VoIP System Guide

Technical User Guide Document
v760

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1 About This Document

This document aims to present the Digital Enhanced Cordless Telecommunications (DECT) solutions from RTX and describe in detail their configuration, customization, management, and operation.

This manual will enable you to set up components in your network and deploy a fully functional system. However, it is not intended as a comprehensive reference to all details and specific steps on how to configure other vendor specific components/devices in a SME VoIP System. For a specific customer mode, please contact the respective vendor for documentation.

1.1 Terms and abbreviations

ABBREVIATION	DESCRIPTION
DECT	Digital Enhanced Cordless Telecommunications
SME	Small and Medium Scale Enterprise
VoIP	Voice over Internet Protocol
LAN	Local Area Network
PoE	Power over Ethernet
NB	Narrowband
WB	Wideband
MMI	Man Machine Interface
NAT	Network Address Translation
P2P	Peer-to-Peer
PTT	Push-to-Talk

1.2 Document History

The table below maps the history of this document, what firmware version it matches and the updates that have been made to it.

REVISION	VERSION	AUTHOR	ISSUE DATE	UPDATES
1.0	V750	LIP	13-9-2024	Updated to match V750 Firmware.
2.0	v760	LIP	38-01-2025	Updated to match V760 Firmware

1.3 What is new?

What new features have been added to the current revision.

REVISION	FEATURE
1.0	7.1.5 Servers – Table updated to match v750
	7.1.7.10 Ring Tone Selection
	7.1.17 Warning shows when unsupported repeater
2.0	7.1.4.6 Menu configuration
	7.1.6.2 VLAN Settings - Table updated – Tag L2 packets

	7.1.7.5 Text messaging – Table updated – Alarm soring
	7.1.14.1 Multicell settings – Table updated - Hierarchical relay

1.4 Related Documentation

List of documentation referred to in this guide.

REFERED DOCUMENT	REVISION	VERSION	COMMENTS
Firmware update guide. (9430-8660-HS).pdf	1.0	v700	
RTX 725x Headsets system guide.pdf	4.3	v760	
RTX provisioning guide.pdf	1.5	v700	
RTX VoIP DECT System Traffic Planning and Deployment .pdf	1.0	v700	

1.5 Documentation Feedback

We always strive to provide excellent customer service and therefore, we value your comments and suggestions concerning our documentation. If you have any comments about this guide, please enter them through the Feedback link on the RTX website. We will use your feedback to improve the documentation.

2 DECT system overview

In a typical telephony system, the network setup is the interconnection between base stations, “fat” routers, repeaters, handsets, etc. The handsets use the DECT technology to communicate with the base stations. The base stations use a third-party call system to communicate and control the call functions. The repeaters extend the rage of the signal of the base stations, thus providing a wider range for handling the user’s calls. These devices ensure user efficiency across different industries, as an RTX solution can be used in various businesses - from general offices and call centers to storage facilities, hospitals, manufacturing, and retail.

The backbone of the network depends on the deployment scenario, but a ring or hub topology is usually used. The network has centralized monitoring and maintenance system.

A graphical overview of the architecture of the SME VoIP system setup is illustrated below *(Fig.1)*. Depending on the choice of bases (RTX9431 or RTX8663), the system can support from 1 to 249 bases in the same network, and up to 1000 registered handsets. Based on PoE interface, each base station is easy to install without additional wires, other than the LAN cable. The system supports the IP DECT repeater, RTX4027, with support up to 5 channels for simultaneous call sessions.

