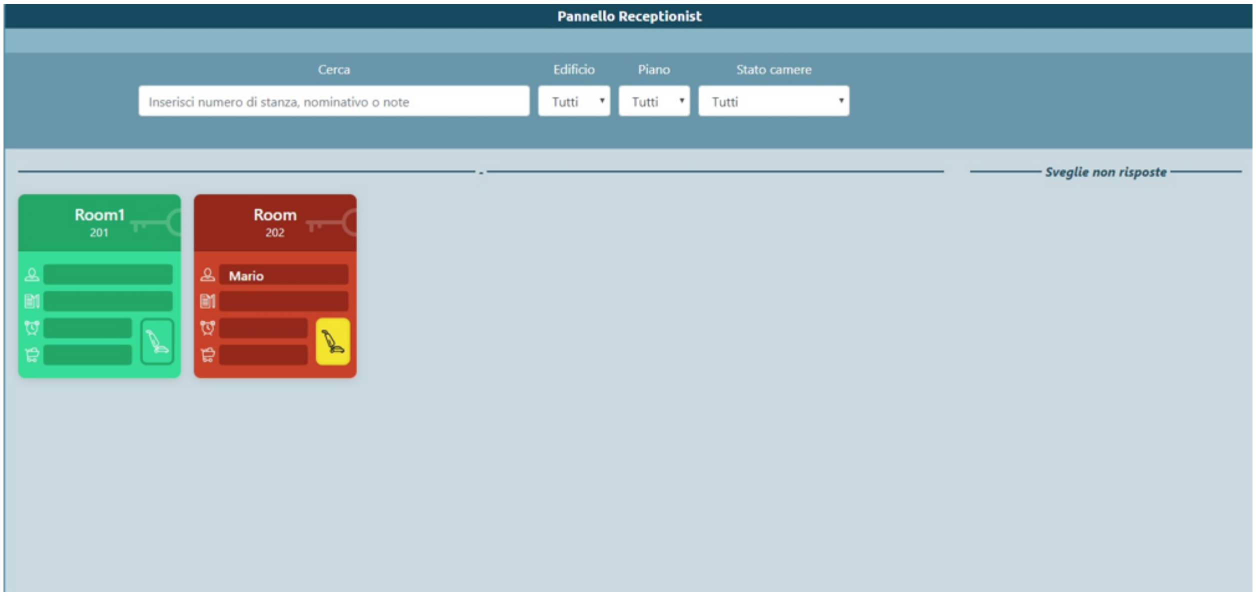
This panel collects all the rooms and is used by the hotel receptionists to manage the rooms and related services. It is divided into two parts: the column on the right that contains notifications for unanswered alarms from guests and the main area on the left, where all rooms are displayed in matrix form, grouped according to the attributes "building" and "floor" of location. The "no answer alarm" alerts appear automatically when all the call attempts associated with the alarm of a room end unsuccessfully; the alert shows the name and number of the room and the time at which the alarm was programmed. By clicking on the X inside the alert, it will be deleted; simultaneously, the corresponding alert will be moved from the panel of the active alerts to the panel of the terminated alerts (see the previous section on the alarm service). Each room configured in the central unit appears in the left section of the dashboard as a widget (or box); the color of the widget indicates the room occupation status: a "green" room is free while a guest occupies a "red" room. In the latter case, the four text fields inside the room box indicate in order:

- The name of the guest(s)
- Any notes associated with the room
- The first scheduled alarm clock for that room
- The total cost of the calls made by the room from the time of check-in to the current time (available starting with Firmware 4.9.6).

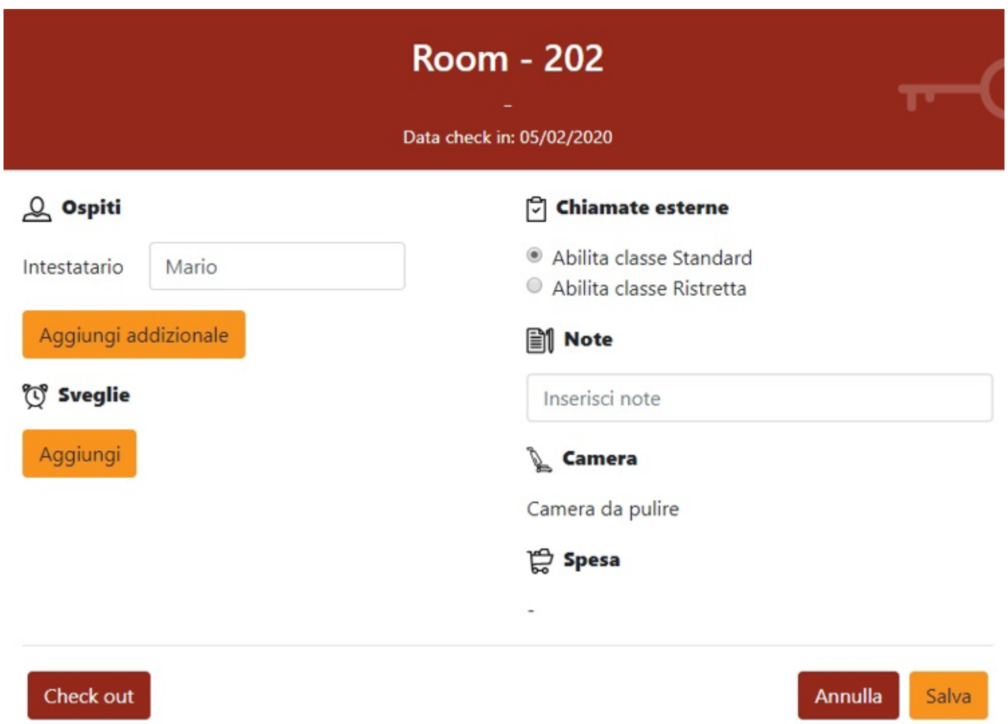
The top bar allows you to apply real-time filtering in the room view based on:

- Building and/or floor
- Cleanliness status
- Free text matching on the "guest names" and "notes" fields of the room
- Finally, inside the widget, there is a button with a vacuum cleaner icon: this button has the dual function of indicating and being able to switch the cleaning status of the room

In the case of a dirty room, the button is on (yellow). By clicking on the button the receptionist can switch the status from "dirty" to "clean" and vice versa. The cleaning status of the rooms is updated every 10 seconds, any change in the status via telephone code from the room is reflected in real time on the status displayed on the panel.



Clicking inside the widget of a room, you can enter the detail status panel, where you can perform check-in and check-out operations and view room data more broadly.



Selecting a free room, it is possible to check-in by clicking on the button with the same name. During check-in, it is necessary to indicate the name of at least one guest in the room, but it is also possible to add the names of additional guests, and any textual notes. Together with the check-in, it is possible to choose the two enabling classes (to external calls) to attribute to the room. This setting is editable afterward, going back to the detail panel of the room.

During the check-in operation, it is possible to enter the following data associated with the room:

- **Owner (required)**: name of the person to whom the room is registered. Additional guests can be added by selecting "Add Additional". When filtering the rooms from the dashboard, the free text search operates on all the names entered

- **Alarms**: date and time to set the room alarm if the owner requested it. You may add one or more alarms by selecting "Add". This service can only be set by the operator
- **External calls**: enable external calls from rooms with standard or restricted class. This service generates of a detailed reporter of the calls made from a room phone since the moment of check-in, including the cost calculated from the duration according to a specific rate configurable.
- **Notes**: free text, allows you to enter reminders. When filtering rooms from the dashboard, the free text search also operates on the content of this field.

Once the configuration is completed, clicking on "Save" the check-in is performed (the widget changes from green to red), and the relative timestamp is associated with the room to calculate the calls competence. In addition to these settings, in the detail panel of the room, it is possible to see the cleaning status of the room and (from firmware 4.9.6) the cumulative count of the cost of the calls made from inside the room from the check-in moment until the current moment. If you select an occupied room, you can click on the "Check out" button to perform the corresponding action (after confirmation), after which the room status returns to available ("free").

Reservation History List

In this panel it is possible to view the history of reservations for each room, displaying the date and time of check-in and check-out, the name of guests, and any notes. For each room, it is also possible to download the XLSX report with the list of calls made by the guests of the specific room within the period of occupancy. The panel contains both the current occupations (the check-out timestamp is empty) and the past ones. The guest information and notes present at the check-out time are saved in the reservation history.

Documentation of debits (Billing Service)

The Debit Documentation service present on KalliopePBX (currently available as a component included in the Hotel Module license), allows the calculation of the costs of calls made from each extension associated with a room, through the definition of a cost profile that can be differentiated according to the destination (with a specific prefix, exact numbers, all other numbers) and the outgoing line used. Learn more through this page:

Call charges (Billing Service)

Note: This service is available starting from firmware version 4.9.6 as an additional feature included in the "Starter Kit" license of the Hotel module

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Description

The Call charges service on KalliopePBX allows the calculation of the costs of calls made from each extension associated with a room, through the definition of a cost profile that can be differentiated according to the destination (with a specific prefix, exact numbers, all other numbers) and the outgoing line used.

The Call charges configuration consists in the definition of one or more charging profiles, which are of a set of routes – identified by prefix, exact selection or all other numbers – each associated with a charging rule. The charging rule makes it possible to calculate the cost attributable to the call based on its duration, using four parameters:

- The cost of the connection fee;
- The duration in seconds included in the connection fee;
- The cost of the subsequent connection fees;
- The duration in seconds of each subsequent connection fee;

These 4 parameters allow the creation of Flat pricing profiles (e.g. a fixed response price that includes the entire duration of the call), per second without response time (by setting the period of the steps to 1 second), or the classic step profiles. Once at least one billing profile has been defined, it is necessary to associate it with the outgoing lines used by the rooms (each outgoing line can be associated with a different charging profile) so that a cost, calculated using this profile, is associated with the calls made. At the moment, there are no differentiated tariffs for time bands or days.

Configuration

The availability of the call charges service is subject to the presence of the Kalliope Billing Module license. In the case of Multi-Tenant nodes, it is necessary to assign the license to all Tenants in which it is required to use the service.

The call charges service is configured in the "PBX Applications" –> "Billing Module". The visibility of the panel and its tabs is conditioned by the permissions associated with the user's role

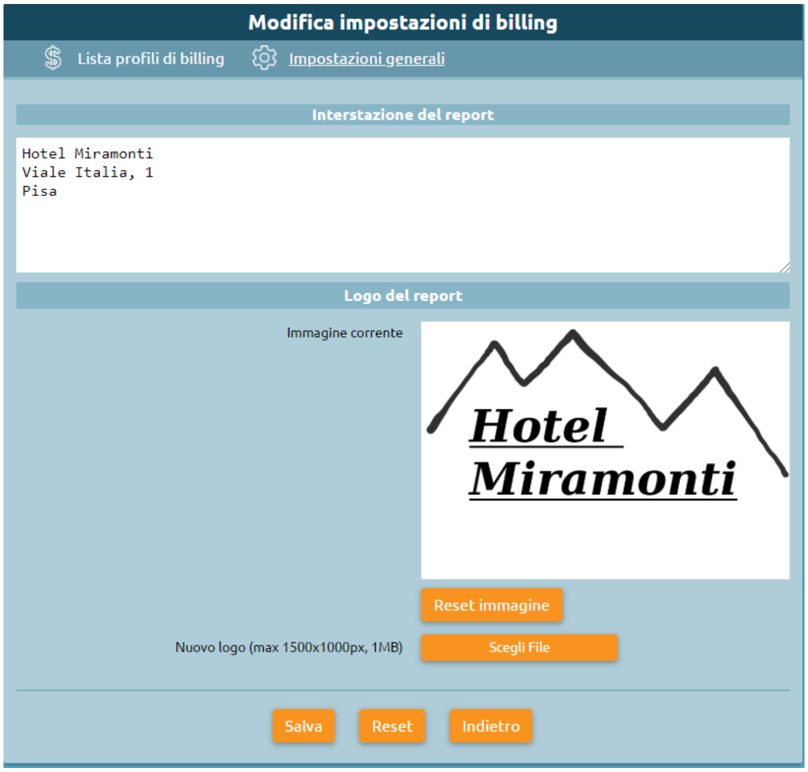
The panel has two tabs:

- General Settings
- Billing Profiles List

In the "General Settings" panel, it is possible to upload an image (e.g. logo of the structure) and a header; these are used to customize the report generated by the system with the list of calls made and the related costs.

Note: The report is generated in XLSX format; the header and the image are inserted inside the header of the spreadsheet, so in the "Normal" view, they are not shown on the screen, but only in the "Page Layout" view or in print preview.

Note: LibreOffice or OpenOffice software, in current versions, does not support the insertion of images inside the page header, so the image will not be displayed if one of these applications is used to open the report.



The "Billing Profiles List" panel lists the billing profiles created by the user, which must then be associated with the outgoing line so that the cost to be charged for each call made is correctly evaluated.

By selecting "add new traffic profile" the administrator, or delegated user, can configure multiple charging profiles. Each profile contains a list of charging rules that will be matched based on the number called. The rules are grouped according to the selection match mode: exact match, prefix match, or any. In the case of prefix dialing, a "longest match" type of matching is performed, then the charging rule is selected for the most extended selection among those present that matches the called number. When the rules are saved, they are automatically sorted according to this principle to make the visual feedback more immediate. As an example, if the following prefix destinations are configured:

- 0
- 00

- 001
- 050
- 003
- 0033

the order in which a called number will be found will be:

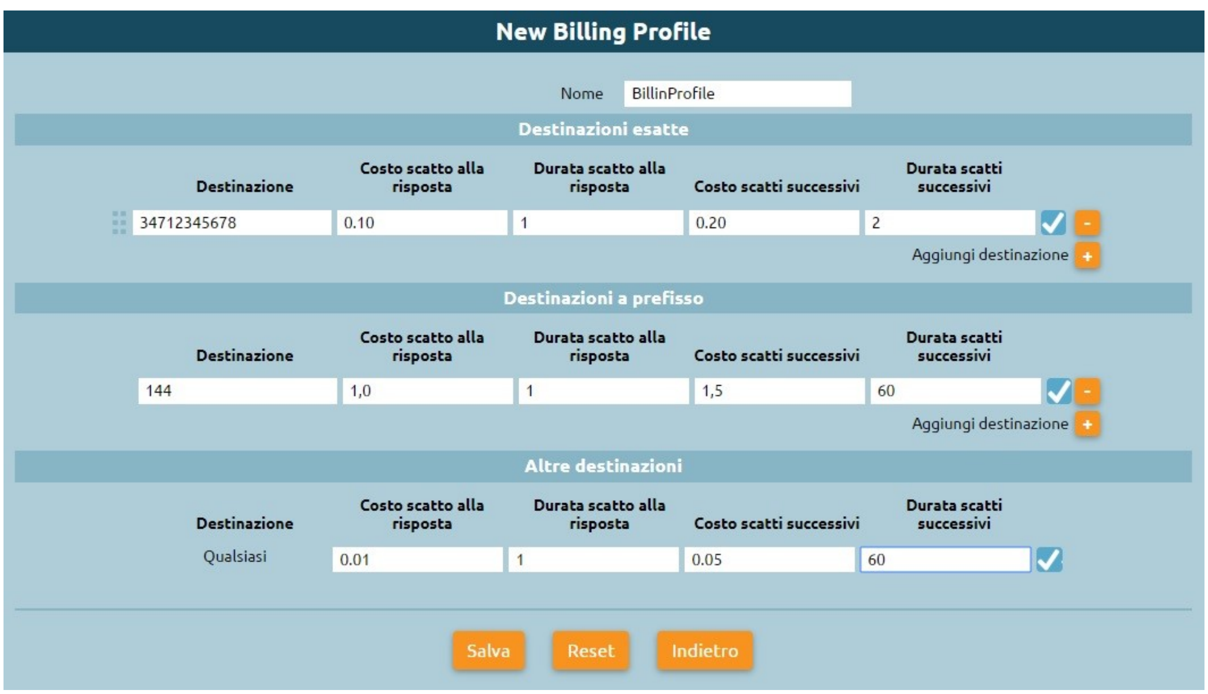
1. 001
2. 003
3. 0033
4. 00
5. 050
6. 0

Therefore, a call destined to the number 001555123456 will be charged according to rule 1, a ring destined to 0041234567 will use rule 4, while all urban calls (unlike those destined to Pisa numbers) - 050 - will use rule 6.

Finally, each rule is completed by the charging parameters, where costs are expressed in Euros (and fractions) and times in seconds:

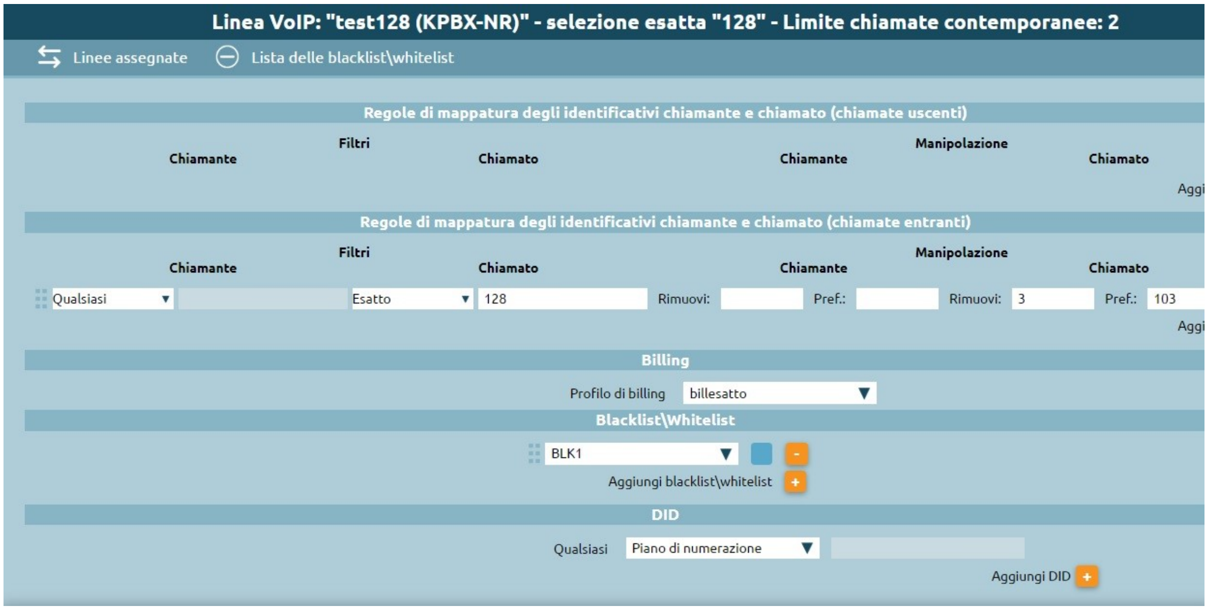
- Cost of the connection fee;
- Conversation time included in the connection fee;
- Cost of subsequent connection fees;
- Duration of the subsequent connection fees.

Only outgoing calls with call times higher than zero seconds are included in the report.



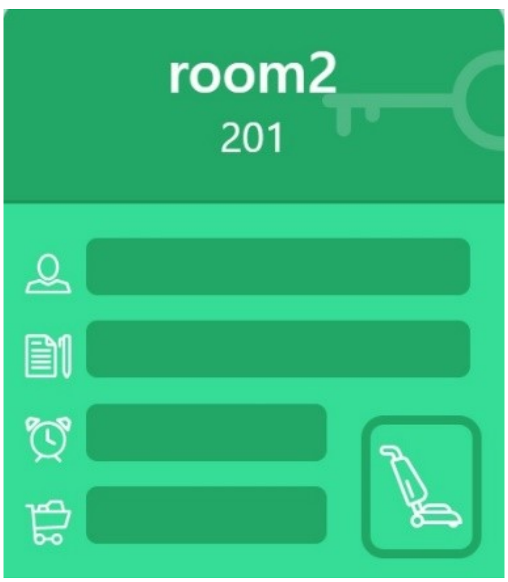
Once one or more billing profiles have been created, it is necessary to associate them with the output lines from the configuration panel of each of them (Gateway, Trunk or Terminations in the case of a Single-Tenant central unit or

Assigned Lines in the case of a Tenant of a Multi-Tenant node) by selecting the billing profile in the section of the same name.



Data visualization and report generation

The total cost of calls made from a room is displayed in real-time on the widget of the room itself, present in the receptionist dashboard. In this way, it is immediate to visualize anomalous consumptions by a room and eventually modify the enabling class to block or limit further outgoing calls.



Entering the detail of the room, there is a button ("Download Call Report"), which allows downloading the report in XLSX format containing the details of calls made from the moment of check-in to the moment of report generation (including date and time of the ring, the destination number - anonymized in the last three digits - and the duration of the conversation) and the related costs.

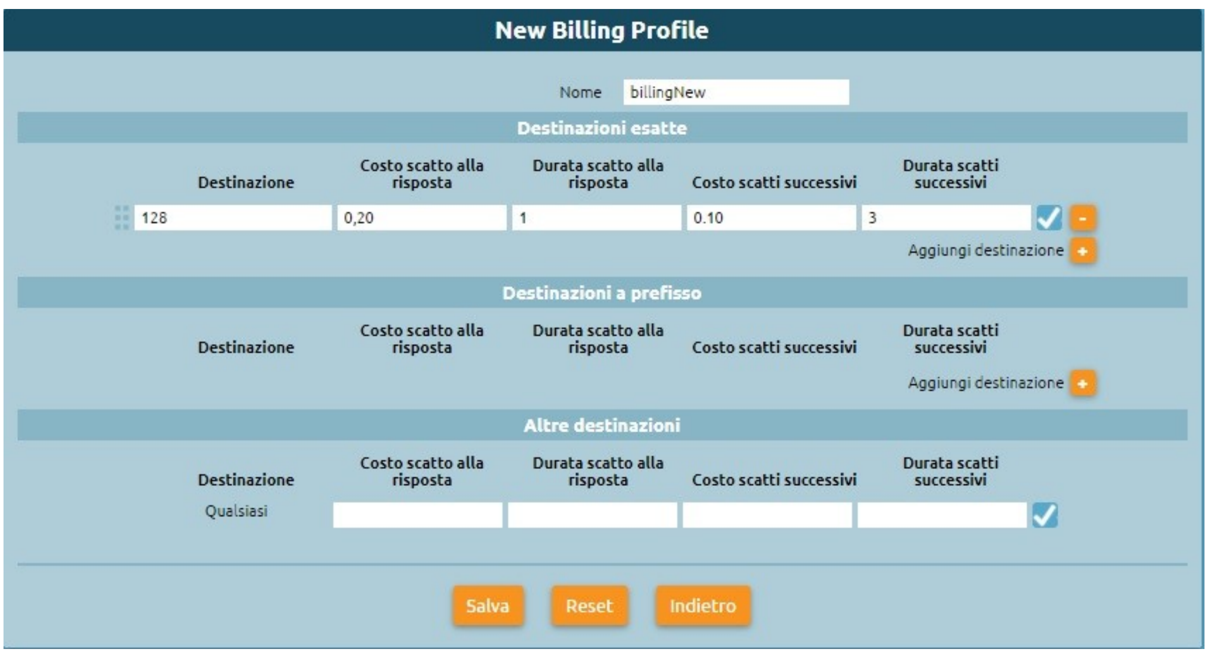
Di seguito si riporta un esempio del report generato (al netto della formattazione di pagina):

Camera 202				
Dettaglio costi telefonici periodo 10/03/2020 – adesso				
Giorno	**Ora**	**n° chiamato**	**Durata**	**Costo**
10/03/2020	*18:54:14*	*0501234xxx*	*00:03:14*	*1,50 €*
10/03/2020	*21:12:13*	*3319001xxx*	*00:02:15*	*2,50 €*
11/03/2020	*07:25:33*	*3319001xxx*	*00:01:55*	*2,50 €*
11/03/2020	*20:55:10*	*0501234xxx*	*00:03:05*	*1,50 €*
12/03/2020	*07:44:24*	*3319001xxx*	*00:02:44*	*2,50 €*
				Totale
				10,50 €

Billing Profile Cloning

As seen above, the "List Billing Profiles" panel lists the billing profiles created by the user. By selecting "add new billing profile" the administrator, or the delegated user, can configure multiple billing profiles, but it is also possible to clone an existing billing profile.

Selecting the document icon, the "New Billing Profile" panel will open.



The panel has the same settings of the profile we want to clone and editing the name or modifying the destinations data, by selecting "Save", the new profile will be created.

Import

It is also possible to perform the massive import of the billing profiles. By selecting the item "mass import" in the "Billing Profiles List" panel, it is possible to upload an Excel file containing one or more billing profiles.

The Excel file must respect the format of the following template:

name	type	selection	connectionfee	connectionduration	nextunitsfee	nextunitsduration	enabled

- Name: Indicative name of the billing profile
- Type: Indicates whether the billing rule is associated with exact (exact), prefix or any other number (any)
- Selection: Number or prefix with which the rule is associated
- ConnectionFee: Cost of connection fee
- ConnectionDuration: Duration of connection fee
- nextUnitsFee: Cost of subsequent connection fees
- nextUnitDuration: Duration of subsequent connection fees
- Enabled: Flag indicating whether the billing profile is active or not

Export

From the "Billing Profiles List" panel, it is also possible to export the billing profiles created in an XML format file. Selecting "export" in fact, an Excel file is created that contains the data of the billing profiles created as the following:

name	type	selection	connectionfee	connectionduration	nextunitsfee	nextunitsduration	enabled
billing	any		0,01	1	0,05	60	t
billing	exact	34712345678	0,1	1	0,2	2	t
billing	prefix	144	1	1	1,5	60	t
billingNew	any						t
billingNew	exact	128	0,2	1	0,1	3	t
billingesatto	any						f
billingpref	any						f
billingzero	any						f

4.2.3 Kalliope LAM

Note: This service is tied to optional licenses and is available starting with firmware version 4.11.X and subsequent updates.

Description

alliope LAM Module is an add-on of KalliopePBX. You can get it by a speciific license that offers a set of functions to support the business continuity of a company, regardless of the location of its employees and customers. KalliopeLAM allows you to organize virtual meetings in a physical meeting room just as you would. Based on an open-source engine and entirely in the cloud, it is a web-based platform (click and go), meeting participants access the meeting through their web browser by clicking on the link contained in the invitation email. The mobile app version for Android and iOS operating systems is also available and can be downloaded for free from Google Play and the App Store. For more information:

Kalliope LAM app Mobile

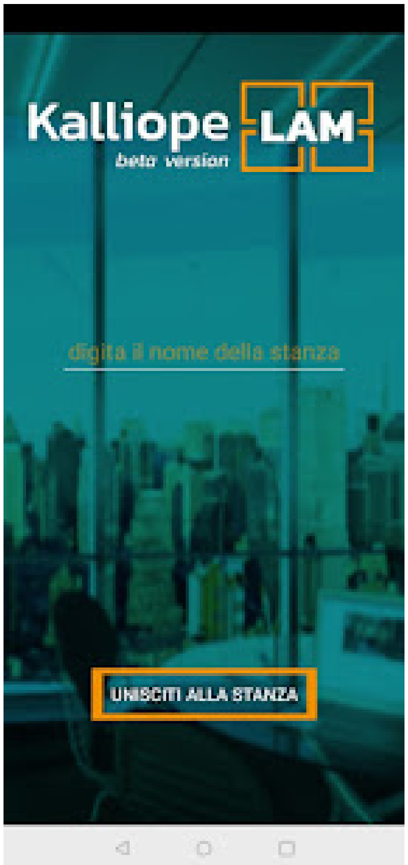
With the KLAM app, you won't miss any more meetings.

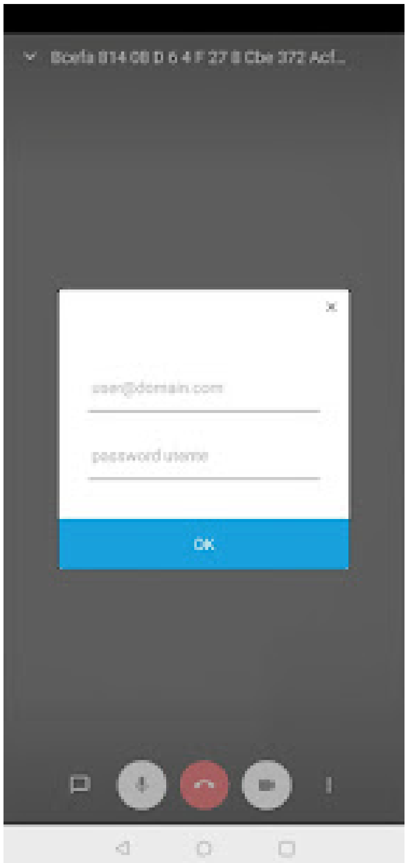
KalliopeLAM is an accessory license that can be integrated with an already existing and activated KalliopePBX PBX. It can be purchased for a monthly fee with a yearly renewable contract. The app is free and included in the license.

Have you already tried KalliopeLAM on your PC? Well, the same experience you know and enjoy is now also available on your smartphone. Available for both Android and iOS. Whether you're sitting at your desk or out on an appointment, the KLAM app provides accessibility and reliability to take your work with you. Attend your meetings wherever you are.

HOW DO I CONNECT TO THE MEETING?

Accessing the meeting is simple: download the app, click on the link you receive from the meeting organizer and take part in the meeting! The only thing you have to do is enter your name and a password (if required), all as if you were sitting at your desk.


Kalliope
beta version
LAM
digita il nome della stanza
UNISCITI ALLA STANZA


user@domain.com
password userne
OK

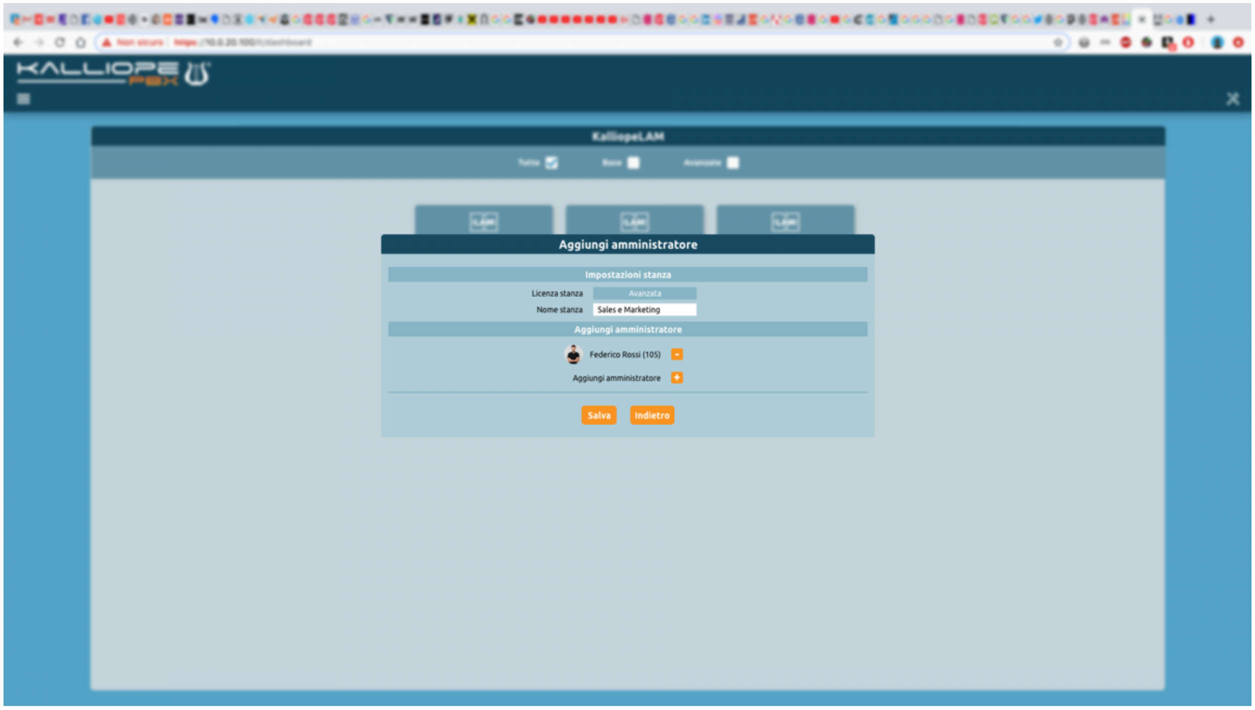
WHAT I CAN DO DURING A MEETING:

- Share your screen
- Invite other participants
- Chat during the meeting
- Raise your hand if you want to speak
- Share YouTube video

Configuration

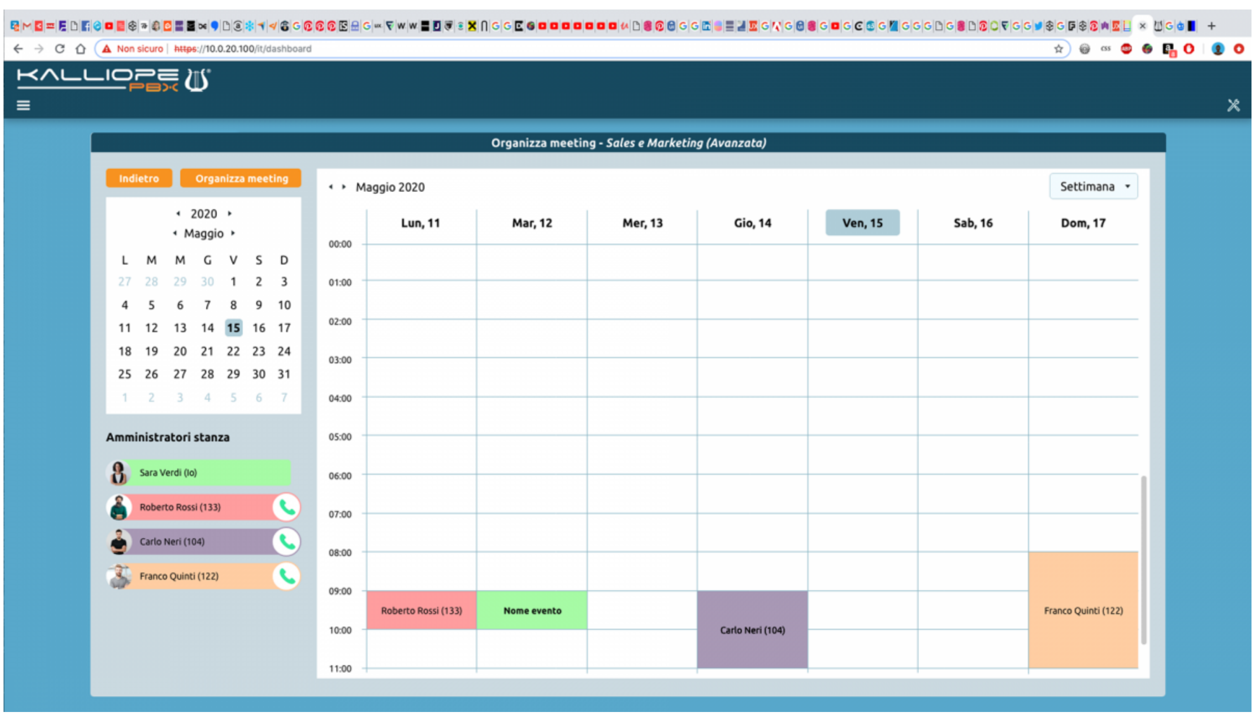
The management of the KalliopeLAM platform by users, for the creation of events or the assignment of permissions, is fully integrated into the web interface of the PBX. The KalliopeLAM solution is not licensed for users but for rooms, each license enables a videoconference room, a real virtual meeting room for which it will be possible:

- Give the room a specific name
- Define the ownership of the room: assign its use to one or more users. There is no limit to the number of administrators for each room, as long as they are internal to the PBX. The user who has management permissions for the entire KalliopeLAM service has a complete overview of the virtual meeting rooms; he can choose a room.

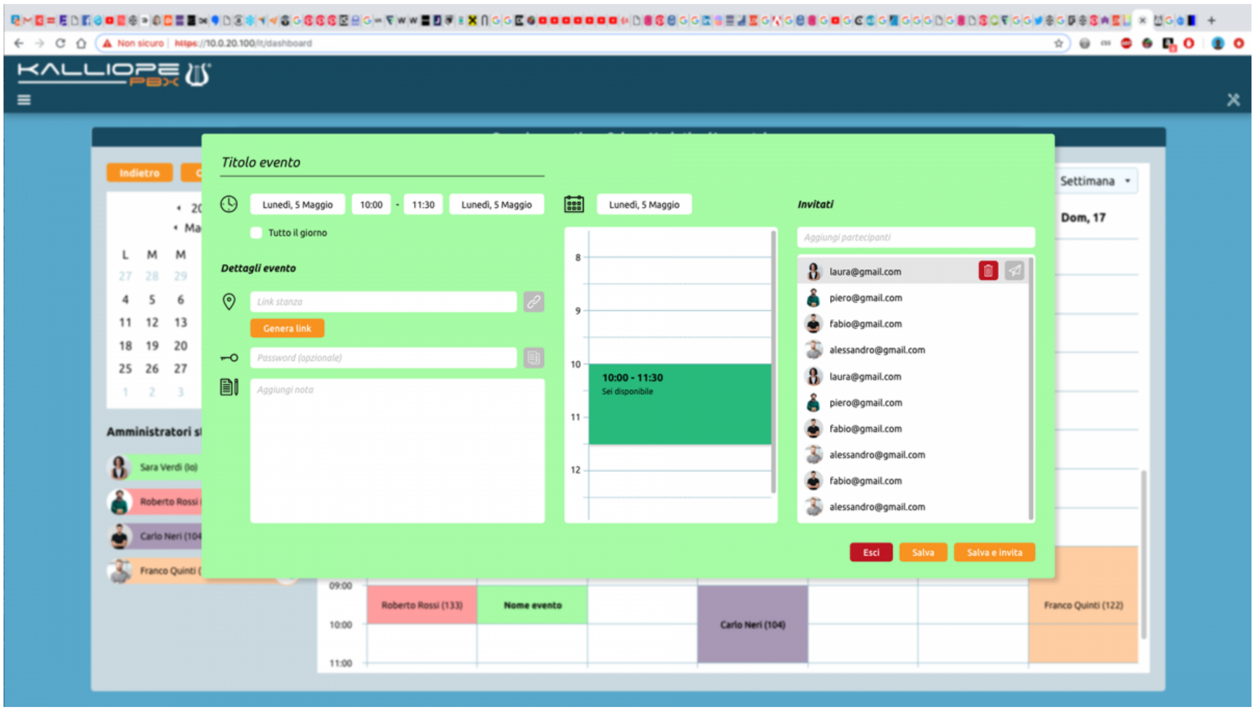


The user can:

- Access the “meeting schedule calendar” for a particular room and choose to schedule a meeting



- Create a new meeting by entering the following fields: meeting name, time, login password (optional), list of attendees



- Send invitations and calendars to all participants of a meeting by clicking the "Save and Send" button

Main features

- Webconference
- Meeting protection with password
- Waiting-room
- Desktop sharing
- Chat
- Raise a hand
- Partner statistics
- YouTube video sharing

Protecting the privacy of your meetings

Several mechanisms have been implemented to ensure the privacy and protection of your meetings on the KalliopeLAM platform.

- New videoconference instances for each new meeting: the different istances will be valid only in the period indicated for the booking, start and end of the meeting. These instances (represented in the meeting sharing URL) consist of 42 alphanumeric characters, and in the case of a moderator invitation, there will be an additional 493-character token at the end of the URL
- Waiting room enabled by default: the moderator is notified that a guest has requested to enter and can decide to accept or not. The moderator can disable the waiting room on a metting-by-meeting basis.

- Definition of password protection for access to the meeting : The password can be defined in two different moments: either during the scheduling of the meeting and during the conference itself
- If the moderator leaves the meeting, guests are disconnected
- The moderator can elect other users as "moderator"

CHAPTER

FIVE

KALLIOPE NEXUS

5.1 Kalliope Nexus

5.1.1 Analytics

Description

The Kalliope Analytics module collects all data collected from CDRs (Call Detail Records) in order to offer a detailed analysis of the use of company telephone resources. The reports that are offered were born from the need to offer a simple, fast and intuitive method to understand what the call trends were within your company.

This module allows you to have:

- professional, yet intuitive reports
- accurate and effective charts
- clear and comprehensive dashboards
- customizable reports
- exportable data tables

Structure

The module is structured into two main sub-modules, both of which have sections. The first sub-module collects data from the CDR and processes it, while the second does so for data from the Call Center. In both cases, the structure of the sections is very similar, with an initial dashboard and then the various report sections. The data that are collected are then processed and made available to the various report groups (CDR or Call Center) in the different sections into which they are divided.

VoIP report CDR

Dashboard

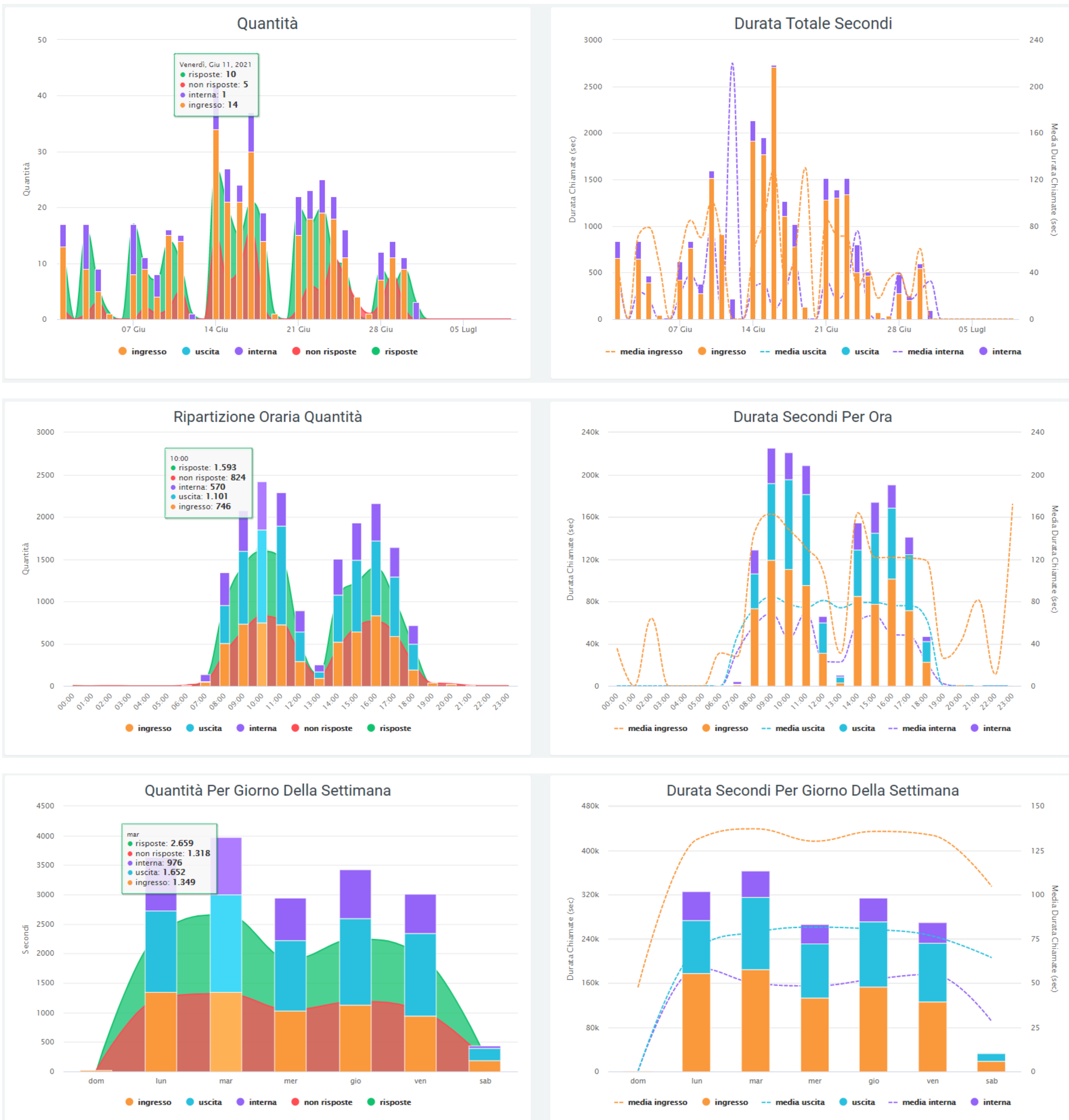
Charts

The data collected in this section are contained in "Voip -> CDR" and concern the calls passing through the telephone exchange.

The Dashboard page displays graphs showing call trends within the affected telephone exchanges. You can view the number and duration of incoming, outgoing, and internal calls, also showing how many calls were answered and how many were not. To highlight specific portions of a graph, simply press on it and scroll by the portion of interest.

The data can be broken down:

- By daily breakdown: the amount of calls received in a day
- By hourly breakdown: the duration of calls for each day of the period entered, number of calls by time of day, duration in seconds by time of day
- By weekday breakdown: the call quantity for each day of the week and duration in seconds of calls according to the day of the week

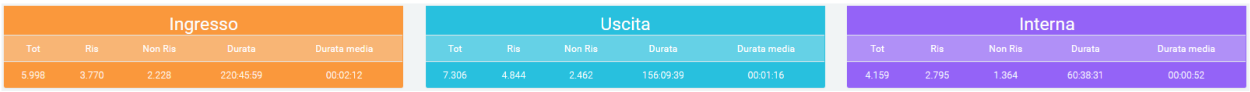


By scrolling over the graphs with the mouse, it is possible to observe the precise values of amount of total, answered and unanswered calls (as shown in the figure above).

The data in the graphs, shown in the legend, can be removed from the display by clicking on the colored dot next to the name. To view them again, simply click on them again.

The same data can be found tabulated below the last graph, broken down by incoming, outgoing, and internal calls (include both calls made and received by an operator), each broken down into:

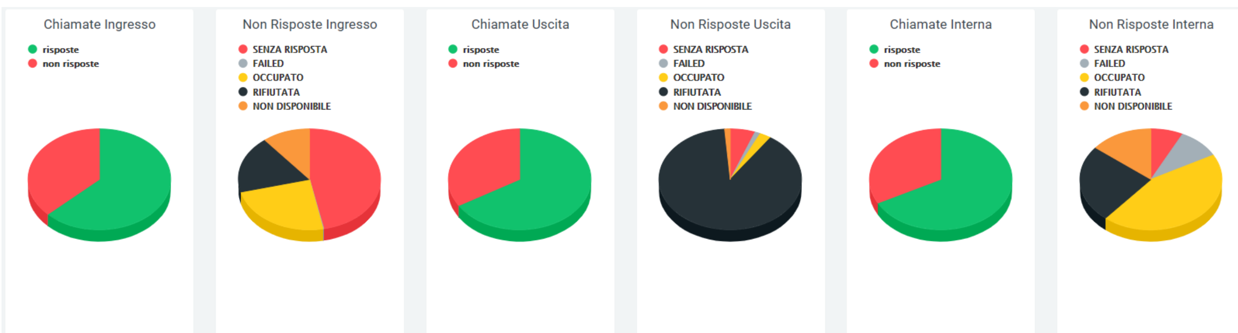
- Total calls
- Answered calls
- Unanswered calls
- Total duration of all calls
- Average call duration



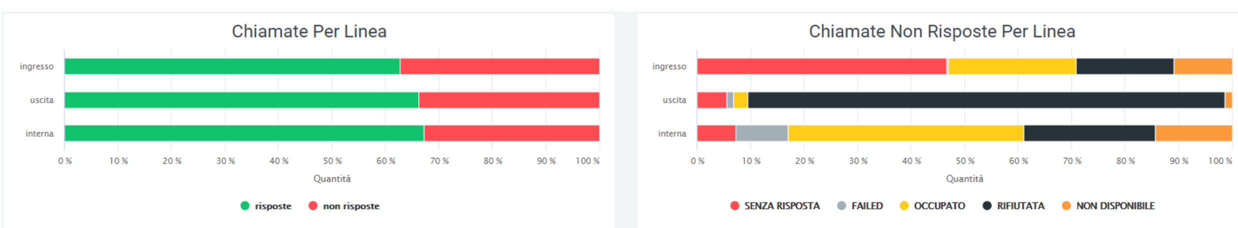
Ingresso					Uscita					Interna				
Tot	Ris	Non Ris	Durata	Durata media	Tot	Ris	Non Ris	Durata	Durata media	Tot	Ris	Non Ris	Durata	Durata media
5.998	3.770	2.228	220:45:59	00:02:12	7.306	4.844	2.462	156:09:39	00:01:16	4.159	2.795	1.364	60:38:31	00:00:52

Pie charts

Below the tables are pie charts depicting the number of incoming, outgoing and internal calls, divided into "answered" and "unanswered." For the latter, there is a further breakdown with the reasons for non-response, which can be: unanswered, failed, busy, declined, unavailable.



The same data are also represented in a horizontal band graph. Again, scrolling over it with the mouse will display the quantities and percentages. Again, scrolling over it with the mouse will display the quantities and percentages.



Tables

All the data represented in the previously illustrated graphs can be found broken down within dedicated tables. Each table has the option of being downloaded as an excel file by clicking on the "Export XLS" label in the upper right corner of the table. For each breakdown (daily, hourly, or weekly) the table shows the number of answered and unanswered calls broken down by incoming, outgoing, and internal calls. In addition to this, the total call time is reported for internal and outgoing calls, and for incoming calls:

- The total call times
- The total hold time

- The total talk time
- The percentage of hold time to total time

Totale

Export XLS

DATA	TOTALE RISPOSTE	TOTALE NON RISPOSTE	INGRESSO N. RISPOSTE	INGRESSO N. NON RISPOSTE	INGRESSO T. TOTALE	INGRESSO T. ATTESA	INGRESSO T. CONVERSAZIONE	INGRESSO % TEMPO ATTESA	USCITA N. RISPOSTE	USCITA N. NON RISPOSTE	USCITA T. TOTALE	INTERNA N. RISPOSTE	INTERNA N. NON RISPOSTE	INTERNA T. TOTALE
01/06/2021	578	327	155	82	09:20:43	02:04:21	07:10:51	22.4%	304	153	10:26:55	119	92	03:15:30
02/06/2021		16		16	00:46:29	00:46:20		100%						
03/06/2021	551	263	161	101	08:36:07	01:54:19	06:34:44	22.46%	256	108	08:41:04	134	54	02:47:23
04/06/2021	516	221	175	60	08:39:06	01:36:03	06:58:34	18.66%	222	121	06:26:54	119	40	02:12:24
05/06/2021	66	57	25	24	01:31:59	00:35:52	00:54:30	39.69%	34	33	00:53:50	7		00:05:05
06/06/2021		4		4	00:06:58	00:06:58		100%						

Ripartizione oraria

Export XLS

ORA	TOTALE RISPOSTE	TOTALE NON RISPOSTE	INGRESSO N. RISPOSTE	INGRESSO N. NON RISPOSTE	INGRESSO T. TOTALE	INGRESSO T. ATTESA	INGRESSO T. CONVERSAZIONE	INGRESSO % TEMPO ATTESA	USCITA N. RISPOSTE	USCITA N. NON RISPOSTE	USCITA T. TOTALE	INTERNA N. RISPOSTE	INTERNA N. NON RISPOSTE	INTERNA T. TOTALE
00:00	1	2	1	2	00:01:47	00:01:02	00:00:44	58.49%						
01:00														
02:00		1		1	00:01:04	00:01:04		100%						
03:00														
04:00														
05:00														

Ripartizione settimanale

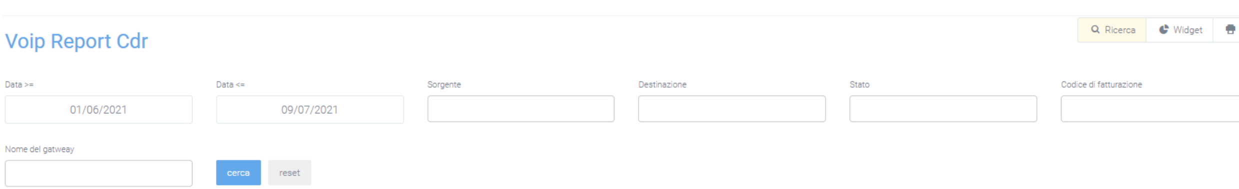
Export XLS

GIORNO	TOTALE RISPOSTE	TOTALE NON RISPOSTE	INGRESSO N. RISPOSTE	INGRESSO N. NON RISPOSTE	INGRESSO T. TOTALE	INGRESSO T. ATTESA	INGRESSO T. CONVERSAZIONE	INGRESSO % TEMPO ATTESA	USCITA N. RISPOSTE	USCITA N. NON RISPOSTE	USCITA T. TOTALE	INTERNA N. RISPOSTE	INTERNA N NON RISPOSTE	INTERNA T. TOTALE
dom	1	16	1	16	00:13:27	00:11:53	00:00:59	92.36%						
lun	2378	1257	863	488	49:12:59	12:22:01	36:12:28	25.46%	926	453	26:37:57	589	316	14:42:23
mar	2659	1318	876	473	51:23:29	11:59:34	38:51:43	23.58%	1117	535	36:08:12	666	310	13:30:48
mer	1902	1042	615	411	37:05:51	09:35:04	27:04:45	26.14%	807	392	27:10:13	480	239	09:39:32
gio	2245	1187	700	430	42:35:25	09:06:43	32:48:03	21.74%	975	492	32:43:31	570	265	12:06:15
ven	1976	1039	623	320	34:57:46	07:25:42	26:57:18	21.6%	901	498	29:44:35	452	221	10:15:31

Filters

This facade can be changed by activating filters by clicking on the "Search" button in the upper right corner, with which you can select:

- A time interval in which call progress is displayed, setting a start date by "Date >=" and an end date by "Date <=."
- Data source
- Call destination
- Call status
- Billing code
- Gateway name



Once the filter fields are filled in as desired, press on the "search" button in light blue to save and display the screen with the activated filters, otherwise press on "reset."

Widget

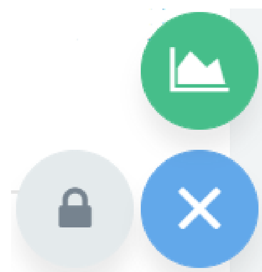
For explanation on creating, organizing and managing Widgets, visit:

Widget

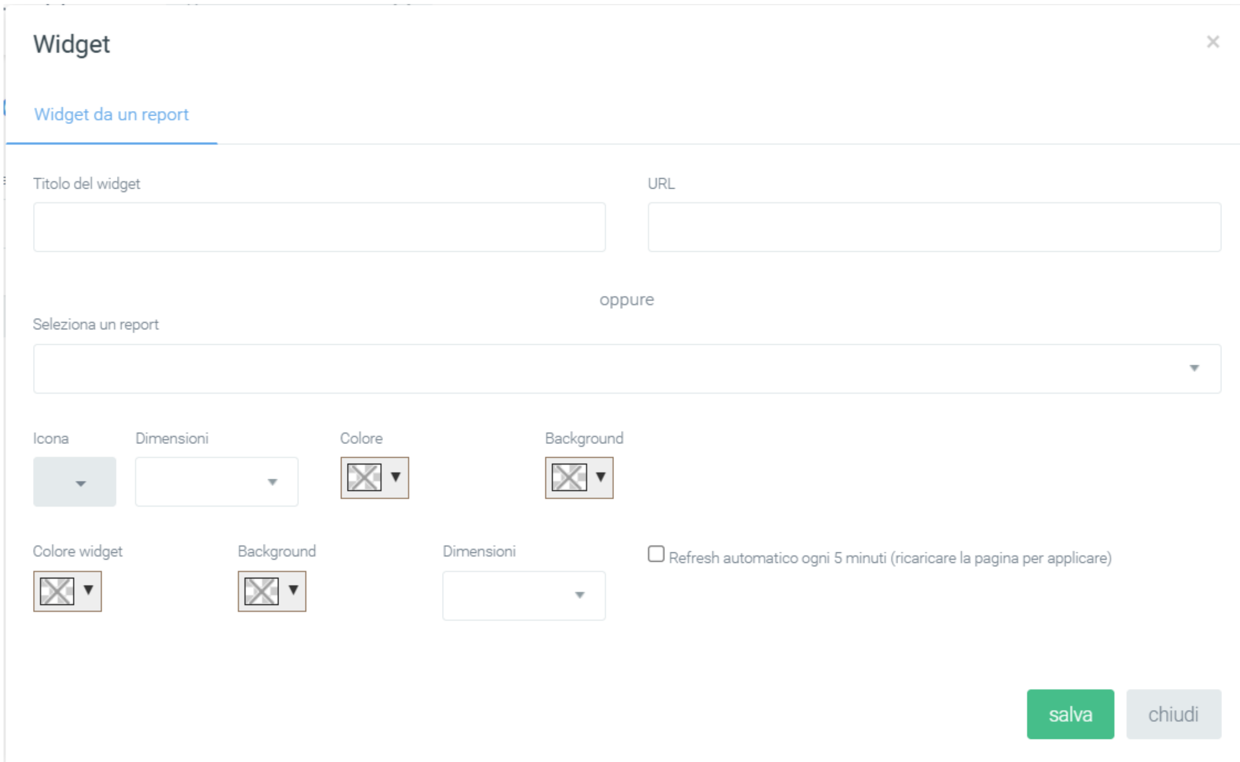
Thanks to the "widget" button, it is possible to add panels containing additional information. After clicking the "widget" button in the upper right corner, a light blue "+" and a padlock will appear.

To move an existing widget, simply unlock the lock by clicking on it and then move it as desired within the screen.

To add a widget, you will first need to click on the "+" and then on the chart icon, and then select the widget to be added.



After that, a screen will open where the data for creating the widget must be entered:



After filling in the necessary information, click save to create the widget, otherwise click close or the "x" in the upper right corner to not create it.

You can always print the entire graph page view via the printer icon button located at the top of the page on the right.

Report queue (operator, date, time, trend)

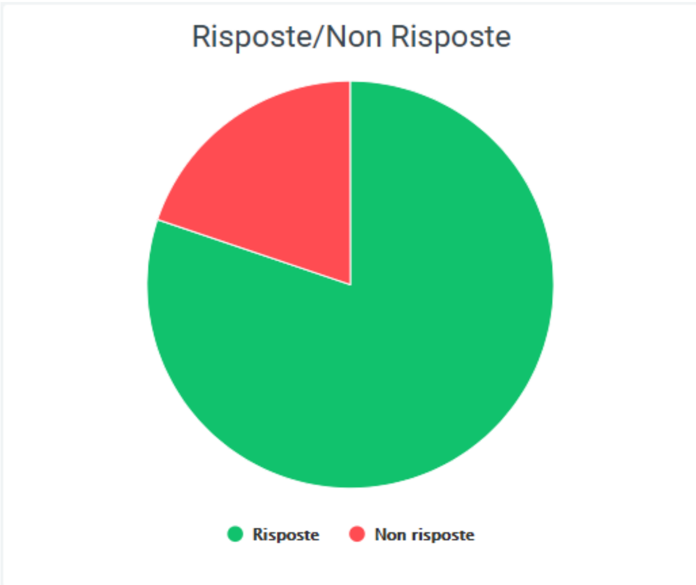
In the "Report Queue" section, under the "Voip report CDR" form, through the table and graphs the call trends are represented.

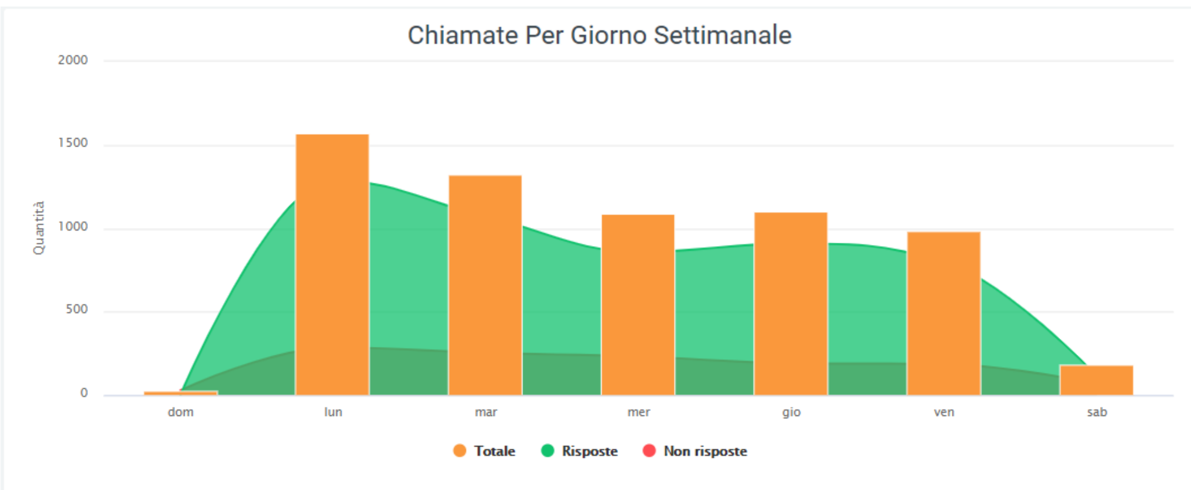
Charts

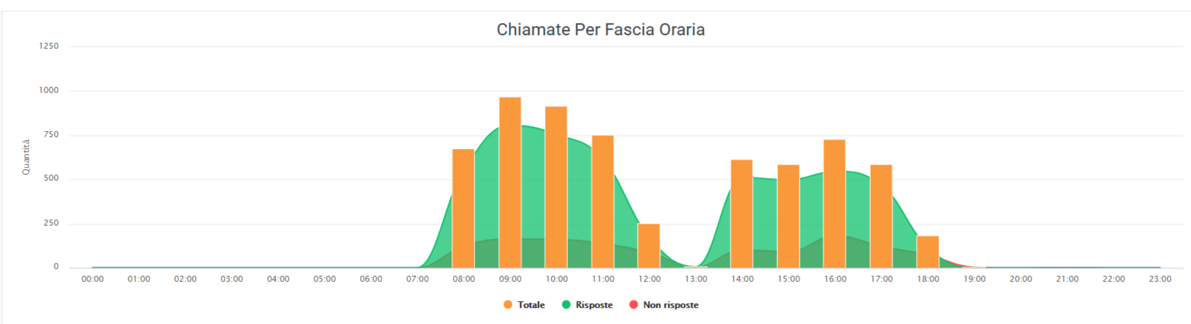
Three types of graphs can be displayed in this report: the first pie chart contains information on the number of answered and unanswered calls. If no queue is entered in the filters, the graph will refer to all queues.

In addition to the pie chart, there are two graphs that contain data on answered, unanswered, and total calls divided by days of the week and by time slot.

This allows for immediate analysis on call trends.


Risposte/Non Risposte
Risposte
Non risposte


Chiamate Per Giorno Settimanale
2000
1500
1000
500
0
Quantità
dom
lun
mar
mer
gio
ven
sab
Totale
Risposte
Non risposte


Chiamate Per Fascia Oraria
1250
1000
750
500
250
0
Quantità
00:00
01:00
02:00
03:00
04:00
05:00
06:00
07:00
08:00
09:00
10:00
11:00
12:00
13:00
14:00
15:00
16:00
17:00
18:00
19:00
20:00
21:00
22:00
23:00
Totale
Risposte
Non risposte

Tables

Below all the graphs shown above, there is a table called "Number of Calls Handled" where all the data regarding calls broken down by queue are collected. Before the tabulated data, under the title, there are a series of values showing the total amount of calls, total call time, total talk time, average talk time, average queuing time, average hold time, average time for operator response, and average talk time. It is also possible to export the table to excel format using the button at the top right of the table, "Export XLS."

Numero chiamate gestite

Export XLS

quantità: 6.282	tempo totale: 11gg 22:42:08	tempo di conversazione: 10gg 03:01:48	media tempo totale: 00:02:44	media tempo accodamento: 00:00:00	media tempo attesa: 00:00:25	media tempo risposta operatore: 00:00:12	media tempo conversazione: 00:02:19

In the table, each row belongs to a queue so that the same information can be compared on different queues. Next to the name of each, there is a pie chart representing answered and unanswered calls from the relevant queue. After the pie chart, several more detailed pieces of information follow: the total number of calls received by the queue, the number of answered and unanswered calls, the trend of those answered with their times, NCC calls, calls that have been transferred, calls that have been cancelled, and calls that have not been answered, all accompanied by qualitative information regarding the times taken.

Finally, the lower part of the table shows the totals for each column.

Numero chiamate gestite

Export XLS

quantità: 6.282	tempo totale: 11gg 22:42:08	tempo di conversazione: 10gg 03:01:48	media tempo totale: 00:02:44	media tempo accodamento: 00:00:00	media tempo attesa: 00:00:25	media tempo risposta operatore: 00:00:12	media tempo conversazione: 00:02:19

	TOTALE	RISPOSTE					NON RISPOSTE	NCC							TRASFER			
	quantità	quantità	media durata chiamata	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	quantità	durata	media durata chiamata	tempo conversazione	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	durata	media durata chiamata	tempo conversazion
Giime	350	248	00:02:37	00:02:25	00:00:00	00:00:11	102	248	10:50:20	00:02:37	10:01:04	00:02:25	00:00:00	00:00:11				
IVR-Giorno-1Caldaie	3.477	3.038	00:03:12	00:02:55	00:00:00	00:00:17	439	3.033	6gg 18:19:18	00:03:12	6gg 03:50:47	00:02:55	00:00:00	00:00:17	5	00:19:32	00:03:54	00:17:3
IVR-Giorno-2Condiz	1.052	952	00:02:51	00:02:44	00:00:00	00:00:06	100	947	1gg 21:11:14	00:02:51	1gg 19:27:20	00:02:45	00:00:00	00:00:06	5	00:05:06	00:01:01	00:04:2
IVR-Giorno-3Contratti	291	196	00:03:38	00:03:31	00:00:00	00:00:06	95	195	11:52:25	00:03:39	11:31:11	00:03:32	00:00:00	00:00:06	1	00:00:54	00:00:54	00:00:5
IVR-Giorno-4SanSec	137	61	00:02:56	00:02:50	00:00:00	00:00:05	76	61	02:59:02	00:02:56	02:53:34	00:02:50	00:00:00	00:00:05				
IVR-Giorno-5Amministr	354	211	00:02:58	00:02:51	00:00:00	00:00:06	143	208	10:23:25	00:02:59	10:00:39	00:02:53	00:00:00	00:00:06	3	00:02:47	00:00:55	00:02:2
Operatore	619	332	00:03:51	00:03:02	00:00:00	00:00:48	287	331	21:17:18	00:03:51	16:50:22	00:03:03	00:00:00	00:00:48	1	00:03:37	00:03:37	00:01:3
SanSec	2						2											
Totali	6.282	5.038	00:03:09	00:02:53	00:00:00	00:00:15	1.244	5.023	11gg 00:53:02	00:03:09	10gg 02:34:57	00:02:53	00:00:00	00:00:15	15	00:31:56	00:02:07	00:26:5

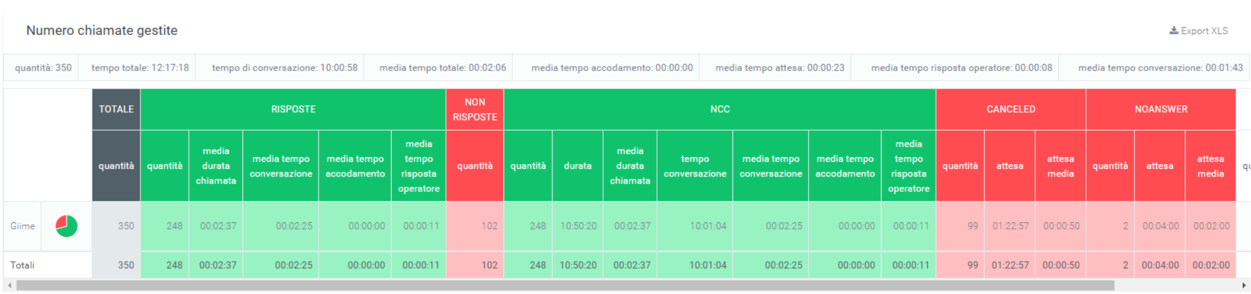
Numero chiamate gestite

Export XLS

quantità: 6.282 | tempo totale: 11gg 22:42:08 | tempo di conversazione: 10gg 03:01:48 | media tempo totale: 00:02:44 | media tempo accodamento: 00:00:00 | media tempo attesa: 00:00:25 | media tempo risposta operatore: 00:00:12 | media tempo conversazione: 00:02:19

NCC						TRASFERIMENTO							CANCELED			NOANSWER			QUEUE_JOINEMPTY		
durata	media durata chiamata	tempo conversazione	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	durata	media durata chiamata	tempo conversazione	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	attesa	attesa media	quantità	attesa	attesa media	quantità	attesa	attesa media
10:50:20	00:02:37	10:01:04	00:02:25	00:00:00	00:00:11								99	01:22:57	00:00:50	2	00:04:00	00:02:00			
g 18:19:18	00:03:12	6gg 03:50:47	00:02:55	00:00:00	00:00:17	5	00:19:32	00:03:54	00:17:39	00:03:31	00:00:00	00:00:22	225	04:00:44	00:01:04	14	01:11:43	00:05:07	200		
g 21:11:14	00:02:51	1gg 19:27:20	00:02:45	00:00:00	00:00:06	5	00:05:06	00:01:01	00:04:24	00:00:52	00:00:00	00:00:08	13	00:02:17	00:00:10	87	00:56:51	00:00:39			
11:52:25	00:03:39	11:31:11	00:03:32	00:00:00	00:00:06	1	00:00:54	00:00:54	00:00:51	00:00:51		00:00:03	12	00:04:46	00:00:23	83	01:08:42	00:00:49			
02:59:02	00:02:56	02:53:34	00:02:50	00:00:00	00:00:05								6	00:00:49	00:00:08	70	00:23:24	00:00:20			
10:23:25	00:02:59	10:00:39	00:02:53	00:00:00	00:00:06	3	00:02:47	00:00:55	00:02:29	00:00:49	00:00:00	00:00:06	14	00:05:54	00:00:25	129	01:46:44	00:00:49			
21:17:18	00:03:51	16:50:22	00:03:03	00:00:00	00:00:48	1	00:03:37	00:03:37	00:01:34	00:01:34		00:02:03	239	06:00:26	00:01:30	47	04:05:18	00:05:13			
													1	00:00:01	00:00:01	1	00:00:20	00:00:20			
g 00:53:02	00:03:09	10gg 02:34:57	00:02:53	00:00:00	00:00:15	15	00:31:56	00:02:07	00:26:57	00:01:47	00:00:00	00:00:19	609	11:37:54	00:01:08	433	09:37:02	00:01:19	200		

In case a filter is entered on the queues, only one queue will be displayed in the table.



Numero chiamate gestite

Export XLS

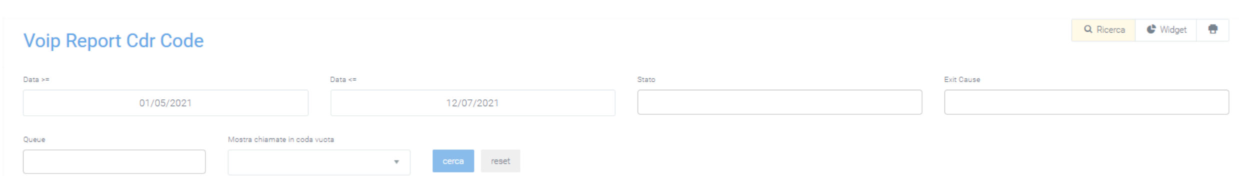
quantità: 350 | tempo totale: 12:17:18 | tempo di conversazione: 10:00:58 | media tempo totale: 00:02:06 | media tempo accodamento: 00:00:00 | media tempo attesa: 00:00:23 | media tempo risposta operatore: 00:00:08 | media tempo conversazione: 00:01:43

	TOTALE	RISPOSTE					NON RISPOSTE	NCC							CANCELED			NOANSWER		
	quantità	quantità	media durata chiamata	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	quantità	durata	media durata chiamata	tempo conversazione	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	attesa	attesa media	quantità	attesa	attesa media
Giime	350	248	00:02:37	00:02:25	00:00:00	00:00:11	102	248	10:50:20	00:02:37	10:01:04	00:02:25	00:00:00	00:00:11	99	01:22:57	00:00:50	2	00:04:00	00:02:00
Totali	350	248	00:02:37	00:02:25	00:00:00	00:00:11	102	248	10:50:20	00:02:37	10:01:04	00:02:25	00:00:00	00:00:11	99	01:22:57	00:00:50	2	00:04:00	00:02:00

Filters

There is also a mask that groups filters that can be used to improve the analysis of the desired data. To activate the filters, click on the "search" button in the upper right corner. A mask will open with:

- Start and end date
- The status of the calls
- The exit causes
- The queues
- Ability to show calls in empty queue



Once you have selected the desired filters, click search to perform the search. You can also add a widget with the appropriate button and print the information on the screen.

Queue operators report

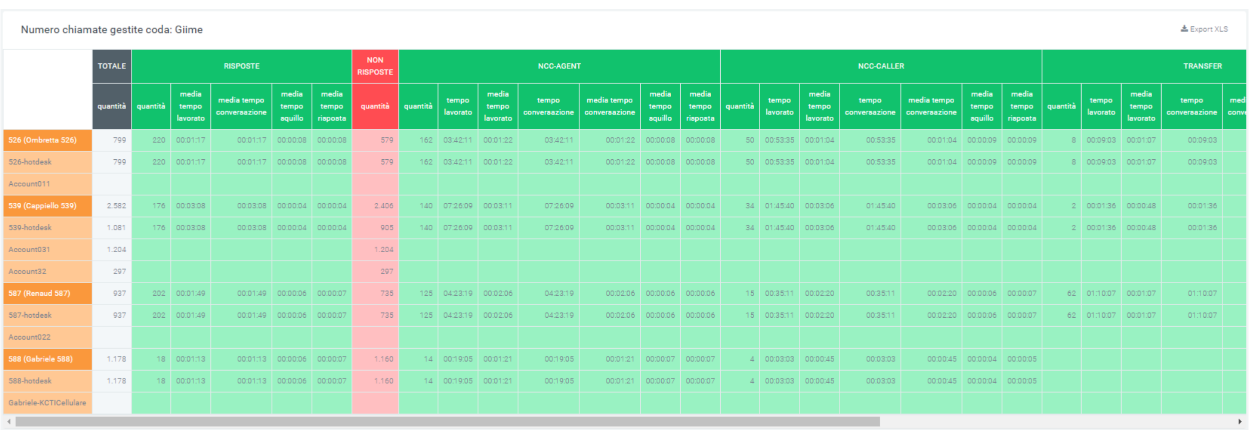
In this section, within the “Voip report CDR” module, you can view, in detail, call trends for individual operators, broken down into their respective queues. The information that is displayed concerns how many calls were answered and how they went, but also how many were not answered and the reasons why. This makes it easier to perform an analysis on each operator accurately. The following are displayed: total calls (amount of calls), answered, unanswered, NCC-Caller (call closed normally), transfer (caller transferred), answered elsewhere (call was answered by another operator), cancelled (caller put the call down after a waiting period), forward, refused (call was rejected by the operator, timeouf (call after a certain waiting time, is dropped)

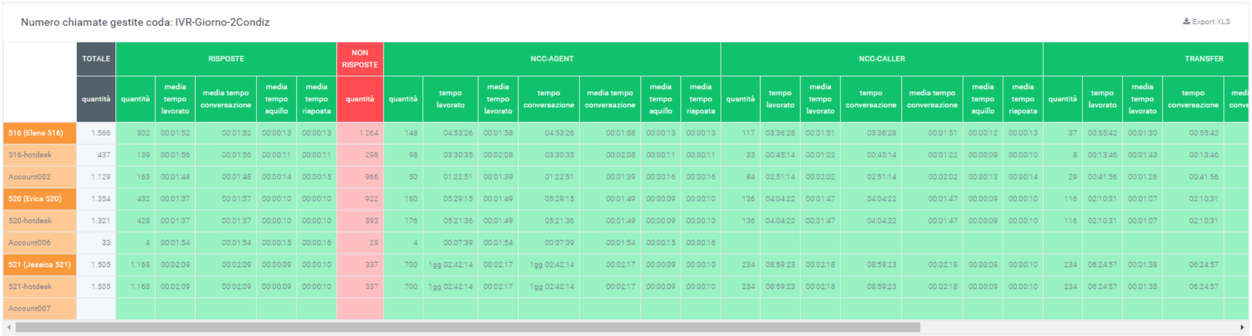
Tables

Total	Responses	Non-Responses	NCC-Caller	Transfer	Answered Elsewhere	Cancelled	Forward	Refused	Timeout
	Quantity / Average time worked / Average talk time / Average ring time / Average response time	Quantity / Average Time Worked / Average Time Worked / Average Talk Time / Average Ring Time / Average Response Time	Quantity / Time worked / Average time worked / Average time talk / Average time ring / Average time answer	Quantity / Time worked / Average time worked / Average time talk / Average time ring / Average time answer	Quantity / Average ringing time	Quantity / Average ring time	Quantity / Average ringing time	Quantity / Average ringing time	Quantity / Average ringing time

Each row corresponds to an operator with its corresponding accounts. If a filter was entered on a single operator, only the operator concerned would be displayed.

As for the other queues, they would be displayed in the same table, but with different data and operators.





	TOTALE	RISPOSTE					NON RISPOSTE	NCC-AGENT						
	quantità	quantità	media tempo lavorato	media tempo conversazione	media tempo squillo	media tempo risposta	quantità	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo squillo	media tempo risposta
516 (Elena 516)	1.566	302	00:01:52	00:01:52	00:00:13	00:00:13	1.264	148	04:53:26	00:01:58	04:53:26	00:01:58	00:00:13	00:00:13
516-hotdesk	437	139	00:01:56	00:01:56	00:00:11	00:00:11	298	98	03:30:35	00:02:08	03:30:35	00:02:08	00:00:11	00:00:11
Account002	1.129	163	00:01:48	00:01:48	00:00:14	00:00:15	966	50	01:22:51	00:01:39	01:22:51	00:01:39	00:00:16	00:00:16
520 (Erica 520)	1.354	432	00:01:37	00:01:37	00:00:10	00:00:10	922	180	05:29:15	00:01:49	05:29:15	00:01:49	00:00:09	00:00:10
520-hotdesk	1.321	428	00:01:37	00:01:37	00:00:10	00:00:10	893	176	05:21:36	00:01:49	05:21:36	00:01:49	00:00:09	00:00:10
Account006	33	4	00:01:54	00:01:54	00:00:15	00:00:16	29	4	00:07:39	00:01:54	00:07:39	00:01:54	00:00:15	00:00:16
521 (Jessica 521)	1.505	1.168	00:02:09	00:02:09	00:00:09	00:00:10	337	700	1gg 02:42:14	00:02:17	1gg 02:42:14	00:02:17	00:00:09	00:00:10
521-hotdesk	1.505	1.168	00:02:09	00:02:09	00:00:09	00:00:10	337	700	1gg 02:42:14	00:02:17	1gg 02:42:14	00:02:17	00:00:09	00:00:10
Account007														

	NCC-CALLER							TRANSFER			
	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo squillo	media tempo risposta	quantità	tempo lavorato	media tempo lavorato	tempo conversazione
516 (Elena 516)	117	03:36:28	00:01:51	03:36:28	00:01:51	00:00:12	00:00:13	37	00:55:42	00:01:30	00:55:42
516-hotdesk	33	00:45:14	00:01:22	00:45:14	00:01:22	00:00:09	00:00:10	8	00:13:46	00:01:43	00:13:46
Account002	84	02:51:14	00:02:02	02:51:14	00:02:02	00:00:13	00:00:14	29	00:41:56	00:01:26	00:41:56
520 (Erica 520)	136	04:04:22	00:01:47	04:04:22	00:01:47	00:00:09	00:00:10	116	02:10:31	00:01:07	02:10:31
520-hotdesk	136	04:04:22	00:01:47	04:04:22	00:01:47	00:00:09	00:00:10	116	02:10:31	00:01:07	02:10:31
Account006											
521 (Jessica 521)	234	08:59:23	00:02:18	08:59:23	00:02:18	00:00:09	00:00:10	234	06:24:57	00:01:38	06:24:57
521-hotdesk	234	08:59:23	00:02:18	08:59:23	00:02:18	00:00:09	00:00:10	234	06:24:57	00:01:38	06:24:57
Account007											

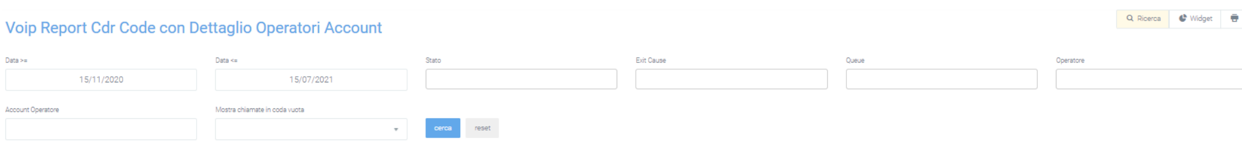
There is an "Export XLS" button in the upper right corner of each table to export the tables to excel (XLS) format.

Filters

To make data analysis more effective, it is possible to use filters that will allow only the necessary data to be derived from a search. To activate them, simply click on the "search" button at the top right and a mask will open with:

- Start date
- End date
- Status
- Exit causes (reasons for the end of calls)
- Queue
- Operator
- Operator Account
- Display calls in empty queue

By clicking on "search", filters are enabled and a search with them is carried out. In case you want to delete them, just click on the "reset" button.

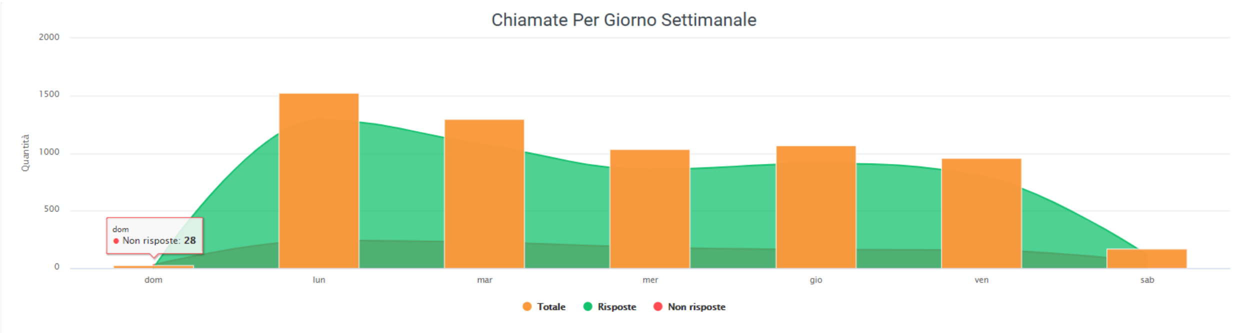


Queue data report

This section encapsulates the information found in the "Report Queue" section, but the breakdown is by day, based on the time frame entered through "start date" and "end date" filters.

Charts

The first graph represents total calls, answered and unanswered, along the time frame of one week. This allows a first general analysis to be performed quickly and easily. You can also select only some of the three pieces of information in the graph by clicking on the name of the one you do not want to display within the legend.



Tables

At the bottom of the section, below the graph, are tables for each queue broken down by date. Each row corresponds to a day within the period entered in the filters and is divided into columns, which show all the information needed to perform an accurate analysis of call trends for each specific queue.

The information contained within the table includes:

Total quantity	Answered calls	Non-answered	NCC	Cancelled	No answered
	Quantity / Average call duration / Average talk time / Average operator response time.		Quantity / Duration / Average call duration / Average talk time / Average queuing time / Average operator response time	Quantity / Time worked / Total wait / Average wait	Quantity / Total wait / Average wait

Depending on the queue, there may also be an additional transfer part with:

- quantity
- duration
- average call duration
- conversation time
- average conversation time
- average queuing time
- average operator response time

Numero chiamate gestite coda: Giime

Export XLS

quantità: 91	tempo totale: 02:48:47	tempo di conversazione: 02:01:18	media tempo totale: 00:01:51	media tempo accodamento: 00:00:00	media tempo attesa: 00:00:31	media tempo risposta operatore: 00:00:08	media tempo conversazione: 00:01:19

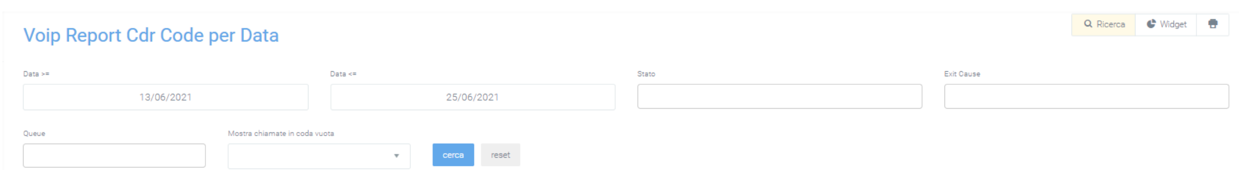
	TOTALE	RISPOSTE					NON RISPOSTE	NCC							CANCELED			NOANSWER			
	quantità	quantità	media durata chiamata	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	quantità	durata	media durata chiamata	tempo conversazione	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	attesa	attesa media	quantità	attesa	attesa media	quantit
13/06/2021																					
14/06/2021	8	8	00:01:50	00:01:40	00:00:00	00:00:10		8	00:14:46	00:01:50	00:13:21	00:01:40	00:00:00	00:00:10							
15/06/2021	17	12	00:02:19	00:02:08	00:00:00	00:00:10	5	12	00:27:49	00:02:19	00:25:45	00:02:08	00:00:00	00:00:10	5	00:02:07	00:00:25				
16/06/2021	10	5	00:02:45	00:02:32	00:00:00	00:00:13	5	5	00:13:49	00:02:45	00:12:42	00:02:32	00:00:00	00:00:13	5	00:04:00	00:00:48				
17/06/2021	5	3	00:01:55	00:01:36		00:00:19	2	3	00:05:47	00:01:55	00:04:48	00:01:36		00:00:19	2	00:01:12	00:00:36				
18/06/2021	11	10	00:01:58	00:01:45	00:00:00	00:00:12	1	10	00:19:44	00:01:58	00:17:36	00:01:45	00:00:00	00:00:12				1	00:02:00	00:02:00	
19/06/2021	1	1	00:04:36	00:04:29		00:00:07		1	00:04:36	00:04:36	00:04:29	00:04:29		00:00:07							
20/06/2021																					
21/06/2021	10	8	00:02:18	00:02:09	00:00:00	00:00:09	2	8	00:18:28	00:02:18	00:17:15	00:02:09	00:00:00	00:00:09	2	00:01:05	00:00:32				
22/06/2021	9	9	00:02:09	00:01:52	00:00:00	00:00:16		9	00:19:23	00:02:09	00:16:54	00:01:52	00:00:00	00:00:16							
23/06/2021	3	2	00:01:14	00:01:10		00:00:04	1	2	00:02:29	00:01:14	00:02:20	00:01:10		00:00:04	1	00:00:37	00:00:37				
24/06/2021	9	2	00:03:17	00:03:07		00:00:10	7	2	00:06:35	00:03:17	00:06:14	00:03:07		00:00:10	7	00:11:01	00:01:34				
25/06/2021	8						8								7	00:13:18	00:01:54				
Totali	91	60	00:02:13	00:02:01	00:00:00	00:00:12	31	60	02:13:26	00:02:13	02:01:24	00:02:01	00:00:00	00:00:12	29	00:33:20	00:01:08	1	00:02:00	00:02:00	

Each table can be exported to an excel sheet via special "Export XLS" button in the upper right corner.

Filters

By clicking on the "search" button at the top right of the section, filters can be enabled that include:

- Start date
- End date
- Status
- Exit cause
- Queue
- Calls in empty queue



Queue Time Report

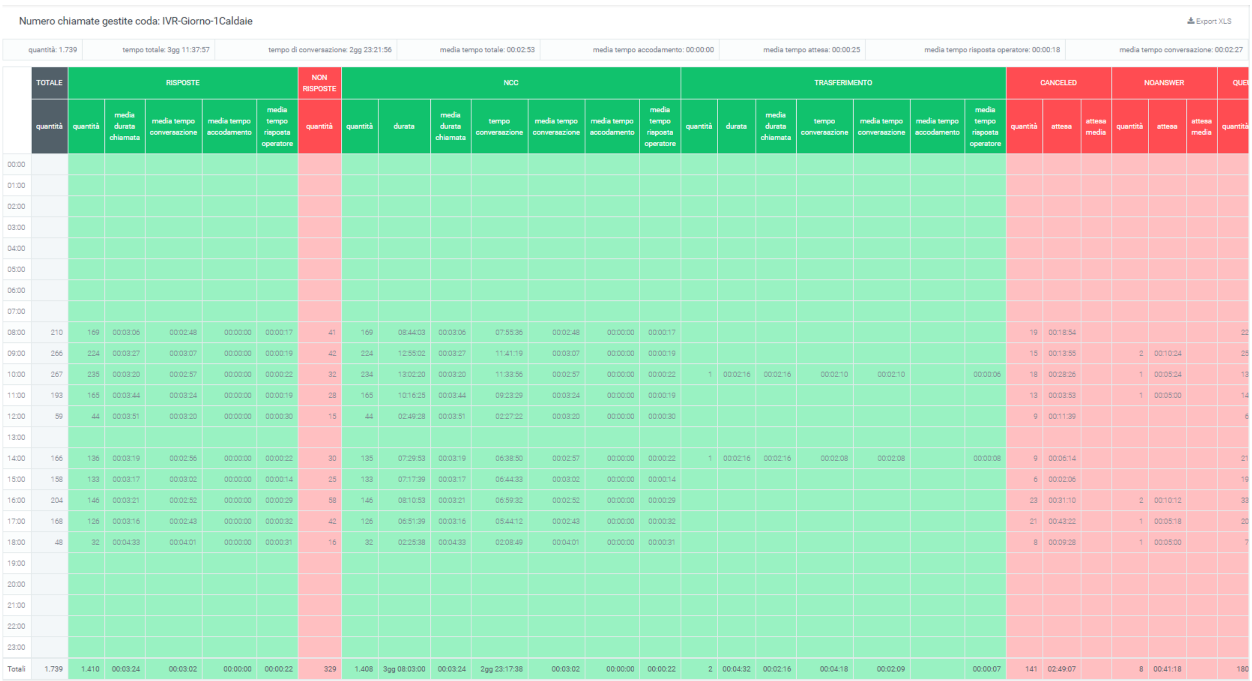
Tables

This section contains only the tables divided by time slots, which contain all the information necessary for the analysis of calls in individual queues.

The information contained in the tables includes:

Total quantity	Answered calls	Non-answered	NCC	Cancelled	No answered	Transfer
	Quantity / Average call duration / Average talk time / Average queuing time / Average operator response time.		Quantity / Duration / Average call duration / Talk time / Average talk time / Average queuing time / Average operator answer time	Quantity / Total wait / Average wait	Quantity / Total wait / Average wait	Quantity / Duration / Average call duration / Talk time / Average talk time / Average queuing time / Average operator response time

Each table is exportable as an excel child via special "Export XLS" button in its upper right corner.



Numero chiamate gestite coda: IVR-Giorno-1Caldaie

Export XLS

quantità: 1.739 | tempo totale: 3gg 11:37:57 | tempo di conversazione: 2gg 23:21:56 | media tempo totale: 00:02:53 | media tempo accodamento: 00:00:00 | media tempo attesa: 00:00:25 | media tempo risposta operatore: 00:00:18 | media tempo conversazione: 00:02:27

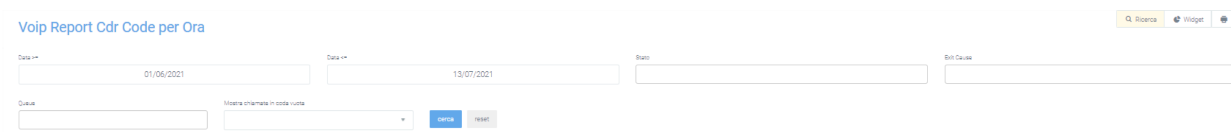
	TOTALE	RISPOSTE					NON RISPOSTE	NCC						
	quantità	quantità	media durata chiamata	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	quantità	durata	media durata chiamata	tempo conversazione	media tempo conversazione	media tempo accodamento	media tempo risposta operatore
00:00														
01:00														
02:00														
03:00														
04:00														
05:00														
06:00														
07:00														
08:00	210	169	00:03:06	00:02:48	00:00:00	00:00:17	41	169	08:44:03	00:03:06	07:55:36	00:02:48	00:00:00	00:00:17
09:00	266	224	00:03:27	00:03:07	00:00:00	00:00:19	42	224	12:55:02	00:03:27	11:41:19	00:03:07	00:00:00	00:00:19
10:00	267	235	00:03:20	00:02:57	00:00:00	00:00:22	32	234	13:02:20	00:03:20	11:33:56	00:02:57	00:00:00	00:00:22
11:00	193	165	00:03:44	00:03:24	00:00:00	00:00:19	28	165	10:16:25	00:03:44	09:23:29	00:03:24	00:00:00	00:00:19
12:00	59	44	00:03:51	00:03:20	00:00:00	00:00:30	15	44	02:49:28	00:03:51	02:27:22	00:03:20	00:00:00	00:00:30
13:00														
14:00	166	136	00:03:19	00:02:56	00:00:00	00:00:22	30	135	07:29:53	00:03:19	06:38:50	00:02:57	00:00:00	00:00:22
15:00	158	133	00:03:17	00:03:02	00:00:00	00:00:14	25	133	07:17:39	00:03:17	06:44:33	00:03:02	00:00:00	00:00:14
16:00	204	146	00:03:21	00:02:52	00:00:00	00:00:29	58	146	08:10:53	00:03:21	06:59:32	00:02:52	00:00:00	00:00:29
17:00	168	126	00:03:16	00:02:43	00:00:00	00:00:32	42	126	06:51:39	00:03:16	05:44:12	00:02:43	00:00:00	00:00:32
18:00	48	32	00:04:33	00:04:01	00:00:00	00:00:31	16	32	02:25:38	00:04:33	02:08:49	00:04:01	00:00:00	00:00:31
19:00														
20:00														
21:00														
22:00														
23:00														
Totali	1.739	1.410	00:03:24	00:03:02	00:00:00	00:00:22	329	1.408	3gg 08:03:00	00:03:24	2gg 23:17:38	00:03:02	00:00:00	00:00:22

	TRASFERIMENTO							CANCELED			NOANSWER			QUE
	quantità	durata	media durata chiamata	tempo conversazione	media tempo conversazione	media tempo accodamento	media tempo risposta operatore	quantità	attesa	attesa media	quantità	attesa	attesa media	quantità
00:00														
01:00														
02:00														
03:00														
04:00														
05:00														
06:00														
07:00														
08:00								19	00:18:54					22
09:00								15	00:13:55		2	00:10:24		25
10:00	1	00:02:16	00:02:16	00:02:10	00:02:10		00:00:06	18	00:28:26		1	00:05:24		13
11:00								13	00:03:53		1	00:05:00		14
12:00								9	00:11:39					6
13:00														
14:00	1	00:02:16	00:02:16	00:02:08	00:02:08		00:00:08	9	00:06:14					21
15:00								6	00:02:06					19
16:00								23	00:31:10		2	00:10:12		33
17:00								21	00:43:22		1	00:05:18		20
18:00								8	00:09:28		1	00:05:00		7
19:00														
20:00														
21:00														
22:00														
23:00														
Totali	2	00:04:32	00:02:16	00:04:18	00:02:09		00:00:07	141	02:49:07		8	00:41:18		180

Filters

Clicking on the "search" button at the top right of the section enables filters that include:

- Start date
- End date
- Status
- Exit cause
- Queue
- Calls in empty queue



After entering preferences, click on "search" to perform a filtered search. If you enter the filter for specific queues, only the tables of the selected queues will be shown.

Billing Codes Report (date, operator)

It is possible to observe the trend of calls recorded in the CDR broken down by billing codes (used in campaigns), which is why only outgoing calls will be analyzed.

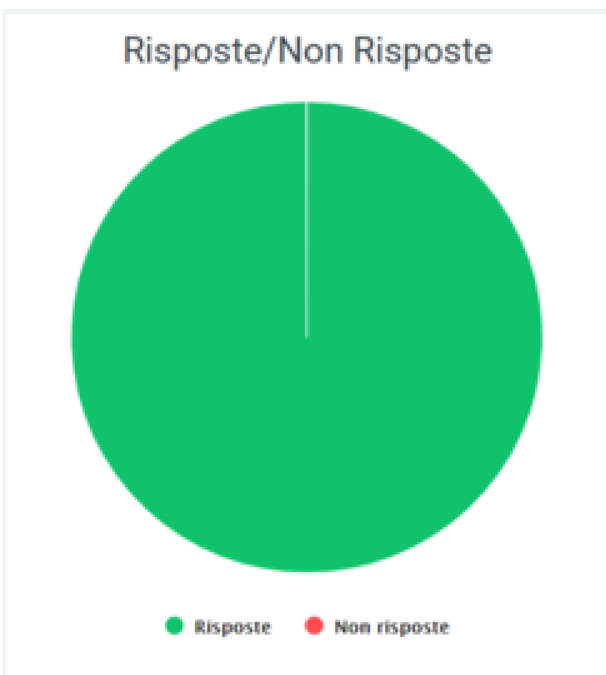
Charts

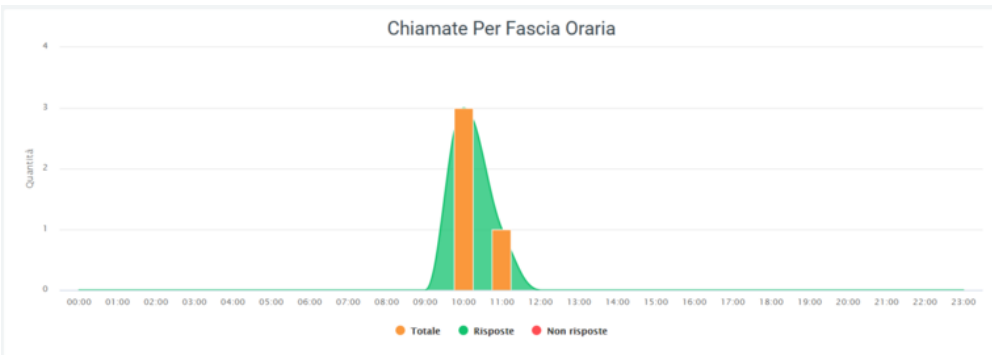
To get a visual representation of the data just click on the "graphs" button.

Three types of graphs can be displayed in this report:

- A pie chart (on the right) showing the number of calls answered and not by the billing codes entered through the filters.
- A graph showing call trends, again for selected billing codes, by day of the week (Sunday through Saturday). In orange is shown the total number of calls, and in green and red, the curves for answered and unanswered calls, respectively.

A graph showing average call trends by billing code broken down by time slot (00:00 to 23:00)





In all three graphs, it is possible to remove some data from the view via the legend by clicking on the name or colored dot. It is also possible to see in detail the number of affected calls by scrolling over the various graphs with the mouse, in the affected areas.

Tables

Below all the graphs shown above, there is a table called "Number of Calls Handled" where all the data regarding calls broken down by billing codes are collected. Before the tabulated data, under the title, there are a series of values showing the total amount of calls, total call time, total billing time, average talk time, average total time, average billing time, average hold time, and average talk time. It is also possible to export the table to excel format using the button at the top right of the table, "Export XLS."

Numero chiamate gestite

Export XLS

quantità:	tempo totale:	tempo di fatturazione:	tempo di conversazione:	media tempo totale:	media tempo fatturazione:	media tempo attesa:	media tempo conversazione:
4	00:01:24	00:00:17	00:00:21	00:00:21	00:00:04	00:00:16	00:00:05

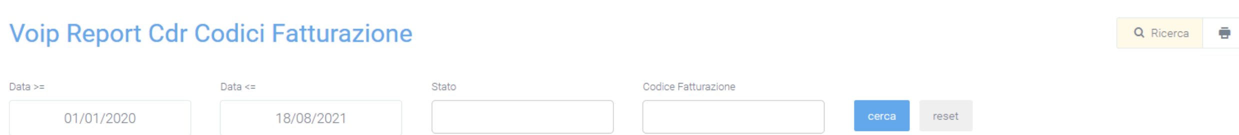
	TOTALE			OK							
	quantità	durata	tempo fatturazione	quantità	durata	media durata chiamata	tempo di fatturazione	tempo conversazione	media tempo fatturazione	media tempo conversazione	media tempo attesa
9999	1	00:00:19	00:00:05	1	00:00:19	00:00:19	00:00:05	00:00:06	00:00:05	00:00:06	00:00:14
12345	1	00:00:16	00:00:03	1	00:00:16	00:00:16	00:00:03	00:00:04	00:00:03	00:00:04	00:00:12
54321	1	00:00:21	00:00:05	1	00:00:21	00:00:21	00:00:05	00:00:06	00:00:05	00:00:06	00:00:15
141414	1	00:00:28	00:00:04	1	00:00:28	00:00:28	00:00:04	00:00:05	00:00:04	00:00:05	00:00:23
Totali	4	00:01:24	00:00:17	4	00:01:24	00:00:21	00:00:17	00:00:21	00:00:04	00:00:05	00:00:16

In the table, each billing code belongs to a row that allows the same information to be compared on different code. Next to the name of each, there is a pie chart representing answered and unanswered calls by the code concerned. After the pie chart, several more detailed information follows, for total, answered (OK) and unanswered calls. At the bottom of the table, however, are the totals for each column.

Filters

Filters are used to select a specific time frame in which to analyze the performance of billing code by calls, or to select some specific ones. To enable them, click on the "search" button with which a mask will open allowing the following data to be entered:

- a start and end date of the time interval you want to observe
- the status of the calls
- one or more specific billing codes



After entering all the desired data, click on "search" to enable the filters, otherwise on "reset" to delete them.

The printer button, to the right of the "search" button, allows you to print the entire page view, including charts and the table.

Billing Codes Date Report

To analyze call trends for each billing code in more detail, you can take advantage of the "Billing Code Report by Date" section, which allows you to observe call trends for each individual day contained within the entered interval through filters ("start date" and "end date," visible via "search" button in the upper right).

Charts

To get a visual representation of the data just click on the "graphs" button. The graph in this section shows an average regarding the distribution of calls over the days of a week (Sunday through Saturday). The green curve represents the trend of answered calls, while the red curve, the unanswered calls. The orange column shows the total calls on that day.

Using the legend, by clicking on the name or colored dot of any of the information shown, you can remove it and reinsert it into the display. By scrolling with the mouse over the graph, more specific values about what is represented will be visible.

Tables

In the lower part of this section, below the graph, there are tables showing the detail regarding calls for each day, from the first to the last of the entered interval. Each table belongs to a billing code, so by doing so it will be possible to observe the days on which calls were made, with all the relevant details, quantitative and qualitative.