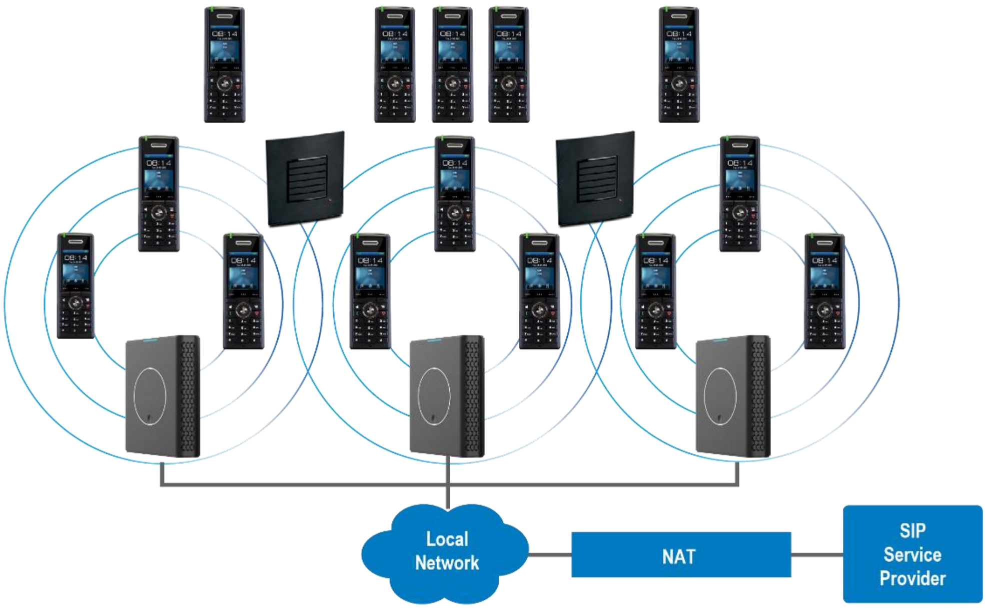


Figure 1 SME VoIP system

2.1 Deployment

An SME system setup can be deployed as follows:

Single cell setup: Telephony network composed of one base station.

Multi cell setup: Telephony network composed of more than one base station.

Synchronization Level: The air core interface between two base stations.

Base stations are connected via Layer 3 and/or VLAN Aware Router depending on the deployment requirements. The Layer 3 router implements the switching function.

The base stations can be mounted on walls or lamp poles so that each device is separated from each other by up to 50m indoor and 300m outdoor. Radio coverage can be extended using repeaters installed with the same distance to the base stations. Repeaters are range extenders and cannot be used to solve local call capacity issues. In this case additional bases must be used.

The base station's antenna mechanism is based on space diversity feature which improves coverage. The base station uses complete DECT MAC protocol layer and IP media stream audio encoding feature to provide up to 10 simultaneous calls.

2.2 Advantages

Implementing a VoIP system provides many advantages such as:

1. Simplicity. Integrating functionalities leads to reduced maintenance, troubleshooting and significant cost reductions.

2. Flexibility. Single network architecture can be employed and managed. Furthermore, the architecture is amenable to different deployment scenarios, including isolated buildings for in-building coverage, location with co-located partners, and large to medium scale enterprises deployment for wide coverage.

3. Scalability. SME network architecture can easily be scaled to the required size depending on the customer's requirements.

4. Performance. The integration of different network functionalities leads to the collapse of the protocol stack in a single network element and thereby eliminates transmission delays between network elements. Furthermore, it reduces the call setup time, packet fragmentation and aggregation delays.

2.3 VoIP product series

The RTX VoIP solutions are designed to serve businesses and organizations of all sizes. The products include DECT base stations, handsets, and repeaters *(Fig.2)*. The following sections aim to present and describe the characteristics of the various devices.

Figure 2 RTX VoIP products

2.3.1 Base stations

The base stations convert IP protocol to DECT protocol and transmit the traffic to and from the end-nodes (i.e. wireless handsets) over a channel. They have 12 available channels.

In a dual/multi cell setup, each base station has:

- 8 channels that have associated DSP resources for media streams.
- 4 channels which are reserved for control signaling between IP base stations and the SIP/DECT end nodes (or handsets).

In a dual and multi cell environment, the base stations are grouped into a cluster. Within the cluster, base stations are synchronized to enable a seamless handover, when a user moves from one base station coverage to another. The 4 control signaling channels are used to carry signals that enable a handset to initiate a handover process.

In a multi cell environment, it is not necessary for base stations to communicate directly with each other in the system. This means that a base station may only need to communicate with the next in the chain. It is advisable for a base station to identify more than one base station to guarantee synchronization in the case that one of the bases fails.

There are two types of base stations offered by RTX: RTX8663 Multi cell and RTX9431 Dual cell *(Fig.3)*.

Figure 3 RTX base station models

The RTX8663 base is a scalable multi cell IP DECT base station for Small, Medium and Large Enterprises. The multi cell system offers a unique redundant solution that ensures the best up time. The system is scalable from 1 base and 40 users, up to 4000 bases and 16000 users. The scalability is achieved by used patent protect technology. The system easily integrates towards a cloud solution that ensures easy maintenance, high security, and flexibility.

The second type of base offering is RTX9431, which is the Next Generation Single Cell base station. It can easily be upgraded to a dual cell that extends range and call capacities. The dual cell uses the same patent-protected technology as the RTX8663 multi cell system. The dual cell offers 100% security and plug and play for the second base.

Both base stations can be used in a standalone mode via the power cable (without PoE).

The main differences between the two devices can be seen in the table in *Appendix 9.7 Calls Capacity*.

* It does not matter if you are running Single cell or Dual cell setup, the maximum number of SIP registrations for RTX9431 is 40 (with a maximum of 20 handsets). On the contrary, the RTX8663 allows having up to 40 SIP registrations <u>per base.</u>

These numbers make it possible for different combinations of terminals with SIP extensions.