Numero chiamate gestite codice di fatturazione: 9999 — Export XLS

quantità: 1	tempo lavorato:	tempo di conversazione: 00:00:06	media tempo lavorato: 00:00:19	media tempo attesa: 00:00:13	media tempo conversazione: 00:00:06

	TOTALE	RISPOSTE				NCC					
	quantità	quantità	media tempo lavorato	media tempo conversazione	media tempo attesa	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo attesa
09/03/2021											
10/03/2021	1	1	00:00:19	00:00:06	00:00:13	1	00:00:19	00:00:19	00:00:06	00:00:06	00:00:13
11/03/2021											

For each day there is a wealth of information regarding answered and unanswered calls, showing information regarding why they were or were not answered. At the beginning of each row is a "TOTAL" column reporting the total number of calls made by the billing code, whether those calls were answered or not.

At the top of each table is a total of some information such as:

- the total amount of calls
- total time worked

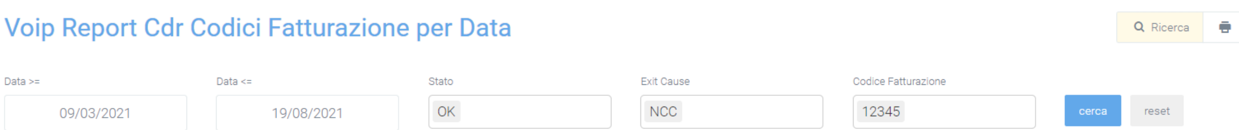
- total talk time
- average time worked
- average waiting time
- average of conversation time

Each table can be exported to excel format through the “export xls” button in the upper right corner.

Filters

In order to display only certain information in the graph and tables, filters must be used. To enable them, click on the “search” button in the upper right corner and enter the desired data:

- start and end date of the time interval in which to view the data
- call status
- exit causes, i.e., the reasons why the call ended
- billing code to be displayed



Once the data entry is finished, click on the “search” button to enable the desired filters, otherwise click “reset” to delete them.

Widget

For explanation on creating, organizing and managing Widgets, visit:

Operator Billing Codes Report As in the other reports regarding billing codes, this section will also show call trends by billing code, but with the difference that here you will be able to observe the detail for each operator.

Tables

There is a table for each code and each row belongs to an operator (with the darker color) and its accounts (with the lighter color). For each of these it is possible to observe the detail of calls made, answered and unanswered, with reasons for them.

Numero chiamate gestite codice fatturazione: 9999 | Export XLS

quantità: 1	tempo lavorato: 00:00:19	media tempo lavorato: 00:00:19	tempo di conversazione: 00:00:06	media tempo conversazione: 00:00:06	media tempo attesa: 00:00:13

	TOTALE	RISPOSTE				NON RISPOSTE	NCC					
	quantità	quantità	media tempo lavorato	media tempo attesa	media tempo conversazione	quantità	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo attesa
201 (Damiano Malizia)	1	1	00:00:19	00:00:13	00:00:06		1	00:00:19	00:00:19	00:00:06	00:00:06	00:00:13
201-APP												
201-DECT												
201-KCTI												
201-PHONE												
Damiano_DECT												
Damiano_Fisso												
Damiano_KCTIP	1	1	00:00:19	00:00:13	00:00:06		1	00:00:19	00:00:19	00:00:06	00:00:06	00:00:13

At the top (outside the grid) a total of some information such as:

- the total amount of calls
- total time worked
- average time worked
- total talk time
- average talk time
- average of waiting time

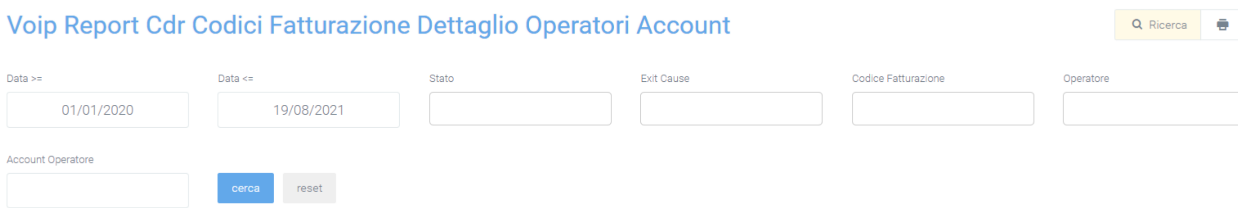
In the table, on the left you will be able to see the operators with their code. Then entering the table, before the whole set of information for answered and unanswered calls, there is a column called "total" which contains the total number made by the operator.

For each table it is possible to export a copy in excel format via special "Export XLS" button in the upper right corner.

Filters

To extract only some essential information, filters can be used by clicking on the "search" button in the upper right-hand corner and with which an appropriately dedicated mask will open. Here you will be able to enter information such as:

- a start and end date of the time interval in which you want to consider the data
- the status of the calls you want to view
- their exit cause, i.e., why the calls ended
- one or more billing codes that you want to observe
- the operator involved
- the account used by the operator



After entering all the data, click on the "save" button to enable the filters, otherwise press "reset" to delete them.

You can also print the page view by simply clicking on the button located at the top right of the page, next to the "search" button.

Report Coincidences

Coincidences refers to the number of calls that pass through the exchange in the same period of time, from answering to closing a call. Depending on the number of simultaneities the company has, it may receive a certain number of calls, i.e., occupy a certain number of lines, at the same time.

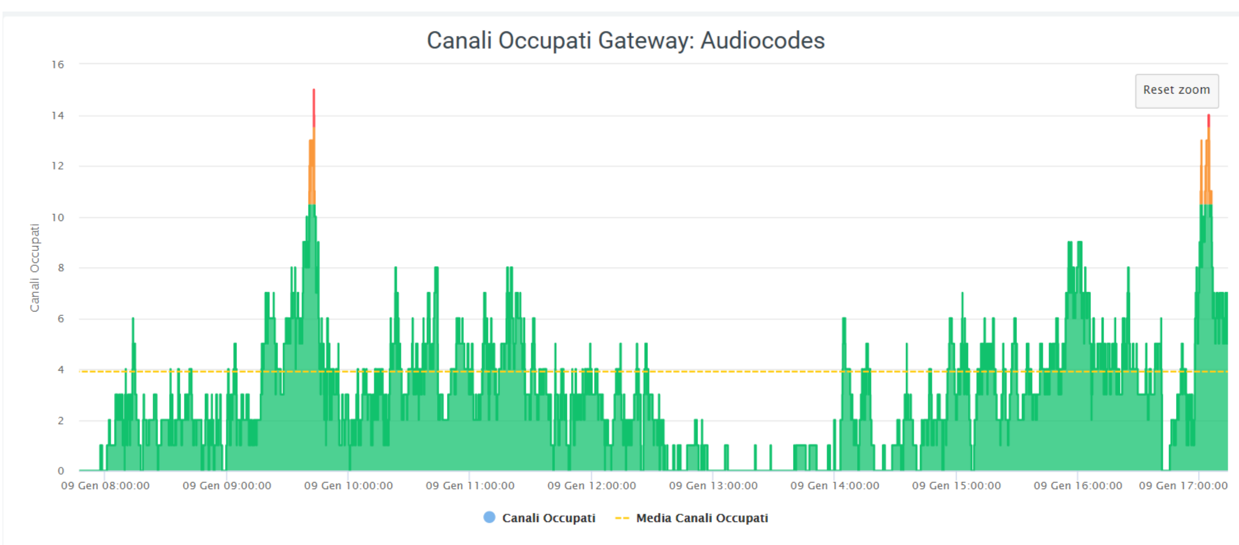
This section makes it possible to observe their progress through a graphical representation, allowing the company to understand whether it needs additional channels or if it already has too many that are not being used and are therefore an unnecessary expense.

Charts

Once you enter the section, you can see represented on the respective graphs, the channels occupied for the various gateways over the specified time interval.

As on other graphs, it is possible to remove from the information display via the legend by simply clicking on the name or symbol to the left. By scrolling with the mouse over the graph, it is also possible to view the specific data at that particular time.

In addition, on this graph it is possible to zoom in by holding down the right mouse button and moving it to select the part to be zoomed in, once selected let go of the right mouse button. To return to the initial view, click on the "reset zoom" button in the upper right corner of the graph.



You can see how the calculation of simultaneities changes, even reaching some peaks (this is based on call traffic). To observe whether the number of channels in the various gateways are enough, just look at the graph:

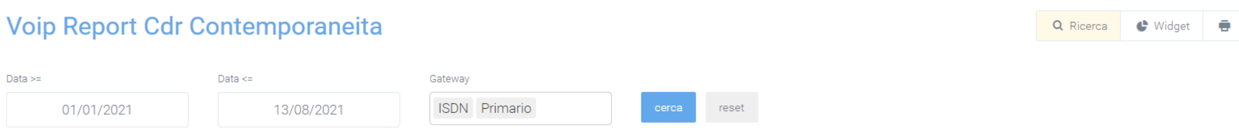
- the color remains green if the number of simultaneities remains below 75% of the number of channels
- the color turns orange when you are approaching total channel occupancy, i.e., 75 to 90 percent
- the color turns red when you are capped, i.e., when all (or almost all) channels are occupied: 90% and up.
- The dashed line in yellow stands for the average channel occupancy during the interval.

This helps to perform a more accurate analysis toward channel availability throughout the day.

Filters

To make a more specific analysis, it is possible to filter the information represented on the graphs. To do this, simply click on the "search" button in the upper right-hand corner, with which a mask will open where you can enter:

- A time period in which you want to observe the data (this via start and end date)
- One or more specific gateways to observe



Once you have entered all the information, click on the "search" button to enable the filters, otherwise on "reset" to clear them completely.

Widget

For explanation on creating, organizing and managing Widgets, visit:

Coincidences Out

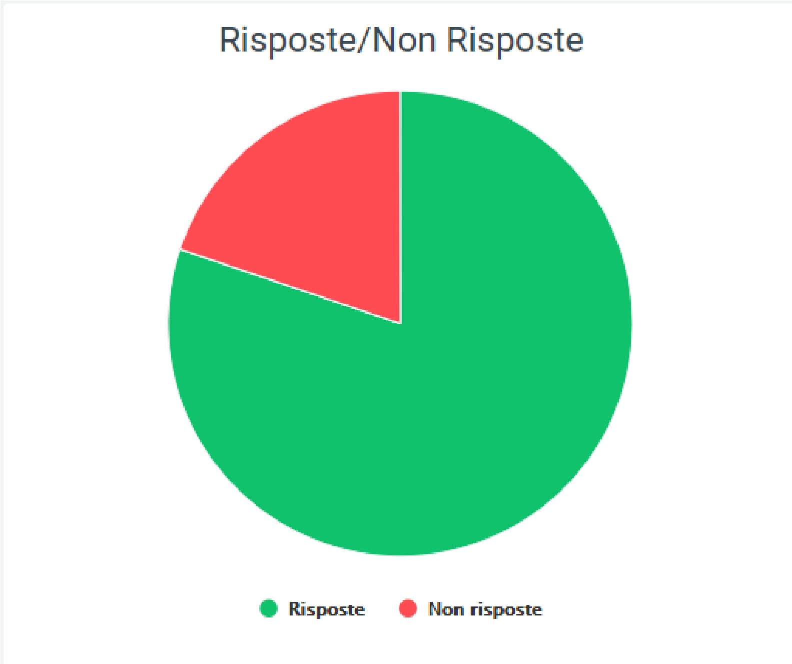
Unlike the "Coincidences" section, which takes into account all voice lines in the exchange, the "Coincidences Out" section counts only outgoing calls passing through "configurations." The representation of the graphs is based on the representation of the gateway "configurations" data. Report Groups (date, time)

Report Groups

In this section it is possible to visualize the progress of calls divided into the various groups of the telephone exchange through the use of simple and clear graphs, but highlighting the information necessary for accurate analysis. In addition to the graphs, there is a table for each group, which contains detailed information.

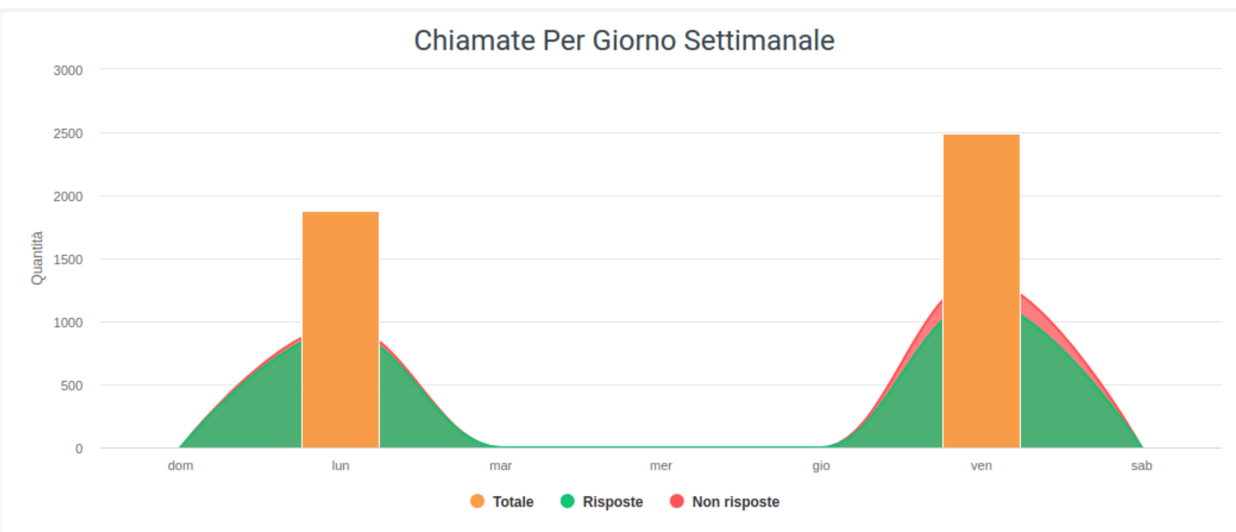
Charts

The first graph that can be observed is a pie chart representing the number of calls answered and not answered by the groups. By scrolling over it with the mouse, it will be possible to observe, depending on the position of the mouse, the numbers in detail.

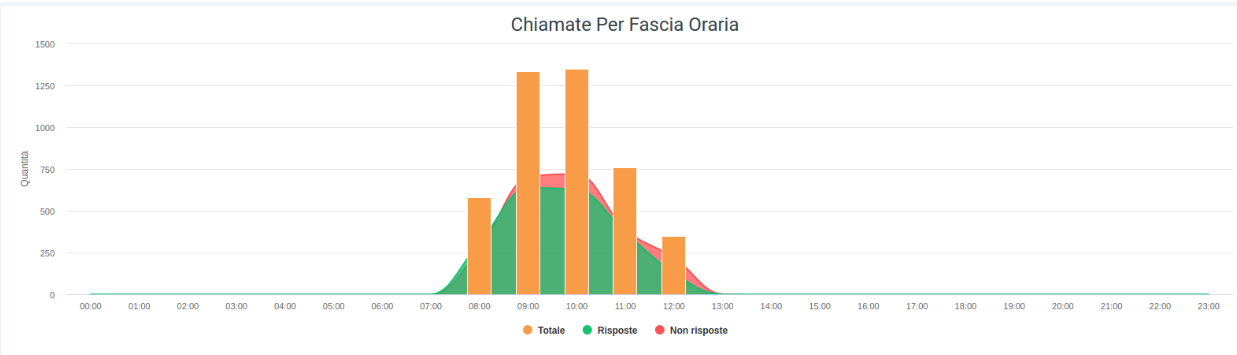


Using the legend at the bottom, it is possible to remove and enable the display of a graph information by clicking on the corresponding colored dot.

The second graph represents the number of calls broken down by days of the week. The total number of calls and the trend of answered and unanswered calls is visible. Again, through the legend it is possible to select which information to display and which not to display.



The last graph represents the trend of calls broken down by time slot. This allows you to observe the trend of answered, unanswered, and total calls over the course of a day. You can use the legend to view only some of the information.



Tables

Below is a table called "Number of Calls Handled" which contains information regarding the calls in each group. On the leftmost part is the name of the group, followed by a small pie chart representing call trends for that particular group.

In the middle part of the table are:

- Total: the total amount of calls arrived at the group
- **Answered**
  - quantity: the quantity of answered calls
  - converse time: total time of all answered calls
  - Average talk time: average time among all answered calls
- Unanswered
- Count CFWD (Conditional Forward)
- Count UFWD (Unconditional Forward)
- Count FORK2MOBILE (Fork to mobile).

The last row of the table contains the sum of all data in the respective column. For the average talk time, an average will be taken between all group averages.

These tables are exportable in excel format, via the "Export XLS" button in the upper right corner.

Numero chiamate gestite

Export XLS

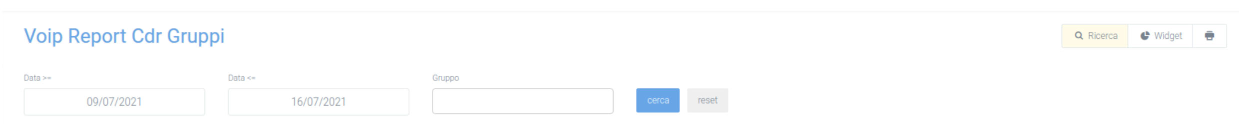
		TOTALE	RISPOSTE			NON RISPOSTE	COUNT CFWD	COUNT UFWD	COUNT FORK2MOBILE
		quantità	quantità	tempo conversazione	media tempo conversazione	quantità	quantità	quantità	quantità
###		2				2			
ADRIA gruppo Centralino Generale		34	28	00:52:44	00:01:53	6			
AFFI gruppo Centralino Generale		69	16	00:53:23	00:03:20	53			
AFFI gruppo Servizio Tirocini		2	2	00:03:39	00:01:49				
ARZIGNANO gruppo Centralino Generale		54	32	01:35:28	00:02:59	22			
BADIA gruppo Centralino Generale		52	23	01:41:15	00:04:24	29			
BASSANO gruppo Centralino Generale		119	78	02:14:53	00:01:43	41	3		15

	TOTALE	RISPOSTE			NON RISPOSTE	COUNT CFWD	COUNT UFWD	COUNT FORK2MOBILE
	quantità	quantità	tempo conversazione	media tempo conversazione	quantità	quantità	quantità	quantità
VENEZIA gruppo Centralino Generale	336	173	01:50:49	00:00:38	163	21		1
VENEZIA gruppo Servizi Amministrativi	21	6	00:03:33	00:00:35	15			
VERONA gruppo Centralino Generale	115	48	02:12:41	00:02:45	67	41		
VERONA gruppo Servizi Amministrativi	2	2	00:06:17	00:03:08				
VERONA gruppo Servizio Collocamento	11	5	00:09:17	00:01:51	6			
VERONA gruppo Servizio Imprese	1	1	00:00:50	00:00:50				
VERONA gruppo Servizio Tirocini	6				6			
VICENZA gruppo Centralino Generale	213	166	03:38:09	00:01:18	47			23
VILLAFRANCA gruppo Centralino Generale	58	21	01:04:16	00:03:03	37	1		
VITTORIO VENETO gruppo Centr Generale	50	31	01:12:50	00:02:20	19	96		
Totali	4.371	2.073	2gg 22:39:38	00:02:02	2298	331		105

Filters

To perform the analysis of a specific group in a specified time period, the filter mask can be used, which provides:

- start date filter: the system takes into account data present from the entered date onwards;
- end date filter: data stored after the entered date are not used;
- Group filter: only the groups entered in this filter will be displayed;



Filters are retrievable via the "Search" button in the upper right corner with which the mask will open, as in the picture, where it will be possible to enter the data to be investigated. After entering, click on "search," while to delete all filters, click on the "reset" button.

Widget

For the explanation on creating, organizing and managing Widgets, please visit the page:

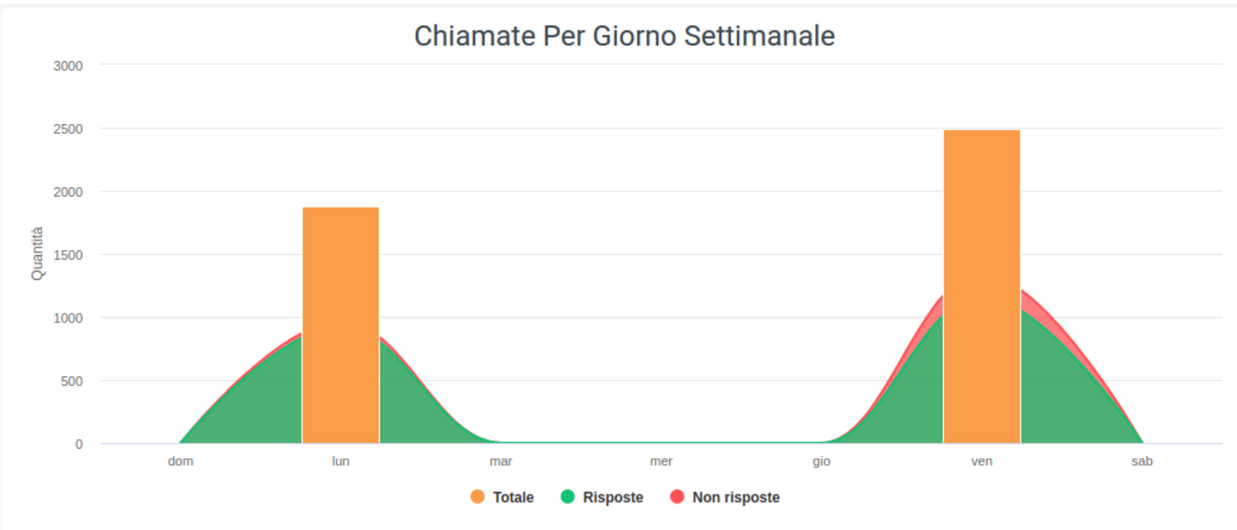
Report Date Groups

Unlike the "Groups Report" section that groups together total call information for each group, the "Groups Report by Date" allows this information to be analyzed separately for each individual day.

Charts

With the graph below, it is possible to observe the trend and quantity of calls with a breakdown by days of the week. In addition, by scrolling over the graphs with the mouse, the number of total calls, answered calls, and unanswered calls can be observed.

The data contained in the graph, shown in the legend, can be removed from view by clicking on the colored dot next to the name. To view them again, simply click on them again.

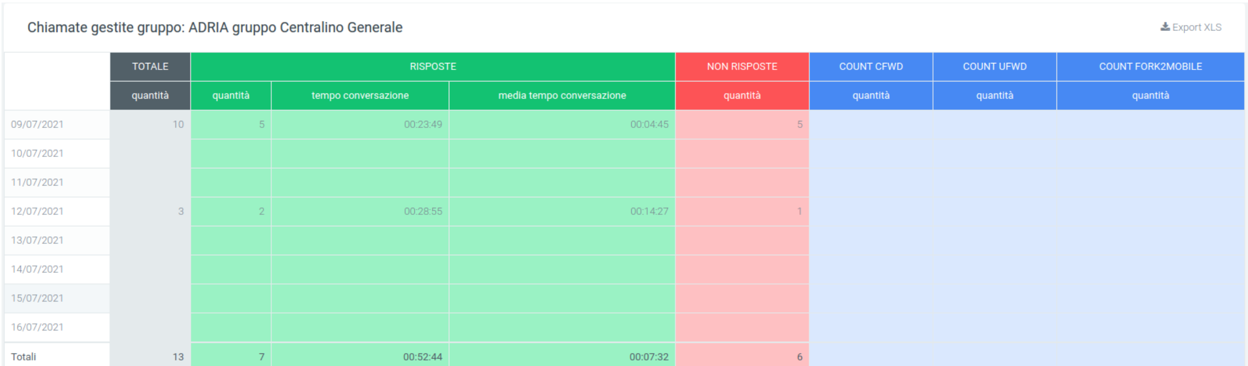


Tables

At the bottom of the screen you will see a table for each group, broken down by days of the week. For each of these you will be able to analyze the same types of data found in the group report table.

Again, the last row of the table contains the sum of all the data in the respective column. For the average talk time, an average will be taken between all group averages.

These tables are exportable in excel format, via the "Export XLS" button in the upper right corner.



Chiamate gestite gruppo: ADRIA gruppo Centralino Generale

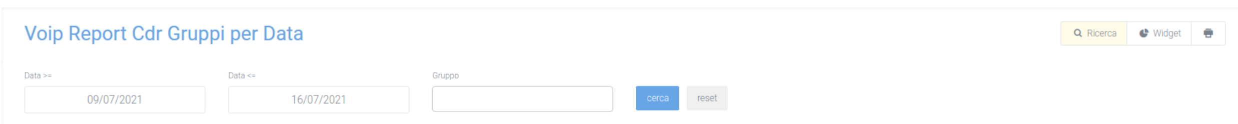
Export XLS

	TOTALE	RISPOSTE			NON RISPOSTE	COUNT CFWD	COUNT UFWD	COUNT FORK2MOBILE
	quantità	quantità	tempo conversazione	media tempo conversazione	quantità	quantità	quantità	quantità
09/07/2021	10	5	00:23:49	00:04:45	5			
10/07/2021								
11/07/2021								
12/07/2021	3	2	00:28:55	00:14:27	1			
13/07/2021								
14/07/2021								
15/07/2021								
16/07/2021								
Totali	13	7	00:52:44	00:07:32	6			

Filters

If there is a need to analyze data on a specific date or time frame or for one or more specific groups, simply use the filter mask, which will provide:

- start date filter: the system takes into account data present from the date entered onwards;
- end date filter: data stored after the entered date are not used;
- Group filter: only the groups entered in this filter will be displayed;



Report Groups Time

In the "Report Groups by Time" section you can view call trends, this time broken down into time slots. You can then analyze the data by checking, for example, in which time slots there is more call traffic or in which slot the groups work best.

Tables

For each group there will be a table broken down by time slots containing the same types of data as in the group report tables.

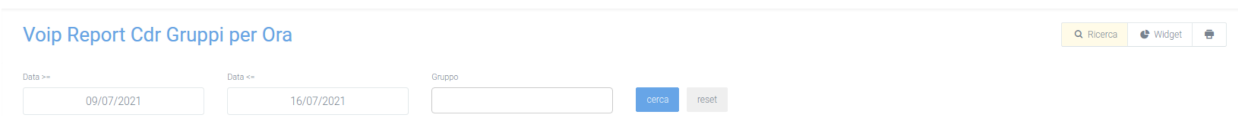
Chiamate gestite gruppo: BASSANO gruppo Centralino Generale

Export XLS

	TOTALE	RISPOSTE			NON RISPOSTE	COUNT CFWD	COUNT UFWD	COUNT FORK2MOBILE
	quantità	quantità	tempo conversazione	media tempo conversazione	quantità	quantità	quantità	quantità
00:00								
01:00								
02:00								
03:00								
04:00								
05:00								
06:00								
07:00								
08:00	14	4	00:33:19	00:08:19	10			1
09:00	33	17	00:39:43	00:02:20	16			4
10:00	16	9	00:38:22	00:04:15	7	1		4
11:00	14	7	00:12:10	00:01:44	7	1		4
12:00	6	5	00:11:19	00:02:15	1	1		2
13:00								
14:00								
15:00								
16:00								
17:00								
18:00								
19:00								
20:00								
21:00								
22:00								
23:00								
Totali	83	42	02:14:53	00:03:12	41	3		15

Filters

Again, if there is a need to analyze a precise date or time frame or to search for information for one or more specific groups, simply use the filter mask, which will provide:



- start date: the system takes into account the data present from the entered date onwards
- End date: data stored after the entered date are not used
- Group: only data from the groups entered in this filter are displayed

Report Operators (account)

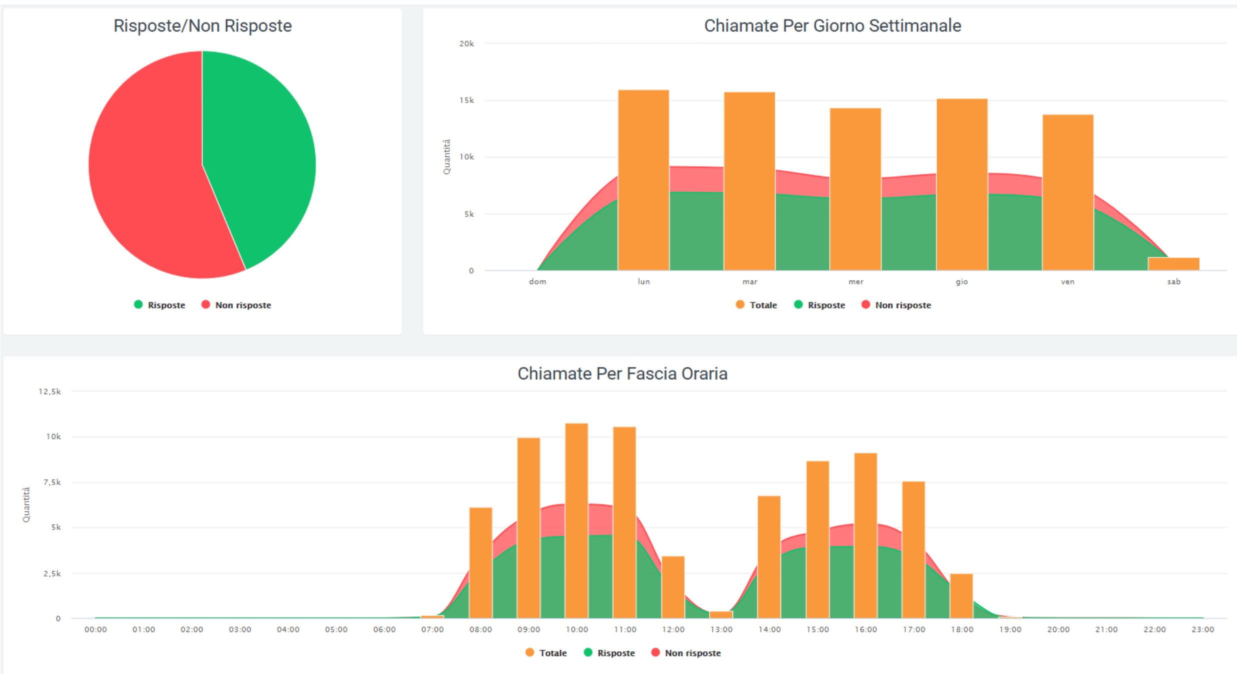
The following section reports call trends for each individual operator.

Charts

The total trend of answered and unanswered calls is graphically represented in a pie chart where, by scrolling over with the mouse, the quantities can be observed in detail.

In addition, there is the possibility of removing a data item from the visualization via the legend by clicking over the name or colored dot. To put it back into the visualization, simply click on it again. These two functions, detail and visibility, are also available in the same way on the other two graphs.

The other two graphs represent one the total call trends by breakdown by days of the week, the other by time slot. These two, in addition to the number of answered and unanswered calls, also report the total number of calls, as can be seen in the following picture.



Tables

After the graphical display of call trends, the data sonported in a table divided by operators, each with its own detailed information. Before reporting the data for each operator, the total data for each field can be observed:

- the quantity of calls
- the total time
- the talk time
- average total time
- average ringing time
- average answer time
- average talk time

After this first part, the table shows for each row the respective operator with the exact information. On the left is the name of the operator, followed by the total amount of calls handled by it. Next, information on answered and unanswered calls is shown before going into the details of each of these macro-groups.

- **Answers:**
    - NCC-AGENT
    - NCC-CALLER
    - NCC
    - ANSWER
    - TRANSFER
    - TRANSFER
- **Unanswered:**
    - ANSWERED ELSEWHERE
    - BUSY
    - CANCELLED
    - CFWD
    - CONGESTION.
    - FORWARD
    - NOANSWER
    - PICKUP
    - REFUSED
    - TIMEOUT
    - NAVIABLE

Each of these subgroups will have a further subdivision into details.

For answered calls, the details present will be:

- quantity
- time worked
- average time worked
- conversation time
- average ring time
- average answer time
- average talk time

For unanswered calls instead:

- quantity
- time worked
- average time worked

At the end of the table, at the bottom, the totals of each detail column will be shown.

Using the appropriate button located at the top right of the table, it will be possible to export it to excel "XLS" format.

Numero chiamate gestite

Export XLS

quantità: 76.258	tempo totale: 49gg 06:00:41	tempo di conversazione: 43gg 14:00:46	media tempo totale: 00:00:55	media tempo squillo: 00:00:04	media tempo risposta: 00:00:04	media tempo conversazione: 00:00:49

	TOTALE	RISPOSTE					NON RISPOSTE
	quantità	quantità	media tempo lavorato	media tempo squillo	media tempo attesa	media tempo conversazione	quantità
500 (Voicemail Giime)	6	6	00:00:36		00:00:01	00:00:35	
515 (Francesca 515)	7.804	3.647	00:01:57	00:00:03	00:00:13	00:01:47	4.157
516 (Elena 516)	2.405	731	00:01:58	00:00:06	00:00:09	00:01:55	1.674
518 (Laura 518)	11.312	4.633	00:02:05	00:00:02	00:00:12	00:01:55	6.679
519 (Anna 519)	9.474	3.620	00:02:07	00:00:02	00:00:12	00:01:57	5.854
520 (Erica 520)	3.037	1.612	00:01:54	00:00:03	00:00:08	00:01:49	1.425
521 (Jessica 521)	4.448	3.040	00:01:38	00:00:03	00:00:08	00:01:34	1.408
524 (Cristina 524)	951	632	00:03:18	00:00:02	00:00:06	00:03:14	319
525 (Danilo 525)	230	123	00:02:57	00:00:01	00:00:07	00:02:51	107
526 (Ombretta 526)	1.059	486	00:01:28	00:00:04	00:00:06	00:01:26	573
529 (Meli 529)	56	32	00:01:17	00:00:06	00:00:06	00:01:17	24
530 (ElisaT 530)	7.632	3.289	00:02:13	00:00:02	00:00:12	00:02:03	4.343
531 (Alessia 531)	1.363	802	00:02:02	00:00:05	00:00:08	00:02:00	561
533 (Molinari 533)	917	530	00:03:48	00:00:02	00:00:04	00:03:47	387
534 (Gemma 534)	110	53	00:01:39	00:00:02	00:00:04	00:01:38	57
535 (Antonio 535)	5						5

	NCC-AGENT							NCC-CALLER							
	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo squillo	media tempo risposta	media tempo conversazione	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo squillo	media tempo risposta	m co
500 (Voicemail Giime)														
515 (Francesca 515)	238	11:57:13	00:03:00	11:57:13	00:00:11	00:00:11	00:03:00	460	21:27:24	00:02:47	21:27:24	00:00:11	00:00:12	
516 (Elena 516)	125	04:29:00	00:02:09	04:29:00	00:00:12	00:00:12	00:02:09	81	02:17:41	00:01:41	02:17:41	00:00:11	00:00:11	
518 (Laura 518)	317	16:09:14	00:03:03	16:09:14	00:00:11	00:00:11	00:03:03	250	12:25:38	00:02:58	12:25:38	00:00:11	00:00:11	
519 (Anna 519)	483	1gg 01:11:28	00:03:07	1gg 01:11:28	00:00:08	00:00:09	00:03:07	169	08:44:33	00:03:06	08:44:33	00:00:09	00:00:09	
520 (Erica 520)	114	03:35:17	00:01:53	03:35:17	00:00:09	00:00:10	00:01:53	84	02:23:43	00:01:42	02:23:43	00:00:09	00:00:10	
521 (Jessica 521)	458	17:39:27	00:02:18	17:39:27	00:00:09	00:00:10	00:02:18	156	05:57:25	00:02:17	05:57:25	00:00:10	00:00:10	
524 (Cristina 524)														
525 (Danilo 525)														
526 (Ombretta 526)	119	02:56:55	00:01:29	02:56:55	00:00:08	00:00:08	00:01:29	43	00:44:43	00:01:02	00:44:43	00:00:08	00:00:08	
529 (Meli 529)														
530 (ElisaT 530)	521	20:41:50	00:02:23	20:41:50	00:00:08	00:00:09	00:02:23	139	05:14:19	00:02:15	05:14:19	00:00:09	00:00:09	
531 (Alessia 531)	126	04:04:10	00:01:56	04:04:10	00:00:09	00:00:10	00:01:56	51	01:44:02	00:02:02	01:44:02	00:00:09	00:00:10	
533 (Molinari 533)														
534 (Gemma 534)														
535 (Antonio 535)														

Filters

By using filters, it will be possible to improve the data analysis activity by focusing only on those of interest. In order to obtain them, it will be sufficient to click on the "search" button at the top right with which a mask containing the relevant filters will be opened. Through this enter the data in the appropriate boxes:

- start and end date: only data present in this time frame will be included
- status: where the status of the calls to be analyzed should be indicated
- exit causes: the reason why the call is terminated
- operator: this is to make sure to have a report, even a graphical one, exclusive to the selected operator (or more than one).
- source: where the call came from
- destination: where the call will end

Once the relevant data has been entered, click on "search" to start the filtered search, otherwise on "reset" to delete all data entered.

Widget

For explanation on creating, organizing and managing widgets, visit:

Operators Account Report

This section encapsulates the information found in the "Operators Report" section, but the breakdown is by account.

Tables

In this section we can find a table containing the working time and the number of calls handled by each operator account. This allows us to keep track of each operator's work performance and to assess which account is being used the most (each account may be a different device on which a call may be answered).

As the first piece of information is the operator's account code and name, then the total time worked in the time interval entered through the filters. Below the account code and name (in dark gray), there are other names (light gray box): these are the various accounts connected to the operator. After these you have a breakdown of calls by incoming, outgoing, and internal calls, each with its own detailed information.

Input:

- quantity
- time worked
- average time worked
- average response time

Output:

- quantity
- time worked
- quantity OK (successful calls)
- average time worked OK (with successful calls)
- quantity KO (unanswered calls)
- average time worked KO (with calls not answered)

Internal:

- quantity
- time worked
- average time worked

Each of these boxes will show the exact data regarding a specific operator account.

Numero chiamate gestite e Tempo di Lavoro

Export XLS

		ingresso				uscita						interna		
	TEMPO TOTALE LAVORATO	quantità	tempo lavorato	media tempo lavorato	media tempo risposta	quantità	tempo lavorato	quantità OK	media tempo lavorato OK	quantità KO	media tempo lavorato KO	quantità	tempo lavorato	media tempo lavorato
500 (Voicemail Giime)	00:03:39					1	00:00:14	1	00:00:14			5	00:03:25	00:00:41
###	00:03:39					1	00:00:14	1	00:00:14			5	00:03:25	00:00:41
500-hotdesk														
515 (Francesca 515)	3gg 10:38:10	769	1gg 07:55:56	00:02:29	00:00:10	1.922	1gg 13:46:48	1.311	00:01:27	611	00:00:34	402	12:55:26	00:01:55
###	1gg 17:05:16	48	02:51:26	00:03:34		1.922	1gg 13:46:48	1.311	00:01:27	611	00:00:34	139	00:27:02	00:00:11
515-hotdesk	1gg 03:20:49	532	20:39:23	00:02:19	00:00:10							174	06:41:26	00:02:18
Account001	14:12:05	189	08:25:07	00:02:40	00:00:12							89	05:46:58	00:03:53
516 (Elena 516)	20:59:43	407	12:27:34	00:01:50	00:00:08	214	05:04:05	173	00:01:41	41	00:00:18	150	03:28:04	00:01:23
###	07:14:12	44	02:07:17	00:02:53		214	05:04:05	173	00:01:41	41	00:00:18	41	00:02:50	00:00:04
516-hotdesk	08:07:48	268	07:28:01	00:01:40	00:00:09							58	00:39:47	00:00:41
Account002	05:37:43	95	02:52:16	00:01:48	00:00:12							51	02:45:27	00:03:14

Filters

The filters that can be entered are the same as those listed above:

- start and end date
- call status
- exit causes
- operator
- operator account
- call source
- call destination

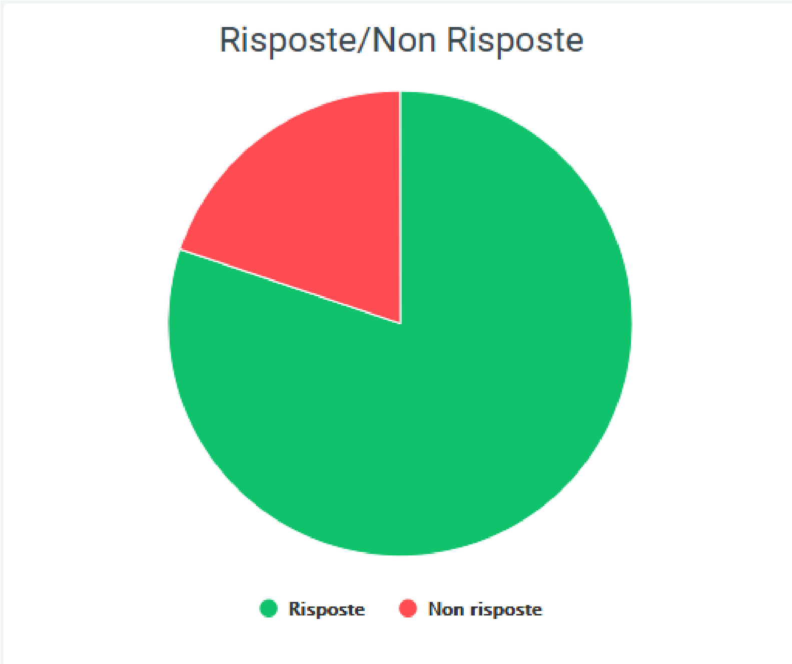
Report Organizations (operator)

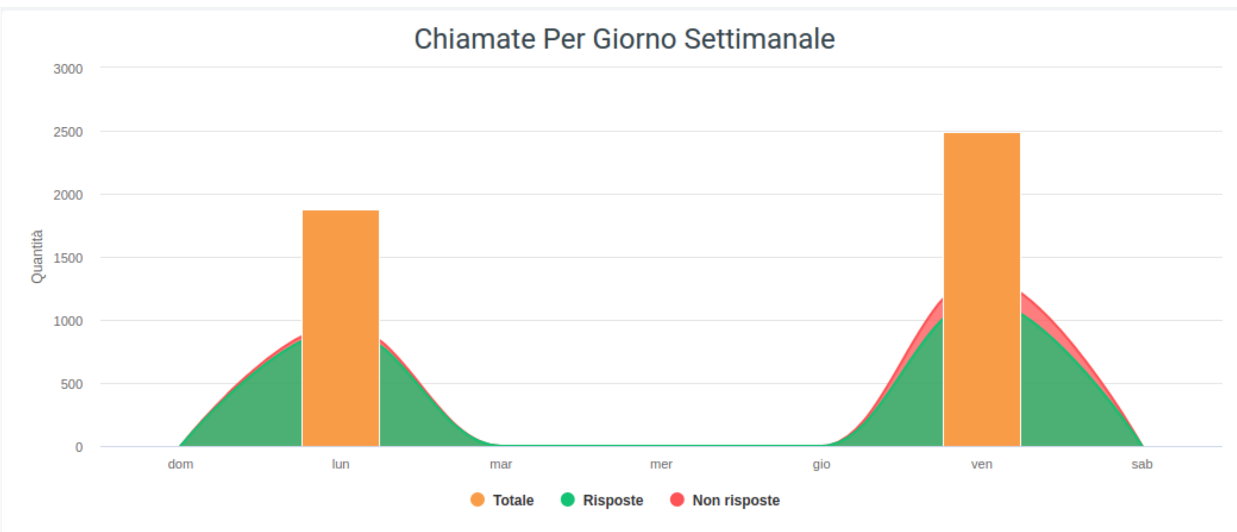
This section allows you to analyze call trends for each organization, bringing back both graphical and tabular reports, so that you get first a visual and then a more detailed analysis.

Charts

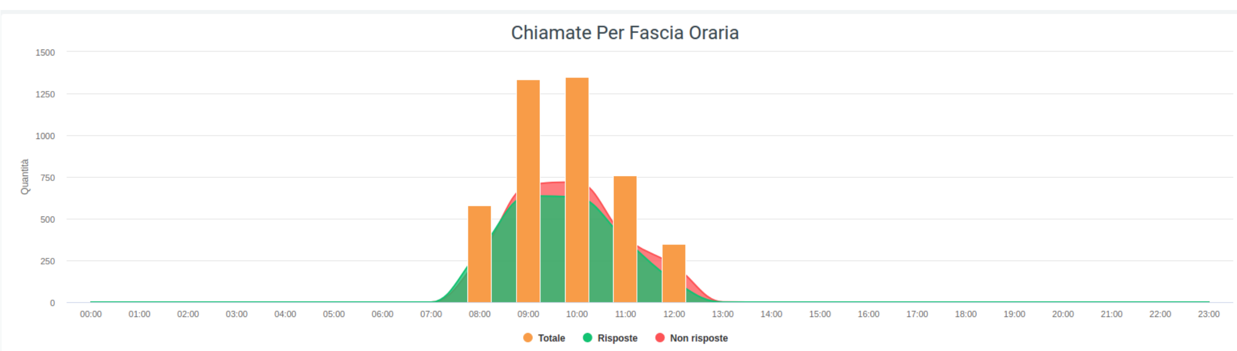
There are 3 graphs in this section and they show call trends in total or distributed calls over a certain period of time:

- pie chart: (on the right) represents answered and unanswered calls, within the time frame defined in the filters, from all organizations in the company. It performs a sum total.
- Weekly graph: represents call trends over the week. It shows a curve for calls and their status, while reporting a column for the total number of calls.





- hourly graph: shows the pattern of calls over the course of a day, thus with an hourly breakdown. Also used in this are the curves for call status and the column for total number.



In all three cases, it is possible to observe the detailed number of calls by simply scrolling with the mouse over the graph. You can also remove or reinsert one or more pieces of information into the view by clicking on the name or colored dot of the data item in the legend.

Tables

The table below shows, broken down by organization, call details, including information such as status, with all related information, and exit causes, i.e., why a call ended. First, at the top of the table, there are totals of the data in the table, namely:

- the total quantity
- the total time worked
- the total talk time
- the average time worked
- the average time the phone rang
- the average answer time
- the average talk time

On the left side of the table, before the grid containing all the data, is the name of the organization with an associated small pie chart representing the status of calls received and made by operators belonging to that organization. Immediately following this is the total amount of calls, before all the detailed data.

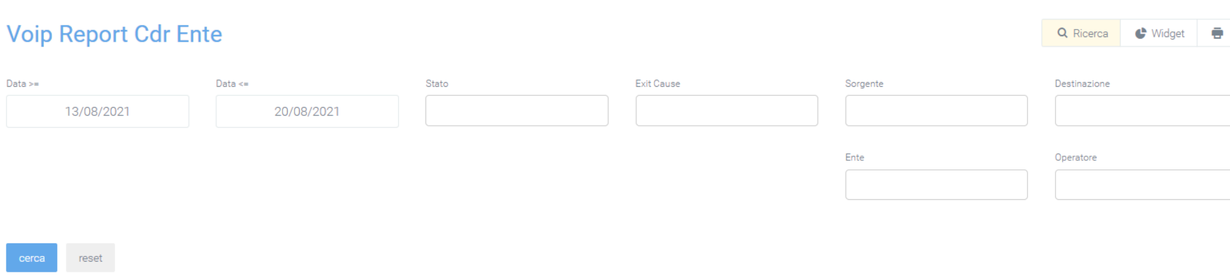
As in any table, a total calculation of each column will be found at the bottom.

As always, it is possible to export the table to excel format via special "Export XLS" button in the upper right corner (of the table).

Filters

As described initially, to enable the filters simply click the "search" button in the upper right corner with which a mask will open in which all the necessary data should be entered:

- start and end date of the time interval in which the data will be shown
- status
- exit causes, i.e. the reasons why a call is terminated
- source
- destination
- organization (more than one can be entered)
- operator



After entering all the desired data, click on "search" to enable the filters, otherwise on "reset" to remove them.

Widget

For explanation on creating, organizing and managing Widgets, visit:

Report Organizations Operator

Each organization consists of several operators, each holding different accounts. This section allows you to keep track of call trends for each of these by displaying a table for each organization, where each row belongs to an operator account.

Tables

The table will present information regarding total calls, answered calls, and unanswered calls, broken down by exit causes. For successful calls, the information contained in the exit causes will be:

- the quantity
- the waiting time
- the total and average work and talk time
- the average time for which the phone rang
- the average of the answer time

Regarding unanswered calls, the information for each exit cause is basically three:

- quantity
- average time worked
- total time worked

First, however, above the table is a grid containing a total of some information, viz:

- the total quantity
- the total time worked
- the total talk time
- the average time worked
- the average time the phone rang
- the average answer time
- the average talk time

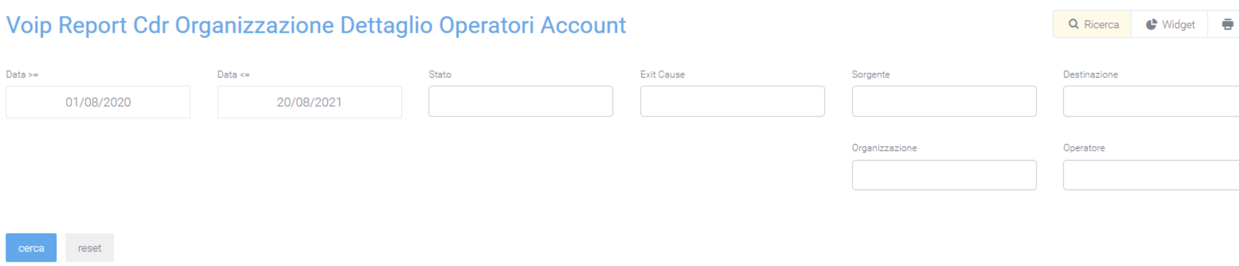
In the leftmost part are the operators (those with the darkest color) and their accounts (below the operator and lighter in color). Next, before going into the details of each exit cause, is a total of calls, both answered and unanswered.

Each table is exportable in excel format, individually via its own "Export XLS" button located at the top right of each one.

Filters

To enhance the analysis experience by selecting only certain information to display or certain specific time periods, filters can be enabled. To do this, click on the "search" button, with which a specially designed mask will open, and enter the desired data from those proposed:

- start and end date of the time interval in which the data will be shown
- call status
- exit causes
- source
- destination
- organization
- operator



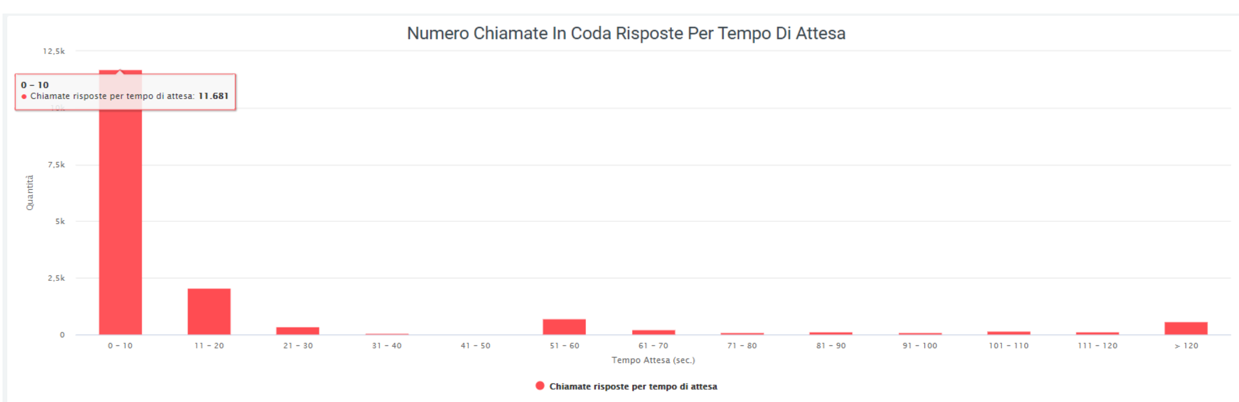
After entering all the desired data, click on "search" to enable the filters, otherwise on "reset" to delete them.

Waiting Times per Queue Report (incoming).