Every "SIP account" occupies one "extension instance". However, as a terminal instance cannot exist without a SIP instance, a terminal associated with a SIP account, occupies only one extension instance in total. A terminal associated with two SIP accounts occupies two extension instances, etc. This means that in order to have 40 SIP accounts, you can have the following combinations:

- 40 handsets, each with 1 line (i.e. 1 SIP account) – only for RTX8663
- 20 handsets, each with 2 lines (i.e. 2 SIP accounts)
- 10 handsets, each with 4 lines (i.e. 4 SIP accounts)
- 8 handsets, each with 5 lines (i.e. 5 accounts)

RTX9431 NOTE: *When local call group mode is enabled, the maximum number of extensions is decreased to 30. However, the maximum number of handsets remains 20.*

RTX8663 NOTE: *If there are over 150 base stations in a multi cell, the "Multicast" data sync should be used.*

2.3.2 Handsets

A handset is a lightweight, ergonomic, and portable unit, compatible with Wideband Audio (G.722), DECT, GAP standard and CAT-iq audio compliant. RTX's handsets include a color display with graphical user interface and can provide the user with most of the features available in a wired phone. In addition to this, they support roaming and handover capabilities.

RTX offers several types of handsets with various characteristics *(Fig.4).*

Figure 4 RTX handset models

- **RTX8830** is an IP65 handset intended for industrial, hospital and hotel use.
- **RTX8630** is a mid-range handset intended for any work environment.
- **RTX8631** is a ruggedized handset with Bluetooth intended for office use.
- **RTX8632** is an IP65 handset with Bluetooth intended for industrial use.
- **RTX8633** is an antibacterial IP65 handset with Bluetooth intended for hospital use.
- **RTX8431/RTX8436** is an entry level handset intended for any work environment.

The main differences between the handsets can be seen in the table below. For more details, please refer to the datasheets on www.rtx.dk

DESCRIPTION	RTX8431	RTX8436	RTX8630	RTX8631	RTX8632 / RTX8633	RTX8830
Display	1,77'' TFT, 128x160	2" TFT, 176x220	2'' TFT, 176x220	2'' TFT, 240x320	2'' TFT, 240x320	2'' TFT, 176x220
Size (mm)	123 x 46 x 21	128 x 48 x 21	140 x 48 x 25	118 x 46 x 21	122 x 50 x 21	152 x 48 x 28
Weight (including batteries)	98g	110g	136g	86g	95g	175g
Talk time	22H(NB), 14H(WB)		18h(NB), 12h(WB)	17h(NB), 12h(WB)		
Battery	1100 mAh Li-ion					
Alarm Key	No	No	No	Yes	Yes	Yes
Bluetooth	No	No	No	Yes	Yes	Yes
Other	4 extra programmable keys				RTX8633 has antibacterial coating	

***NOTE:** From v730 supports Bluetooth 5.1 on handsets that have this feature.*

2.3.3 Repeaters

Repeaters are devices which extend the DECT signal to ensure a wider DECT range for handling the calls. Currently, RTX offers RTX4027 repeater which ensures more coverage and range *(Fig.5)*. This guarantees perfect quality and interruption-free calls in large premises. Each repeater adds up to 50m indoor and 300m outdoor of extra range. Combined with encrypted daisy chaining, this adds 150m indoor or 900m outdoor range to a base. The RTX4027 is compatible with all RTX system solutions - extending range both indoor and outdoor.

RTX4027

Figure 5 RTX repeater

Since the device has the same hardware overview as the RTX9431 base station, the repeater can be identified by the following characteristics:

- The repeater has a double lined blue symbol on the top of the device *(Fig.5)*
- The label at the back of the device
- The missing LAN port.

The main characteristics of the repeater can be seen in the table below. For more technical details, please refer to RTX's website www.rtx.dk

DESCRIPTION	RTX4027
Size	120(W) x 35(D) x 135(H)mm
Coverage	Up to 50m indoor and 300m outdoor Receiver sensitivity: -92dBm
Antennas	2 internal omni-directional
Simultaneous call handling	2 (wide band) or 4 (narrow band)

2.4 User interface

The handsets and base stations provide the user with an easy-to-use digital interface. This section aims to present the different user interfaces available on the devices.

2.4.1 Base station configuration interface

Both base station models come with the same user-friendly SME VoIP Configuration Interface. It is a web-based administration platform used for configuring and programming the base station and relevant network end-nodes *(Fig.6)*. For example, handsets can be registered and de-registered from the system by using the configuration interface. Furthermore, this web-based platform can be used as a setup tool for uploading software/firmware to the base stations, repeaters and handsets. Moreover, it provides relevant system logs that can be used by the administrator to help troubleshoot the system, if it faces unforeseen operational issues.

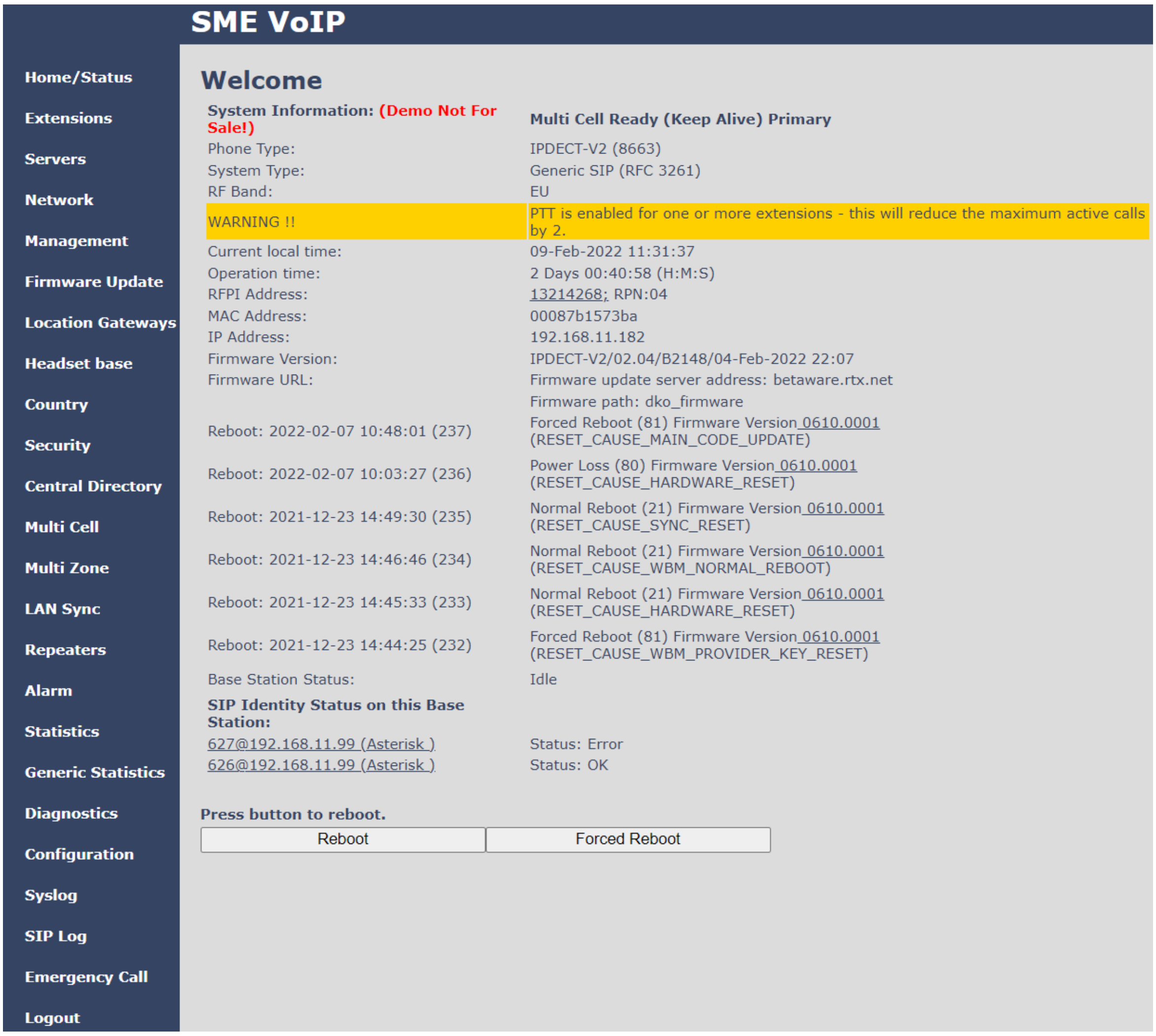


Figure 6 Base station's user interface

The web-based administration page is compatible with the following browsers:

- Chrome 68+
- Edge 42+
- Firefox 61+
- Safari 11.1.2+

The interface is accessed by typing the IP of the base station. On the left-hand side is the blue navigation panel which contains links to different configuration menus. On the right-hand side is the view which lists all the data and configuration parameters. The view changes according to the selected menu from the panel.

2.4.2 Handsets

All type of handsets have the same design, interface logic and icons. However, there are two types of user interface – Classic and Modern MMI *(Fig.7)*. The difference between the 2 MMIs is the interface outlook and the design of the icons. It is up to the client to decide which user interface to integrate in the device. Please note, that the interface cannot be changed via the menu or a specific setting. The transition is only possible with the use of the specific firmware file.