In this section it is possible to observe the report of the number of calls broken down by waiting times, by graphical display and table, of queues specified through the filter mask.

Charts

In the graph, the exact number of calls by waiting time of the specified queue is observed. It can be seen that the wait times in seconds increase as one proceeds to the right of the graph, starting from a range of 0 to 10 seconds, reaching a maximum of over 120. The height of each graph represents the number of calls within a range.



Tables

The table below shows the same data, i.e., the amount of calls by average waiting time. In addition, there is a percentage of accuracy in answering, based on the ratio of the amount of calls in a waiting time range to the total amount. The total is visible in the rightmost part of the table.

The table is exportable via special “Export XLS” button in the upper right corner.

Numero chiamate in coda risposte per tempo di attesa (secondi)

Export XLS

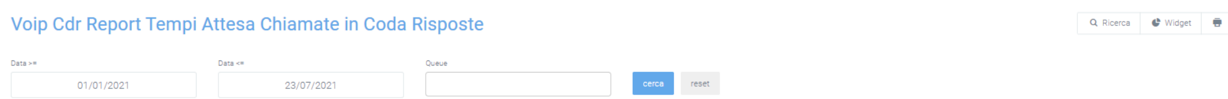
	0 - 10	11 - 20	21 - 30	31 - 40	41 - 50	51 - 60	61 - 70	71 - 80	81 - 90	91 - 100	101 - 110	111 - 120	> 120	TOTALE
QTA	11681	2067	340	69	15	713	239	98	121	99	157	133	591	16323
PERC	71,56 %	12,66 %	2,08 %	0,42 %	0,09 %	4,37 %	1,46 %	0,60 %	0,74 %	0,61 %	0,96 %	0,81 %	3,62 %	

Filters

The special feature of this section is the ability to view the number of calls by waiting time, of each specified queue via filters and to do this you will need to:

- Click on the “search” button in the upper right corner
- Click on the “queues” item
- Select one or more interested queues
- Click the “search” button

The time frame in which the data is represented is also variable via filters. After opening the mask as described above, under the “Date >=” and “Date <=” headings the start and end dates of the relevant time frame should be entered.



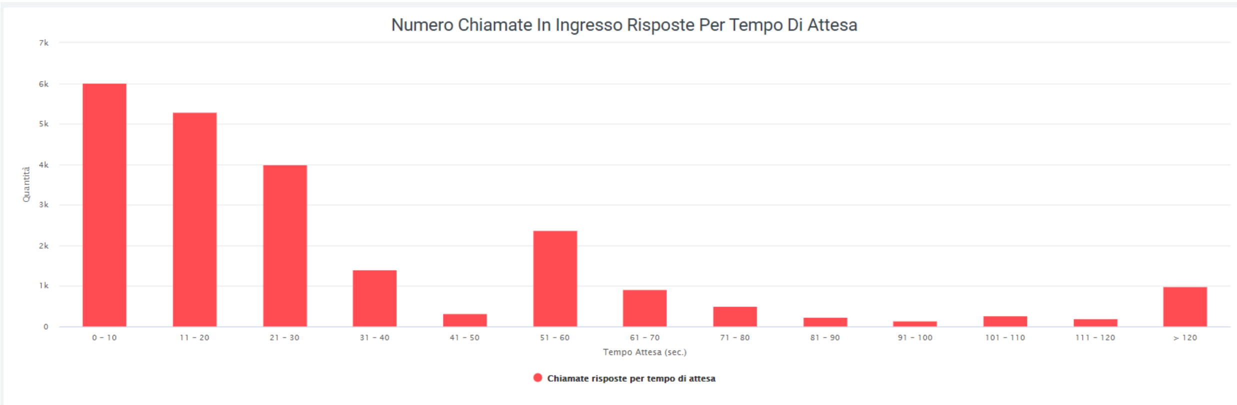
If, under “queues,” no preference is entered, the sum of all queues for each waiting time interval will be displayed.

Incoming Wait Times Report

In the following section it is possible to observe a report of the waiting times for incoming calls, with either a graphical display or by means of a table. In both cases, the quantities of calls answered within a certain waiting time are shown.

Charts

As in the previous graph, it can be seen that the wait times in seconds increase as one proceeds to the right of the graph, starting from a range of 0 to 10 seconds and reaching a maximum of over 120. Scrolling with the mouse over the graphs will show the exact number of calls.



Tables

The table below shows the same data as the previous table.

Numero chiamate in ingresso risposte per tempo di attesa (secondi)

Export XLS

	0 - 10	11 - 20	21 - 30	31 - 40	41 - 50	51 - 60	61 - 70	71 - 80	81 - 90	91 - 100	101 - 110	111 - 120	> 120	TOTALE
QTA	6003	5295	3992	1412	332	2372	909	506	236	137	273	191	987	22645
PERC	26,51 %	23,38 %	17,63 %	6,24 %	1,47 %	10,47 %	4,01 %	2,23 %	1,04 %	0,60 %	1,21 %	0,84 %	4,36 %	

The table can be exported using the “Export XLS” button in the upper right corner.

Filters

Clicking on the “search” button at the top right of the page will open a filter mask where a time range can be entered in which the data will be displayed. This can be done by pressing on either box and entering the desired start and end dates.

Once the data is entered, press on the “search” button to start the filtered search, otherwise on “reset” to delete the filters entered.

Widget

For explanation on creating, organizing and managing Widgets, visit:

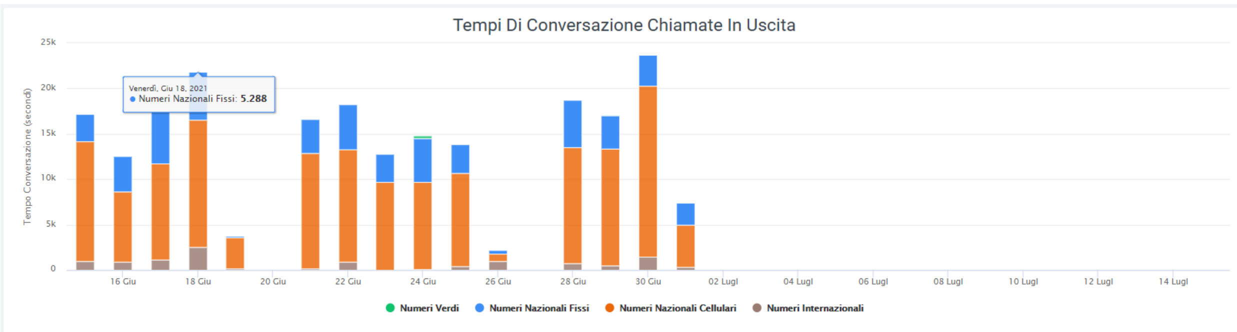
Conversation Times Report

This section reports the talk times, in seconds, of all outgoing calls from the telephone exchange to which the platform is connected. The data are first reported with a daily breakdown graph where conversation times to:

- toll-free numbers
- domestic fixed numbers
- domestic cellular numbers
- international numbers

Charts

By scrolling over the graphs with the mouse, it will be possible to view the exact number in seconds of the talk time of a specific field from those listed before. It will also be possible to remove the display from these via the legend. To do so, simply click on the colored dot of the information you want to remove from display. To reinsert it, click again on the dot, which will now have turned gray.



Tables

At the bottom of the page, on the other hand, it is possible to observe a table containing the talk times represented in the previous graph, but with a monthly time breakdown. As in the graph, here too it is possible to observe the trend of times in the various numbers (toll-free, fixed, cellular and international). The last row of the table encloses the conversation time totals for each category in the table.

Tempi di conversazione chiamate in uscita

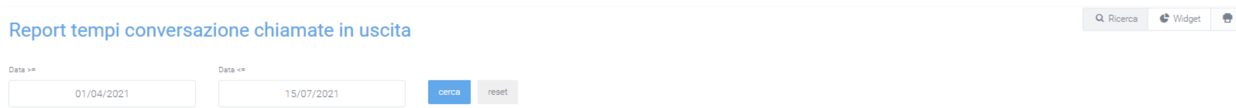
Export XLS

	Numeri Verdi	Numeri Nazionali Fissi	Numeri Nazionali Cellulari	Numeri Internazionali
04-2021	01:31:48	1gg 05:01:24	4gg 01:59:22	11:01:16
05-2021	00:40:36	1gg 08:28:08	4gg 05:38:01	09:07:19
06-2021	00:19:06	1gg 00:33:22	3gg 08:46:09	07:12:03
07-2021		00:41:42	01:17:39	00:04:48
Totali	02:31:30	3gg 14:44:36	11gg 17:41:11	1gg 03:25:26

This can be exported to XLS format using the appropriate button at the top right of the table.

Filters

Pressing on the search button at the top right of the page will open a filter mask on which the time interval can be entered that will be represented on the chart and shown in the table. Simply enter the start and end date, then press "search." To delete all the filters entered, click on the "reset" button.



Widget

Widgets For an explanation of creating, organizing, and managing Widgets, visit:

Report Organizational Unit (operator)

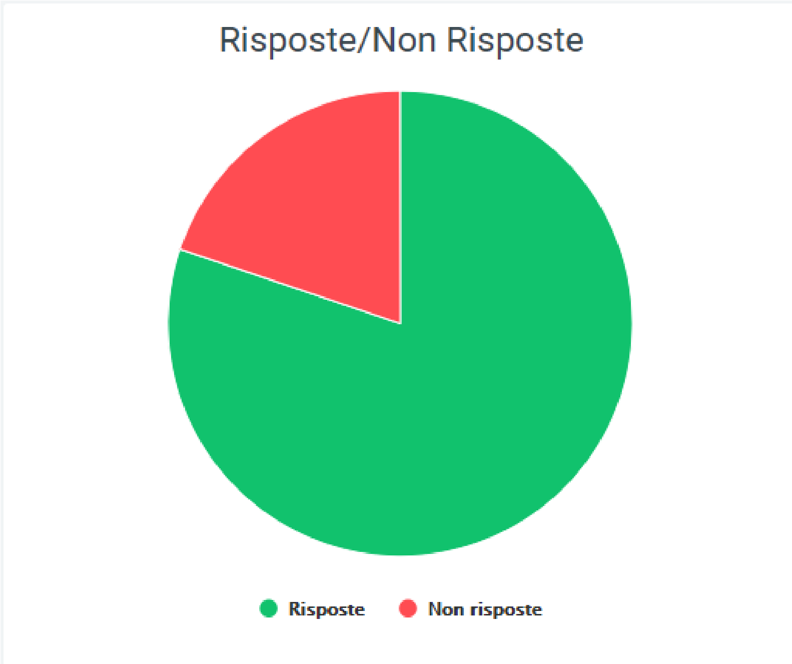
In case there is a need to analyze call trends for each organizational unit, this section allows this by showing reports with graphs and tables, so as to have an initial visual analysis through graphs and then a more detailed one through tables.

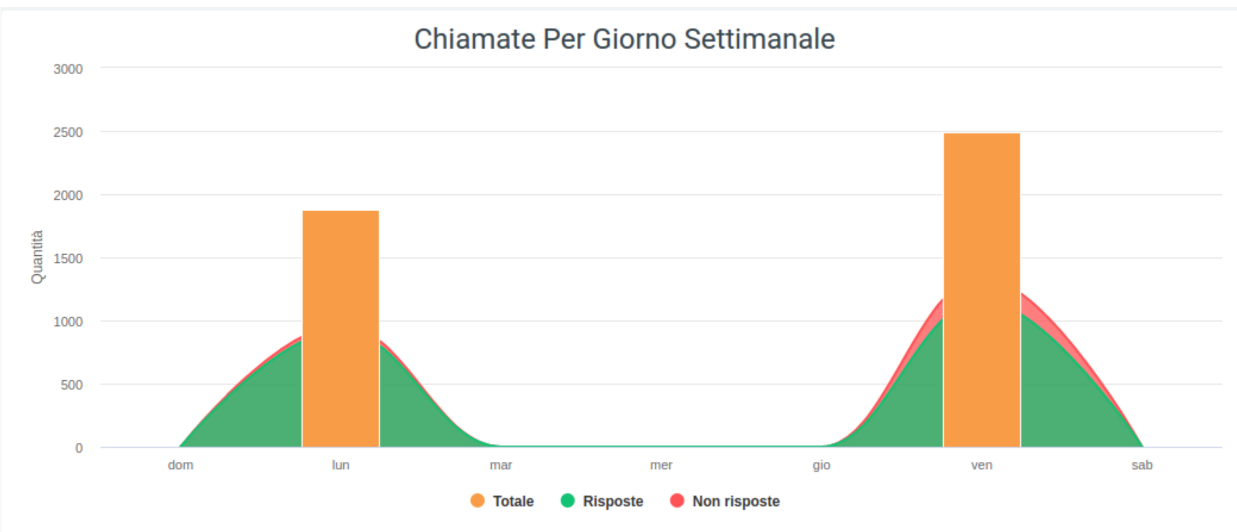
Charts

Answered, not answered

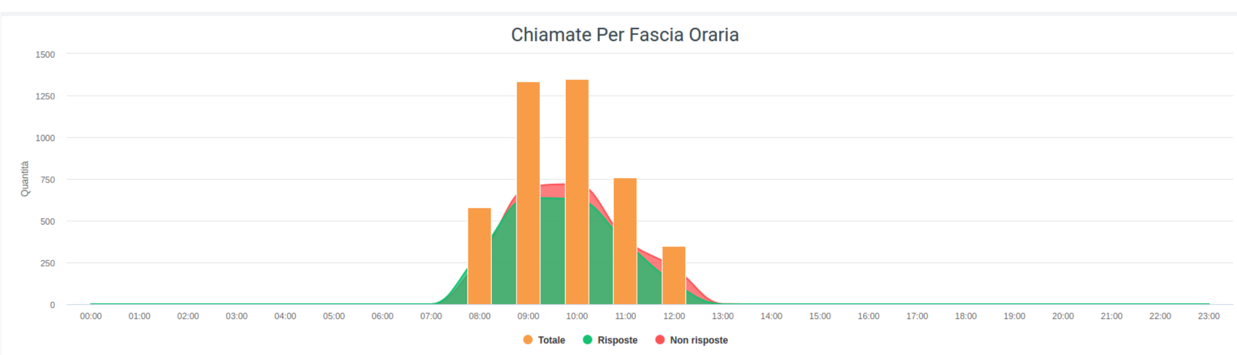
There are 3 graphs in this section and they show the trend of calls in total or distributed over a certain period of time:

- pie chart: represents answered and unanswered calls, within the time frame defined in the filters, from all organizations in the company. It performs a total sum.
- Weekly graph: represents call trends over the week. It shows a curve for calls and their status, while reporting a column for the total number of calls.





- hourly graph: shows the pattern of calls over the course of a day, thus with an hourly breakdown. Also used in this are the curves for call status and the column for total number.

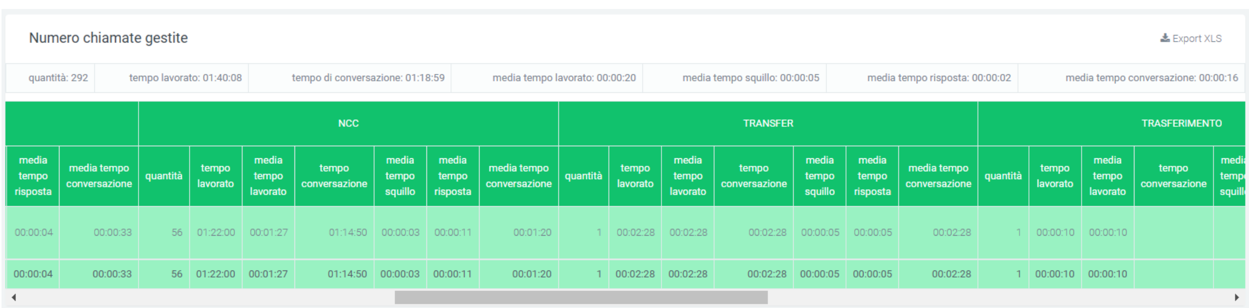


In all three cases, it is possible to observe the detailed number of calls by simply scrolling with the mouse over the graph. You can also remove or reinsert one or more pieces of information into the view by clicking on the name or colored dot of the data item in the legend.

Tables

The table below shows, broken down by organizational unit, call details, including information such as status, with all related information, and exit causes, i.e., why a call ended. First of all, at the top of the table, there are totals of the data in the table, viz:

- the total quantity
- the total time worked
- the total talk time
- the average time worked
- the average time the phone rang
- the average answer time
- the average talk time



Numero chiamate gestite

Export XLS

quantità: 292	tempo lavorato: 01:40:08	tempo di conversazione: 01:18:59	media tempo lavorato: 00:00:20	media tempo squillo: 00:00:05	media tempo risposta: 00:00:02	media tempo conversazione: 00:00:16

BUSY			CANCELED			CONGESTION			NOANSWER			REFUSED			TIMEOUT			UNAVAILABLE			?		
quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato
4	00:00:56	00:00:14	65	00:05:37	00:00:05	4	00:00:05	00:00:01	10	00:01:24	00:00:08	13			51			33	00:04:04	00:00:07	25	00:01:43	00:00:04
4	00:00:56	00:00:14	65	00:05:37	00:00:05	4	00:00:05	00:00:01	10	00:01:24	00:00:08	13			51			33	00:04:04	00:00:07	25	00:01:43	00:00:04

On the left side of the table, before the grid containing all the data, is the name of the organizational unit with an associated small pie chart representing the call status for that specific unit. Immediately following this is the total amount of calls received and made, before all the detailed data.

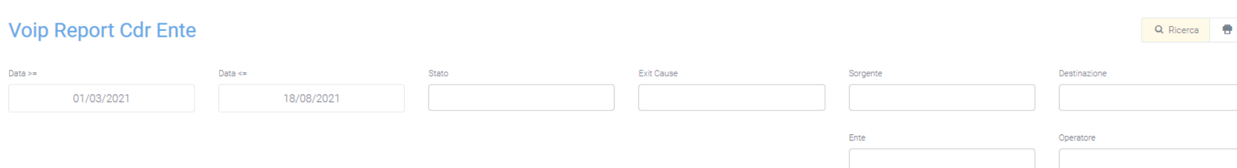
As with any table, a total calculation of each column will be found at the bottom.

As always, it is possible to export the table to excel format via special "Export XLS" button at the top right (of the table).

Filters

As described initially, to enable the filters simply click the "search" button in the upper right corner with which a mask will open in which all the necessary data should be entered:

- start and end date of the time interval in which the data will be shown
- status
- exit causes, i.e. the reasons why a call is terminated
- source
- destination
- organizational unit (more than one may be entered)
- operator



After entering all the desired data, click on "search" to enable the filters, otherwise on "reset" to delete them.

The printer button, to the right of the "search" button, allows you to print the entire page view, including graphs and the table.

Organizational Unit Report by Operator

This section allows you to keep track of call trends, not only broken down by organizational unit, but also by individual operator and their accounts.

Tables

Everything is contained in a table (one for each unit) where there will be information regarding total, answered and unanswered calls broken down by exit causes. For successful calls, the information contained in the exit causes will be:

- the quantity
- the waiting time
- the total and average working and talk time
- the average time for which the phone rang
- the average answer time

Regarding unanswered calls, the information for each exit cause is basically three:

- quantity
- average time worked
- total time worked

Above the table is a grid containing a total of some information, viz:

- the total quantity
- the total time worked
- the total talk time
- the average time worked
- the average time the phone rang
- the average answer time
- the average talk time

Numero chiamate gestite ente: COMMERCIALE

Export XLS

quantità: 242	tempo lavorato: 01:34:20	tempo di conversazione: 01:18:15	media tempo lavorato: 00:00:23	media tempo squillo: 00:00:06	media tempo risposta: 00:00:02	media tempo conversazione: 00:00:19

	TOTALE	RISPOSTE					NON RISPOSTE	NCC-AGENT							NCC-CALLER					
	quantità	quantità	media tempo lavorato	media tempo squillo	media tempo risposta	media tempo conversazione	quantità	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo squillo	media tempo risposta	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	m te s
201 (Damiano Malizia)	242	60	00:01:24	00:00:03	00:00:09	00:01:18	182	6	00:00:35	00:00:05	00:00:35	00:00:05	00:00:03	00:00:03	2	00:01:06	00:00:33	00:01:06	00:00:33	00
###	5	1	00:00:10				4													
201-APP	38	16	00:03:31	00:00:04	00:00:07	00:03:28	22													
201-DECT	76	9	00:00:58	00:00:04	00:00:07	00:00:55	67	2	00:00:04	00:00:02	00:00:04	00:00:02	00:00:02	00:00:03	1	00:00:02	00:00:02	00:00:02	00:00:02	00
201-KCTI	26	9	00:00:33	00:00:03	00:00:15	00:00:21	17													
201-PHONE																				
Damiano_DECT	46	6	00:00:40	00:00:04	00:00:04	00:00:40	40	4	00:00:31	00:00:07	00:00:31	00:00:07	00:00:03	00:00:04	1	00:01:04	00:01:04	00:01:04	00:01:04	00
Damiano_Fisso																				
Damiano_KCTIP	9	5	00:00:31		00:00:02	00:00:29	4													
damiano.malizia_fisso	42	14	00:00:29	00:00:03	00:00:14	00:00:19	28													

Numero chiamate gestite ente: COMMERCIALE

Export XLS

quantità: 242	tempo lavorato: 01:34:20	tempo di conversazione: 01:18:15	media tempo lavorato: 00:00:23	media tempo squillo: 00:00:06	media tempo risposta: 00:00:02	media tempo conversazione: 00:00:19

	NCC							TRANSFER							TRASFERIMENTO					
media tempo risposta	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo squillo	media tempo risposta	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo squillo	media tempo risposta	quantità	tempo lavorato	media tempo lavorato	tempo conversazione	media tempo conversazione	media tempo squillo
00:00:04	50	01:19:43	00:01:35	01:14:06	00:01:28	00:00:03	00:00:10	1	00:02:28	00:02:28	00:02:28	00:02:28	00:00:05	00:00:05	1	00:00:10	00:00:10			
															1	00:00:10	00:00:10			
	16	00:56:29	00:03:31	00:55:42	00:03:28	00:00:04	00:00:07													
00:00:02	6	00:08:38	00:01:26	00:08:13	00:01:22	00:00:05	00:00:09													
	9	00:04:58	00:00:33	00:03:13	00:00:21	00:00:03	00:00:15													
00:00:07								1	00:02:28	00:02:28	00:02:28	00:02:28	00:00:05	00:00:05						
	5	00:02:39	00:00:31	00:02:26	00:00:29		00:00:02													
	14	00:06:59	00:00:29	00:04:32	00:00:19	00:00:03	00:00:14													

Numero chiamate gestite ente: COMMERCIALE

Export XLS

quantità: 242	tempo lavorato: 01:34:20	tempo di conversazione: 01:18:15	media tempo lavorato: 00:00:23	media tempo squillo: 00:00:06	media tempo risposta: 00:00:02	media tempo conversazione: 00:00:19

	ANSWERED ELSEWHERE			BUSY			CANCELED			NOANSWER			REFUSED			TIMEOUT			UNAVAILABLE			?		
media tempo risposta	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato	quantità	tempo lavorato	media tempo lavorato
	21			2	00:00:25	00:00:12	56	00:04:52	00:00:05	6	00:00:47	00:00:07	12			50			29	00:03:47	00:00:07	6	00:00:27	00:00:04
							1												3					
	1						10	00:02:32	00:00:15				2			7			2	00:01:04	00:00:32			
	15						22						2			25			3	00:00:11	00:00:03			
	5			1	00:00:05	00:00:05	2	00:00:15	00:00:07				2			1			6	00:00:05	00:00:00			
							9	00:00:25	00:00:02	6	00:00:47	00:00:07	6			12			1			6	00:00:27	00:00:04
							4	00:00:07	00:00:01															
				1	00:00:20	00:00:20	8	00:01:33	00:00:11							5			14	00:02:27	00:00:10			

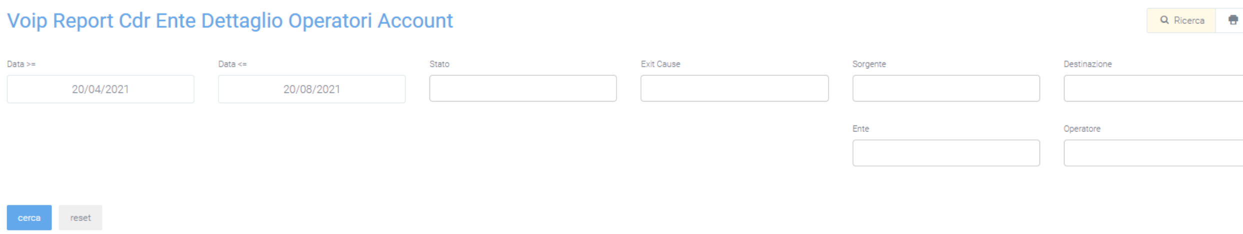
In the leftmost part are the operators (those with the darkest color) and their accounts (below the operator and lighter in color). Next, before going into the details of each exit cause, is a total of calls, both answered and unanswered.

Each table is exportable in excel format, individually via its own "Export XLS" button located at the top right of each.

Filters

To enhance the analysis experience by selecting only certain information to display or specific time periods, filters can be enabled. To do so, click on the "search" button, with which a specially designed mask will open, and enter the desired data from those proposed:

- start and end date of the time interval in which the data will be shown
- call status
- exit causes
- source
- destination
- organizational unit
- operator



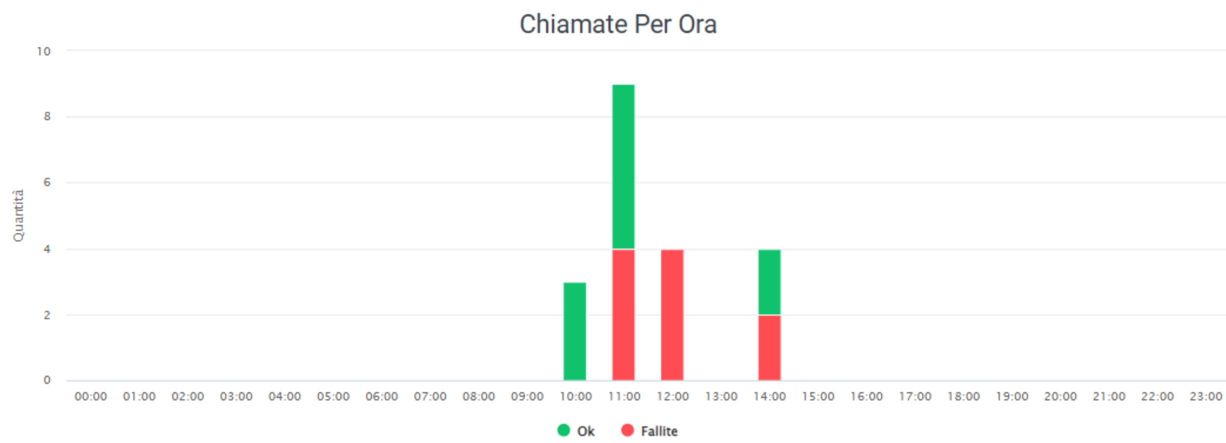
After entering all the desired data, click on "search" to enable the filters, otherwise on "reset" to delete them.

The printer button, to the right of the "search" button, allows you to print the entire page view, including graphs and the table.

Checktime Report

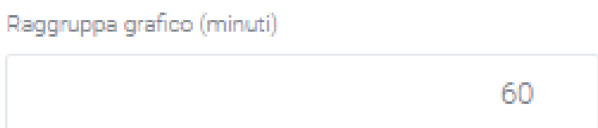
Charts

This report allows you to view, through two graphs, how many calls went through time control (those in green) and how many did not go through because they were out of hours (in red). The first pie chart shows the ratio of calls passed (OK - green) to those not passed (FAILED - red). The second graph, on the other hand, shows, broken down by time slot that can be customized using filters, the amounts of calls passed and not passed by the time control.



In case you want to delve into the detail of every call that did not pass the time control, just enable the option in System Preferences/VoipToCall called "Import failed calls into time control." In this way, all calls that do not pass the time control will end up in Voip to Call, where you can, in addition, observe their detail.

Filter for grouping



In the case just shown (image above), grouping will be done by hours (60 min. for each grouping).

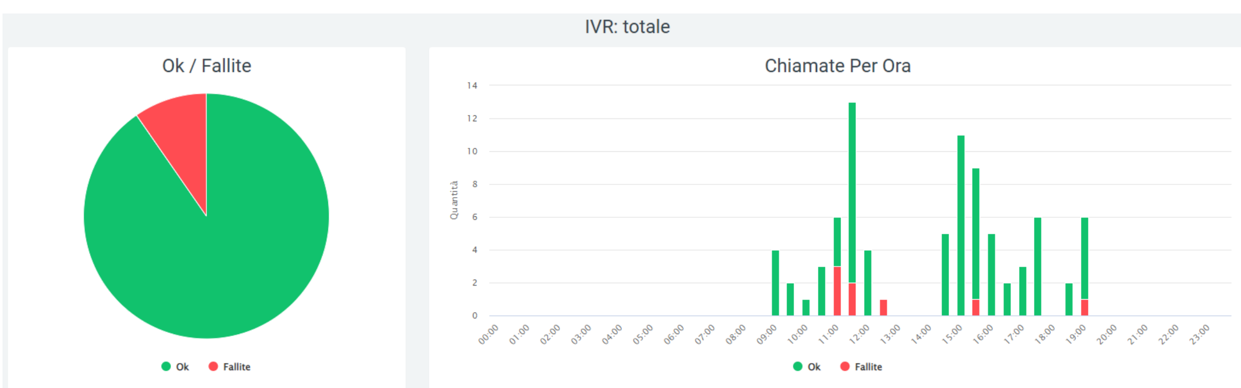
IVR Menu Report

This report allows you to view call failure or success statistics before going to the IVR menu. The division is made by types of IVR menus that you want to analyze.

Charts

The first pie chart shows the ratio of passed (OK - green) to failed (FAILED - red) calls.

The second graph on the other hand shows, divided by time slot that can be customized using filters, the amounts of calls passed and not passed to the IVR menu.



Filters

Clicking on the "search" button at the top right of the section enables filters that include:

- Data range
- IVR
- Group graph (minutes)

Widget

For explanation on creating, organizing and managing Widgets, visit:

Report Provinces

Quest report allows you to view data related to:

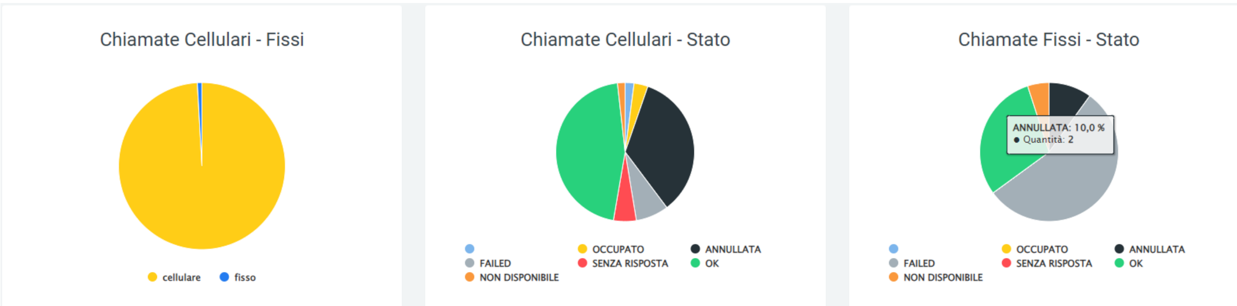
- Phone Calls - Fixed
- Phone Calls - Status
- Fixed Calls - Status

For Phone Calls - Status and Fixed Calls - Status you can view whether the call results:

- Okay
- Canceled
- Unanswered
- Unavailable
- Busy
- Failed
- ??

Charts and Tables

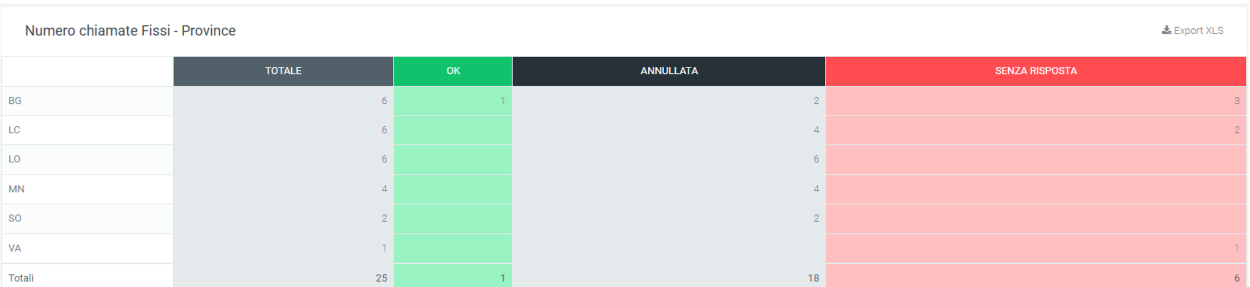
The graphs below show statistics for the above data, and the available tables cover the number of cellular calls-fixed and landlines-provinces.



Numero chiamate Cellulari - Fissi

Export XLS

	TOTALE	OK	ANNULLATA	SENZA RISPOSTA	NON DISPONIBILE	OCCUPATO	FAILED	??
cell	2.251	1.024	775	120	40	73	172	47
fisso	20	6	2		1		11	
Totali	2.271	6	2		1		11	



Numero chiamate Fissi - Province

Export XLS

	TOTALE	OK	ANNULLATA	SENZA RISPOSTA
BG	6	1	2	3
LC	6		4	2
LO	6		6	
MN	4		4	
SO	2		2	
VA	1			1
Totali	25	1	18	6

Each table can be exported to XLS format using the button in the upper right corner on the table itself.

Filters

By clicking on the "search" button at the top right of the section, filters can be enabled that include:

- Date range
- State
- Province

Widget

For explanation on creating, organizing and managing Widgets, visit:

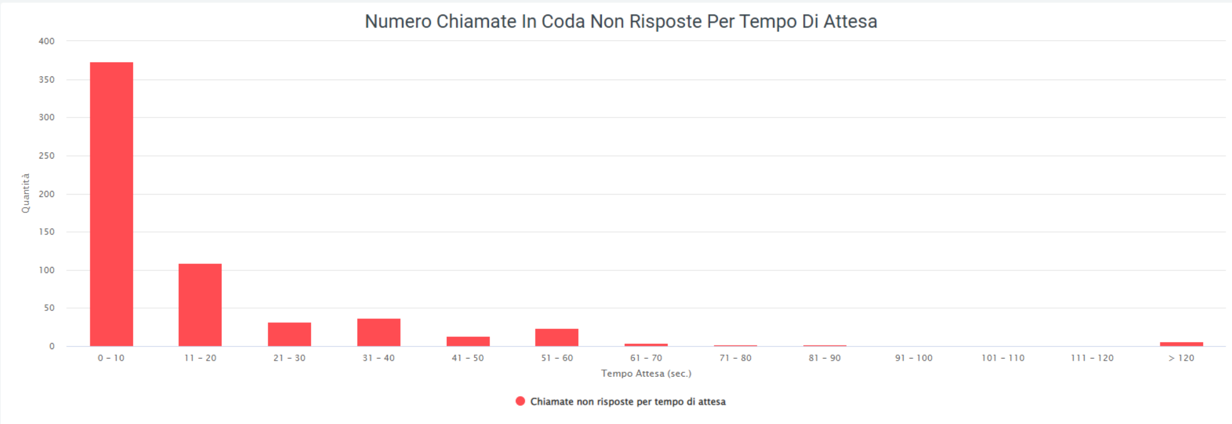
Time Abandonment Queue Report

This report shows the number of unanswered queued calls after waiting time.

Charts and Tables

The graph shows the exact number of unanswered calls per specified queue waiting time before abandoning the call.

The table below shows the above data by call quantity and percentage.



Numero chiamate in coda non risposte per tempo di attesa (secondi)

Export XLS

	0 - 10	11 - 20	21 - 30	31 - 40	41 - 50	51 - 60	61 - 70	71 - 80	81 - 90	91 - 100	101 - 110	111 - 120	> 120	TOTALE
QTA	373	109	32	37	13	24	4	2	2	1	1	1	6	659
PERC	56,60 %	16,54 %	4,86 %	5,61 %	1,97 %	3,64 %	0,61 %	0,30 %	0,30 %	0,15 %	0,15 %	0,15 %	0,91 %	

It can be seen that the wait times in seconds increase as we proceed to the right of the graph, starting from a range of 0 to 10 seconds and reaching a maximum of over 120. The height of each graph represents the number of calls within a range.

Filters

Clicking on the "search" button at the top right of the section enables filters that include:

- Data range
- Queue

Widget

For explanation on creating, organizing and managing Widgets, visit:

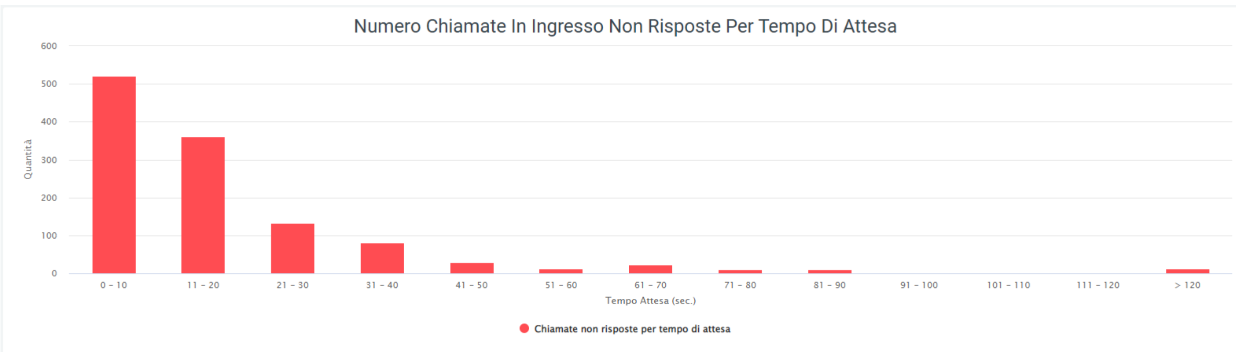
Time Abandonment Entry Report

This report shows the number of unanswered incoming calls after the hold time.

Charts and Tables

The first graph shows the number of unanswered calls by waiting time before abandoning the call.

The table below shows the above data by call quantity and percentage.



Numero chiamate in ingresso non risposte per tempo di attesa (secondi)

Export XLS

	0 - 10	11 - 20	21 - 30	31 - 40	41 - 50	51 - 60	61 - 70	71 - 80	81 - 90	91 - 100	101 - 110	111 - 120	> 120	TOTALE
QTA	521	361	132	82	30	13	23	10	10	1	0	1	12	1196
PERC	43,56 %	30,18 %	11,04 %	6,86 %	2,51 %	1,09 %	1,92 %	0,84 %	0,84 %	0,08 %		0,08 %	1,00 %	

In the graph, it can be seen that the wait times in seconds increase as we proceed to the right of the graph, starting from a range of 0 to 10 seconds and reaching a maximum of more than 120. The height of each graph represents the number of unanswered calls within a range. Scrolling with the mouse over the graphs will show the exact number of unanswered calls.

Filters

By clicking on the "search" button at the top right of the section, filters can be enabled, which includes:

- Date range

Widget

For explanation on creating, organizing and managing Widgets, visit:

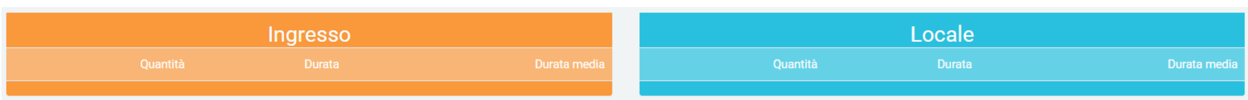
VoIP Call Center report

Dashboard

Service description

By means of graphical and tabular representations, the dashboard section of the "Voip Report Call Center" module allows the user to visualize the trend of incoming calls from queues and local calls within the call center.

Upon opening the dashboard, the first information that can be observed are two tables with the quantity, total duration, and average duration of incoming and local calls.



Charts

The graphs in this section allow an analysis of the trend of calls, investigating how many of these are answered (served) and how many are not (not served). The first four pie charts show, respectively:

- the ratio of served to unserved calls
- the detail of the status of outgoing calls from the queues
- the status of calls served
- the status of unanswered calls

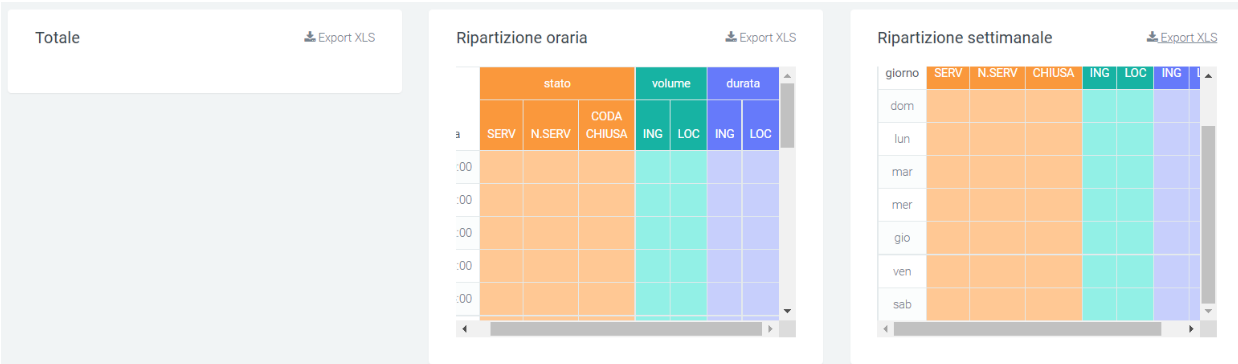
Next, six columnar graphs representing call trends in more detail are visible:

- the first shows the number of incoming and local calls, including the amount of answered and unanswered calls.
- the second shows the total and average duration of incoming and local calls, day by day.
- the third shows the amount of calls broken down by time slot.
- the fourth shows the duration in seconds of calls, broken down by time slot.
- the fifth, like the third, shows the amount of calls, but broken down by day of the week.
- the last graph shows the duration in seconds of calls broken down by day of the week.

For all the graphs described, it is possible to view the details of the data by scrolling with the mouse over the desired section of the graph. It is also possible to remove the display of information via the legend by clicking on the name or on the relevant colored dot.

Tables

At the bottom of the page, after the graphs, there are tables containing more detailed data regarding calls are shown: status, call volume, and duration. There are three tables: one contains day-by-day totals, another breaks down by time slot, and the last table breaks down by day of the week.



Filters

The data represented are included within a time frame indicated by filters. To vary this period, click on the "search" button in the upper right corner and enter the required data, viz:

- date >=, which indicates the starting date of the data that will be extracted from the database
- date <=, which indicates the maximum date of the data that will be extracted from the DB

Widget

For explanation on creating, organizing and managing Widgets, visit:

Report Code

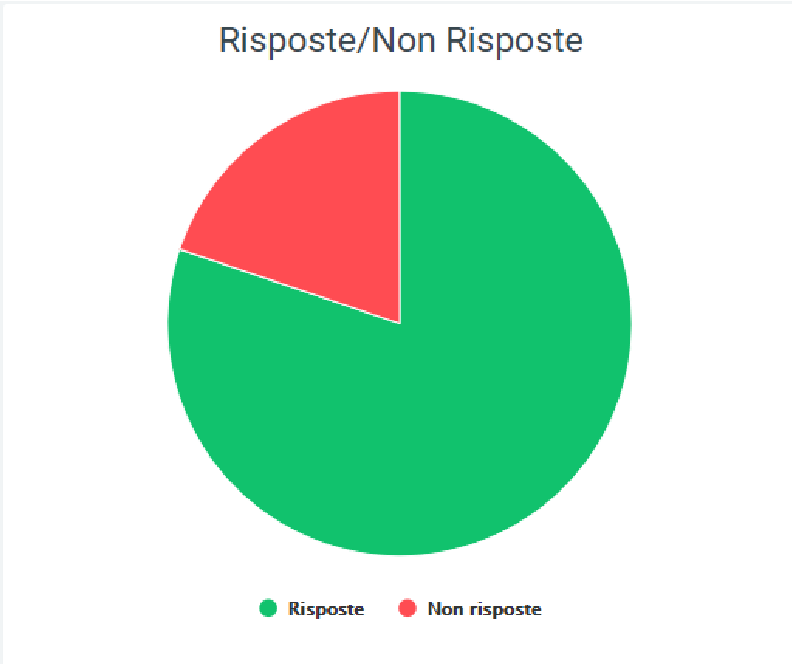
In the "Report Queues" section, under the "Voip report Call Center" form, you can see graphs depicting Call Center call trends and a table broken down by queues showing the detail.

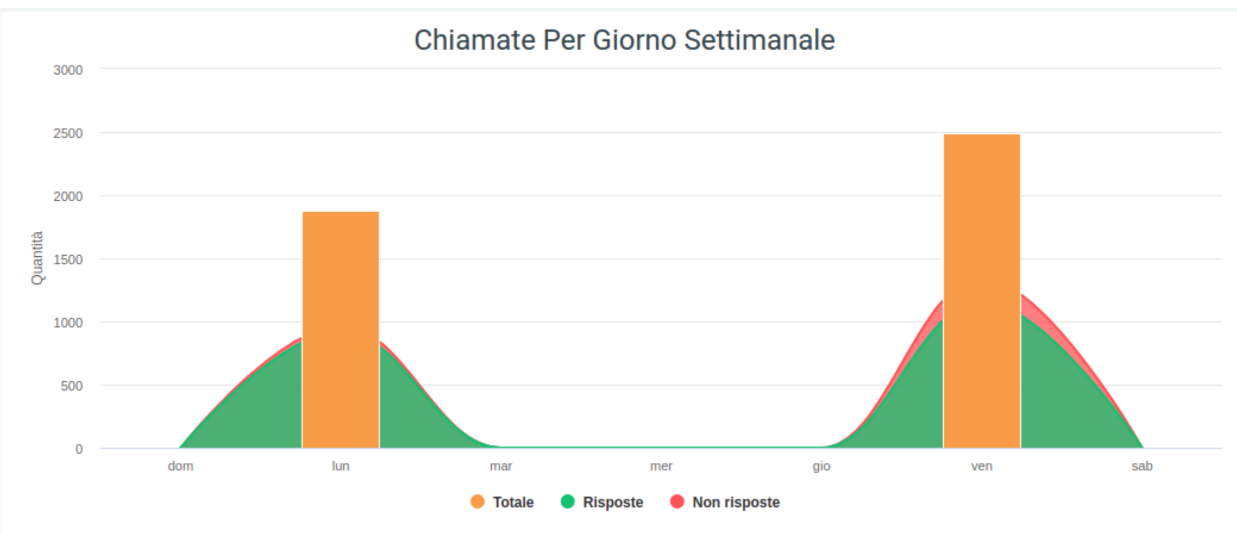
Charts

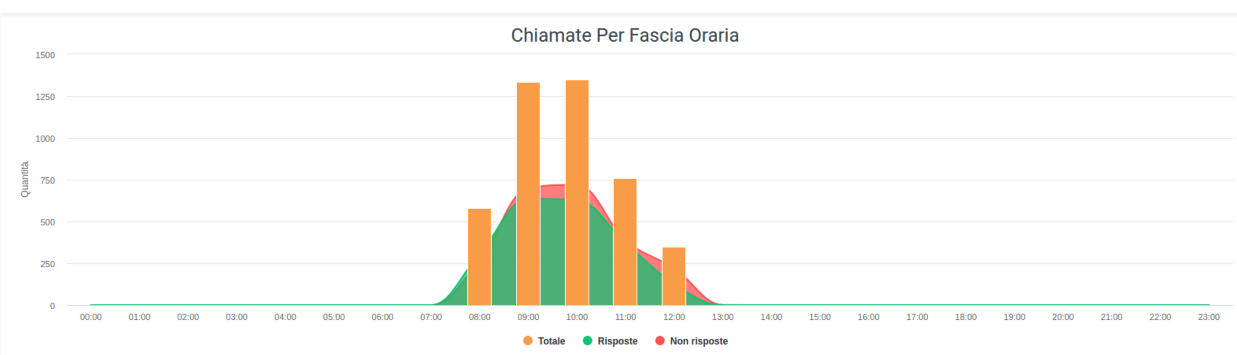
Three types of graphs can be viewed in this report: the first pie chart contains information on the number of answered and unanswered calls. If no queue is entered in the filters, the graph will refer to all queues.

In addition to the pie chart, there are two graphs that contain data on answered, unanswered, and total calls divided by days of the week and by time slot.

This allows for immediate analysis on call trends.


Risposte/Non Risposte
Risposte
Non risposte

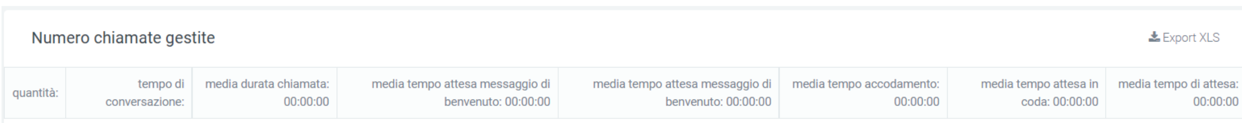

Chiamate Per Giorno Settimanale
Quantità
3000
2500
2000
1500
1000
500
0
dom
lun
mar
mer
gio
ven
sab
Totale
Risposte
Non risposte


Chiamate Per Fascia Oraria
Quantità
1500
1250
1000
750
500
250
0
00:00
01:00
02:00
03:00
04:00
05:00
06:00
07:00
08:00
09:00
10:00
11:00
12:00
13:00
14:00
15:00
16:00
17:00
18:00
19:00
20:00
21:00
22:00
23:00
Totale
Risposte
Non risposte

Tables

Below the graphs shown above, there is a table called "Number of Calls Handled" where all data regarding calls broken down by queue are collected. Before the tabulated data, below the title, there are a series of values showing:

- the total amount of calls
- the total talk time
- the average duration of a call
- the average hold time with the welcome message
- the average queuing time
- the average waiting time in queue
- the average waiting time



Numero chiamate gestite — Export XLS

quantità:	tempo di conversazione:	media durata chiamata: 00:00:00	media tempo attesa messaggio di benvenuto: 00:00:00	media tempo attesa messaggio di benvenuto: 00:00:00	media tempo accodamento: 00:00:00	media tempo attesa in coda: 00:00:00	media tempo di attesa: 00:00:00

It is also possible to export the table to excel format using the appropriate button at the top right of the table, "Export XLS."

In the table, each row belongs to a queue to allow you to compare the same information on different queues. Next to the name of each, there is a small pie chart representing answered and unanswered calls from the relevant queue. After the pie chart, several more detailed pieces of information follow: the total number of calls received by the queue, the number of answered and unanswered calls, the trend of those answered with their times, NCC calls, calls that have been transferred, calls that have been canceled, and calls that have not been answered, all accompanied by qualitative information regarding the time taken.

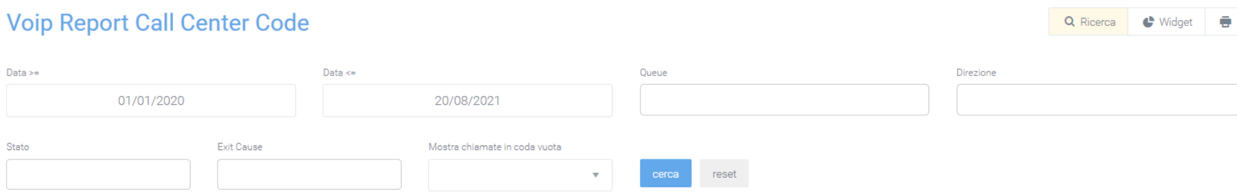
With more detailed data from the Call Center, a new status can be displayed: Cancelled Nowait, which indicates calls that have been terminated by the customer before the welcome message finishes playing.