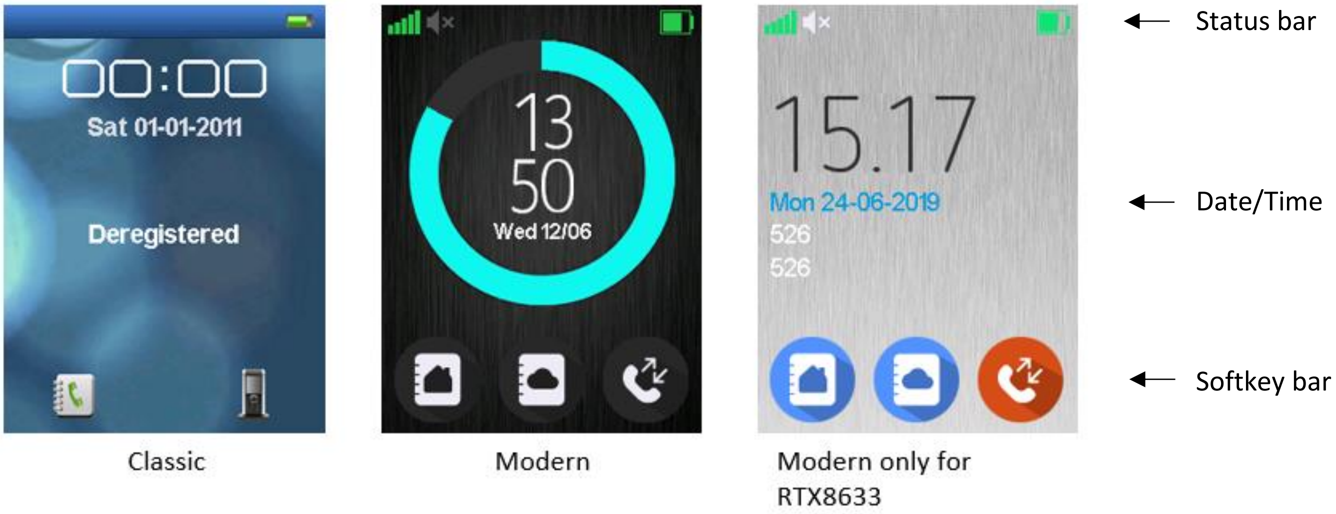


Figure 7 Handsets' user interface

2.4.3 Repeaters

The repeaters do not have a user interface.

3 Unboxing

Before unboxing, please examine the shipping package for evidence of physical damage. If there is a proof of mishandling prior to opening, you must report it to the relevant support center of the regional representative or operator. Furthermore, make sure that all relevant components are present in the package before proceeding to the installation guide.

Damage inspection guide:

- Examine all relevant components for damage.
- Make a "defective on arrival – DOA" report or RMA to the operator. Do not move the shipping carton until the operator has examined it. If possible, send pictures of the damage. The operator/regional representative will initiate the necessary procedure to process the RMA. They will guide the network administrator on how to return the damaged package, if necessary.
- If no damage is found, then unwrap all the components and dispose of empty package/carton(s) in accordance with country specific environmental regulations.

The following sections present the contents of the package for each RTX device.

3.1 Base stations

Every shipped base unit package/box contains the following items:

RTX8663	RTX9431	RTX9431 PoE only
1x Base unit	1x Base unit	1x Base unit
2x Mounting screws and 2x Anchors	1x Ethernet cable 1m	1x Ethernet cable 1m
1x Plastic wall mount	1x Power supply single plug	1x Quick guide
1x Quick guide	1x Quick guide	1x Safety sheet
1x Safety sheet	1x Safety sheet	

NOTE: *The following information is valid only for RTX9431 base station:*

Depending on the manufacturer P/N, the DC adaptor type may vary as listed below:

Manufacturer	DC adaptor plug type by countries
S008ACM0500200	Multi-plug
S010WB0500200	UK
S010WV0500200	EU
S010WU0500200	US
S010WS0500200	AU

Spare accessories:

- PSU single plug
- PSU multi plug

3.2 Handsets

Every shipped handset unit package/box contains the following items:

RTX8431/RTX8436	RTX8630 – RTX8633	RTX8830
1x Handset unit	1x Handset unit	1x Handset unit
1x Battery cover	1x Battery cover	1x Battery
1x Handset belt clip	1x Handset belt clip	1x Handset belt clip
1x Charger	1x Charger	1x Charger
1x A/C adaptor	1x A/C adaptor	1x A/C adaptor
1x Battery	1x Battery	1x Belt clip cover
1x Plastic cover to replace the belt clip		1x Pull cord cover
		1x Swirl for belt clip
		2x Mounting screws

3.3 Repeaters

Every shipped repeater package/box contains the following items:

RTX4027
1x Power supply
1x Repeater
1x table stand
1x Safety sheet

4 Hardware characteristics

The following sections aim to introduce you to the hardware outlook and features of the VoIP products.