Finally, at the bottom of the table are the totals for each column.

Filters

There is also a mask that groups filters that can be used to improve the analysis of the desired data. To activate the filters, click on the "search" button in the upper right corner. A mask will open with:

- start and end date of the time interval for which the data will be shown
- queue, i.e., the name of the queue involved
- direction (whether it is an incoming call or a local call)
- status
- exit cause, i.e., how the call ended
- show calls in empty queue: in case some calls were registered with empty queues, with this option it will be possible to display them anyway



Once the desired filters are selected, click “search” to enable them, or “reset” to remove them.

Widget

For explanation on creating, organizing and managing Widgets, visit:

```
.. toctree::
```

Widget.rst

Caller Queue Report

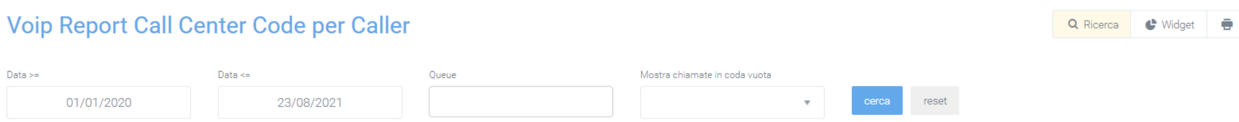
This section shows, in two graphs, the trend of incoming calls in the queues. The first is a pie chart showing the ratio of answered to unanswered calls, while the second is a column chart showing the same states, but with the detail broken down by days included in the time interval entered through filters.

In both you can view the detail of the data by scrolling with the mouse over the desired section of the graph. It is also possible to remove from the information display via the legend, by clicking on the name or the relevant colored dot.

Filters

In order to see only certain information represented in the graphs, filters must be used. To enable them, click on the “search” button in the upper right corner with which the dedicated mask will open in which to enter data, such as:

- start and end date of the time interval for which the data are reported
- name of the queue concerned
- show calls in empty queue: in case some calls have been recorded with empty queues, with this option it will be possible to include them anyway in the graphs



Data Queue Report

This section encapsulates the information found in the "Report Code" section, but breaks it down by day, based on the time frame entered through "start date" and "end date" filters.

Charts

The first graph represents total calls, answered and unanswered, along the time frame of one week. This allows a first general analysis to be performed quickly and easily. You can also select only some of the three pieces of information in the graph by clicking on the name of the one you do not want to display within the legend.

Tables

At the bottom of the section, below the graph, are tables for each queue broken down by date. Each row corresponds to a day within the period entered in the filters and is divided into columns that include all the information needed to perform an accurate analysis of call trends for each specific queue. The information contained within the table includes:

Total quantity	Answered Calls	Not Answered	NCC	Cancelled	Cancelled nowait*	No answered
	Quantity / Average call duration / Average talk time / Average operator response time.		Quantity / Duration / Average call duration / Talk time / Average talk time / Average queuing time / Average operator response time	Quantity / Total wait / Average wait	Quantity / Total wait / Average wait	Quantity / Total wait / Average wait

- Indicates calls that have been put down by the customer before the welcome message ends.

Depending on the queue, there may also be an additional transfer portion with:

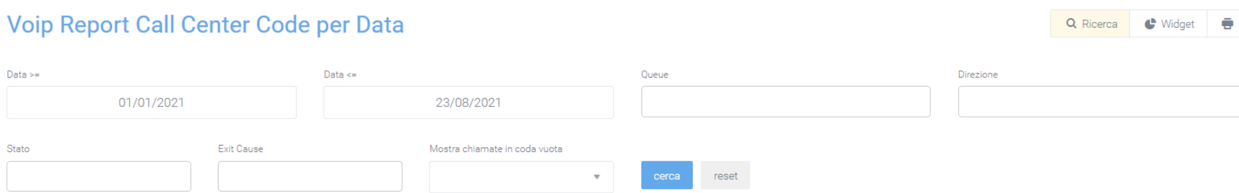
- quantity
- duration
- average call duration
- conversation time
- average conversation time
- average queuing time
- average operator response time

Each table can be exported to an excel sheet via special "Export XLS" button in the upper right corner.

Filters

By clicking on the "search" button at the top right of the section, filters can be enabled that include:

- Start date and end date: to indicate a period in which to analyze the data
- Queue: name of the queue to be displayed
- Direction
- Status
- Exit cause: reason for ending a call
- Display calls in empty queue: if during call registration on the Call Center, some calls are not associated with any queue, they can still be displayed through this specification



By clicking on "search," filters are enabled and a search with them is carried out. In case you want to delete them, just click on the "reset" button.

Operators Queue Report

In this section, within the "Voip report Call Center" module, it is possible to view, in detail, the progress of calls for individual operators, divided into their respective queues. The information that is displayed concerns the number of calls that were answered and the respective outcome, but also how many were not answered and the reasons why. This makes it easier to perform an analysis on each operator accurately.

Tables

All call performance data are grouped in tables divided by queue number, within which are the names of each operator working in that same queue with associated (visible in lighter colors) accounts. The information they contain is:

Total	Responses	Non-Responses	NCC-Caller	Transfer	Answered Elsewhere	Cancelled	Cancelled nowait	Forward	Refused	Timeout
	Quantity / Average time worked / Average talk time / Average ring time / Average answer time	Quantity / Average Time Worked / Average Time Worked / Average Talk Time / Average Ring Time / Average Response Time	Quantity / Time worked / Average time worked / Average time talk / Average time ring / Average time answer	Quantity / Time worked / Average time worked / Average time talk / Average time ring / Average time answer	Quantity / Average ringing time	Quantity / Average ringing time		Quantity / Average ringing time	Quantity / Average ringing time	Quantity / Average ringing time

Each row corresponds to an operator with its corresponding accounts. If a filter were entered on a single operator, only the operator concerned would be displayed.

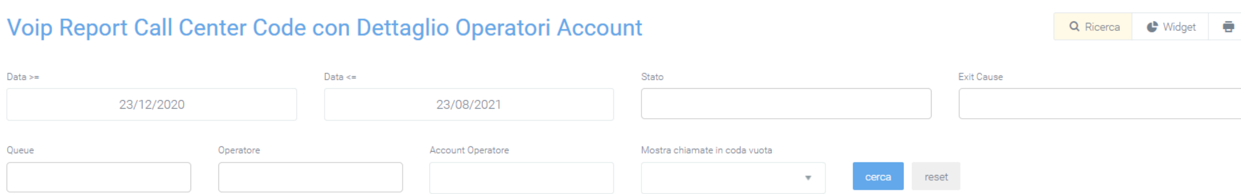
As for the other queues, each has a separate table where the same information will be displayed.

In each table, there is an "Export XLS" button in the upper right corner to export the tables to excel (XLS) format.

Filters

To make data analysis more effective, it is possible to use filters that will make it possible to extract, from a search, only the necessary data. To activate them, simply click on the "search" button at the top right and a mask will open with:

- Start date and end date: to indicate a period in which to analyze the data
- Status: whether the call was successful or not
- Exit cause: reason for the termination of a call
- Queue: name of the queue
- Operator
- Operator account
- Display calls in empty queue: if during call registration on the Call Center, some calls are not associated with any queue, they can still be displayed through this specification



By clicking on "search," filters are enabled and a search with them is carried out. In case you want to delete them, just click on the "reset" button.

Report Code Now

Tables

This section contains only the tables divided by time slots, which contain all the information necessary for the analysis of calls in individual queues. The information contained in the tables includes:

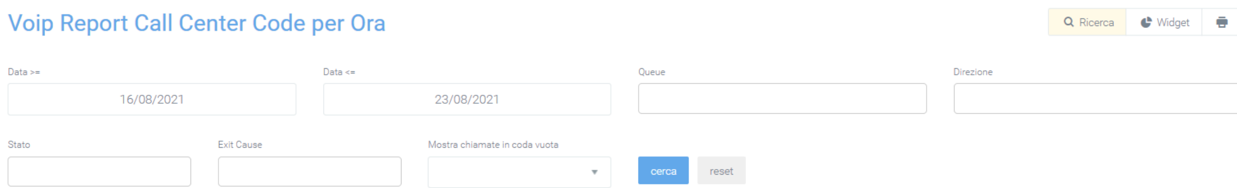
The information contained in the tables includes:

Total quantity	Calls Answered	Not Answered	NCC-Caller	Cancelled	Cancelled nowait	No answered	Transfer
	Quantity / Average call duration / Average talk time / Average operator response time.		Quantity / Duration / Average call duration / Talk time / Average talk time / Average queuing time / Average operator response time	Quantity / Total wait / Average wait	Quantity / Total wait / Average wait	Quantity / Total wait / Average wait	Quantity / Duration / Average call duration / Talk time / Average talk time / Average queuing time / Average operator response time

Filters

By clicking on the "search" button at the top right of the section, filters can be enabled that include:

- Start date and end date: to indicate a period in which to analyze the data
- Queue: name of the queue to be displayed
- Direction
- Status
- Exit cause: reason for ending a call
- Display calls in empty queue: if during call registration on the Call Center, some calls are not associated with any queue, they can still be displayed through this specification



By clicking on "search," filters are enabled and a search with them is carried out. In case you want to delete them, just click on the "reset" button.

Widget

For an explanation of creating, organizing and managing widgets, visit:

Operators Report

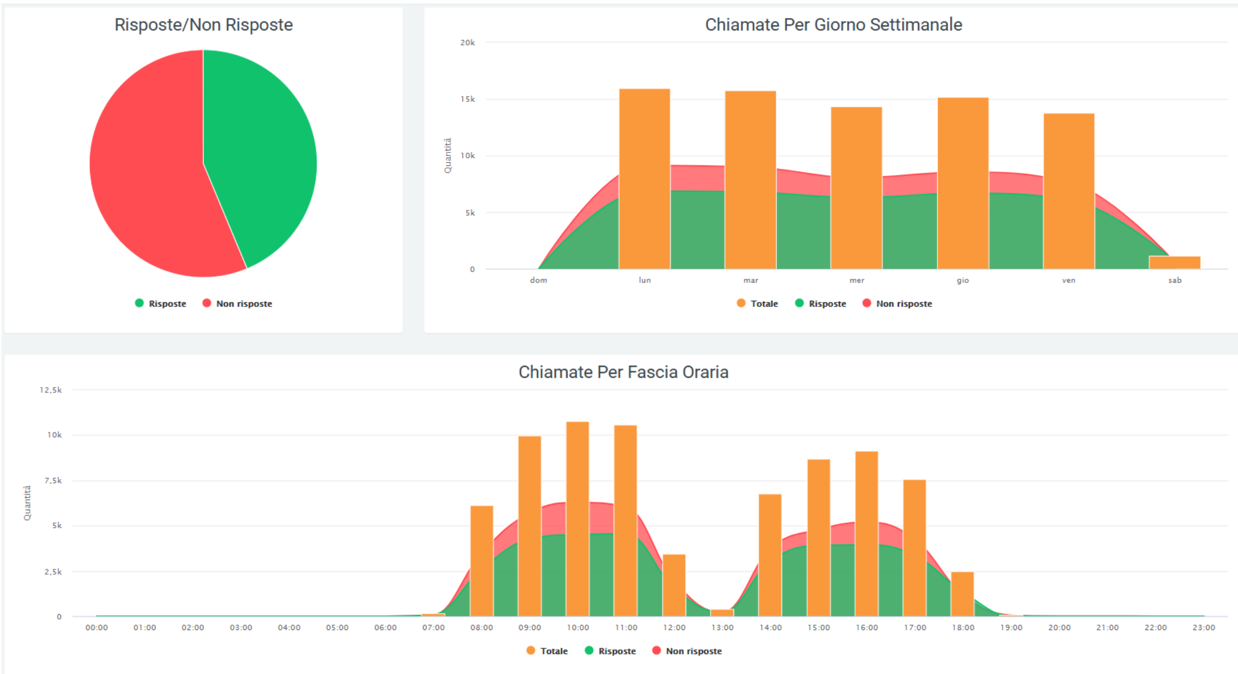
This section reports call trends for each individual operator.

Charts

The total trend of answered and unanswered calls is graphically represented in a pie chart where, by scrolling over it with the mouse, the quantities can be observed in detail.

It is possible to remove a data item from the visualization via the legend by clicking above the name or colored dot. To put it back into the visualization simply click on it again.

These two functions, detail and visibility, are also available in the same way on the other two graphs. Graphs that represent one the total call trends by breakdown in days of the week, the other by time slot. These two, in addition to the number of answered and unanswered calls, also show the total number of calls, as can be seen in the following picture.



Tables

After a graphical display of call trends, this data will be reported in a table broken down by operators, each with their own detailed information. Before reporting the data for each operator, the total data for each field can be observed:

- the quantity of calls
- the total time
- the talk time
- average total time
- average ringing time
- average answer time
- average conversation time

Numero chiamate gestite — Export XLS

quantità:	tempo totale:	tempo di conversazione:	media tempo totale: 00:00:00	media tempo squillo: 00:00:00	media tempo risposta operatore: 00:00:00	media tempo conversazione: 00:00:00

After this first part, the table shows for each row the respective operator with the exact information. On the left is the name of the operator, followed by the total amount of calls handled by it. Next, information on answered and unanswered calls is shown before going into the details of each of these macro-groups.

- **Answers:**
  - NCC-AGENT
  - NCC-CALLER
  - NCC
  - ANSWER

  - TRANSFER
  - TRANSFER
- **Unanswered:**
  - ANSWERED ELSEWHERE
  - BUSY
  - CANCELLED
  - CFWD
  - CONGESTION.
  - FORWARD
  - NOANSWER
  - PICKUP
  - REFUSED
  - TIMEOUT
  - UNAVIABLE

Each of these subgroups will have a further subdivision into details. For answered calls the details present will be:

- quantity
- time worked
- average time worked
- talk time
- average ring time
- average answer time
- average talk time

For unanswered calls instead:

- quantity
- time worked
- average time worked

At the end of the table, at the bottom, the totals of each detail column will be shown.

Using the appropriate button located at the top right of the table, it will be possible to export it to excel "XLS" format.

Filtri

Utilizzando dei filtri, sarà possibile migliorare l'attività di analisi dati focalizzandosi solo su quelli interessati. Per poterli ottenere sarà sufficiente cliccare sul taso "ricerca" in alto a destra con il quale si aprirà una maschera contenente i filtri interessati. Tramite questa inserire i dati negli apposito box:

- data iniziale e finale: verranno compresi solamente i dati presenti in questo lasso di tempo
- stato: dove dovrà essere indicato lo stato delle chiamate da analizzare
- exit cause: il motivo per cui la chiamata è terminata

- operatore: questo per fare in modo di avere un report, anche grafico, esclusivo per l'operatore (o più di uno) selezionato.
- queue: nome dalla coda da visualizzare

By clicking on "search," filters are enabled and a search with them is carried out. In case you want to delete them, just click on the "reset" button.

Widget

For an explanation of creating, organizing and managing widgets, visit:

5.1.2 Missed Calls

With the Missed Calls module, the management of calls that have not been answered is made easy to use for any operator. Indeed, it will be possible to book and make callbacks to initially unserved customers through a few clicks, with methods that prevent multiple operators from calling back the same customer.

The module is designed not only for operators, but also for their administrators. In fact, reports are reserved for them to monitor the way various requests are handled and pages that allow them to force close callback requests or manually assign them to specific operators.

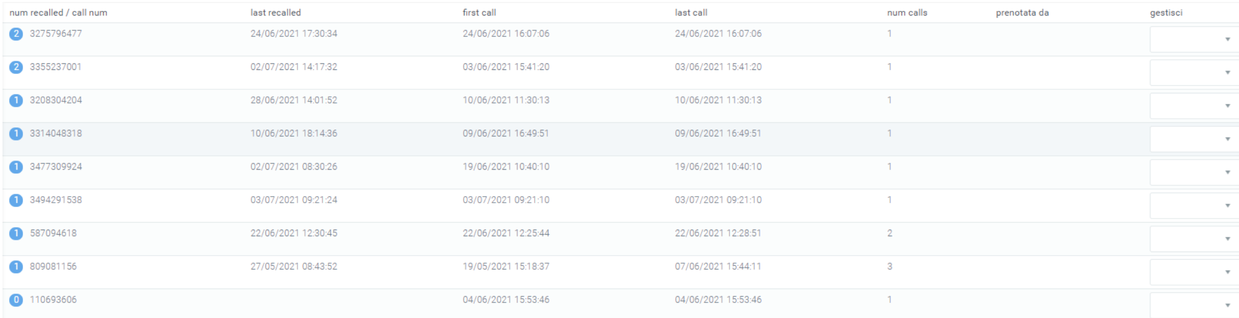
The purpose of this module is the refinement of the relationship between company and customer through the delivery of a markedly improved service.

The module includes:

Assign Call

Within the "Missed Calls" module is "Assign Call," which is a section generally enabled for managers only. It allows you to view all phone numbers related to customers still under management (missed callback) and to assign or unassign callbacks to the various operators. In the mask each record displayed is complete with

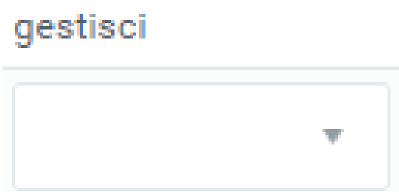
- num recalled: number of times it has been called back
- call num: caller number
- last recalled: date of the last time the number was recalled
- first call: date of the first call
- last call: date of the last call
- num calls: number of times called
- booked by: operator who booked the callback



num recalled / call num	last recalled	first call	last call	num calls	prenotata da	gestisci
2 3275796477	24/06/2021 17:30:34	24/06/2021 16:07:06	24/06/2021 16:07:06	1		
2 3355237001	02/07/2021 14:17:32	03/06/2021 15:41:20	03/06/2021 15:41:20	1		
1 3208304204	28/06/2021 14:01:52	10/06/2021 11:30:13	10/06/2021 11:30:13	1		
1 3314048318	10/06/2021 18:14:36	09/06/2021 16:49:51	09/06/2021 16:49:51	1		
1 3477309924	02/07/2021 08:30:26	19/06/2021 10:40:10	19/06/2021 10:40:10	1		
1 3494291538	03/07/2021 09:21:24	03/07/2021 09:21:10	03/07/2021 09:21:10	1		
1 587094618	22/06/2021 12:30:45	22/06/2021 12:25:44	22/06/2021 12:28:51	2		
1 809081156	27/05/2021 08:43:52	19/05/2021 15:18:37	07/06/2021 15:44:11	3		
0 110693606		04/06/2021 15:53:46	04/06/2021 15:53:46	1		

Management

The management part is generally the responsibility of the managers, who will be empowered to manually assign the operator to perform the callback. To do this, simply click on the mask under "management" and select the operator to be assigned the job.

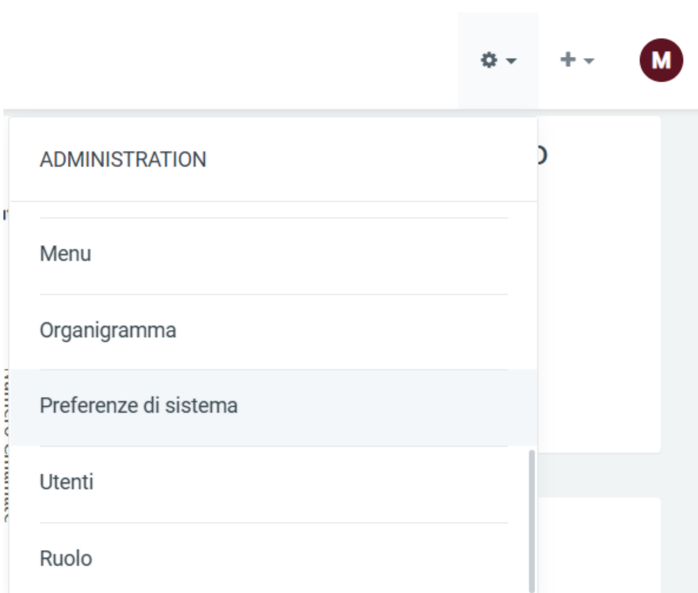


If an operator had already booked to call back, there is an option to cancel the booking by clicking on the button to the right of the name, under manage.

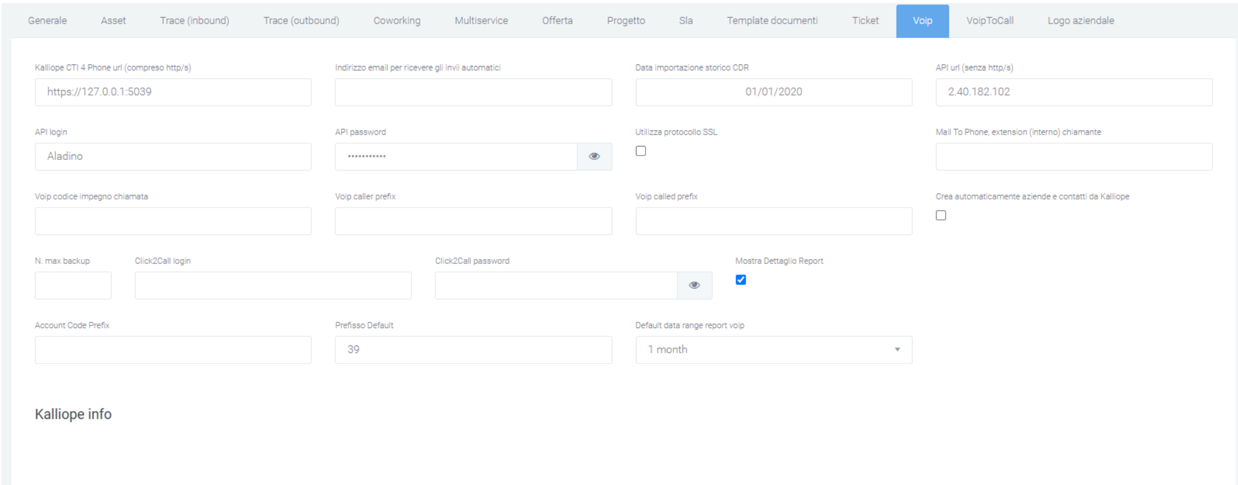
The users that can be selected in the management part are exclusively those users who in their profile have the Voip Extension extension (entered during the creation of the user's profile in the "Voip" tab) configured with the same code present in the kalliope central unit.

It is also possible to enable management of only certain users among those correctly configured. To do this simply:

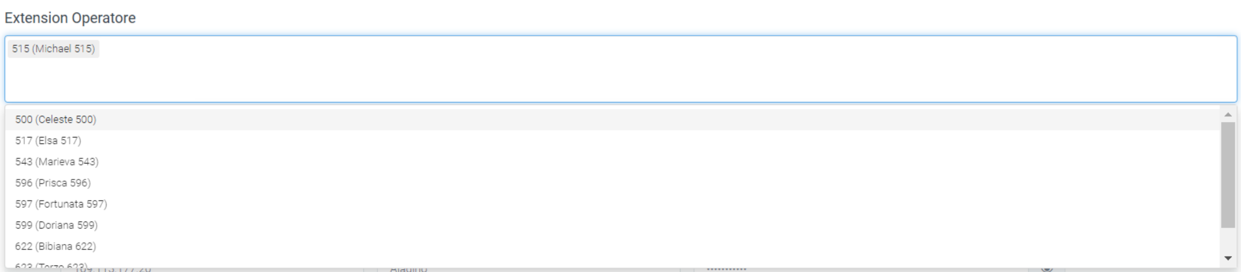
- click on the cogwheel at the top right of the entire cogwheel page.
- click on "system preferences"



- select the "Voip" tab
- scroll down to "Operator Extension" at the bottom of the page



- Select only those operators who will be considered for management

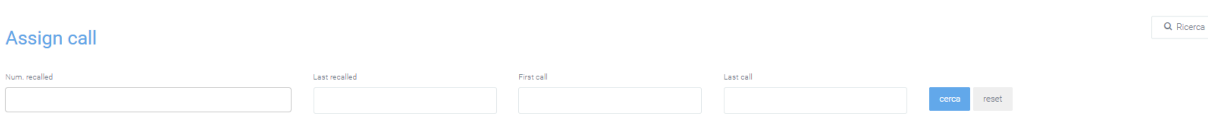


In case this last field is left blank, all correctly configured users would be included.

Filters

The page is complete with a filter that allows searching by:

- Num. recalled
- Last recalled
- First call
- Last call



Dashboard

The data collected in the dashboard are represented through graphs and tables that provide a clear and comprehensive view of the management of call requests.

Tables

A first table shows the calls:

- Totals
- Served inbound
- Handled with callback
- Forced out
- Unanswered callbacks
- Not managed

Gestione richieste: totale

totali	servite in ingresso		gestite con richiamata		chiusura forzata		richiamate senza risposta		non gestite	
294	187	63,61%	42	14,29%	62	21,09%	3	1,02%		

The daily detail of requests is then provided through a table:

- Totals
- Served inbound
- Managed with callback
- Forced closure
- Callbacks without response
- Unmanaged

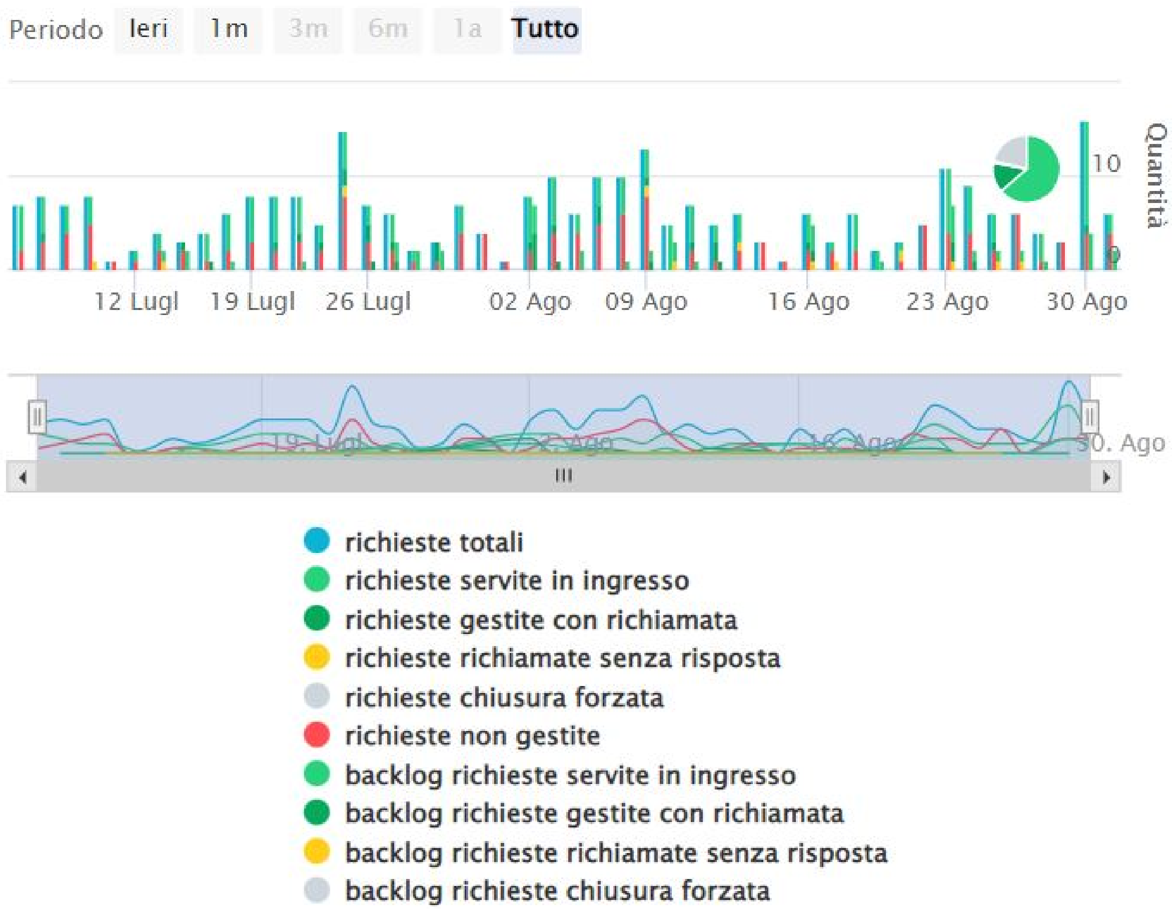
And of daily requests on backlog:

- Served incoming
- Handled with callback
- Forced closure
- Unanswered callbacks

richieste	06/07/2021		07/07/2021		08/07/2021		09/07/2021		10/0	
totali	7		8		7		8			
servite in ingresso	5	71,43%	4	50,00%	3	42,86%	3	37,50%		
gestite con richiamata			1	12,50%						
chiusura forzata										
richiamate senza risposta										
non gestite	2	28,57%	3	37,50%	4	57,14%	5	62,50%	1	1
richieste su backlog	**06/07/2021**		**07/07/2021**		**08/07/2021**		**09/07/2021**		**10/0**	
servite in ingresso										
gestite con richiamata										
chiusura forzata										
richiamate senza risposta							1			

Charts

The daily detail is also represented through a graph showing the trend of requests as follows:



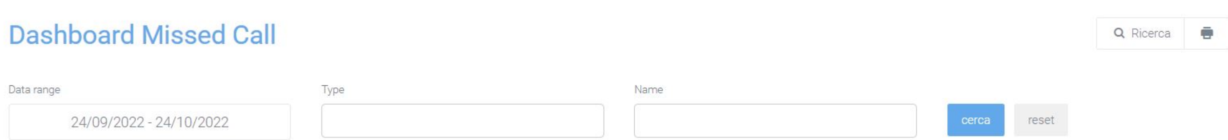
By running the mouse over the graph you can select specific parts to analyze them in more detail. You can always remove data from the display by clicking on the name or colored dot, in the legend.

Filters

This facade can be modified by activating filters by clicking on the "Search" button at the top right, with which you can select:

- Date range: time range you want to specify
- Type: type of request
- Name: specific name of the request

(with the print button you can print all the information on the page such as graphs and tables)



Once the filter fields are filled in as desired, press on the "search" button in light blue to save and display the screen with the activated filters, otherwise press on "reset."

Widget

For explanation on creating, organizing and managing Widgets, visit:

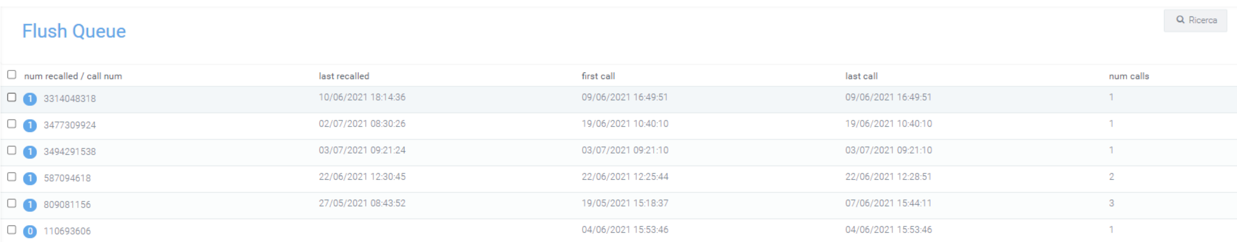
Flush Queue

This section is generally enabled for managers only.

It allows you to view all phone numbers related to customers still being handled (unsuccessful callback), with the ability to delete them by selecting them and clicking the “flush queue” button at the bottom. In this way the numbers will no longer be displayed among the Missed Calls either.

In the mask each record displayed is complete with:

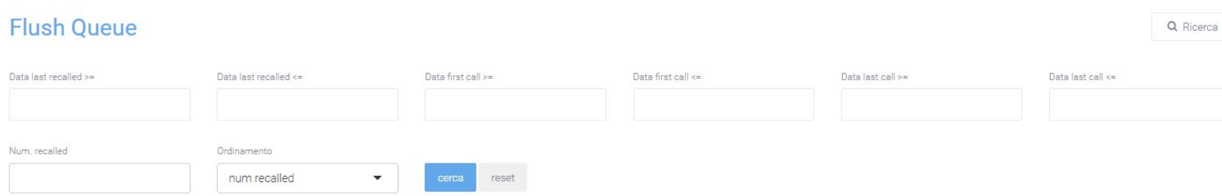
- num recalled: number of times it has been recalled
- call num: caller number
- last recalled: date of the last time the number was recalled
- first call: date of the first call
- last call: date of the last call
- num calls: number of times called



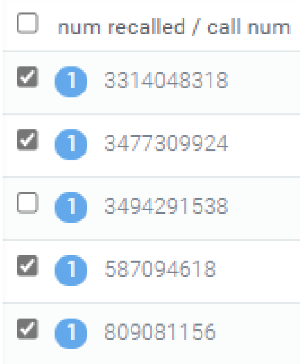
Filters

The mask is complete with a filter that allows searching by :

- Date last recalled >= / <=
- Date first call >= / <=
- Date last call >= / <=
- Num recalled
- Sorting: order by num recalled, last recalled, first call, last call, num calls



In the Mask you can select the various phone numbers for which you were unable to contact the customer. By means of specific button located at the bottom of the list, under “recalled/call num”, you can delete them in mass mode from the mask “Numbers to call back”. This eliminates the possibility for operators to be able to book them for the next callback.



Missed Calls

Within the “Missed Calls” module we find the section of the same name where in one mask all phone numbers that need to be called back are displayed, while in the other all calls booked by the user viewing the page are visible.

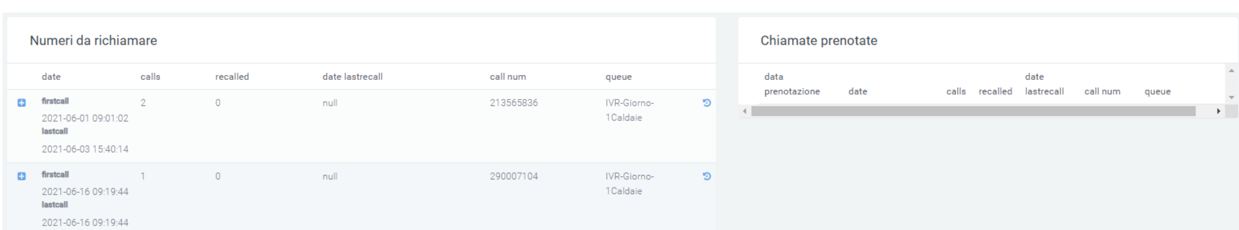
Structure

The mask is divided into 2 sections:

1. **“Numbers to call back”**: contains the list of all numbers that need to be called complete with:

- firstcall date: date of the first call
- date lastcall: date of the last call
- calls: number of calls made
- recalled: number of times it has been recalled
- date lastrecall: date of last callback
- call num: number to be recalled
- queue: name of the queue the customer was in when not answered

An operator, using the button with the “+” symbol in green, can book a specific call, i.e., the callback activity of that specific number. This action will move the number from the “Numbers to Call Back” section to the “Booked Calls” section. Numbers in the “Booked Calls” section will be visible only to the operator who booked the call.



2. **“Booked calls”**: contains the list of all numbers booked by the specific operator logged into the system, complete with:

- reservation dates: date when the reservation was made by the operator
- firstcall dates: date of the first call
- lastcall dates: date of the last call
- calls: number of calls made

- recalled: number of times it was recalled
- date lastrecall: date of last callback
- call num: number to be recalled
- queue: name of the queue the customer was in when not answered

The call center operator who booked the call can immediately make the call by simply clicking on the number to be called (call num). This function is known as Click2Call (C2C). Using the specific button with a "-" symbol in red instead, the operator has the option to cancel the call reservation and then reissue it in visibility in the "Numbers to call back" section available to all Call Center operators.

It is possible to view the details of all calls received and made to and from that specific customer. In the "Booked Calls" section there is the above-mentioned C2C feature that allows the operator to dial the call by pressing on the phone number in the record. A specific colored indicator will recommend the callback if the current time is within the range of the period to be called back, set in the system parameters. It is also possible to activate a specific LINK for opening by parameter preconfigured WEB pages (e.g., CRM page with Caller Number Parameter).

At the moment when the operator initiates the call, the system hides the record and allows it to be re-displayed only if the call was unsuccessful to the customer. In the event that another operator–not the one booked–makes the call and it is not answered, the record will still be hidden. A call is considered "successful" when it is answered by the customer and has a minimum talk time of more than 5 seconds.

To properly import overflow calls into the missed calls section, you must first configure dynamic routing in the Kalliope central office. When creating this you will be prompted for parameters, including the url that will call the central, and the xml file. In the link enter the following:

```
https://nome_azienda.kalliopenexus.cloud/voipToCall/kalliopeToCall/insert/
```

(replace "nome_azienda.kalliopenexus" with the link of the Kalliope Nexus in your possession).

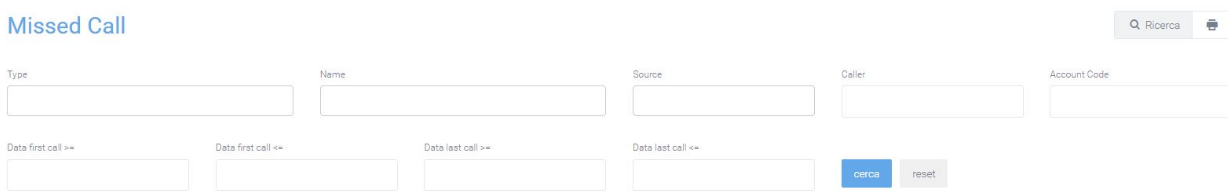
As an XML file instead, enter the following dynaminco routing on Kalliope to retrieve and import the call into voip to call:

```
<?xml version="1.0"?>
<parameters>
<unique_id>%UNIQUE_ID%</unique_id>
<caller>%CALLER_NUM%</caller>
<called>%DNID%</called>
<servicecode>%PARAM1%</servicecode>
<param1>NOME_DELLA_CODA</param1>
<param2>queue</param2>
<param3>%PARAM3%</param3>
<param4>%PARAM4%</param4>
<param5>%PARAM5%</param5>
</parameters>
```

Filters

By pressing the "Search" button in the upper right corner, you can add a filter on the queues, so that only missed calls made to a particular queue are displayed. You can filter by:

- Type: Type of call
- Name: Name
- Source: Source of the call
- Caller: Caller
- Account Code: Account Code
- Date first call >= / <=: Date of the first call
- Date last call >= / <=: Date of last call

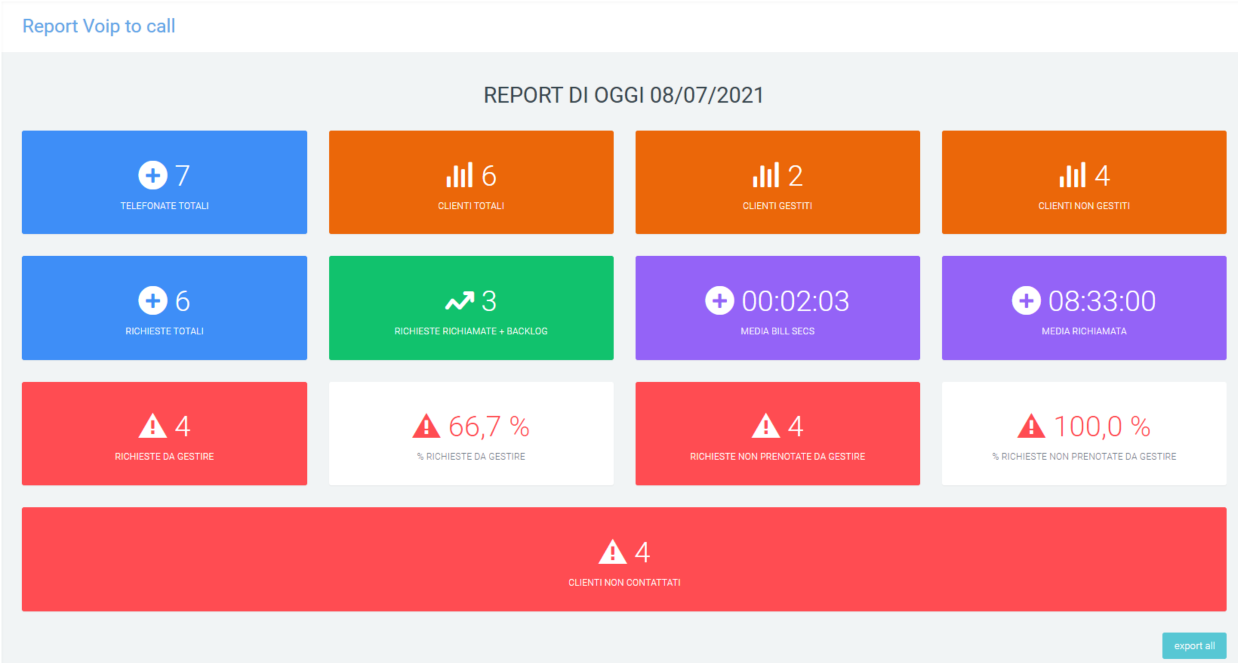


Once the data entry is finished, click on the "search" button to enable the desired filters, otherwise click "reset" to delete them.

Report

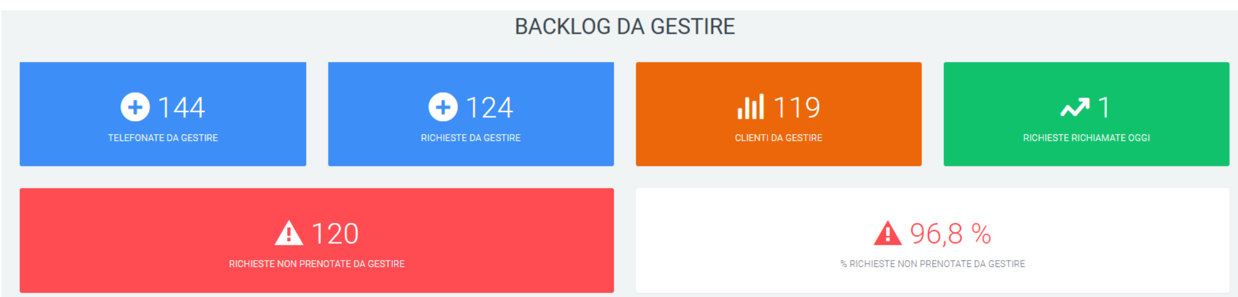
The "Report" section consists of the dashboard made available to Call Center managers who can have all the parameters available in real time to monitor Call Center activity. The data are divided into the two sections Today's Report and Backlog to Manage:

- Today's Report:
  - Total phone calls
  - Total customers
  - Managed customers
  - Unmanaged customers
  - Total Inquiries
  - Inquiries called back + backlog
  - Average Bill Secs
  - Average recalled
  - Requests to be managed
  - Percentage of requests to be handled
  - Unbooked requests to be handled
  - Percentage of unbooked requests to be handled
  - Customers not contacted



The "export all" button in the lower right corner allows the data on the screen to be exported to excel format, even as a history.

- Backlog to manage:
  - Phone calls to manage
  - Requests to be managed
  - Customers to manage
  - Inquiries called today
  - Unbooked requests to be managed
  - Percentage of unbooked inquiries to be handled

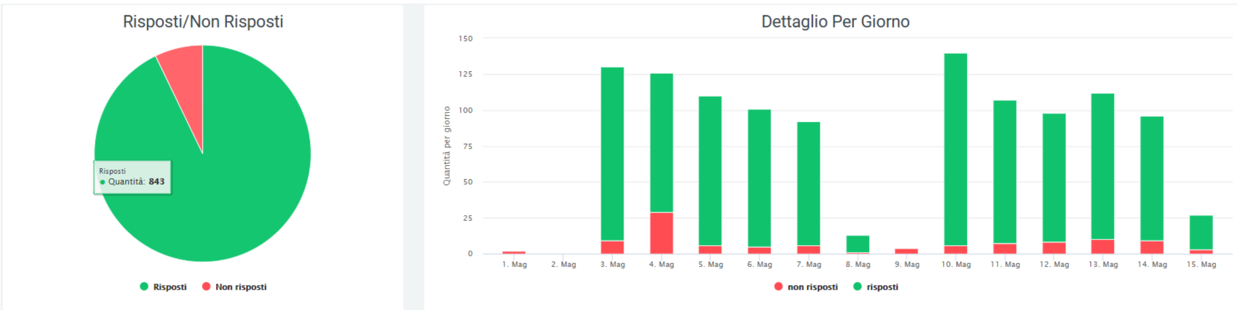


Report Caller Queue

Charts

In this section it is possible to observe a graphical representation, first by pie chart and then by histogram, of the general trend of calls over the selected time period and in the specified queues. The histogram presents a daily breakdown for representation.

As in the other graphs on this platform, the quantity of both answered and unanswered calls can be observed by simply scrolling with the mouse over the relevant graph. It is always possible to remove data from the display by clicking on the name or colored dot, in the legend.



Tables

Underneath the graphs are tables for each queue, containing more specific information regarding call trends with callers, i.e., customers who called. The information in each table is broken down by date and includes:

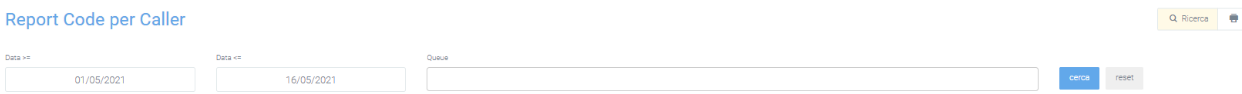
- number of answered calls
- number of unanswered calls
- the total numbers who called
- the total numbers that were answered
- the total numbers that were not answered
- the amount of numbers that were called back and answered
- the amount of numbers that were called back but did not answer
- the amount of numbers that were not called back

IVR-Giorno-1Caldaie	01/05/2021	02/05/2021	03/05/2021	04/05/2021	05/05/2021	06/05/2021	07/05/2021	08/05/2021	09/05/2021	10/05/2021	11/05/2021	12/05/2021	13/05/2021	14/05/2021	15/05/2021	16/05/2021
tel risposte			94	69	82	64	63	10		106	75	78	73	68	18	
tel non risposte	3		3	11	3	3	8	1	1	4	3	1	3		1	
numeri totali	1		84	71	75	65	63	11	1	97	69	69	72	59	16	
numeri risposti			83	63	72	64	59	10		95	67	68	69	59	15	
numeri non risposti	1		1	8	3	1	5	1	1	3	3	1	3		1	
richiamati con successo								1		1						
richiamati senza successo																
non richiamati	1		1	8	3	1	5	0	1	2	3	1	3		1	

Filters

To speed up the search for certain information in precise time periods, filters are used. These can be found within a visible mask by pressing on the "search" button at the top right of the page. The filters that can be used are:

- start and end date of a time interval for displaying data (shows data in the database from one day to another)
- queue, which is the ability to enter one or more queues that you want to display



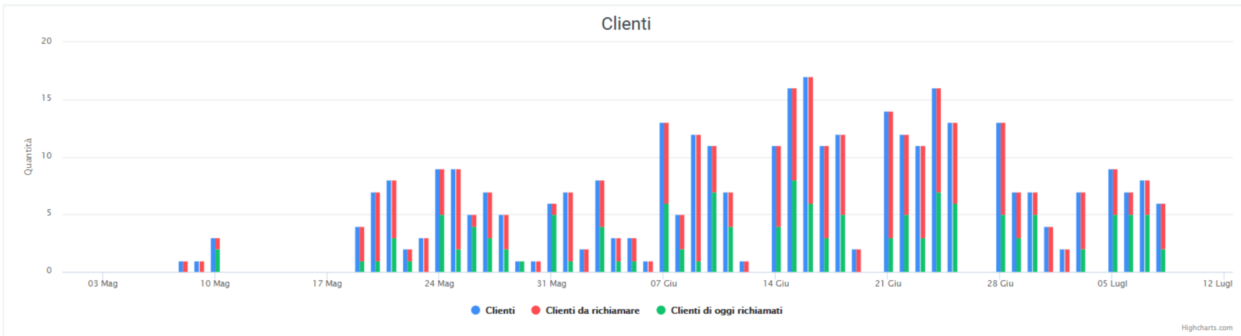
Once all the data have been entered, click on the "search" button to enable the filters entered, otherwise click on "reset" to delete them.

To print the page view, click on the printer-shaped button in the upper right corner, next to the "search" button.

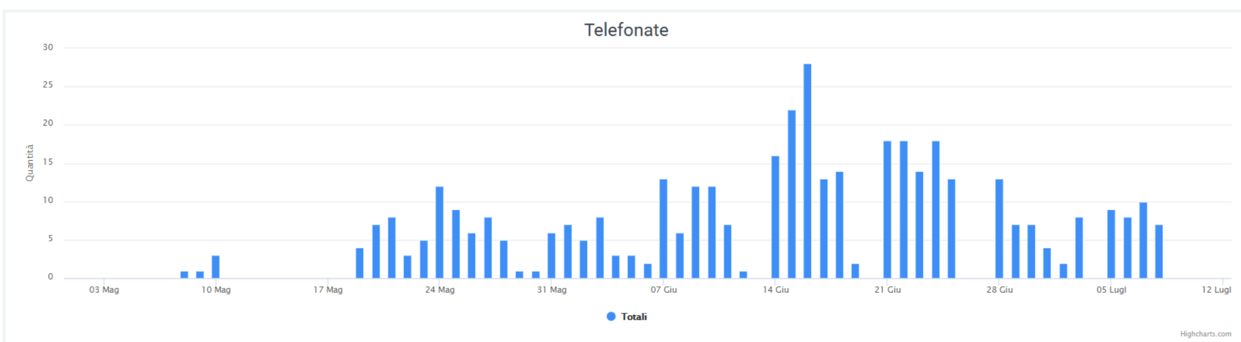
History Report

Two graphs can be viewed in this section of the "Missed Calls" form.

The first graph shows the number of customers who, on a given day, called but were not answered and therefore ended up in Voip to Call. Of this quantity, you can see in green the number of customers who have been called back and in red those who have not yet been called back (and are therefore still visible in Voip to Call).



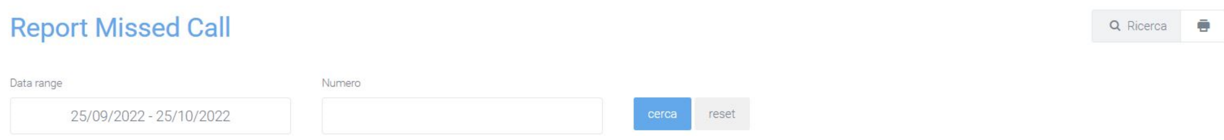
The second graph, on the other hand, reports only the number of calls that: arrived at the exchange, were not answered, and thus ended up in Voip to Call (whether called back or not).



Filters

There is, by pressing on the "search" button in the upper right corner, a filter mask with:

- time range: start and end date in which to represent the data
- number of calling customers



5.1.3 Ticket

Description

Features

The module presents the following functionalities:

- **Customer master data management**: The platform allows you to manage your customer master data and automatically associate each new ticket with the corresponding customer.
- **Type classification management**: Ticketing allows you to classify ticket types for proper display of activities with a type-code_operator logic.
- **Custom field creation**: The platform allows you to independently create new fields to be able to record all the information to improve and speed up the service and support service.
- **Ticket priority management with identifying colors**: Ticketing allows you to manage the priority of tickets with a description and an identifying color for each priority level.
- **Asset management**: In tickets it is possible to enter the Asset for which the request was opened.
- **Contract management**: It is possible to associate the specific contract active for the customer who opened the request with the ticket, monitoring associated SLAs and service delivery mode (e.g., prepaid).
- **Activity planning and deadline tracking**: All activities can be planned and scheduled in the platform's internal calendar, allowing you to monitor scheduled deadlines in real time.
- **Escalation management**: The platform can notify you via e-mail of any escalations related to contractual SLA deadlines.
- **Tracking time recording**: Each operator can record the time he/she has dedicated to the activity allowing at the end of the period to report for each customer the activities performed with the related dedicated time.
- **Automatic ticket mask opening management from caller number**: The platform, thanks to integration with the IPPBX system, facilitates ticket opening by presenting the operator receiving the phone call with the mask pre-filled with the customer's data.
- **Automatic opening management from email**: The platform allows the reading of emails, received on specific system-configured addresses, and the subsequent automatic creation of tickets. The system manages responses entered either in the ticket by sending notifications via email, or from email by entering responses in the ticket.
- **Click to Call Management**: The platform, through integration with the IPPBX system, facilitates calls to customers by taking advantage of click to call features.