4.1 Base stations

4.1.1 LED indicator

Both base stations (RTX9431 and RTX8663) have an LED indicator at the front-end panel *(Fig.8)*. It is used to signal different functional states of the base units and occasionally of the overall network. The indicator is off when a base unit is not powered on.

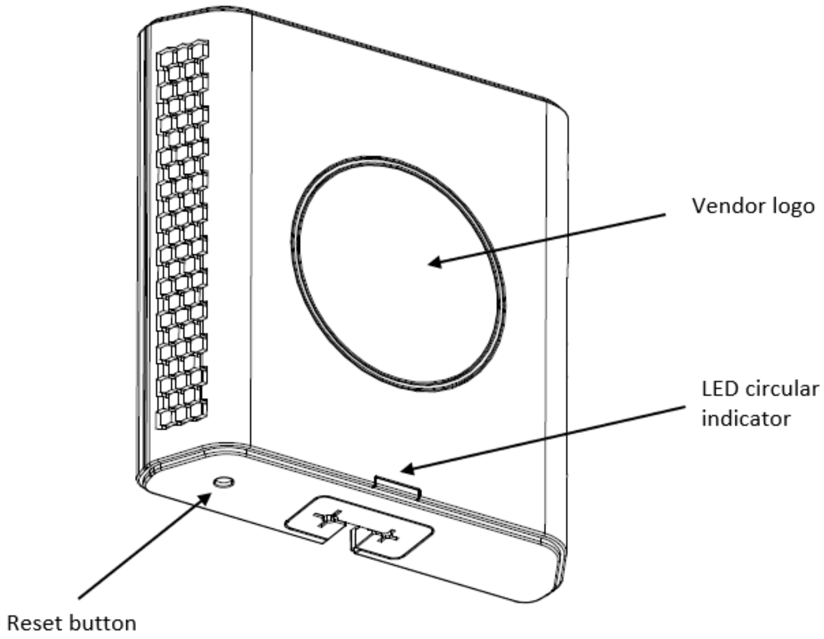


Figure 8 Base hardware characteristics (RTX9431)

LED COLOR	INDICATION	STATE
UNLIT	Off	No power in unit
RED	Steady on	Error condition /Unacceptable RSSI below -90dBm /Critical error (can only be identified by RTX engineers. Symptoms include no system/SIP, etc.) /Factory reset warning or long press on base station's reset button
RED	Flashing	/Factory setting in progress /Firmware upgrade/downgrade in progress

		/Ethernet connection not available or handset SIP registration failed
GREEN	Steady on	Ethernet connection available /Normal operation with good RSSI equal to or better than -75dBm
GREEN	Flashing	Initialization /Searching for base stations
ORANGE	Steady on	Normal operation with pure RSSI between -75dBm and -90dBm.
ORANGE	Flashing	Initialization /Searching for IP

4.1.2 Reset button

In addition to the hardware characteristics, the base stations have a reset button which allows the user to restart or reset the device. On both bases the button is placed below the main unit *(Fig.8)*. Alternatively, the bases can be reset from the web SME Configuration Interface. Furthermore, unplugging and plugging the Ethernet cable back to the PoE port of the base, also resets the base unit. However, we do not recommend using this reset method.

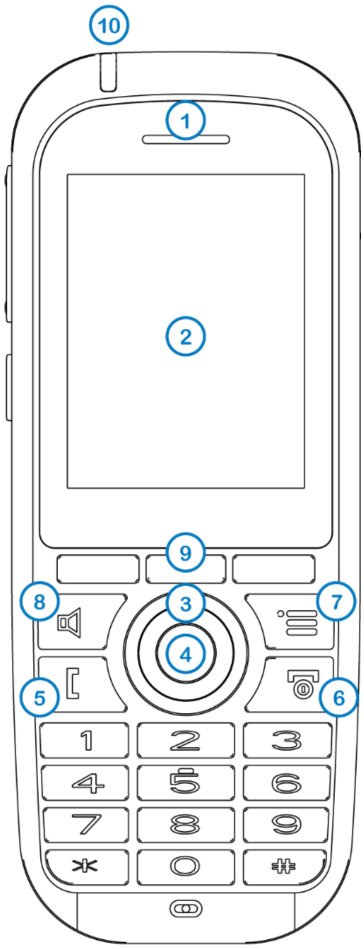
4.2 Handsets

There are some minor hardware differences between the devices, even though all handsets have a similar design. The following list presents an overview of RTX8631 handset, which includes all the main hardware features for the handsets.