- **Attachment Management**: The management of attachments within the platform is extremely intuitive and functional to the activity of processing customer requests. Attachment management is useful for exchanging documents with customers and with users outside the system. Customers can attach one or more files to the email with which they open or respond to the ticket.

Ticket List

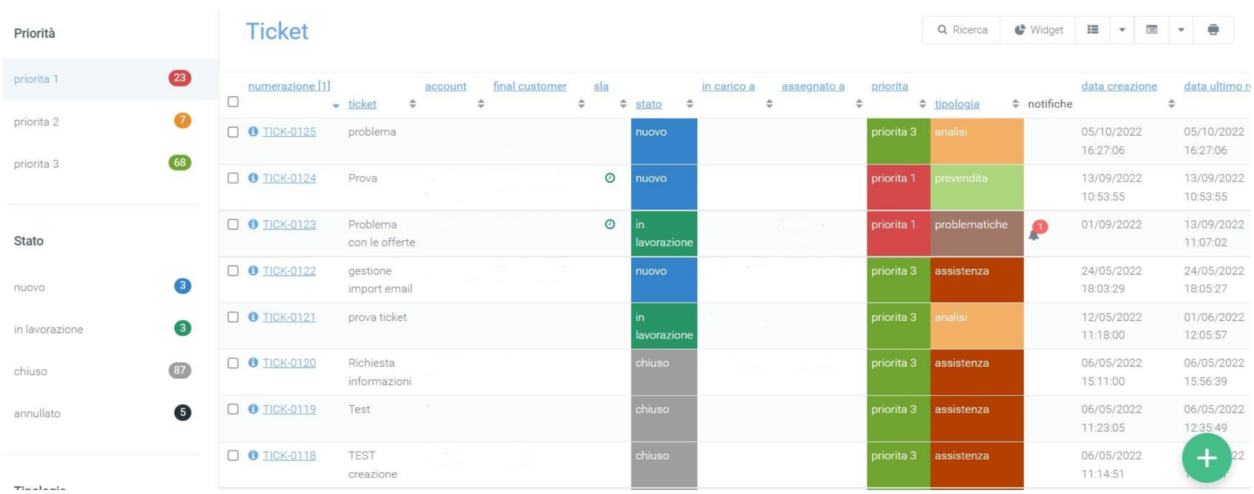
On the Ticket home page, accessible via "Support -> Tickets," a grid can be displayed where all tickets that have been created and that the user is authorized to view are shown. On the left are quick filters to apply divided by priority, status, type, area, tags, estimation and tracking.

On the default table we can find the essential information:

- The ticket number
- The name of the ticket
- The Account
- The Final Customer
- The status of the ticket: new, in process, cancelled, closed
- Who is in charge of the ticket
- To whom the ticket has been assigned
- Its priority: high, medium, low
- The type of the ticket
- Notifications: if you receive an internal note or email in response to the ticket, it is reported through a bell icon in the ticket list. To delete the flagged notification, simply click on the letter depicted in the following icon:

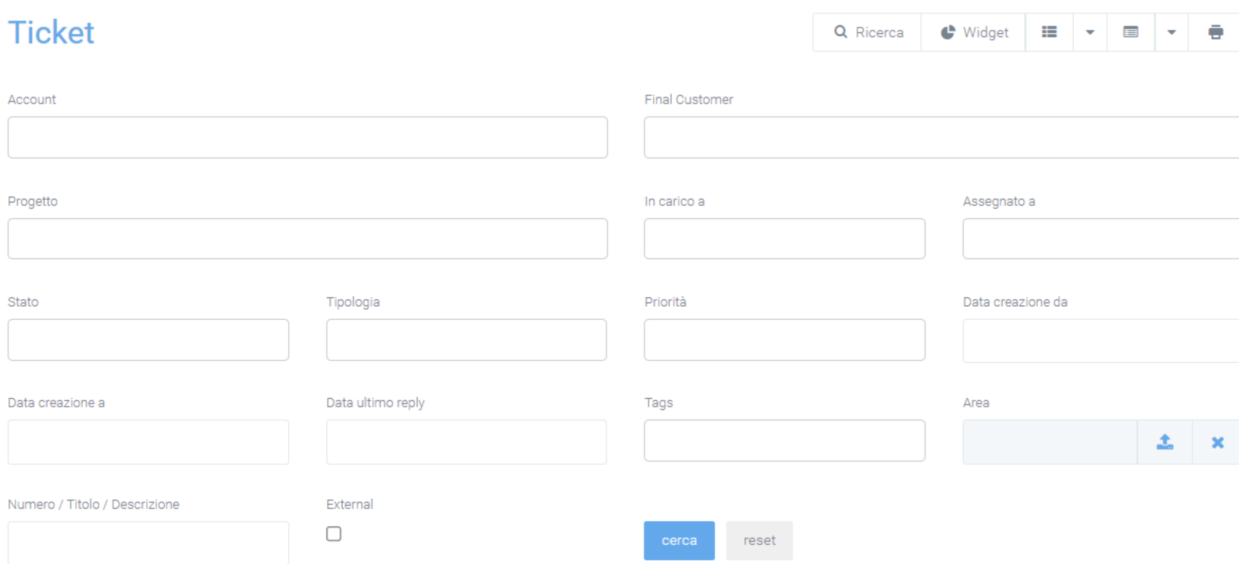
If the letter depicted in the lower left corner is yellow, the notification has not yet been read and appears in the ticket list; if it is green, the notification has been read and disappears.

- The date of the last response
- The estimated processing time in hours
- The tracking, i.e., the recorded time people have worked on the affected ticket.



Search

In case there are many tickets and you wish to perform a more specific search than the one offered through the quick filters, you will be able to press the "Search" button in the upper right corner. More precise and customizable filters will then be displayed as needed.



Available filters are divided by:

- Account: the company reselling a service/product
- Final Customer: the final customer
- Related Projects
- In charge of: the person in charge of the ticket
- Assigned to: who the ticket is assigned to
- Ticket status: new, in process, closed, canceled, suspended
- Type
- Priority

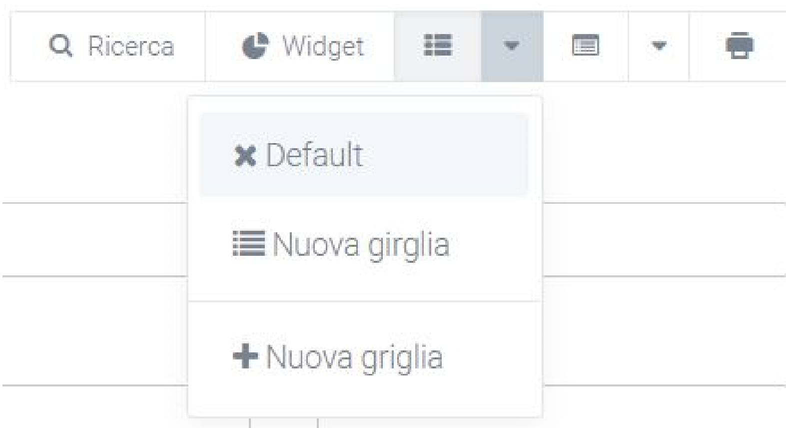
- Creation mode
- Date of creation from: to indicate a range of time in which tickets were created
- Date of last replay
- Tags: keywords inserted in tickets to speed up searching
- Area associated with the tickets
- The ticket number, title, or description
- Select tickets with unread notifications

The last flag, "External," is used to instruct the system to search for tickets destined for external vendors.

Widgets and grids

Each operator can create Widgets to monitor in real time the status of the activities and tasks he or she has to perform in order to stay aligned with scheduled plans. In addition to widgets, specific visualizations can be created via customizable Grids per operator so that the information collected is displayed specifically according to each operator's tasks. All these customization actions can be performed via the buttons following "Search."

To add and use widgets follow the explanation on this page:

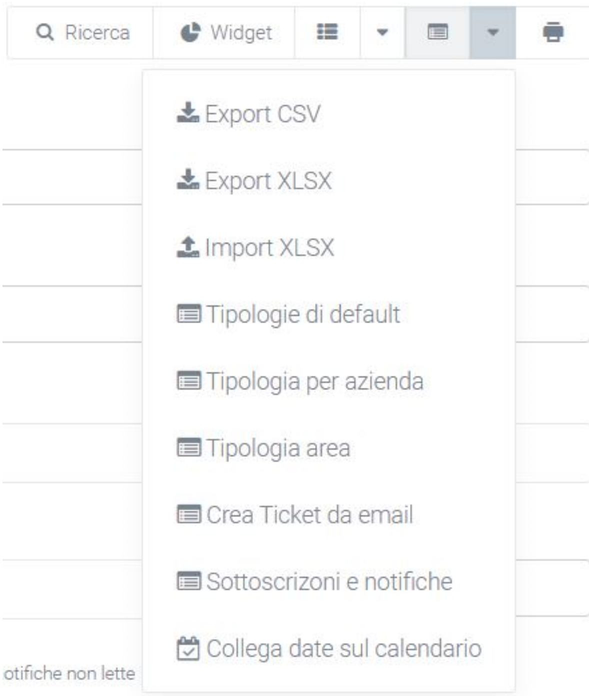


Actions menu

Using the actions menu found next to the grids button, you can export the ticket grid in XLSX and CSV format and import it into XSLX.

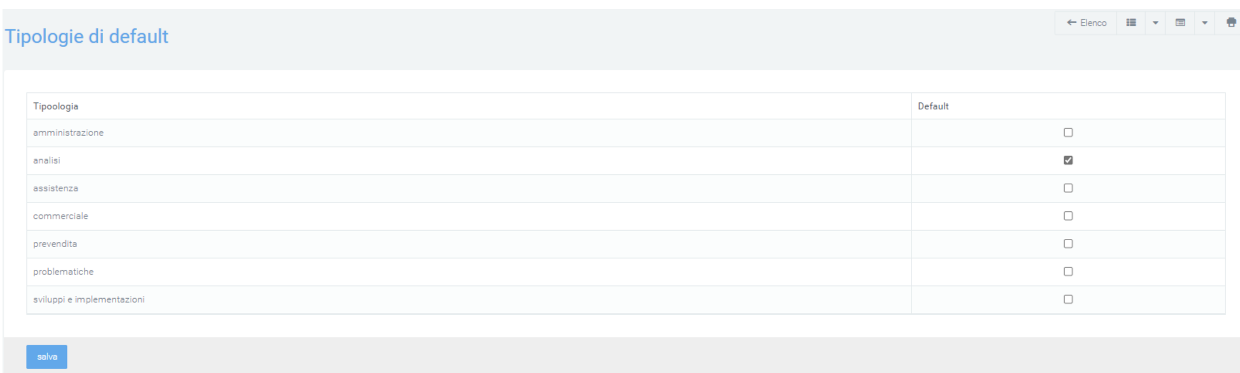
Next you will also find some sections for further customization:

- Default type
- Type by company
- Type by area
- Ticket from email
- Ticket subscriptions and notifications
- Link dates on calendar



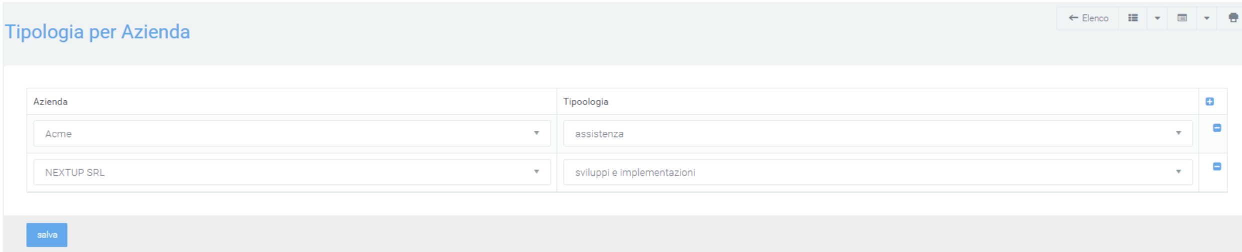
Default type

This section contains all the types that the final company will be enabled to see and select, "default" refers to the case where the company was not enabled to view specific types (assigned in "type by company"). To add or remove them, simply check the box next to the name of the type you want to enable companies to view.



Company type

In this section the ticket types that a specified company will be enabled to select during creation can be selected. Simply select the company under the appropriate heading and assign it a type. To add more, you will need to click on the "+" button to the right of the "type" entry and repeat the assignment process.

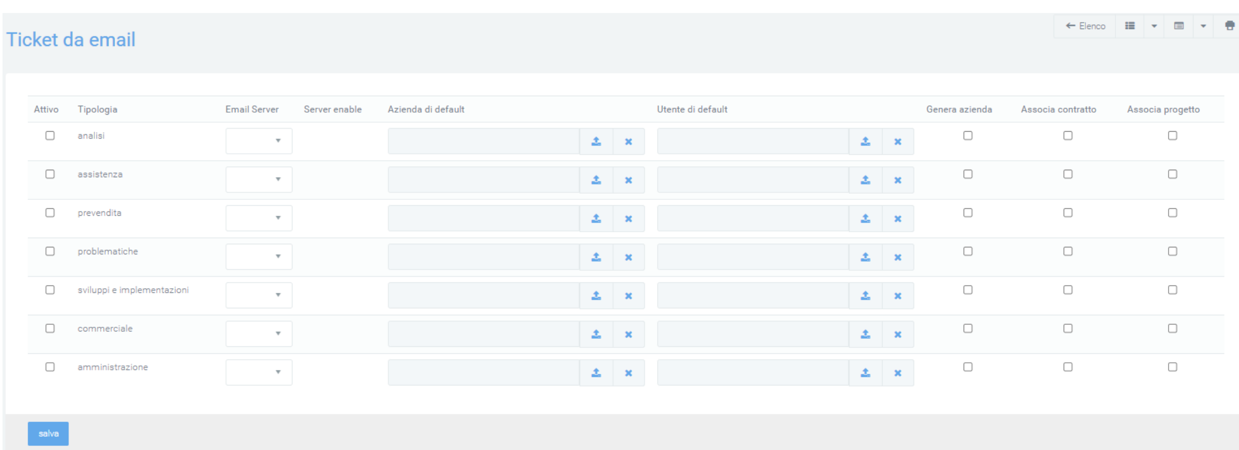


Area type

Through this section, tickets of a specific type can be assigned to a specific area. In this way, the visibility of all tickets of a specific typology will be exclusive to the relevant area. To do this, from the "Area Type" screen, click on the box under "Assignment Area" and select the intended area.

Ticket from email

This section allows you to configure the creation of tickets by the arrival of an email in a specific mailbox. You can decide who to assign a ticket to and what company name it should be associated with, in case the email address is not already registered. It is also possible to generate a company from the email address extension, associate a contract and a project with it, again based on the same extension.



- Each company type can be assigned the server where emails will arrive via the "Email Server" entry.

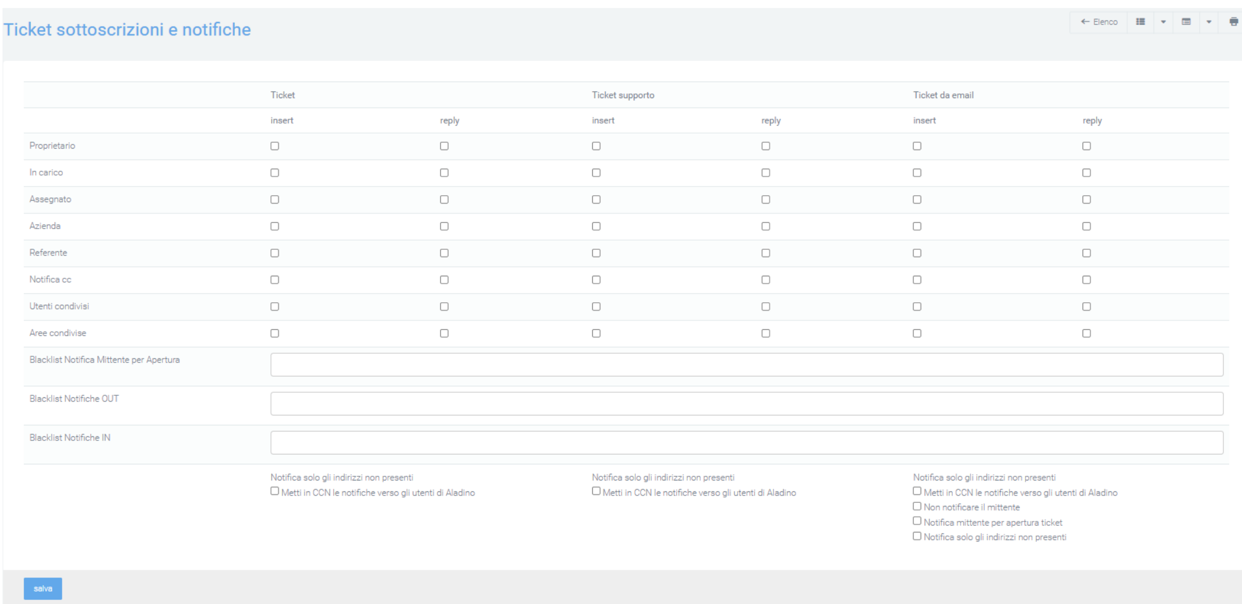
- The name that will be displayed under the "Default Company" heading of the ticket should be entered, in case the sender is not recognized by the system.
- The Default User will be enabled to read the ticket and can be added by clicking on the box under the "Default User" heading.
- The "Generate Company," "Associate Contract," and "Associate Project" buttons, if selected, will use the email extension to do what the button indicates, so a company will be generated with the name of the email extension and a project and contract will be associated with it, again with the name of the extension.

After entering all preferences, click on the light blue "save" button to store the changes.

Ticket subscriptions and notifications

In this section, you can control the sending of notifications for creating (insert) and replying (reply) tickets, support tickets, and tickets generated via email. To do this, simply select via appropriate flags

- which roles should receive the notification (owner, in charge, . . .)
- under what circumstances (after creation or upon ticket reply)
- to what type of ticket the notification should be sent (ticket, support ticket, ticket from email).

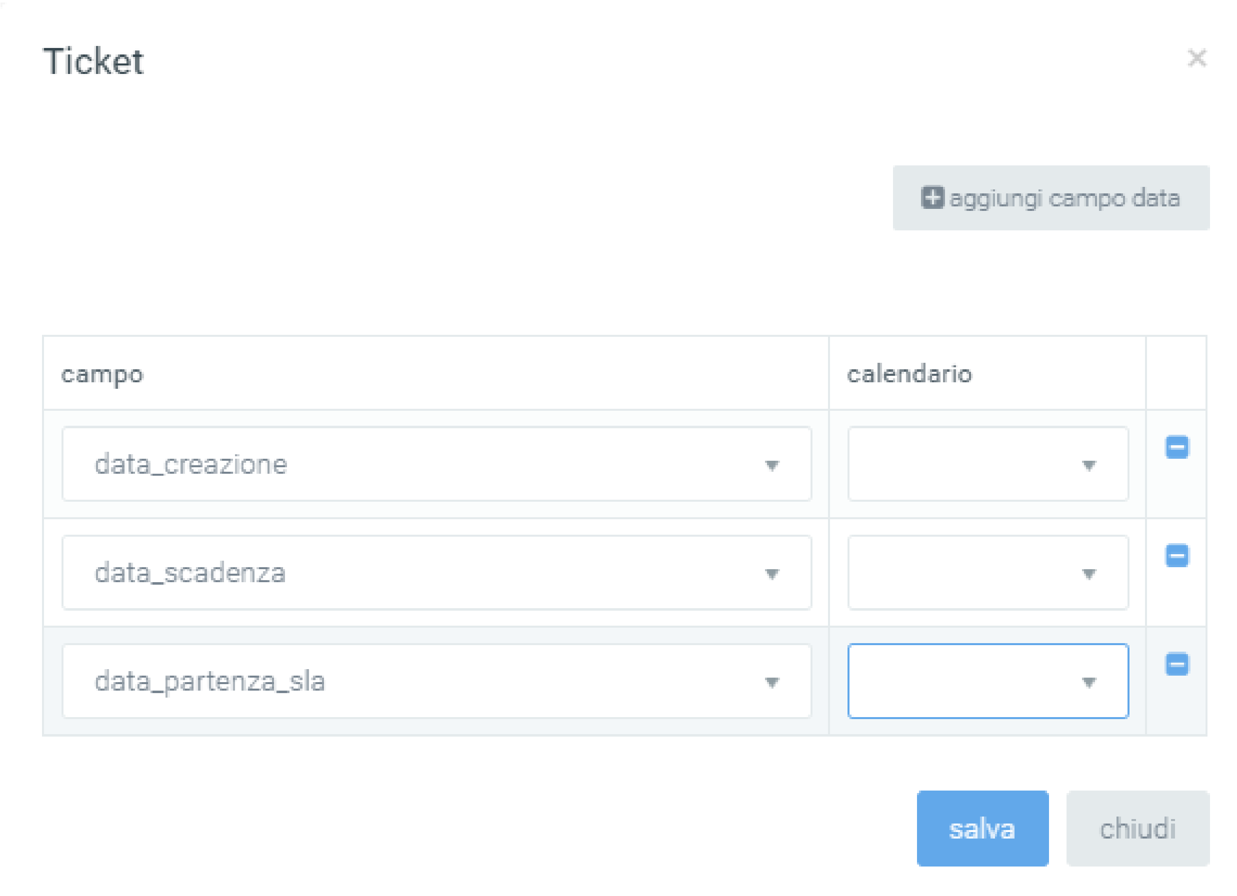


- **"Sender Notification Blacklist for Opening "**: email addresses or domains can be added to which the ticket opening notification should NOT be sent, in case the "Sender Notification for Ticket Opening" flag is active.
- **"Blacklist Notifications "**: it is possible to add email addresses, or just the domain, which should NOT receive notifications regarding the affected tickets, from opening to replying to the ticket, either incoming (IN) or outgoing (OUT).
- The flag **"Put notifications towards Kalliope Nexus users in CCN "** will allow to hide the addresses of all those to whom the initial email will be sent in CC, so that the client, responding to the email, will only respond to the correct one, avoiding errors that would be shared with other addresses as senders.
- The **"Notify sender for ticket opening "** flag, if activated, will notify the system to send an email notifying the sender of the initial email of the ticket opening.
- The **"Do not notify sender "** flag will not send response email notifications to the sender (either in creation, or in response)

- The **"Notify only non-present addresses "** flag is used to avoid repetitive notifications to users. The customer email that will then generate a ticket will have several users as recipients, in addition to the customer's referral email address (e.g., supporto@azienda.com). The generated ticket will send notifications to specific users, among whom there may be users who have already received the original email notification. To avoid repetitive notifications, it is necessary to enable the flag to notify the ticket only to addresses not in the original email.

Link dates on calendar

Clicking on this option will allow you to choose a ticket date field to be entered into a calendar at will.



The fields that will be displayed on the calendar, can be found on the "Scheduler" form, where you can find all the calendars created as personal and system.

Create ticket

Using the green button in the lower right corner with the "+" symbol, you can manually create a new ticket that will be visible to authorized users (preferences entered through role configuration).

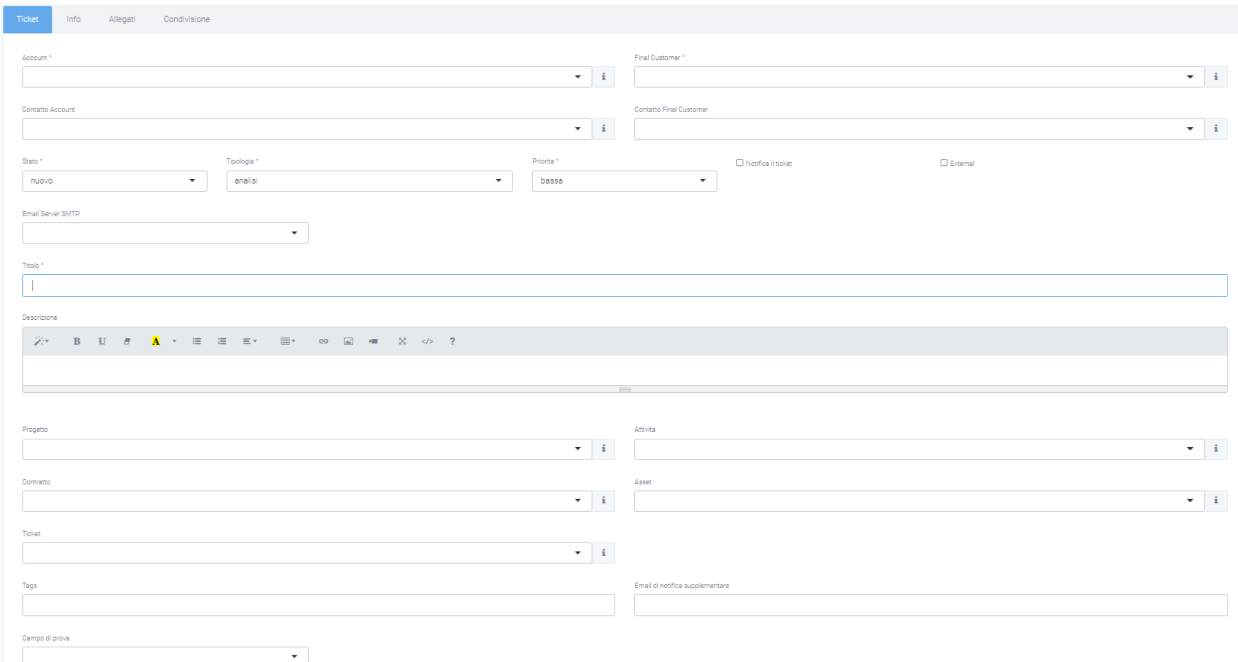
During ticket creation, four tabs can be viewed in the upper left corner:

- Ticket
- Info
- Attachments

- Sharing

Ticket

Within this section you can find various fields to fill out:



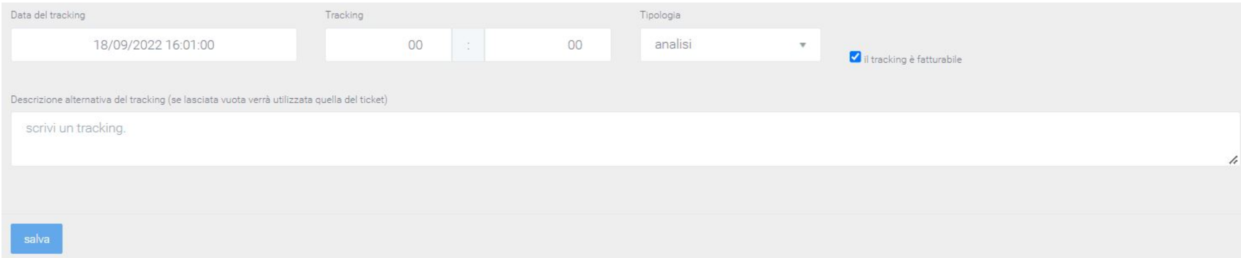
- Account and Final Customer: the two fields "Account" and "Final Customer" indicate the referring company that opened the ticket and the company on which the activity is performed, respectively. By clicking on the "i" on the right side of the drop-down menu, more information regarding the relevant company can be displayed:
  - info
  - asset
  - associated contacts
  - associated contracts
  - offers
  - opportunities
  - orders
  - passwords
  - prepaid
  - projects
  - tickets
- **Account Contact and Final Customer Contact**: These two fields are used to indicate the two contacts associated with the Account and Final Customer. The drop-down menu allows the choice of input fields.
- **Status**: tickets created have a "new" status. This status can be changed according to the activities performed ("in process", "taken in charge", … , "Closed")

- **Typology**: with the typology field it is possible to classify the ticket to facilitate its recognition and taking in charge by the various operators. The types are configurable by the user.
- **Priorities**: priorities, which can be customized by the user, identify the importance of the ticket and are displayed in different colors, also configurable by the user.
- **Email SMTP Server**: through this item you can select the server from which you will want to send the notification email. All emails that will be sent or received will refer to the entered SMTP email server.
- **Notification Ticket**: selecting this field enables the system to send notifications. Notifications are sent according to the configuration made in the appropriate "subscriptions and notifications" section.
- **External**: if checked, this is used to indicate tickets whose resolution is intended for external providers.
- **Title**: in the "title" box should be entered the title under which the ticket will be displayed, so as to provide an idea of what it will be about. In case the ticket is opened by email, the title would correspond to the subject.
- **Description**: the "description" details the customer's requests and any problems to be solved.
- **Contract**: in case there is a contract between the Account and the Final Customer entered, an existing contract can be entered. If there is an SLA (Service Level Agreement) in the selected contract that has minimum or higher priority, the contract timelines will be automatically associated.
- **Asset**: Entering an asset makes it easier to identify the problem and speed up support operations. Other open and/or closed tickets belonging to the entered account/final customer can be indicated, so as to correlate the tickets and help understand the problem.
- **Ticket**: you can indicate other open and/or closed tickets belonging to the account/final customer entered, so as to correlate the tickets and help understand the problem.
- **Tags**: tags can be added through this entry, which can then be used in filters, to improve the search for certain tickets.
- **Additional notification emails**: through this box you can enter additional email contacts who will be notified when the new ticket is created.
- **Additional**: at the bottom of the creation mask, depending on the modules enabled in the platform, additional fields are available to associate the ticket with:
  - Projects
  - Activities
  - Contract
  - Asset
  - Ticket
  - …

At the bottom is a section specifying:

- Date of tracking
- Tracking
- Type
- Tracking is billable
- Alternative tracking description (if left blank the ticket description will be used)

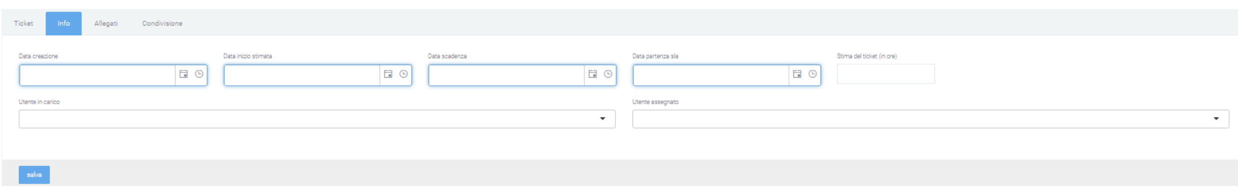
Completing the tracking causes the status of the ticket to change from "new" to "in process".



Info

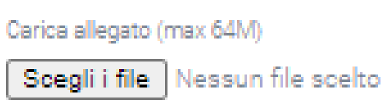
- Creation date: date on which the ticket was opened
- Estimated start date: estimated date for the ticket to begin processing
- Expiration date: date the ticket expires
- SLA start date: automatically calculated date from which the system starts calculating the SLA associated with the ticket.
- Ticket estimate: in this section you can enter an estimate in hours of the time taken to resolve the affected ticket.
- User in Charge and Assigned User: a ticket can be assigned to a user. A user can take charge of the ticket although it has not been assigned to him/her.

Assignment and taking charge are key information for controlling ticket management activities.



Attachments

In this section you can add attachments that will then be visible within the ticket. To add an attachment, click on the "choose file" button and select the file to be uploaded to the ticket. This section also contains any attachments in the emails imported from the system.



Sharing

In this last section you can share the ticket you created even with those who do not have visibility. You can give visibility to individual users or entire areas.

Saving the ticket

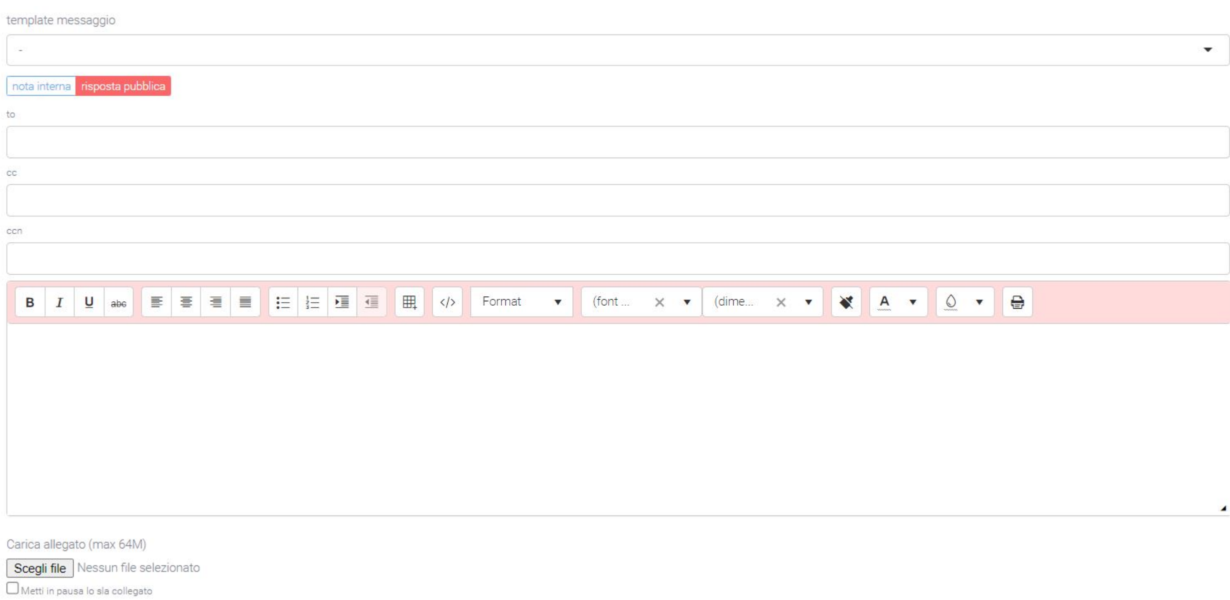
After filling in all required mandatory fields (those marked with an asterisk "*") you can save and create the ticket via the "save" button in the lower left corner. Once created, by pressing on the ticket number link you can view, in addition to Info, Attachments and Sharing, the Tracking if previously configured.

Intake

Once the ticket has been issued, the relevant users will be able to click on it and then, by clicking on the orange button at the bottom right, take it in charge. This will display in the "in charge to" column the name of the user has taken charge of the ticket, the user can decide when to change the status of the ticket.

At the bottom, a message box is displayed, which is useful for communicating notices regarding the ticket. It is possible to set the message as an internal note or public reply. To send the created message simply click on "Reply," the message will then be displayed with the first and last name of the person who sent it, the date and time, the status of the ticket, and the type of reply (internal or public). It is possible to quote received messages via the top right citation icon.

In case "internal note" is selected, no email will be sent. On the other hand, if "public reply" is selected, the message will be notified to the email of the user who created the ticket and the user who is assigned the ticket. It is also possible to add additional email addresses. In addition to typing the text, there is the ability to attach a file and/or pause the SLA via the "Pause linked SLA" checkbox. In addition to the "Reply" button, there are options to "change ticket status" to "working," "closed," "canceled," and an options menu containing the list of people to whom the ticket can be assigned, thus changing the person in charge.



Edit Ticket

To edit information within a ticket, it is necessary to click on the number link and then the "edit" button in the lower left corner. After making changes, click on the "save" button.

Opening ticket from email or website

The platform allows a customer to open a ticket in several ways. The most common are through sending an email to a specific mailbox or through logging into the platform in the Ticket section within the support part. For the automatic opening of tickets, the system relies on the email address to which the messages arrive and, in case this is used for ticket generation, it will automatically create a ticket by reporting the sender's information and the request in the email. The generated ticket will be automatically routed to the relevant area linked to the recipient email address. This allows you to greatly optimize time by avoiding reading each email and manually entering each ticket. In addition, you can indicate in the actions menu-if the ticket arrives from a specific email server-the type and priority of the ticket. In this case, if a contract is present, it is automatically associated. Other ways of entry can be provided, such as, for example, from specific forms on Internet sites.

5.1.4 Billing

> **Warning:** The Billing module is included in the purchase of the Analytics module.

Through the Kalliope Billing module, the Kalliope Nexus platform can manage debit and multi-service documentation.

Pricing

The "Voip Tariffing" module is divided into two sections:

- Pricing: section dedicated to setting the rates that are applied to calls
- Dashboard: section that calculates and displays the total call charges based on the filters entered

Structure

The section in question allows you to go and create, edit and delete rates that associate a value with each call. The latter point is done through the presence of the billing time present on each call and according to the rules set in the tariff, such as:

- company
- prefix
- country
- gateway
- free limit
- status

The subsequent act of calculating these costs is fully automatic through the use of scheduled crons.

The section has two main masks: one for displaying all tariffs created and one for creating or detailing each tariff.

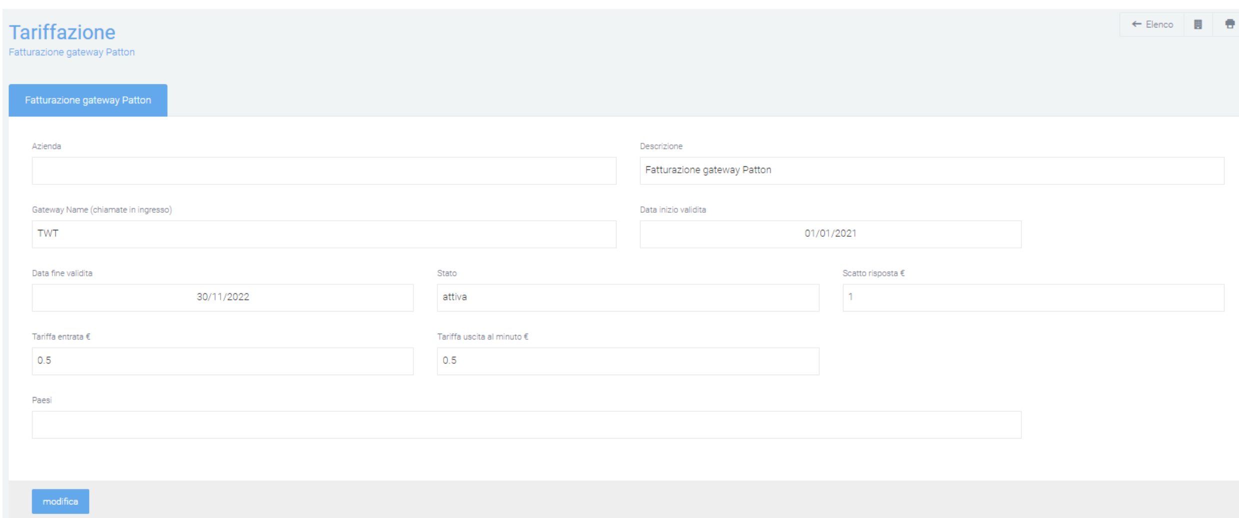
List

The list of tariffs shows:

- description
- start and end date of validity
- inbound/outbound gateway name
- prefix
- in/out rate
- click
- status
- annual free limit (minutes)

descrizione [3]	data inizio validita	data fine validita	gateway name	gateway name configurazione	prefisso [2]	tariffa in	tariffa out	scatto	stato	limite grauito annuale
Fatturazione gateway Patton	01/01/2021	30/11/2022	TWT			0.5	0.5	1	attiva	
fatturazione generica	01/01/2021	30/11/2022				0.5	0.5	0.5	non attiva	
Tariffazione verso reti fisse	05/01/2022	05/01/2023			0		0.65		attiva	
Tariffazioni VERSO reti mobili	05/01/2022	05/01/2023			3		3.9		attiva	
Test tariffazione per Azienda	19/10/2021	05/01/2023				0.5	0.25	0.5	attiva	

Detail:



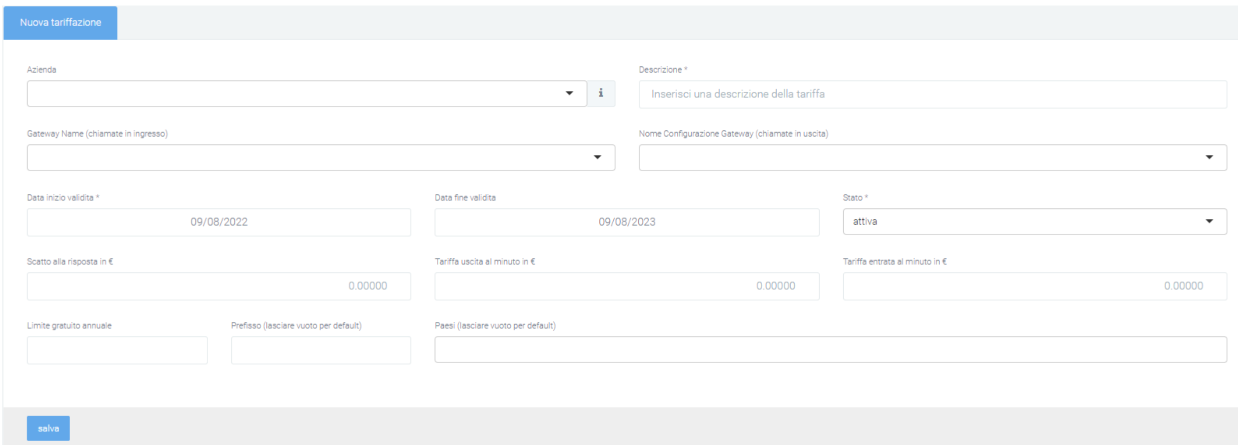
Create new pricing

In order to be able to create and configure new tariffs, which will then go on to value the affected calls, from the "Tariffing" page click on the green "+" button at the bottom right.

Next, the creation mask will be displayed in which all the data necessary for the correct application of the tariff on calls will have to be entered, such as:

- Company: the company for which the tariff applies (if the company is different or not recognized, this tariff will not be applied)
- Description: short title that makes the tariff recognizable in the list on the main page

- Input/output gateway for which the tariff applies
- Start/end validity date: period of validity of the tariff
- Status: current status
- Response charge (in €)
- Outgoing/incoming rate per minute (in €)
- Annual free limit (minutes)
- Area code
- Country: for which the rate applies



Once all fields deemed necessary have been filled in, save the tariff using the appropriate button at the bottom right.

After saving, the specifics of the tariff are visible and you can view the daily consumption divided by:

- Incoming calls
- Outgoing call
- Total

Of each, it is possible to analyze the time spent on incoming calls, the cost and the hypothetical cost (without free rate). A ratio of minutes used to total minutes is also displayed at the bottom:

Consumo: 05/10/2022 - 10/10/2022

	tempo	costo €	costo ipotetico (senza tariffa gratuita) €
entrata		0	0
uscita		0	0
totale		0	0

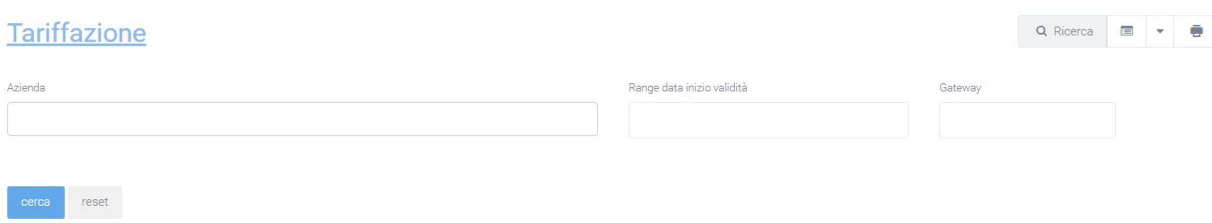
Limite gratuito (minuti)

00:00:00 / 01:40:00 (0%)

Filters

Available in the main mask, filters (click on the "Search" button) and export/import methods. Filters allow selection by:

- Company
- Start date range
- Gateway



Dashboard

Once the charges have been created and activated, attention can be shifted to the Dashboard section where, as calls enter the CDR and are valued, the total costs are calculated and displayed based on the filters entered via the "Search" button.

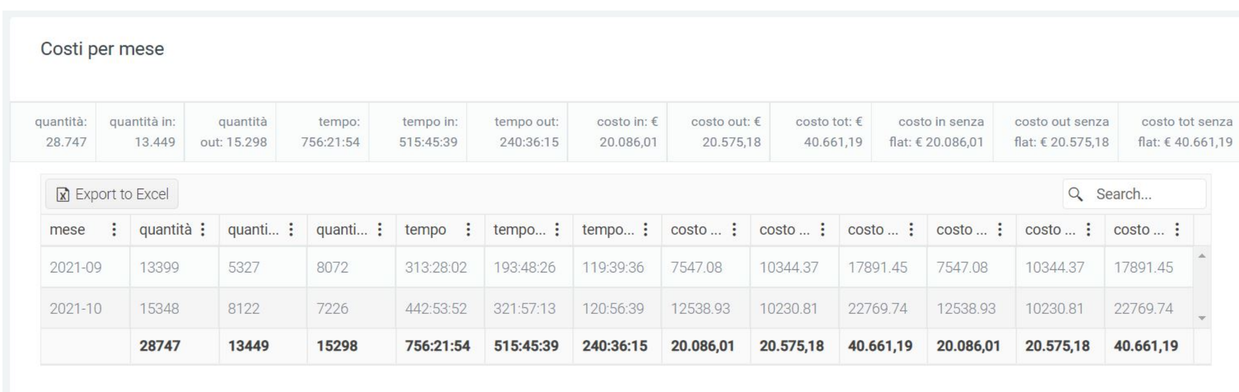
This section is structured in a first part (above the charts) where the totals of quantity, time and costs are displayed in a generic way. This allows the user to have the essential information at hand immediately.

Immediately following, there are two graphs with monthly and daily breakdowns.

For a more detailed analysis of the costs generated by calls made/received, the second part of this section shows quantities, times, and costs with breakdown by:

- month
- date
- organization
- organization
- gateway
- operator
- cost center
- division

Some examples follow:



Costi per mese

quantità: 28.747	quantità in: 13.449	quantità out: 15.298	tempo: 756:21:54	tempo in: 515:45:39	tempo out: 240:36:15	costo in: € 20.086,01	costo out: € 20.575,18	costo tot: € 40.661,19	costo in senza flat: € 20.086,01	costo out senza flat: € 20.575,18	costo tot senza flat: € 40.661,19

Export to Excel

Search...

mese	quantità	quanti...	quanti...	tempo	tempo...	tempo...	costo ...	costo ...	costo ...	costo ...	costo ...	costo ...
2021-09	13399	5327	8072	313:28:02	193:48:26	119:39:36	7547.08	10344.37	17891.45	7547.08	10344.37	17891.45
2021-10	15348	8122	7226	442:53:52	321:57:13	120:56:39	12538.93	10230.81	22769.74	12538.93	10230.81	22769.74
	28747	**13449**	**15298**	**756:21:54**	**515:45:39**	**240:36:15**	**20.086,01**	**20.575,18**	**40.661,19**	**20.086,01**	**20.575,18**	**40.661,19**

Costi per gateway

quantità: 28.747	quantità in: 13.449	quantità out: 15.298	tempo: 756:21:54	tempo in: 515:45:39	tempo out: 240:36:15	costo in: € 20.086,01	costo out: € 20.575,18	costo tot: € 40.661,19	costo in senza flat: € 20.086,01	costo out senza flat: € 20.575,18	costo tot senza flat: € 40.661,19

Export to Excel

Search...

gateway	quantità	quanti...	quanti...	tempo	tempo...	tempo...	costo ...	costo ...	costo ...	costo ...	costo ...	costo ...
	27718	13309	14409	745:58:57	513:28:33	232:30:24	20019.44	19961.24	39980.68	20019.44	19961.24	39980.68
	1029	140	889	10:22:57	02:17:06	08:05:51	66.58	613.93	680.51	66.58	613.93	680.51
	28747	**13449**	**15298**	**756:21:54**	**515:45:39**	**240:36:15**	**20.086,02**	**20.575,17**	**40.661,19**	**20.086,02**	**20.575,17**	**40.661,19**

Multiservice

The Multiservice module stems from the need to manage the valuation of calls billed to its customers as if a service were being offered. Thanks to this, it will be possible to use the platform simultaneously to:

- manage the charges for each call through the "Pricing" module
- manage the value of calls that are to be billed to the customer as a service through this module

There are two sections within this and they allow you to:

- manually manage the services offered in the contract
- view a report containing the quantities and prices of the services used, including calls (which will be calculated automatically by the system)

Billing

In this section you can view the report regarding the various services and incoming and outgoing calls, with their total costs that are billed to each customer.

Structure

The report is structured so that each company has its own dedicated line, with quantities and costs of each service used.

			chiamate in			chiamate out			visura camerale		
ragione sociale	competenza		totale	tempo	costo	totale	tempo	costo	totale	costo	totale costi
NETRESULTS S.R.L.	01/11/2021	28/02/2022			0			0	10	240,00	240,00
NEXTUP SRL	01/12/2021	28/02/2022	67	00:23:50	17,55	25	00:10:33	0		0	17,55

Looking at the table above, the first column shows the company name, the second column shows the contract period on which the services offered are shown, and the next columns show the various services used:

- Total: the total amount of service used (in the case of calls, this will be the total number of calls offered that were made and then used)
- Time: the total time of all calls shown in the total
- Cost: the total cost calculated on the service used or call minutes made (in the case of calls)

At the end of the display of all services used, you will find the total cost to be incurred by the customer indicated at the beginning of the line.

Configuration

In order to start using this module, it is necessary to indicate, in the various contracts that are to be created, the services that are offered with their costs. To do this, when creating the contract, at the end of it you will find a button ([+] new line) that will have to be pressed to add a service to that contract.

Once done, the various fields will need to indicate:

- the service
- the rate of the service
- the maximum limit of "products" of the service that the customer will not have to pay for
- as far as calls are concerned, also the connection charge

Once this initial setup is completed, all that is needed is to manually indicate for each service (except calls) the quantities used by the various clients. Everything else, including the evaluation of each call, will be done automatically.

Filters

By using the filtering system, you are able to get the best out of this module, being able to easily indicate, through 3 fields, the companies you want to display and the time period in which you want the calculations to be done. They can be found in the upper right corner by clicking on the search button.

5.1.5 Trace

Trace Inbound

Description

The Kalliope Trace module makes it possible to handle inbound calls by allowing the operator to collect and display a range of information about the customer associated with the calling number. The latter is automatically recognized and, based on the contacts already saved in the platform, information is provided to the operator who is going to handle the call. Different types of information can be displayed, from the simple name of the customer or company, to open tickets for the latter or calls already recorded.

Saved calls are displayed in one place so that notes or descriptions taken for each call and for which the form screen was used are not lost.

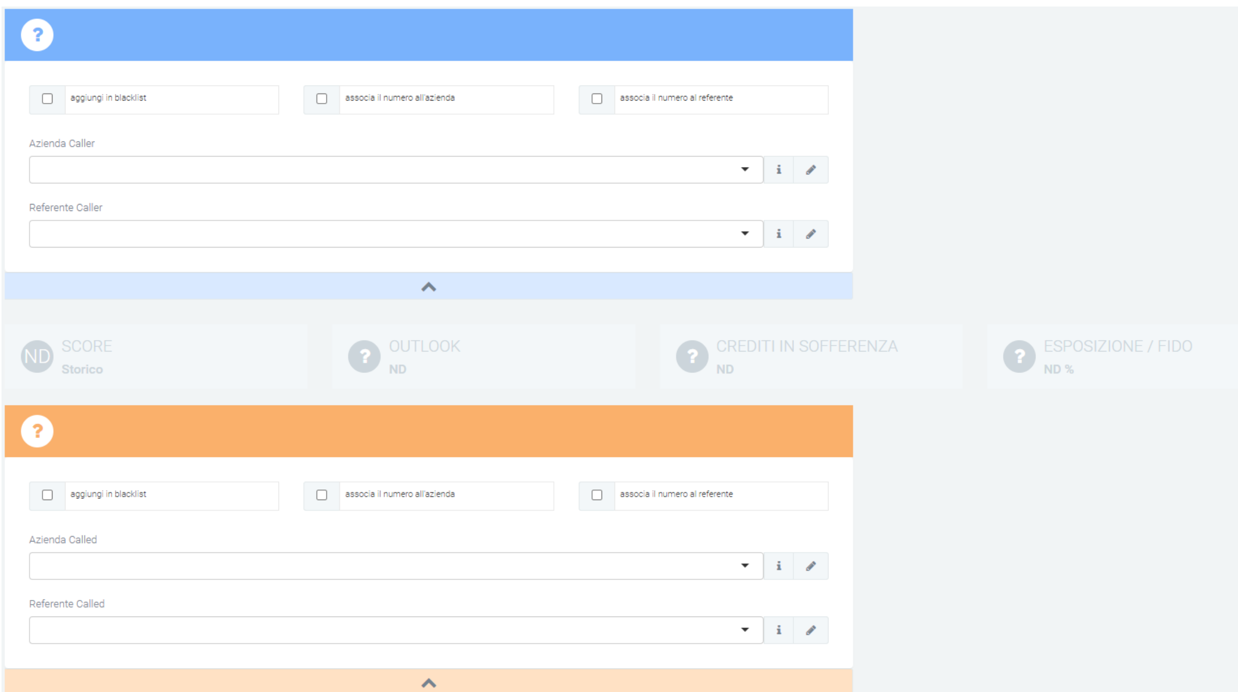
Mask Calls

The first point concerns receiving or answering a call from an operator. Based on a specific configuration made on the KCTI present on one's PC (settings -> automatic actions -> dynamic url opening) and a preference indicated on Kalliope Nexus, an operator receiving or answering the call will see a page open on his monitor in his browser containing the Kalliope Nexus mask. The mask contains some pre-filled fields so as to facilitate recognition of the company and its financial status.

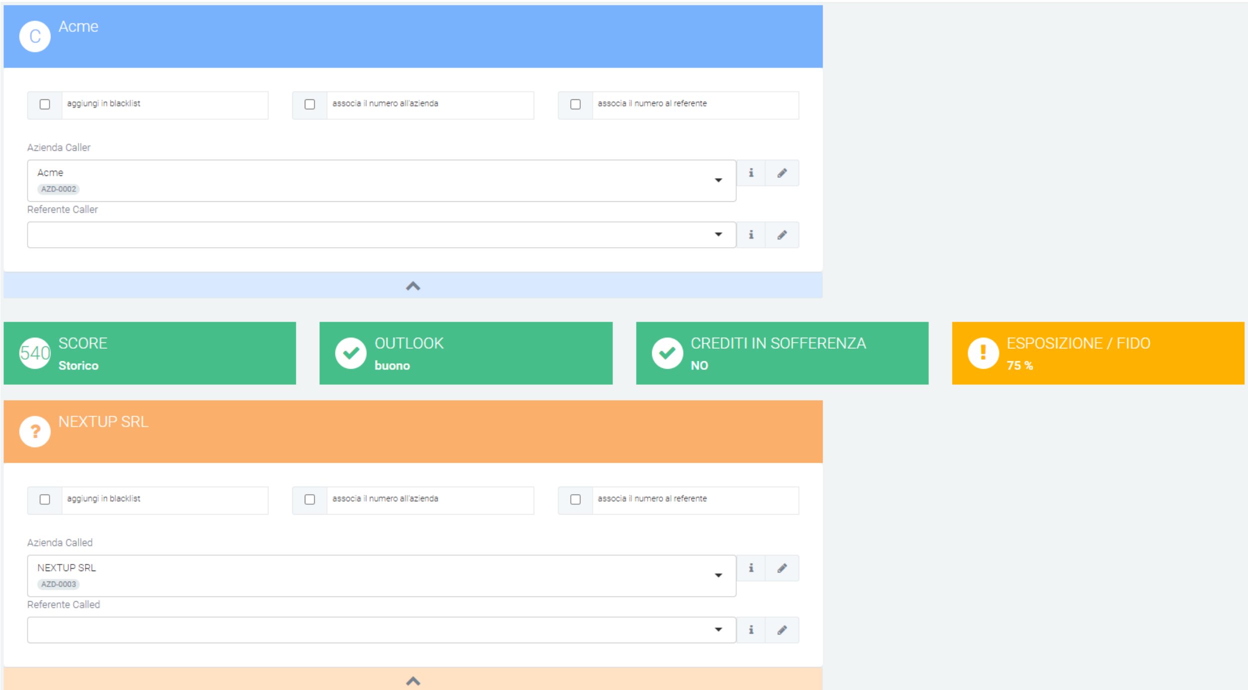
This occurs when the company is registered in the "Companies" section and thus is recognized by the system; conversely, the operator will have the option of simply entering the caller's data and then registering the company.

An example of a call from a number not recognized by the system can be seen in the figure below. In this case it is the operator who creates the connected company by clicking on the pencil icon to the right of the "i". In this way the company name, code and vat number data will have to be entered.

Using the ticks it is possible to: add the number to blacklist, associate the number with the company, and associate the number with the contact person.



In case the caller is already registered in the platform, the previously blank fields will be filled in automatically and the colors of the financial part will change automatically to make it easier to read.



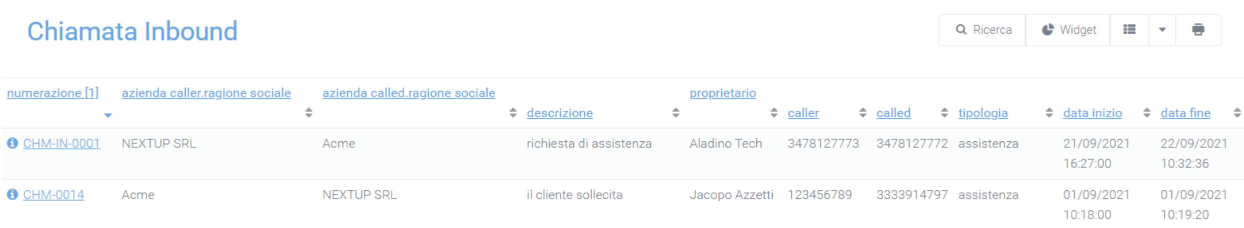
In both cases, the url that will be opened by KCTI due to the "dynamic url opening" configuration will be as follows:

```
indirizzio_di_aladino/chiamataInbound/main/edit/?caller=<callernum>&callername=
↪<callername>&callercompany=<callercompany>&callerunit=<callerunit>&extenNum=<extenNum>&
↪queueName=<queueName>&uid=<uid>&called=<callednum>
```

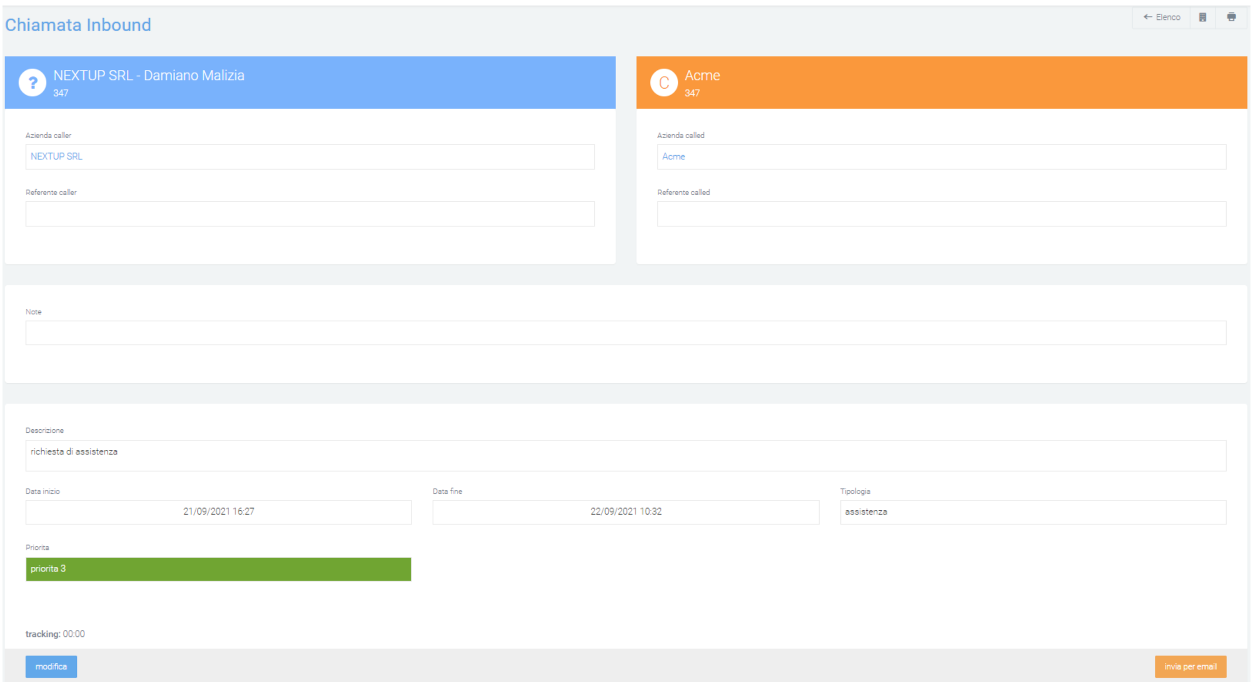
List of received calls

The second main function of this section is the list of incoming managed calls. This allows you to view various information, with the possibility of linking and displaying them in the previously mentioned "call" mask with the relevant data. The information contained in the columns consists of:

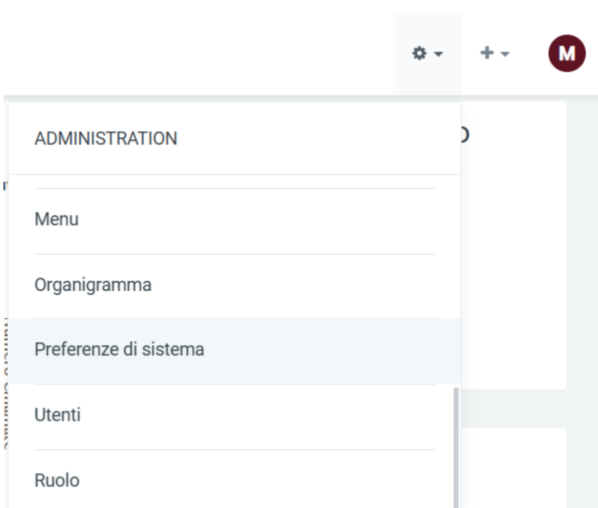
- numbering
- company name of the calling company (CALLER)
- company name of the called company (CALLED)
- description of the call
- owner
- caller
- called (number)
- type
- call start date and time
- call end date and time

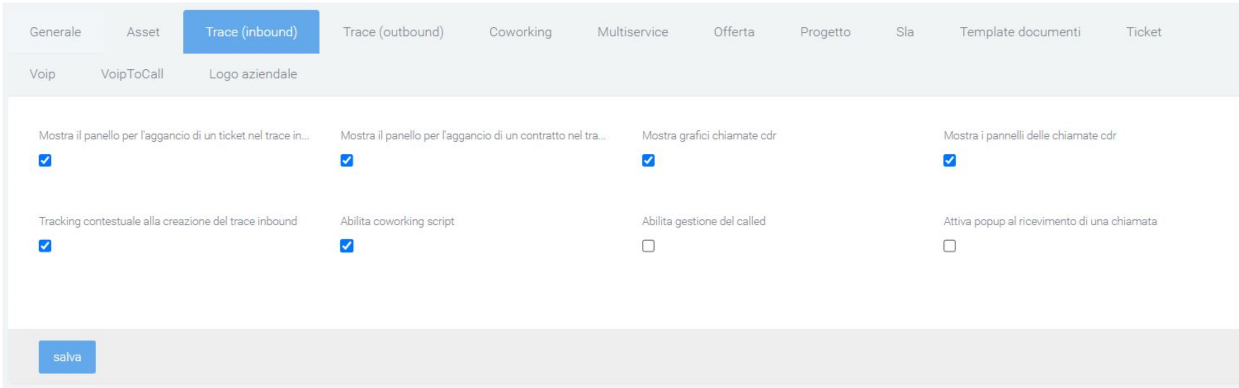


From this page you will be able to view the details of each call by simply clicking on the number that identifies it (highlighted in blue below in the "numbering" column). This will show information about the companies (caller and called), notes taken by the operator, description, associated type, and priority rating entered, as well as the date and time the call started and ended.



Two buttons will be available at the bottom of this mask: one on the left and one on the right. The first (left) "edit," enables the user to make changes to the call, with the possibility also to delete it by clicking on the red "delete" button. Summary graphs can be displayed in this mask, which can be enabled through "System Preferences."





Charts allow you to view: call history report, incoming/outgoing calls, hook a ticket, hook a contract, view a script, etc.

After making changes to the call, click on "save" to save them or on "detail" to exit the edit screen without saving.

The second button instead (right), "send by email" will allow you to select the message template, sender, recipient, and specify subject and description.

Grids and widgets

From the main page of the section, where the list is visible, the display of information in the list can be varied by changing the grid used. To do this, click on the "available grids" button and select the desired grid. If this is not present, it will always be possible to create a new one through the appropriate "+ new grid" button with which you will be redirected to the creation mask of the "Grid" section. After that, carry out the instructions of the indicated section.

To add and use widgets follow the explanation on this page:

Trace Outbound

Service Description.

The Kalliope Trace module allows the management of outbound calls, that is, calls made by the operator to a customer, thus outbound from the telephone exchange. This module sees its usefulness in recording the data of the calls made and automatically opening a "call" mask like that of Inbound Calls, with the only difference being that, in this module, there will be no mask for the calling company (being always the same) but only that of the called company (CALLED).

Call Mask

Again, as with inbound calls, the automation of opening the mask when the call starts is due to the configuration of the telephone exchange, which this time will generate the following url:

```
nome_azienda.aladino.cloud/chiamataOutbound/main/edit/?caller=<callernum>&callername=
↪<callername>&callercompany=<callercompany>&callerunit=<callerunit>&extenNum=<extenNum>&
↪queueName=<queueName>&uid=<uid>&called=<callednum>
```

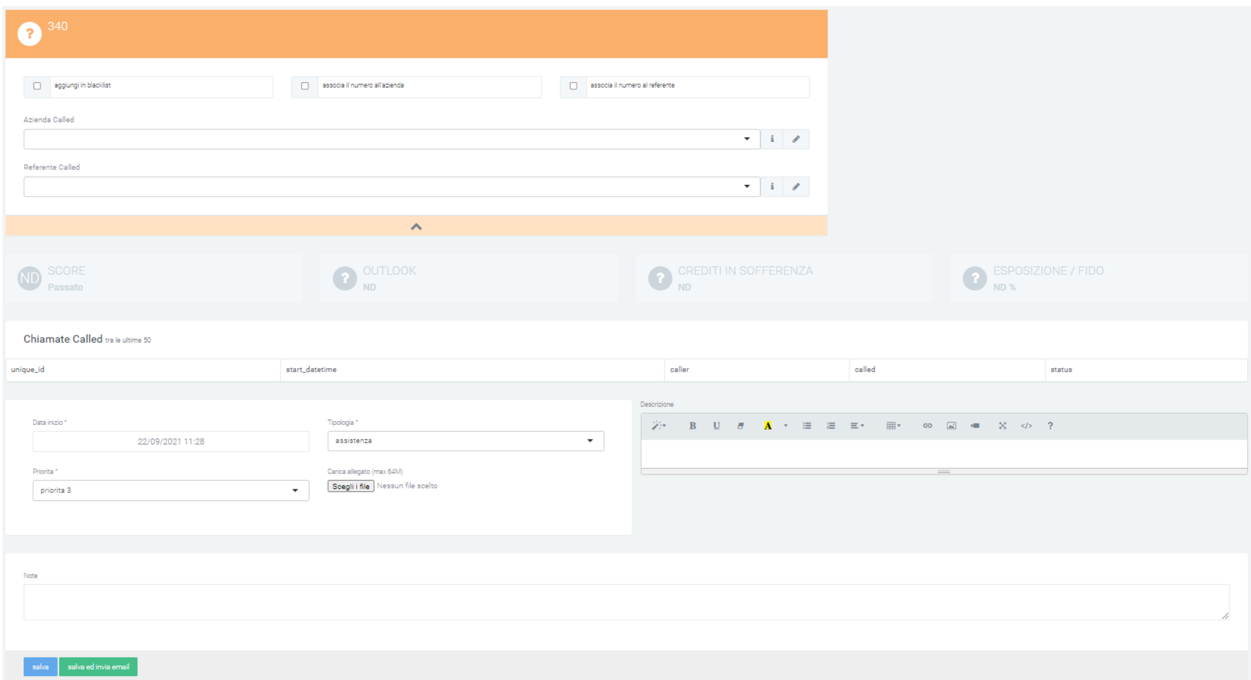
This screen will also be available by clicking the green "+" button in the lower right corner.

As with inbound calls, the operator will be able to create a new company for the contact they called, and thus were not recognized by the system, by simply clicking on the pencil icon to the right of the "i" in the "company called" or "referrer called" row.

The company's financial fields (socre, outlook, bad debt and exposure/overdraft) will be automatically populated, coloring them appropriately to make them intuitive. The user will always be able to edit the fields below, attach documents, write a description or add notes.

Unlike the inbound calls module, this module displays all calls made to the same called among the last 50, with information such as:

- unique id
- start date and time
- caller number
- called number
- status



Once the call is finished, simply press "save" to store it in the list or "save and send email" to send emails (a mask will open with the fields of a classic email to be filled in) in addition to saving.

List

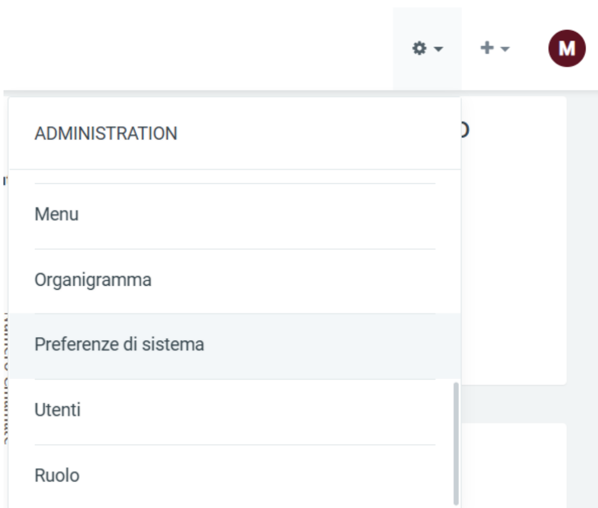
The list that will be visible as soon as the section page is opened, will be the same as in the inbound calls section, except that the caller's company name will not be present. The information that will be contained in the grid will be:

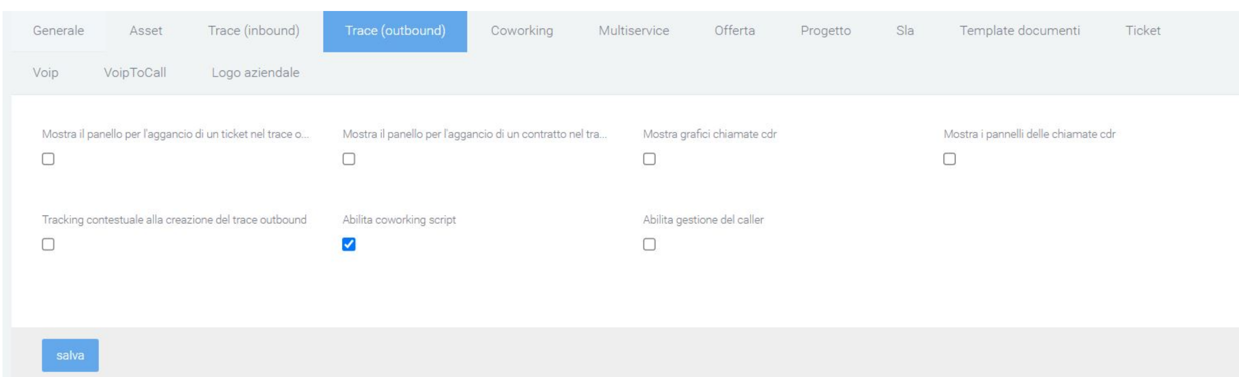
- numbering
- company name of the company called (CALLED)
- description of the call
- owner
- caller
- called (number)
- type

- call start date and time
- call end date and time

You can view the detail mask of each call by clicking on the number in the "numbering" column. This will show information about the company, the notes taken by the operator, the description the associated type and the priority rating entered, as well as the date and time the call started and ended.

As in Inbound calls, two buttons will be available at the bottom of this mask: one on the left and one on the right. The first (left) "edit," enables the user to make changes to the call, with the possibility also to delete it by clicking on the red "delete" button. Summary graphs can be displayed in this mask, which can be enabled through "System Preferences."





Charts allow you to view: call history report, incoming/outgoing calls, hook a ticket, hook a contract, view a script, etc. After making changes to the call, click on "save" to save them or on detail to exit the edit screen without saving.

The second button instead (right), "send by email" will allow you to select the message template, sender, recipient and specify subject and description.

Grids and widgets

From the main page of the section, where the list is visible, the display of information in the list can be varied by changing the grid used. To do this, click on the "available grids" button and select the desired grid. If this is not present, it will always be possible to create a new one through the "+ new grid" button with which you will be redirected to the creation mask of the "Grid" section. Once here, follow the instructions in the indicated section.

To add and use widgets follow the explanation on this page:

5.1.6 Live Chat

Description

The LiveChat module allows you to take advantage of an effective communication channel with your customers. Through the "chatbox" graphical interface integrated into your company's website, you can interact via instant messaging directly with operators.

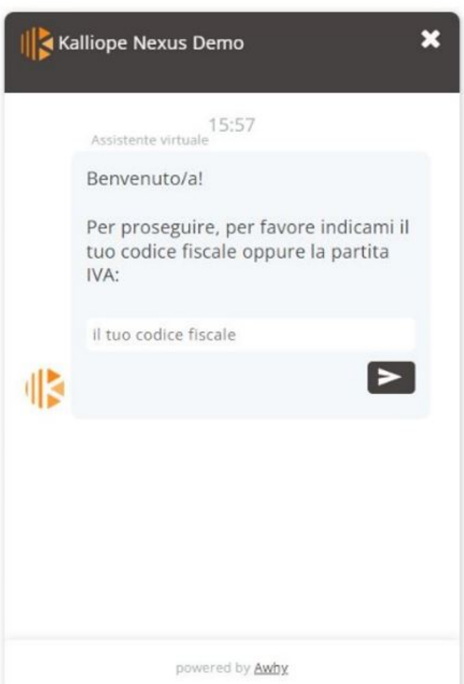
There are three ways to interact:

- Via web
- Via Whatsapp, Telegram, Messenger
- Via telephone

Web-based interaction

Via web, by clicking on the chat icon, the chat with the virtual assistant will be opened on the right where the user is asked to enter the tax code or vat number.

The moment the user enters information already on the system, he/she is identified and called by the name/company name.



If the user types a question that is not in the chatbot's knowledge base, it is possible to converse directly with an operator.

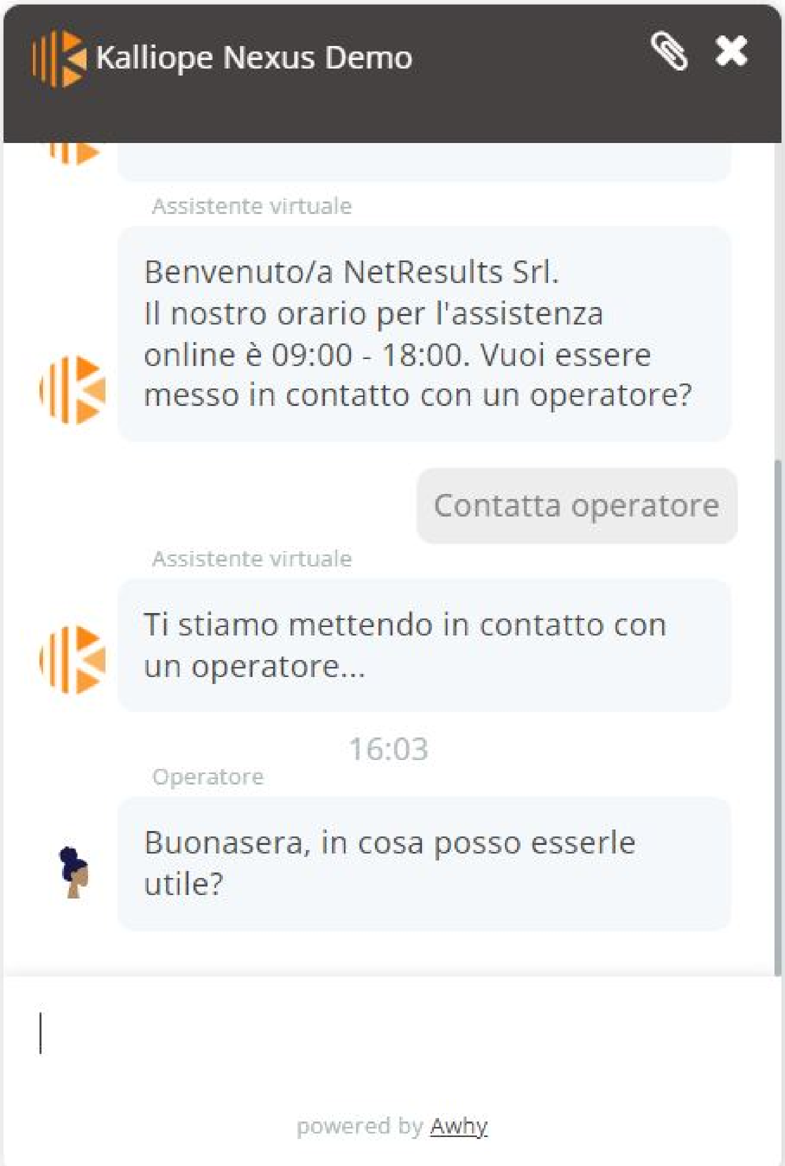
On the operator's screen, upon the user's request, a notification will appear in the following ways:

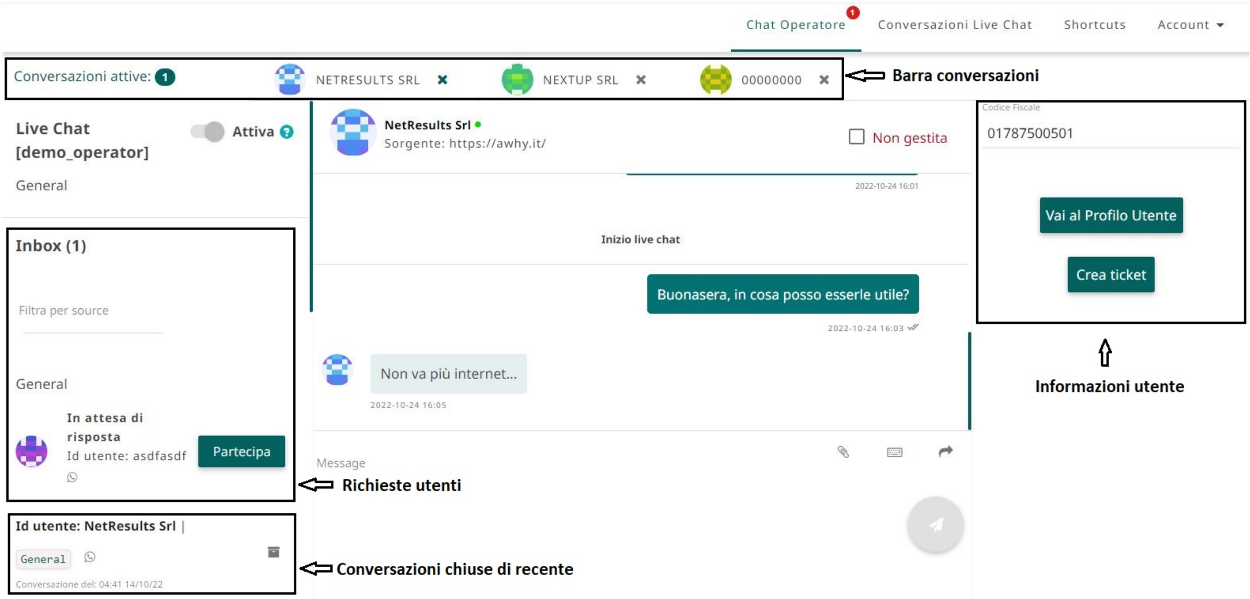
- in the General channel, on the left: the conversation can be joined via the appropriate "Join" button

- above the indication "Operator Chat"

- via the notification bell

The next images show the chat from the user's point of view and the operator's screen


Kalliope Nexus Demo
Assistente virtuale
Benvenuto/a NetResults Srl.
Il nostro orario per l'assistenza online è 09:00 - 18:00. Vuoi essere messo in contatto con un operatore?
Contatta operatore
Assistente virtuale
Ti stiamo mettendo in contatto con un operatore...
16:03
Operatore
Buonasera, in cosa posso esserle utile?
powered by Awhy



The operator's screen allows the operator to observe all available actions for effective management of submitted issues.

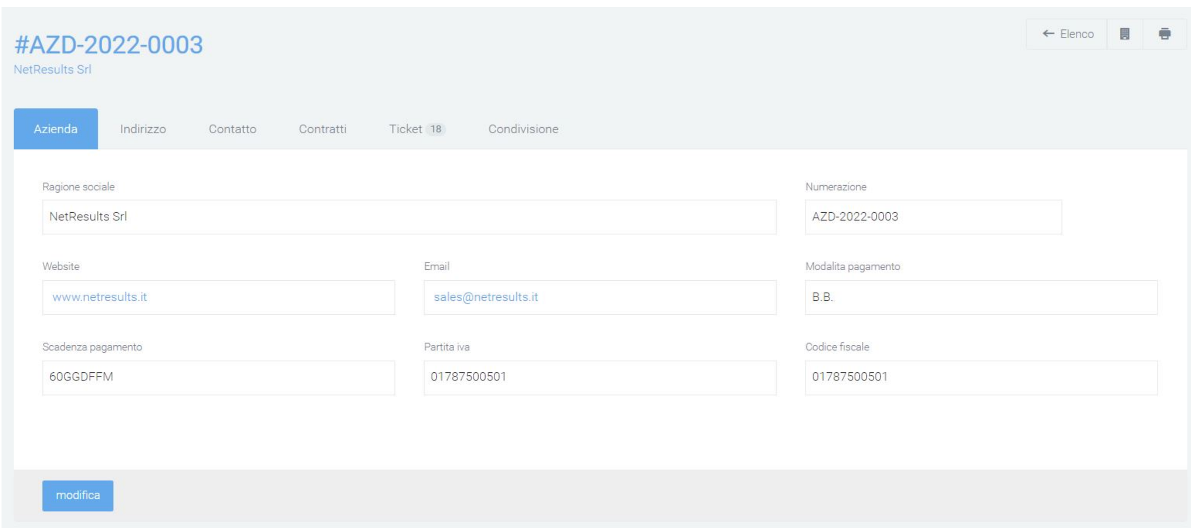
The operator can:

- **Read the conversation with the user and previous conversations had** (if the user is recurring) and respond as in a normal chat
- **Close the chat** by clicking on the "x" at the top next to the user's name/code. Once closed, this will be archived with the previous chats in the "General" channel.

- **Forward the conversation to another operator** via the forward button.
- Check the "unmanaged" box to **report the conversation** in case the user exits the chat before the problem is resolved
- **Attach a file to the conversation**
- **Use/create shortcuts** to facilitate the conversation by clicking on the keyboard icon to the right of the message entry box (see the figure below)

On the operator screen, on the right, there are buttons:

- **"Go to user profile "**: allows inspecting the user as he is identified, via the company description card.



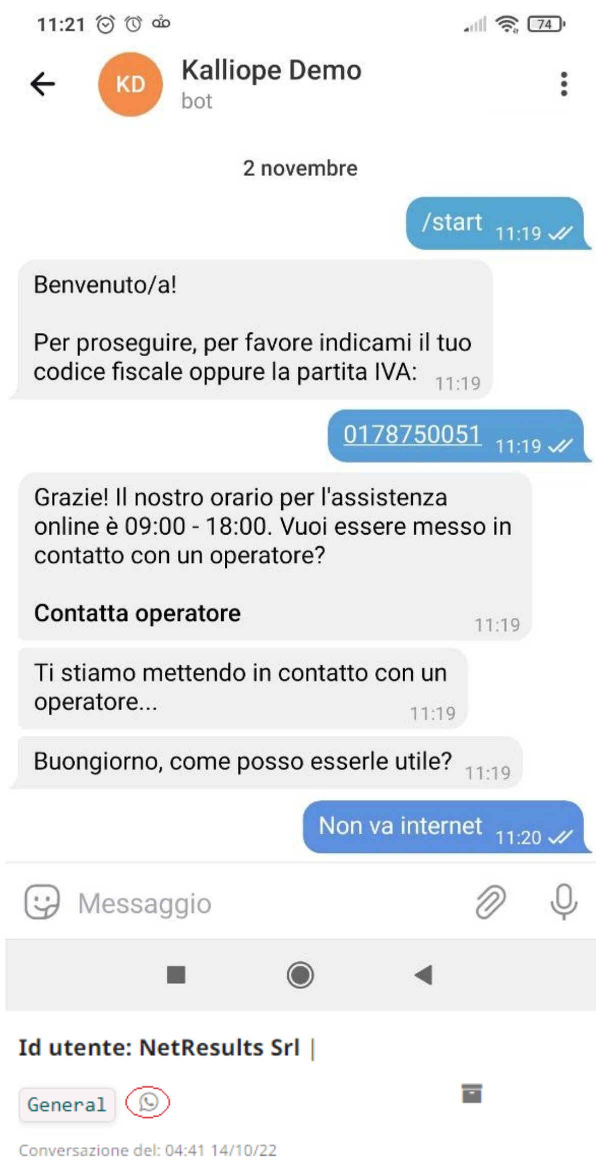
- **Create ticket "**: allows you to create a ticket based on the issues encountered during the conversation.

On the left, there is a list of "recently closed" conversations and conversations that are still active.

Interaction via Whatsapp, Telegram, Messenger

With the "Multi-Channel" option, the customer can connect the livechat engine with Telegram, Messenger and/or Whatsapp social channels.

In the main operator screen seen above, the origin of the request will be indicated via the icon characterizing the social (circled in red):



Interaction via phone

The voicebot module allows you to resolve service-related issues via phone. One proceeds by dialing the number to be called, an IVR is played listing the available options, and, choosing the help and support option, one is guided through the creation of a ticket described by voice. In fact, the service allows you to filter technical support requests by categorizing the type of request and setting up ticket creation automatically. In case you are unable to describe the disservice, you can request to speak to an operator, an extension from the Kalliope central office will then be contacted to resolve the problem. The voicebot is configurable; you can choose whether the customer has to wait for a “beep” to respond or whether the flow is continuous, thus overlapping with the voicebot.

5.1.7 Backup

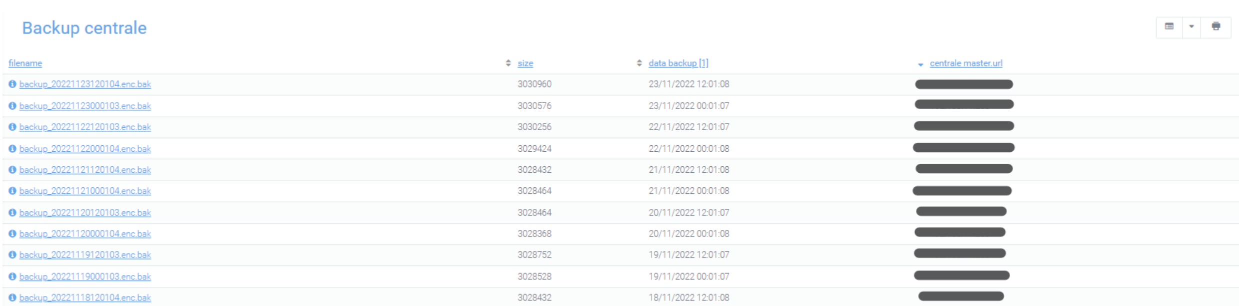
> **Warning:** This section is visible only to admin users and can be reached via the menu path Voip admin > Central Backup.

The backup section allows you to manage and perform backups of your telephone exchange directly through the web interface, so you can have more autonomy in managing this process.

Backup list

Backups can be made manually via the “+” button, once pressed a row will be automatically added to the backup list with the corresponding newly generated backup file of the telephone exchange. In the list, backups are identified by:

- Filename: name of the file
- Size: size of the file
- Backup date
- Master central IP: IP address of the master central



Backup centrale

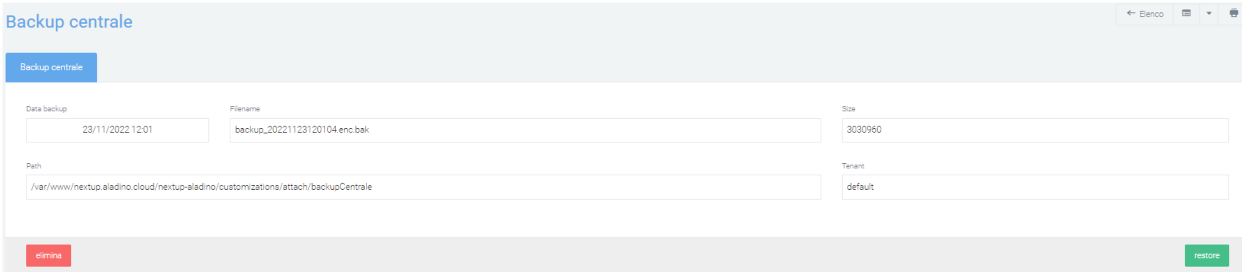
filename	size	data backup [1]	centrale master.url
backup_20221123120104.enc.bak	3030960	23/11/2022 12:01:08	[redacted]
backup_20221123000103.enc.bak	3030576	23/11/2022 00:01:07	[redacted]
backup_20221122120103.enc.bak	3030256	22/11/2022 12:01:07	[redacted]
backup_20221122000104.enc.bak	3029424	22/11/2022 00:01:08	[redacted]
backup_20221121120104.enc.bak	3028432	21/11/2022 12:01:08	[redacted]
backup_20221121000104.enc.bak	3028464	21/11/2022 00:01:08	[redacted]
backup_20221120120103.enc.bak	3028464	20/11/2022 12:01:07	[redacted]
backup_20221120000104.enc.bak	3028368	20/11/2022 00:01:08	[redacted]
backup_20221119120103.enc.bak	3028752	19/11/2022 12:01:07	[redacted]
backup_20221119000103.enc.bak	3028528	19/11/2022 00:01:07	[redacted]
backup_20221118120104.enc.bak	3028432	18/11/2022 12:01:08	[redacted]

Backup detail

Clicking on the file name in the list takes you to the detailed view of the backup file where the same information as in the list is contained, but with the added specification of:

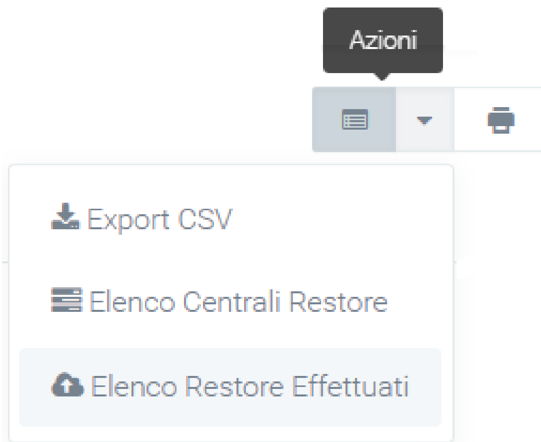
- Tenant: in case the central unit is multitenant, you need to specify the tenant with which you are associated. Instead, the “default” value will be set automatically
- Path: where the backup file is allocated.

Using the green button at the bottom right, you can restore the backup file, to reload it within the central, so that you have all the data available at the time the backup was made. Via the red button at the bottom left, delete, you can delete the backup file.



List of restores performed

In this section, which can be reached via the menu on the right, is the list of restores performed, displayed by: outcome, date and url of the control unit on which it was performed, and whether in case it failed.



Central restore list

In this section you can indicate the secondary power plants on which to activate the restore. You then indicate on which other power plants you can restore a backup file that is already in the system. While the primary central unit that is being backed up to can be viewed by going to: **System Preferences > Voip**.

If it turns out to be the main and only central unit entered, it is shown again in "master central units" with the essential information to make the backup.

Using the "New central" button, you can specify secondary central units, you can decide to make periodic backups for each central unit. Ex. A company may find itself managing multiple telephone exchanges, so you may decide to make periodic backups for each. Periodically, thanks to cron, the central can make backups of the files.

5.1.8 Monitoring

Description

Monitoring your company's IT infrastructure is essential to ensure its proper functioning and to detect any malfunctions. By constantly checking for problems that are detected, opening tickets as needed, and then sending notifications to the affected areas and contacts, it is possible to have total control over processes. All problems that occur are stored and displayed in a single section with a corresponding level of severity. This level can be fully customized via the appropriate section, indicating with which type of problem, or group of problems, a certain severity should be associated. The

Monitoring module thus offers a tool to quickly and efficiently manage all the problems detected on the assets in your company, saving you time and money.

To reach it, you need to press on “Monitoring” in the menu on the left.

Structure and configuration

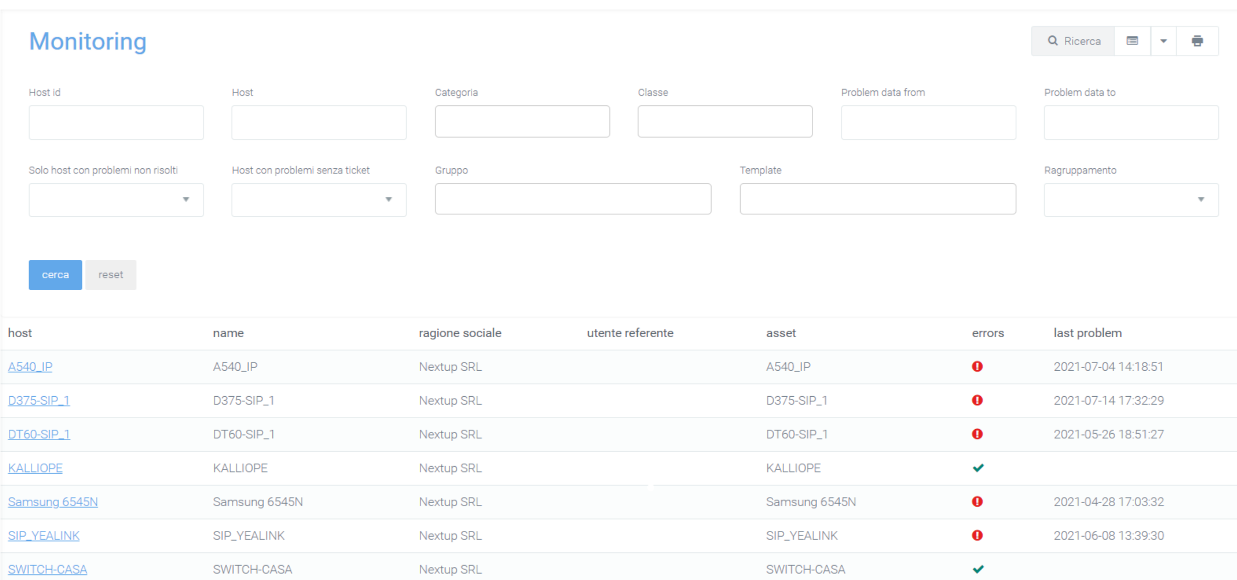
The module is divided into several sections, but the main ones-where basic monitoring information is reported-can be found in the Host and Problems modules. Also of fundamental importance is the Dynamic Classes section, whose structure and configuration follows the two main sections.

HOST

In the Support > Assets section, creating an asset and checking it as “to be monitored” will automatically create a host with the necessary information to be monitored.

In the list of hosts (accessible via Monitoring > Hosts) you can view for each host:

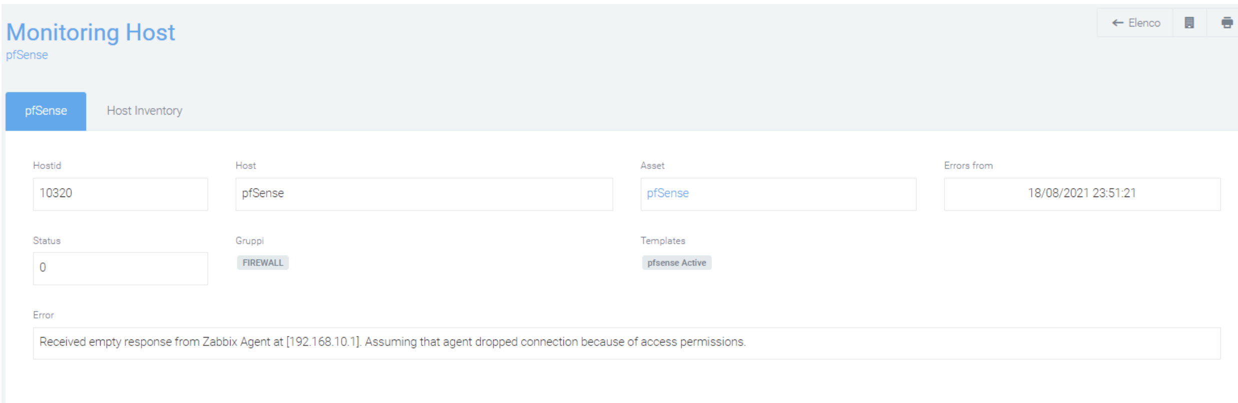
- Host name
- Id of the host
- Category
- Account
- Final customer
- Referring user
- Last problem: date of detection of the last problem



Thanks to the configuration of classes in the “dynamic class” section, it is possible to define times for which asset monitoring should be active: if some equipment at certain times is turned off, monitoring will not report errors.

Host detail

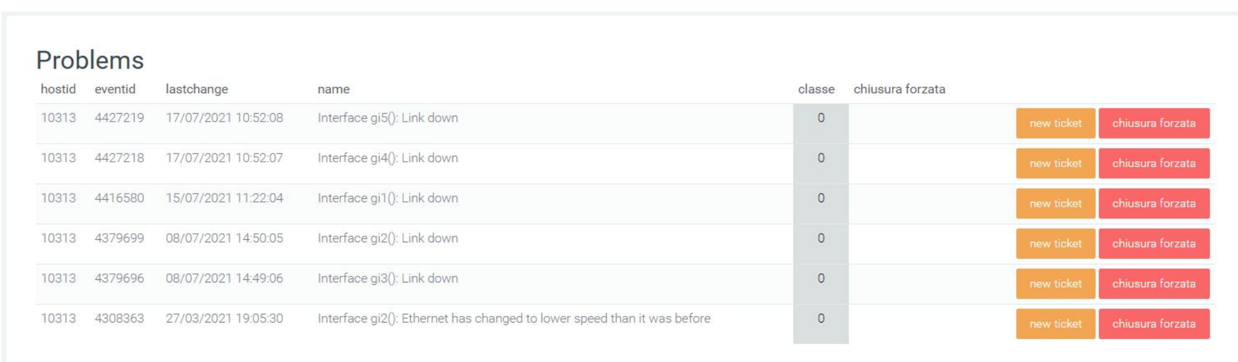
By selecting a host in the list (by clicking on the name), it is possible to enter the host detail mask, where it will be possible to view the monitoring details of the apparatus and the ASSET data. To change the information of a host, you must change the information of the related connected asset. Within the detail mask there is the first section divided into two tabs. In the first one, with the name of the host, some of its essential information is shown. In the second, called "Host Inventory," all the detailed information is shown



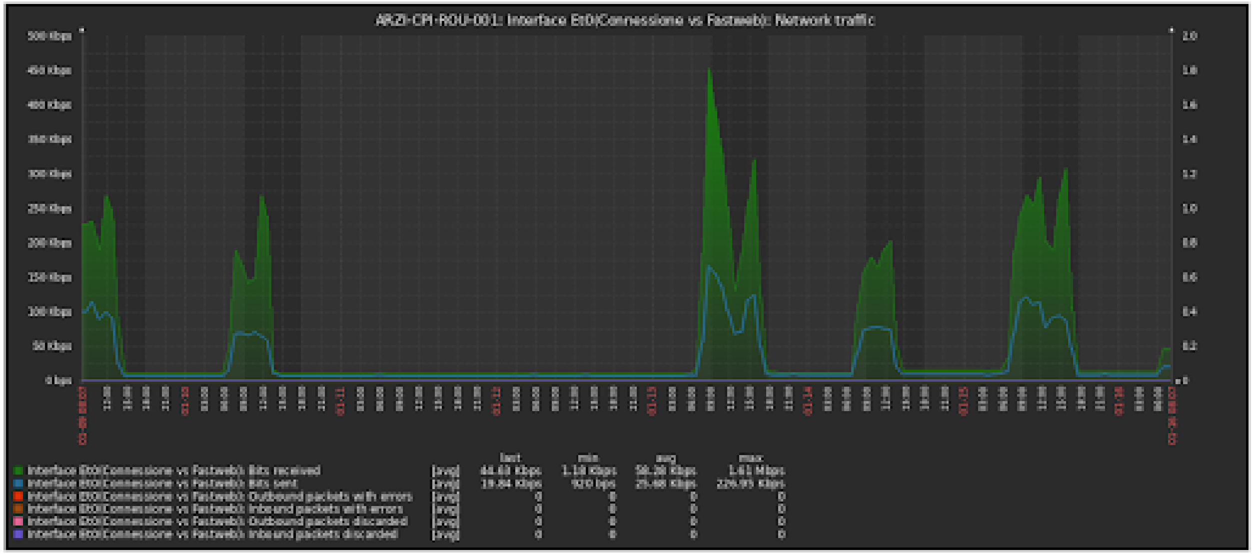
Under these two tabs a list of all the problems detected by the system is visible and for each one it will be possible to open tickets through a special button (new ticket) that, if indicated, will lead to the sending of specific ticket opening notifications to the various contacts. The notification email is structured with a customizable template that will contain all the asset data necessary for the unambiguous identification of the equipment for a rapid resolution of the fault:

- Host id: host identification code
- Eventid
- Lastchange: date of the last change in the host problem
- Name: name of the problem
- Class: indicates the severity of the problem (inserted via dynamic classes)

In addition to the ability to create tickets, it is also possible to perform a forced closure of the problem, so that it no longer appears as such and is not visible in the problems list. Any changes to the ticket will be notified to the various contact persons. Forced closure of the problem prevents it from appearing further on the platform.



In the Graph section, you can select the type of graph you want and the period to be considered to display the time graph of monitored resource utilization for that specific host (e.g., download or upload trend for specific router).



Host Filters

You can filter the data in the list using the "Search" button in the upper right corner and enter:

- Account
- Final Customer
- Host id: host identification code
- Host (name)
- Category of the host to be monitored
- Class: identifies the priority of the detected malfunction
- Problem data from: start date for the search for problems
- Problem data to: end date of the time interval in which the search for problems is performed
- Only hosts with unresolved problems (yes or no)
- Hosts with problems without tickets (yes or no)
- Group
- Tamplate
- Grouping (zip code, municipality, province, identifier)

Problems

This section (accessible via Monitoring > Problems) reviews all problems that have been detected, with their customizable severities in the form provided. It may happen that some problems resolve themselves, in which case there will be a check mark in the table, otherwise - if not - a cross.