NOTE: *RTX8431/RTX8436 has 4 extra programmable keys, also called Function keys, placed below the original keypad*

4.2.1 General overview

1. **Earpiece Speaker**
2. **Screen**
3. **Navigation keys**
   Up/Down/Right/left
4. **Selection key (Center of Navigation keys)**
   Use to confirm/select an option.
5. **Off-hook key**
   Answer a call;
   Use as a shortcut to the Call list.
6. **On-hook key**
   End a call;Return to the previous menu;
   Switch the handset on/off by long press.
7. **Activity Menu Key**
   Enter the main menu of the handset.
8. **Speaker Key**
   Enable/Disable speaker mode during a call.



NOTE: *If enabled during a call, the speaker key will light up in red and the sound will come from the speaker on the back of the handset*

9. **3 Softkeys**
   Used for different commands in the various menus of the handset, which change according to what is displayed on the screen.
10. **LED**
    Indicates incoming call, missed call, messaging, low battery and charging.

11. **Headset connector (3.5mm)**
    A slot used for connecting a headset. The connector is protected against dust by the headset connector cover.

12. **Volume Up Key (Side Key)**
    Increase the volume during a call.

13. **Volume Down Key (Side Key)**
    Decrease the volume during a call.

NOTE: *The volume buttons are used to control the volume when using the earpiece, a headset, and the loudspeaker. The lowest volume level will not mute the handset.*

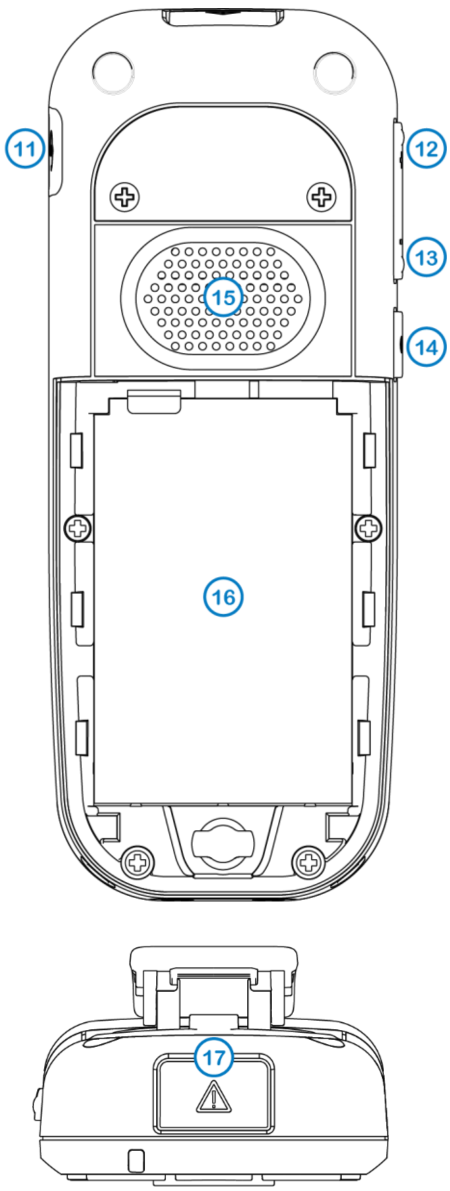
14. **Mute Key (Side Key) / Push-to-talk (PTT) key**
    Turn off the microphone during a call, so that the counterpart cannot hear you
    If the handset is in idle and the button is pressed, it will activate the PTT function.

NOTE: *For more details on the PTT feature, please refer to* ***7.3.4.19. Push to Talk***

15. **External Speaker**
    Used when speaker mode is enabled in a call.

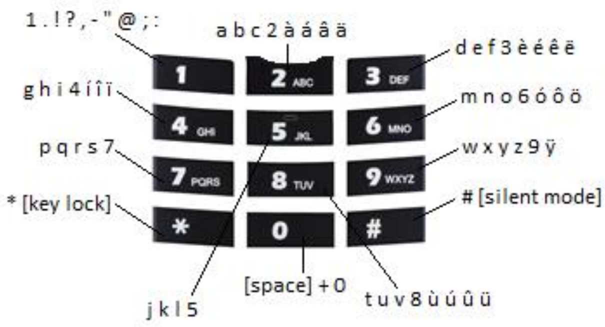
16. **Battery**

17. **Alarm Key**
    Supported only by RTX8631-RTX8633, RTX8830



4.2.2 Keypad

The sketch below illustrates the available characters on the handset's keypad *(Fig.9)*. Depending on the selected menu language, other characters can be available. Therefore, the character order can differ from the one below.



NOTE: *Any key from "2" – "9" can be set as a speed dial*

*key - a shortcut to calling any contact in the contact list, or in the F-Key list. A long press on any of these keys, while the handset is in idle, will call the assigned contact. For additional information on this feature refer to **7.3.1 Contacts.***

4.2.2.1 Number input mode

When in idle, a short key press enters the digits "0" – "9" and the characters "*" and "#".