The problems in the list are classified by classes that identify the priority of the detected malfunction.

Information about the problems listed in the table includes:

- host name: the name of the host

- account
- final customer
- firstchange and lastchange: date of the first and last detected change
- resolution time: time to resolve the problem
- name: name of the problem
- class: rating of the severity of the problem
- hide: indicates whether the problem has been solved (yes) or is still pending (blank)
- date forced closure (if it happened)
- ticket.numbering: ticket number of the ticket linked to the problem
- ticket.title: title of the ticket linked to the problem
- ticket.status: status of the ticket linked to the problem
- intermittent: whether the problem is intermittent or not


Monitoring Problems

Ricerca

host.name	firstchange [1]	lastchange	name	classe [3]	hide [2]	data chiusura forzata	ticket.numerazione	ticket.titolo	ticket.stato
DT60-SIP_1	31/05/2020 15:03:27	31/05/2020 15:07:27	Unavailable by ICMP ping	9	si				
D375-SIP_1	31/05/2020 15:03:29	31/05/2020 15:06:30	Unavailable by ICMP ping	4	si				
D375-SIP_1	31/05/2020 15:42:30	31/05/2020 15:54:29	Unavailable by ICMP ping	4	si				
DT60-SIP_1	31/05/2020 15:43:27	31/05/2020 15:45:27	Unavailable by ICMP ping	9	si				
D375-SIP_1	31/05/2020 15:58:29	31/05/2020 17:31:29	Unavailable by ICMP ping	4	si				
DT60-SIP_1	31/05/2020 17:36:27	03/06/2020 07:42:27	Unavailable by ICMP ping	9	si				
D375-SIP_1	31/05/2020 17:36:30	03/06/2020 07:41:29	Unavailable by ICMP ping	4	si				
D375-SIP_1	03/06/2020 11:32:29	03/06/2020 11:39:29	Unavailable by ICMP ping	4	si				

Click on the host name to enter the "host detail" section (see above).

Filter Problems

Clicking on the "search" magnifying glass will allow you to filter problems by:

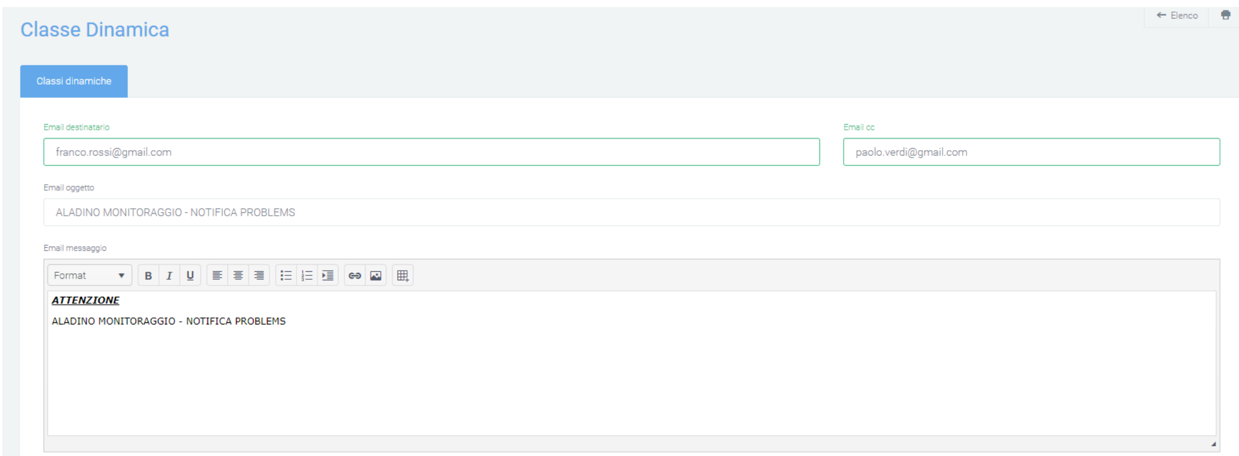
- Account
- Final customer
- Host id: host identification code
- Host name: name associated with the host
- Ticket: possibility to search by ticket title or description
- Problem description
- Category
- Class: priority rating of the malfunction
- Problem data from: start date for searching for problems
- Problem data to: end date of the time interval in which the search for problems is performed
- Only unresolved problems (displaying yes or no)
- Only solved problems but with open tickets (yes or no display)
- Only problems without class (display yes or no)

- Solved problems with resolution time (indicating the time)
- Group
- Template

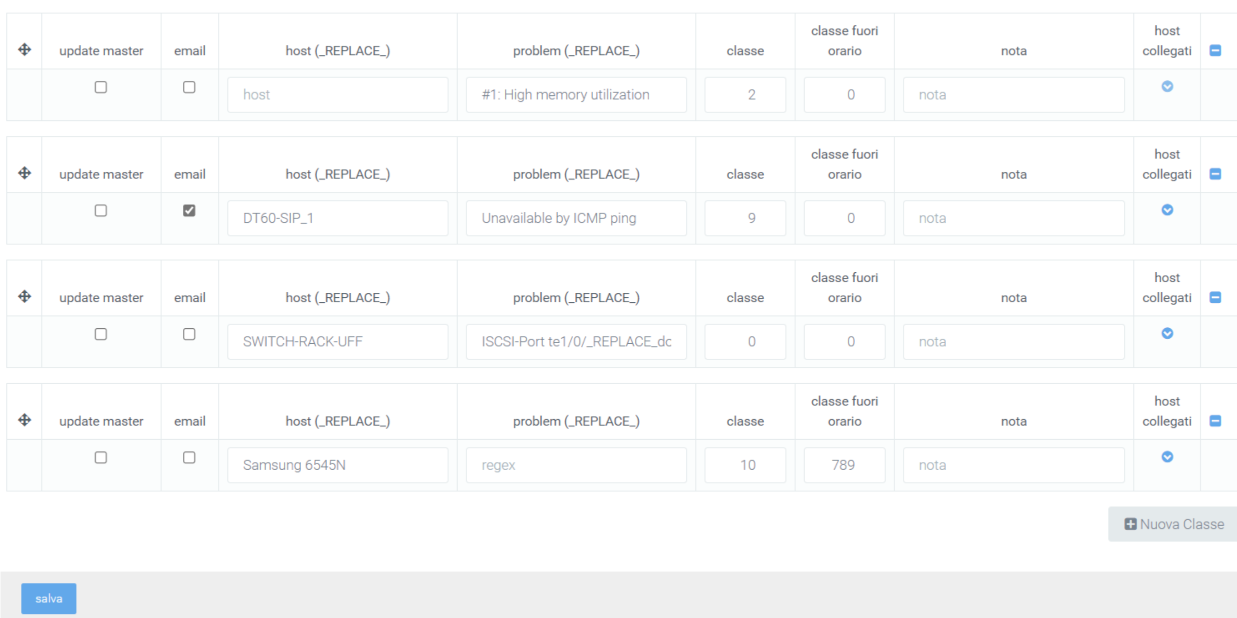
Dynamic classes

It is possible to associate each problem of each host with a specific class. Classes are needed to assign the severity level to each problem so that reports are handled with the correct priorities.

The class is identified by a number and defaults from -1 to 9, where -1 indicates out-of-hours problems and 0 is assigned to problems that you do not want to monitor and for which the system (although it logs them in the database) does not report failures. In the customization of the section, you can configure the recipients of email notifications, specifying Subject and body of the email.



The platform assigns a class to a problem based on specific logic, taking into account that in the monitoring machine the recorded severities range from a minimum of 0 and a maximum of 5. However, it is possible to configure new classes by associating them with problems, via the "+ New Class" button.



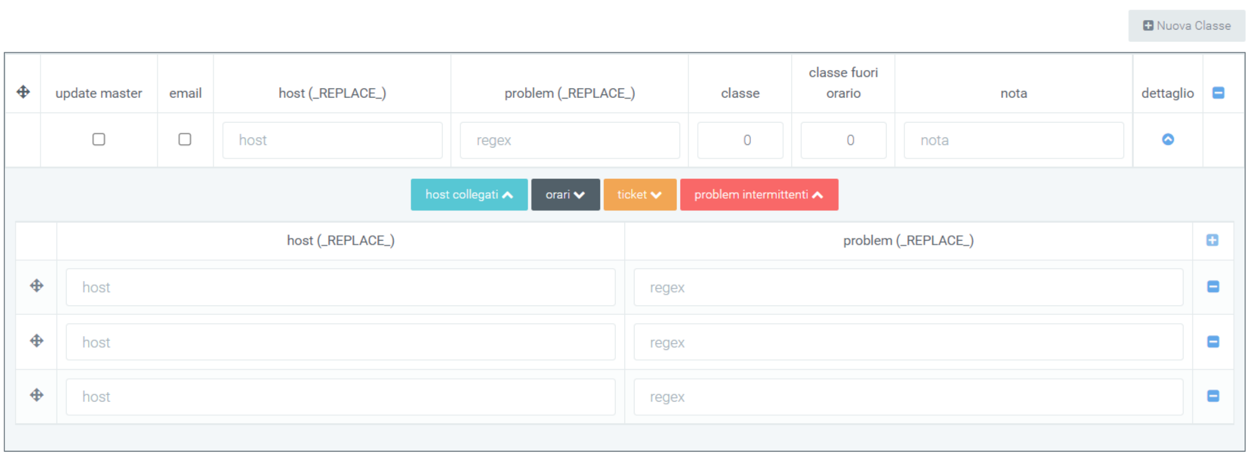
Classification is based on activated filters that work on the host name and error message description. The placeholder "_REPLACE_" can be used to refine the search for 'hosts' and 'regexes' for subsequent class assignment to problems. Examples:

- By entering ESO-CPI-ROU-001 in the 'hosts' field, the system matches only hosts with this name
- By entering **ESO-CPI-ROU_REPLACE_** in the 'host' field the system matches all hosts starting with ESO-CPI-ROU
- By entering _REPLACE_ROU_REPLACE_ in the 'hosts' field, the system matches all hosts that contain ROU

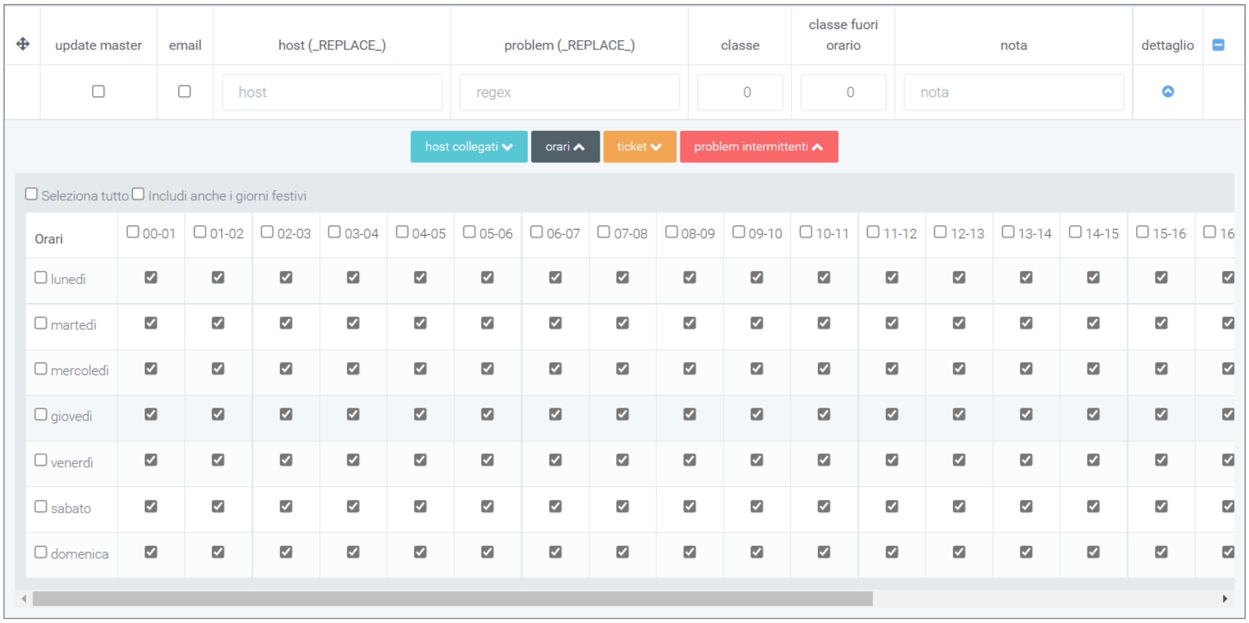
The same logic can be applied to the 'regex' field, which is the description of the error that generates the problem. Each dynamic class that is created provides a detail section in which specifications can be added to specify the cases for which a problem will have one specific class instead of another.

- connected hosts
- schedules
- tickets
- intermittent problems

Pressing on the first button, "connected hosts," will open a configuration mask with the fields "host" and "problem" with a "+" on the right. This is necessaryb to add constraints on the main classification, that is, a given class will be assigned only if there is a problem in the connected hosts, among those entered. Only a host name and error description may be entered, or both linked.



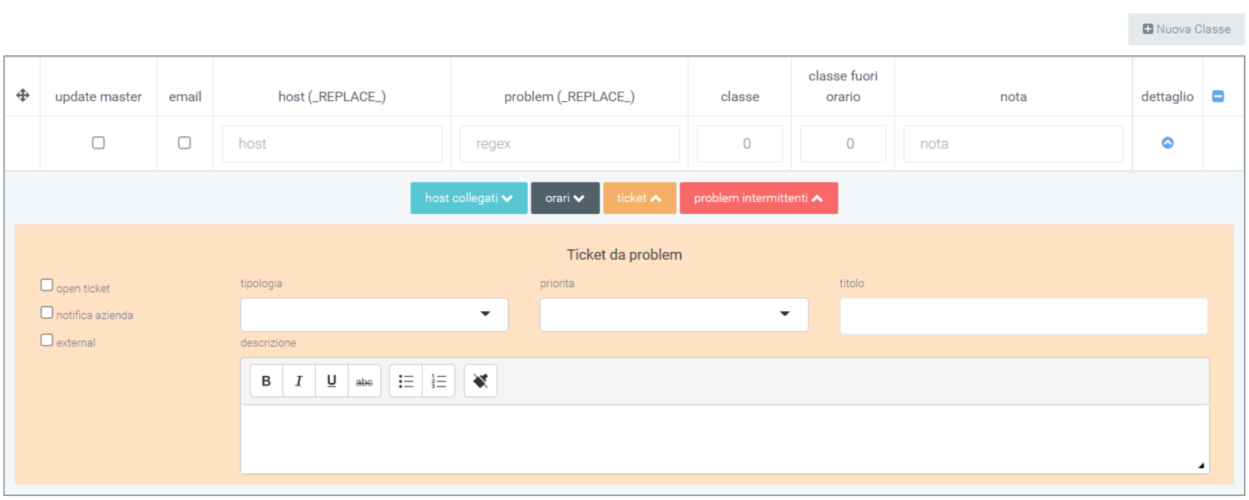
The second button, "schedules," shows a grid divided by days and time slots where it will be possible to enter specific times when a host is to be monitored. To indicate this time period, simply select the flag that intersects day and time slot of interest. Normally, holidays are not counted, but you can change this by selecting the "include holidays" flag in the upper left corner. Instead, to select all days, check the "select all" flag.



By clicking on "ticket," the mask displayed allows you to schedule the opening of a ticket for a problem. The problem for which the ticket will be opened is the one indicated in the main box for which the detail is being created. Some parameters of the ticket should be indicated in this mask:

- type
- priority
- title
- description

On the left are 3 flags, "open ticket," "notify company," and "external," which will be used respectively to: open the ticket upon problem detection, notify the company for which the ticket is being opened, and indicate the open ticket as a ticket in external, i.e., referring to suppliers.



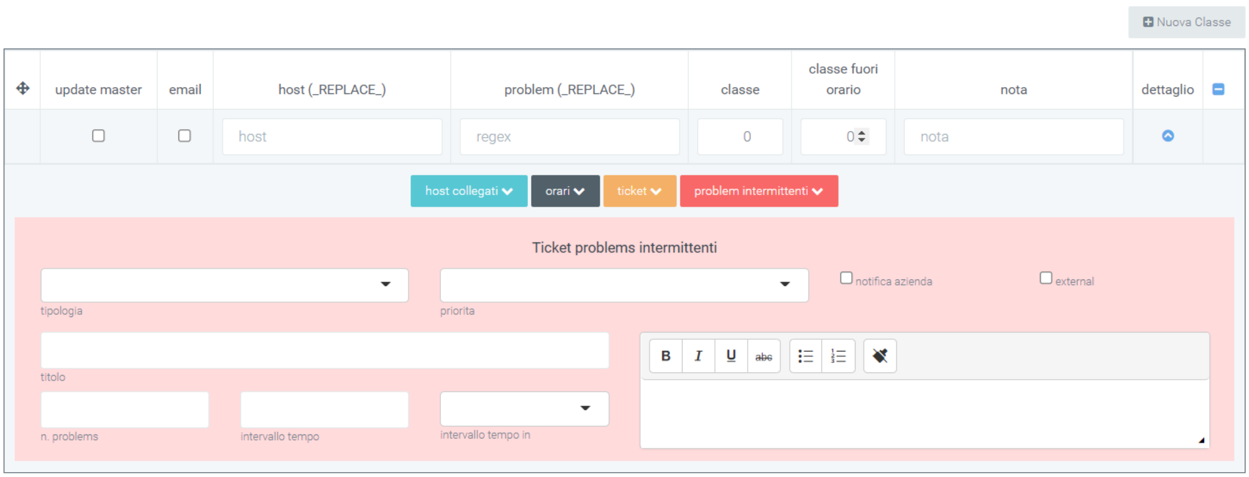
The last button, "intermittent problems," allows you to open tickets for intermittent problems as well. They are detected problems that are resolved independently in a certain time interval; however, some of them may cause disruptions. To handle this type of problems, specific tickets can be created. E.g. If we know that a certain problem is created every time interval (minutes, hours), to avoid a ticket being created tot times, it is convenient to fill out the following section.

After clicking on the button, a mask will be shown in which the following should be entered:

- ticket type
- priority
- whether to notify the company of the ticket opening
- whether it is a ticket for suppliers (external)
- ticket title
- description

The time interval and criteria for which a problem is defined as intermittent, are customizable via 3 cells:

- no. problems: number of problems that occurred in a certain time interval
- time interval: number representing an interval for performing the check for intermittency of a given problem
- time interval in: unit used to indicate the time interval (minutes, hours, days).



Template –> ability to upload or update templates via file.

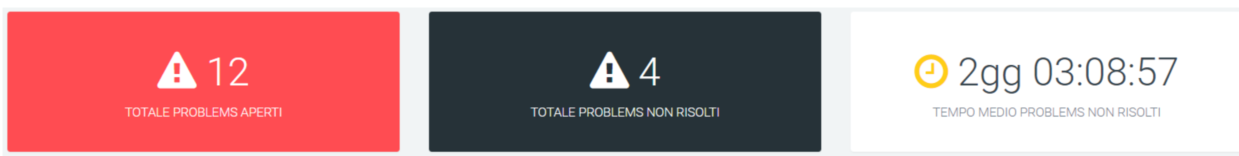
Template repository –> point to / create as if it were a repository, you save templates to then be pointed to within the template section.

Report

In the "Monitoring" section, four main reports are made available that allow you to intuitively understand what is the status of the various issues detected. The reports are divided by category and location; there is also a report reserved for detailed data analysis. The four reports are useful for getting an overview at the generalized level of what is the status of the various problems that are detected. The four reports are divided into:

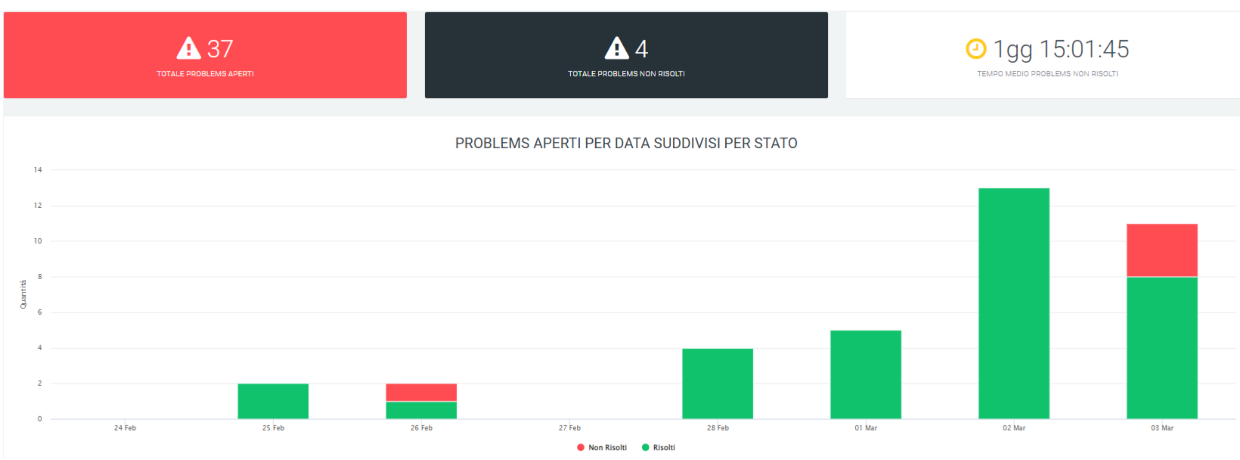
- Reports: the following section is a general section in which two graphs, one pie chart and one histogram, showing the ratio of resolved errors (DONE) to those that are still open (ERRORS) can be viewed

- Analysis Report: in this section you can find a dashboard where the number of problems with their status (unresolved, still open) and the average time for unresolved problems are reported.

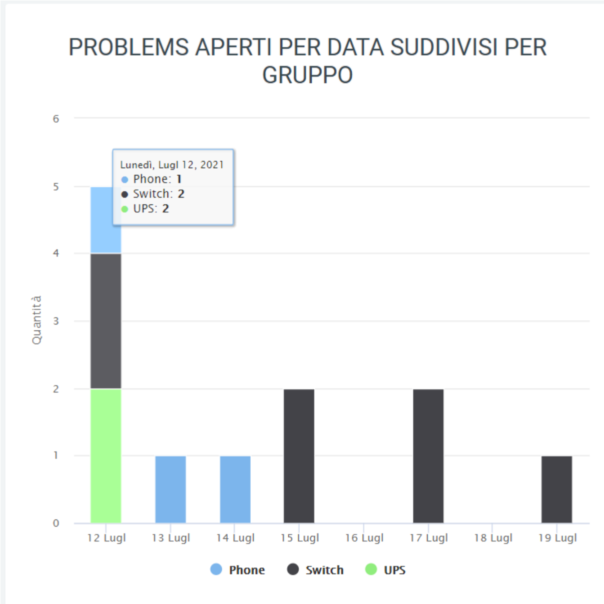


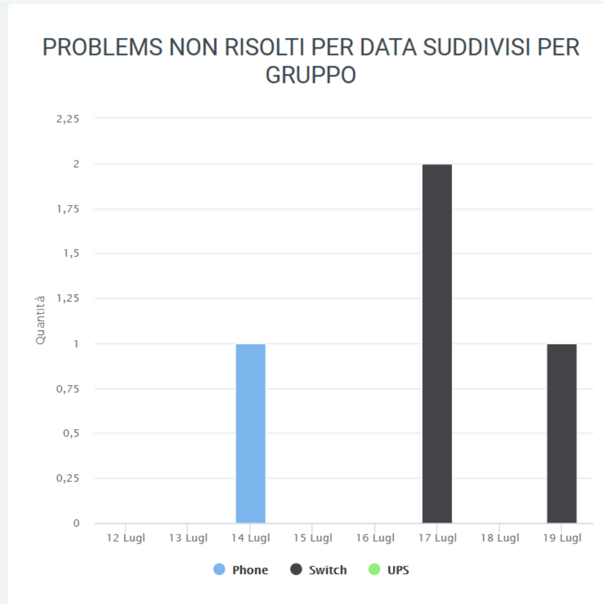
Graphs representing the progress of problems are also present:

- the first represents open problems broken down by date and by status (solved/unsolved)

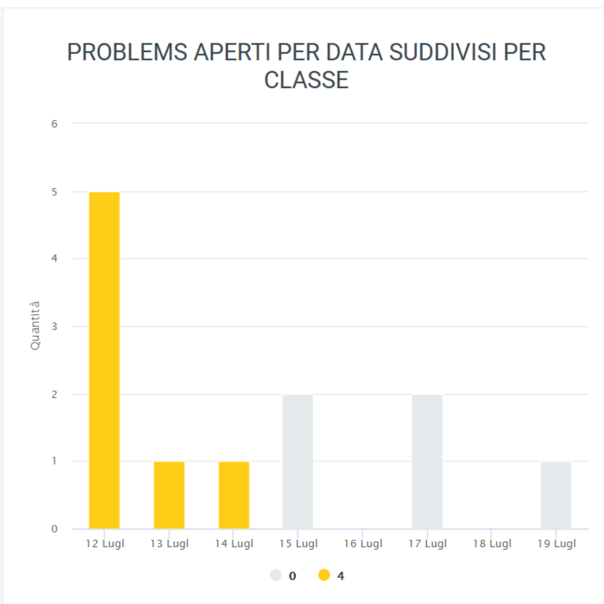


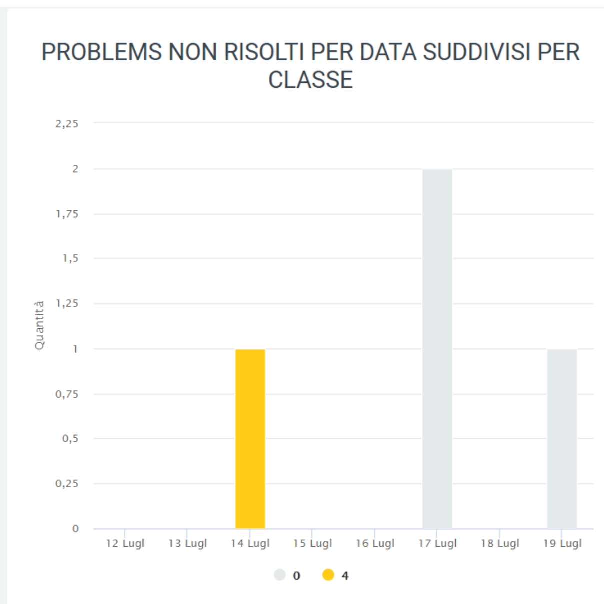
- the second and third graphs represent respectively open problems broken down by group and broken down by date, and unresolved problems, also broken down by group and broken down by date





- the fourth and fifth charts represent open or unsolved problems, again by date, but broken down by class

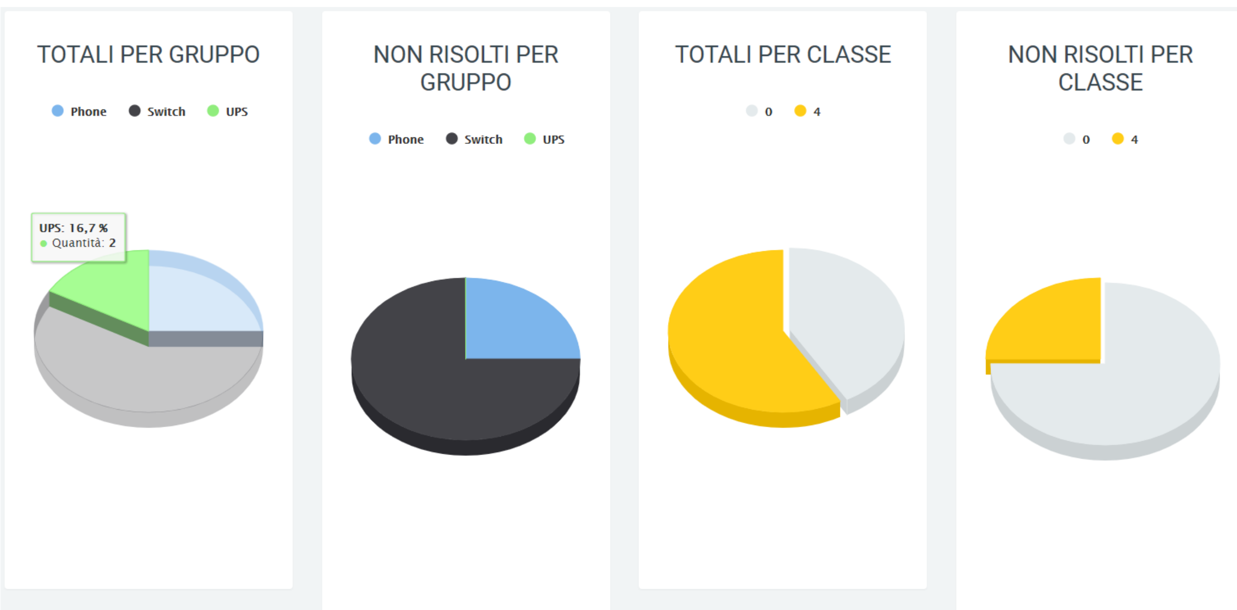




- The sixth graph consists of the total representation of problems by date, with the ability to display: - open problems - problems solved - average of open and closed problems - trend of open problems - trend of solved problems - trend of open and solved problems

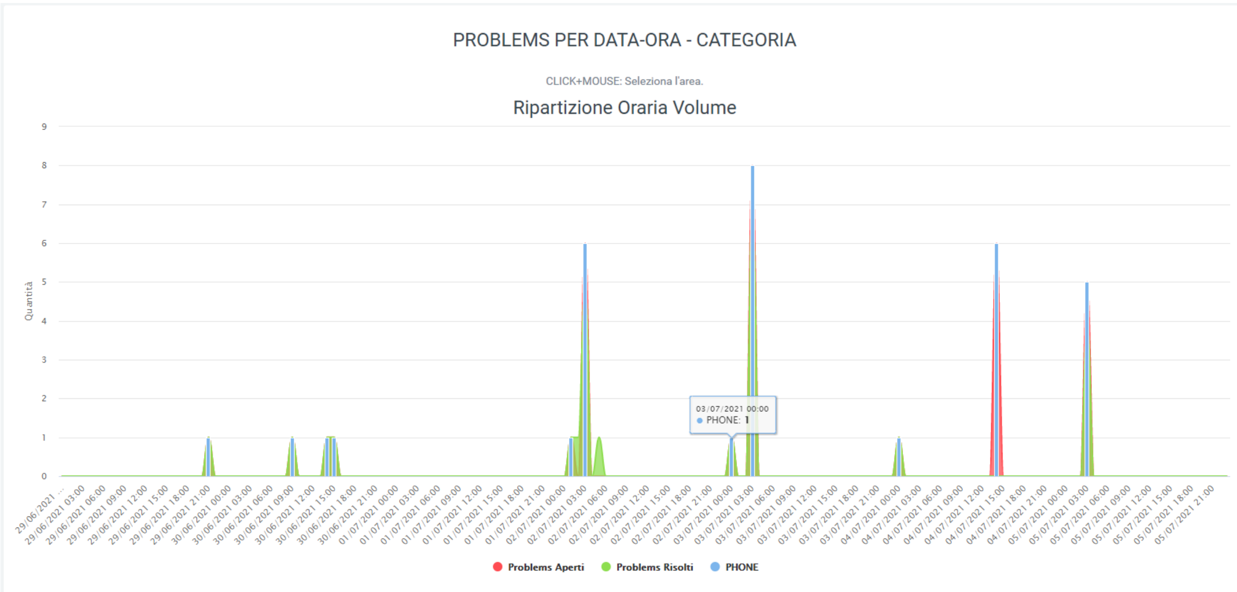
- The last graphs are four pie charts representing: - The total problems broken down by group - The problems not yet solved broken down by group - The total problems broken down by class - The problems not yet solved broken down by class

As in all graphs on the platform, it is possible to remove or add to the display some data via legend and to view quantities by scrolling with the mouse over the graph.



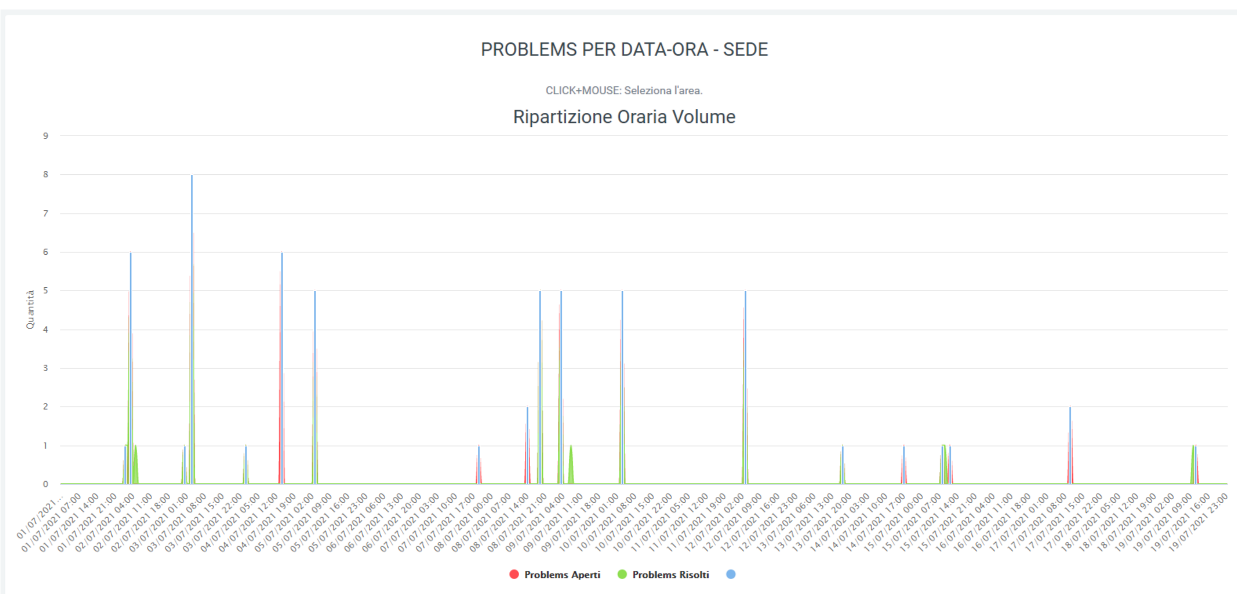
Date-time report by category

In this section, one can observe the progress of problems over a period of time indicated through filters (which can be activated thanks to the "search" button) in dates subdivided in turn by time slots. Again through the filters, one can choose which category of problems to include in the graph. The states (open and solved problems) and categories are displayed, also described in the legend.



Date-time report by location

As in the previous section, it is also possible to observe the progress of problems with a daily and hourly breakdown (where each day has its own hourly breakdown), but with the only difference that it is not the categories but the locations that are displayed. Again using filters ("search" button) it is possible to include the location whose data you want to view, which is also visible in the legend.



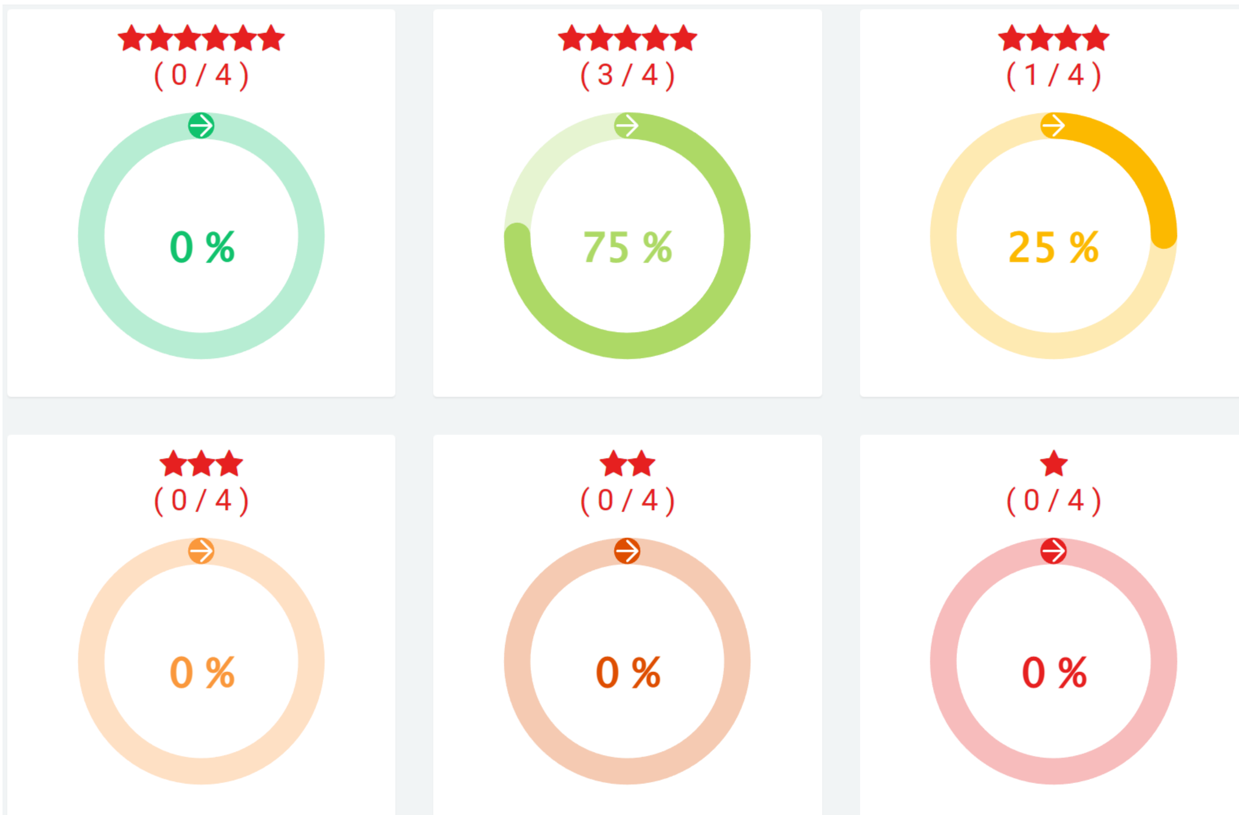
5.1.9 Evaluation Module

Service Description.

The evaluation module allows the analysis of a customer's call experience through the request, at the end of the call, of a value from 1 to 6 to be entered via numeric keypad.

Analysis of the values obtained can be done in the "Voip Routing > Service Report" section of the KalliopeNexus platform where, via graphs and tables, customer ratings are summarized.

Each graph shows out of the total calls, how many customers typed a particular value, indicated by the star icon. The data are also summarized by percentage in order to make the visualization clearer.



At the bottom, there is the "Evaluation Detail" table, which summarizes the same data:

Dettaglio Valutazione

VALUTAZIONE	NUM. CHIAMATE / TOTALE	PERCENTUALE
★★★★★★	0 / 4	0%
★★★★★	3 / 4	75.0%
★★★★	1 / 4	25.0%
★★★	0 / 4	0%
★★	0 / 4	0%
★	0 / 4	0%

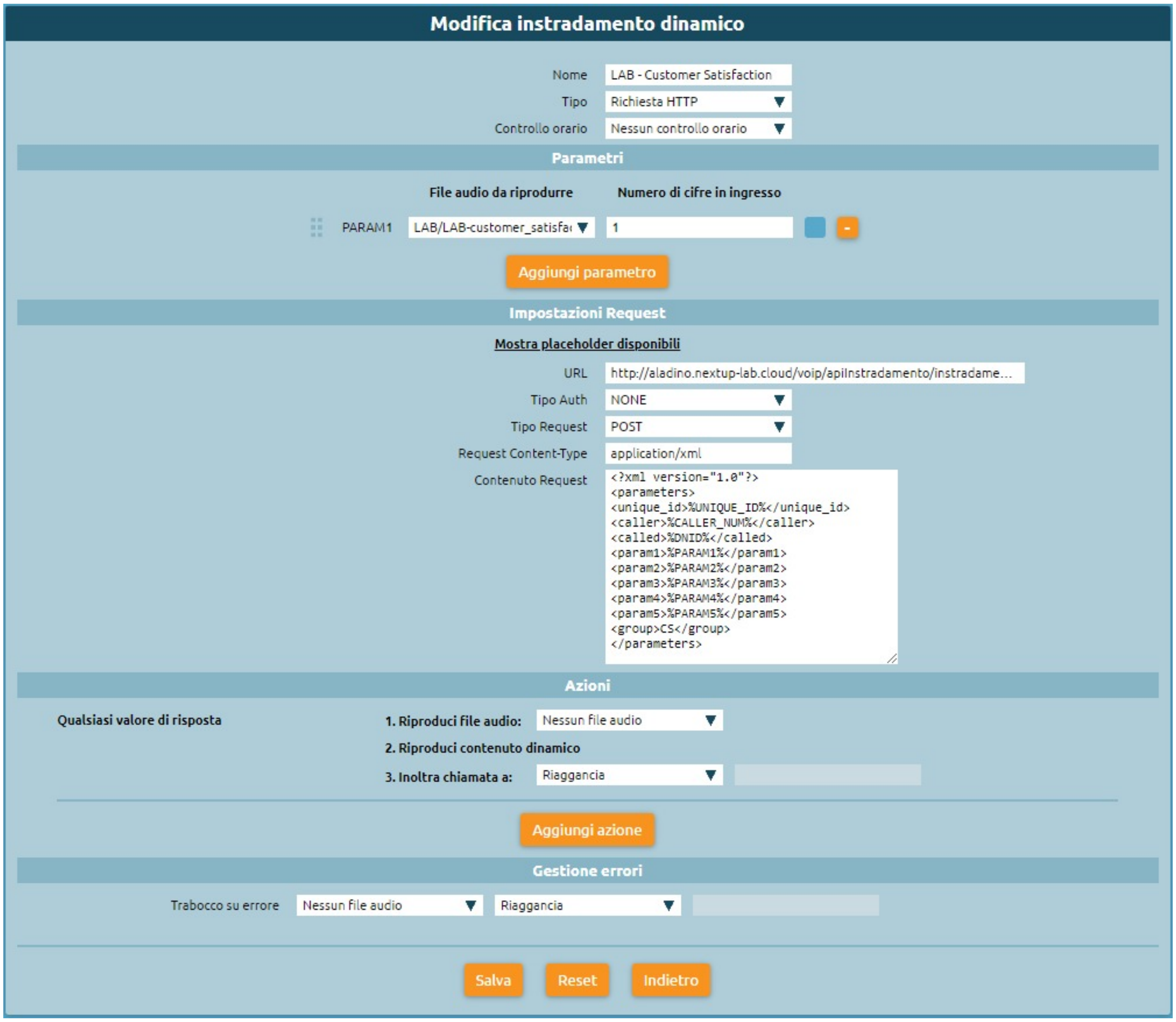
Configuration

To configure the function of the evaluation module, it is necessary to create a new dynamic routing on the Kalliope central unit. In the "Parameters" section, you need to select the audio that will contain the evaluation instructions, and as "Number of input digits" you only need to enter "1." In the "Request Settings" section, indicate:

- **URL**: http://aladino.nextup-lab.cloud/voip/apiInstradamento/instradamento
- **Tipo Request**: POST
- **Request Content Type**: application/xml
- **Contenuto della request**:

```
<?xml version="1.0"?>
<parameters>
<unique_id>%UNIQUE_ID%</unique_id>
<caller>%CALLER_NUM%</caller>
<called>%DNID%</called>
<param1>%PARAM1%</param1>
<param2>%PARAM2%</param2>
<param3>%PARAM3%</param3>
<param4>%PARAM4%</param4>
<param5>%PARAM5%</param5>
<group>CS</group>
</parameters>
```

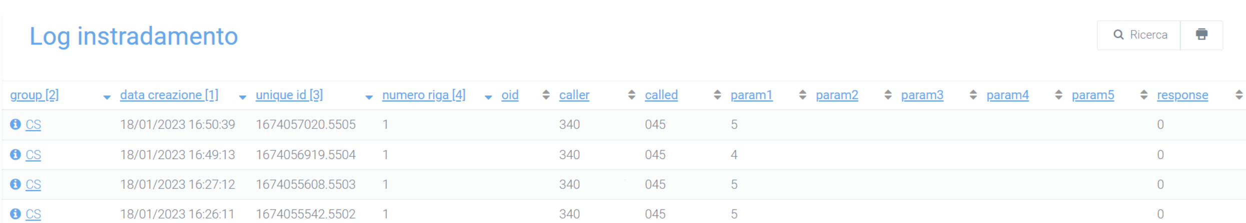
In the "Actions" section, you can leave the fields blank as default or configure them as needed.



Configuration and details on Kalliope Nexus

On the Kalliope Nexus platform it will be necessary to go to the "Voip routing" section to create a new one. The configuration consists only of entering within the "group" field the parameter "CS". The rest of the fields should be left blank. The moment the customer enters the rating from 1 to 6, it will be shown in the report, as explained in the previous section (Service Description).

In the "Voip Routing > Log" section, you can view the routing log, in which you can analyze the detail of the customer's response. In fact, in the log, under the parameter "group" is indicated CS and, in addition to the identification of the calling and called number, the parameter "param1" indicates the rating (from 1 to 6) for each call.



Log instradamento

group [2]	data creazione [1]	unique id [3]	numero riga [4]	oid	caller	called	param1	param2	param3	param4	param5	response
CS	18/01/2023 16:50:39	1674057020.5505	1		340	045	5					0
CS	18/01/2023 16:49:13	1674056919.5504	1		340	045	4					0
CS	18/01/2023 16:27:12	1674055608.5503	1		340	045	5					0
CS	18/01/2023 16:26:11	1674055542.5502	1		340	045	5					0

Note: In case you want to request evaluations of different types from the client, and thus obtain different feedback (not just one question-assessment), you can configure multiple routings that will, however, refer to the same routing (CS). As a result, an overall assessment of all questions will be obtained in the report.

CHAPTER

SIX

API REST DOCUMENTATION

6.1 API Rest

6.1.1 General description

KalliopePBX offers several REST APIs that can be invoked by external systems to execute device actions (e.g. creating extensions, generating and downloading backups, etc.) or consultation actions (e.g. downloading the CDR, etc.).

APIs can be accessed through HTTP or HTTPS with authentication, following the mechanism described below.

6.1.2 Authentication mechanism

The REST API authentication mechanism uses the same user credentials as those needed to access the web. Specifically, each request must contain a custom single-use HTTP header named X-authenticate. The authentication procedure is one-way, i.e. it does not require the server to send a challenge, and all information necessary for authentication is generated from the client side.

Because the headers are single use, knowledge of the X-authenticate header of a request does not allow one to send an illicit request, thus preventing replication attacks.

X-authenticate header construction

The header is comprised of a sequence of data and a digest constructed from that data along with the password of the user making the request. Here is an example of a complete X-authenticate header:

```
X-authenticate: RestApiUsernameToken Username="admin", Domain="default", Digest=
↪"+PJg7Tb3v98XnL6iJVv+v5hwhYjdzQ2tIWxvJB2cE40=", Nonce="bfb79078ff44c35714af28b7412a702b
↪", Created="2016-04-29T15:48:26Z"
```

The information transmitted in the request are:

- **Username**: the username assigned to the user (e.g. admin).
- **Domain**: the domain of the tenant that the user belongs to. In single-tenant systems, the predefined domain is "default".
- **Nonce**: a random hexadecimal string of at least 8 characters generated by the client. A new one must be generated for each request, as the system keeps track of all previously used nonces for their lifetime of 5 minutes. After 5 minutes, it is possible to reuse a nonce; protection from attacks will be guaranteed by the nonce creation timestamp.

- **Created**: the nonce creation timestamp; if the time of reception differs from the time set on KalliopePBX by more than 5 minutes, the request will be considered invalid and refused. This guarantees that an X-authenticate header cannot be reused at any time. The timestamp format is “YYYY-MM-DDThh:mm:ssZ” (UTC timezone).

The Digest field is a hash of the above data along with the user password, generated by applying the SHA-256 hash algorithm to the concatenation (with no delimiters) of Nonce, digestPassword, Username, Domain, and Created:

```
Digest = sha256(Nonce + digestPassword + Username + Domain + Created)
```

Since KalliopePBX saves passwords in the form of non-reversible hashes, the digest must be generated using the hash of the password obtained using the same method as KalliopePBX (digestPassword).

The hash function that produces the digestPassword requires the user password and a salt string, which is unique for each tenant. The salt can be obtained from KalliopePBX with a designated REST API that can be invoked anonymously (i.e. without authentication):

```
rest/salt/<dominio_tenant>
```

where the string <dominio_tenant>, in the case of a single tenant machine, is “default”. In the case of multi-tenant system, in order to use the API available to pbxadmin, it is possible to obtain the relative salt through the API

```
rest/salt/pbxAdmin
```

The formula is as follows (the { e } characters are part of the string to which SHA-256 is applied and must be inserted):

```
digestPassword = sha256(password{<salt>})
```

For example, if the password is “admin” and the salt is “b5a8fdcf2f8d5acdad33c4a072a97d7a”, the resulting digest-Password is

```
digestPassword = sha256(admin{b5a8fdcf2f8d5acdad33c4a072a97d7a}) =␣
↪dd7b0be7fa37d6cbaf0b842bf7532f229cb79ab8d54d509c2aa7eea27a53cd5e
```

So the following information:

```
Nonce: bfb79078ff44c35714af28b7412a702b
digestPassword: dd7b0be7fa37d6cbaf0b842bf7532f229cb79ab8d54d509c2aa7eea27a53cd5e
Username: admin
Domain: default
Created: 2016-04-29T15:48:26Z
```

will result in:

```
Digest =␣
↪base64(sha256binary(bfb79078ff44c35714af28b7412a702bdd7b0be7fa37d6cbaf0b842bf7532f229cb79ab8d54d509c2a
↪04-29T15:48:26Z))
```

which is +PJg7Tb3v98XnL6iJVv+v5hwhYjdzQ2tIWxvJB2cE40=.

Note: sha256binary(..) is the sha256(..) function that outputs a binary string rather than a hexadecimal one.

The complete string to insert in the header is therefore: .. code-block:: console

> X-authenticate: RestApiUsernameToken Username=”admin”, Domain=”default”, Digest=”+PJg7Tb3v98XnL6iJVv+v5hwhYjdzQ2tIWxvJB2cE40=”, Nonce=”bfb79078ff44c35714af28b7412a702b”, Created=”2016-04-29T15:48:26Z”

6.1.3 API list

For firmware versions up to 4.8.x, the updated API list can be viewed at

```
http[s]://KALLIOPE_IP_ADDRESS/api/doc
```

Starting from firmware version 4.9.4 it is also available a Postman collection (that replaces the sandbox integrated in the documentation page); this collection integrates the code to automatically add the required authentication header (it is necessary only to set the IP address of the PBX and the username/password credentials of the user to invoke the API).

Accessing the KalliopeTribe provides a video demonstrating the use of that collection to quickly invoke the Kalliope API.

The collection file can be downloaded from these links:

- `Collection Postman per firmware 4.9.4 - 4.9.8`
- `Collection Postman per firmware 4.9.9 o superiori`
- `Collection Postman per Modulo Hotel, firmware 4.9.9 o superiori`
- `Collection Postman per KalliopeLAM, firmware 4.11.3 o superiori`

You can download the file with API swagger specification from KalliopePBX, here:

```
http[s]://KALLIOPE_IP_ADDRESS/api/doc.json
```

6.1.4 Guide for CDR-related API

Since CDR-related API are the most common, we created a document about this family of API.

You can download it from this `link`.

6.1.5 Classes for generating authentication headers

Listed below are projects in different programming languages for generating and validating authentication headers.

- phpRestApiUtils: implementation in PHP
- RegEx: regular expression useful for validating generated headers