A long key press on "0" generates a "+".

4.2.2.2 Text input mode

When writing a text, a short press on keys "0" – "9" displays the first available character on that specific key. By pressing the "*" key it is possible to change the letter input to upper/lower case or only a numeric value (Abc, abc, ABC, or 123).

Use the "0" key to add a space in the text.

4.2.2.3 Key lock

The keypad can be locked/unlocked by long pressing the "*" key in idle. This feature can prevent accidentally pressing keys and making a call.

NOTE: *It is possible to answer/reject an incoming call while the keypad is locked*

4.2.2.4 Silent mode

The silent mode feature can be activated/deactivated by long pressing the "#" key in idle. When enabled, the handset will turn to silent mode and a "Sound Off" icon will be displayed in the status bar.

NOTE: *When this mode is enabled, it is possible to switch on/off the vibrator on the handset in Settings > Audio Settings > Vibrator (refer to **7.3.4.1 Audio settings** for further information). Please note, that all other audio settings will be greyed out while the handset is in silent mode.*

4.3 Repeaters

Similar to the base stations, the repeater also has an LED indicator and a reset button.

4.3.1 LED indicator

The LED indicator is off when a repeater is not powered on. The figure below *(Fig.10)* illustrates the location of the LED on the repeater.

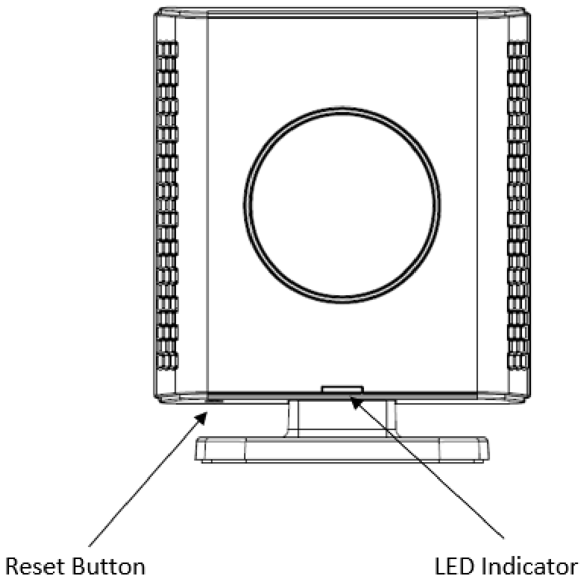


Figure 10 Repeater hardware characteristics

The following table provides more details about the different LED states.

LED COLOR	INDICATION	STATE
UNLIT	Off	Power Off
GREEN	Slow flash	Unblocked, searching for a base station
GREEN	Double flash	Registering procedure. Searching for an open base station to register/subscribe to it.
GREEN	Steady on	Locked to a base station and ready to be used; /Ethernet connection available; /Normal operation with good RSSI equal to or better than -75dBm)
RED	n Flash	n handset using the repeater /Ethernet connection not available or repeater registration failed

RED	Steady on	Error condition; Unacceptable RSSI below -90dBm; Critical error /Registration procedure timed out after 5 min. /When the reset key is long pressed, the LED will light up in 2 sec. to indicate that releasing the key will delete the registration. LED turns off after 4 more sec.
ORANGE	Steady on	Normal operation with pure RSSI between -75dBm and -90dBm
RED/GREEN	Flashing Red/Green	Active call under the coverage of the repeater /Recovery mode – repeater is locked to a base station/repeater without repeater mode activated. /Sync. Source base station/repeater not found (Manual mode). /Timeout during RPN allocation due to busy base station/repeater.

4.3.2 Reset button

Similar to the base stations' hardware, the reset button of the repeater is placed below the main unit. It has the following functionalities:

BUTTON PRESSS	ACTION
00s < x < 02s	Nothing
02s < x < 06s	Delete registration. Old registration is deleted and a new registration procedure has started.
18s < x < 60s	Enable/disable repeater monitor beep tone which is heard in the handset during a call (when it is locked to a repeater). This is used for testing handover and calls.
60s < x < XXs	Nothing

5 Device installation

5.1 Base stations

Before installing the base stations, please first determine the best location that will provide an optimal coverage, by considering the construction of the building, architecture and choice of building materials. Next, proceed to the following sections to get an overview of the mounting and signaling details of the devices.

5.1.1 Mounting

Mount the devices on a wall to cover range between 50 – 300 meters (i.e. 164 to 984 feet), depending on whether it is an indoor or outdoor installation. Mount them as high as possible (min. 2 meters) to clear all nearby objects (e.g. office cubicles and cabinets, etc.). Make sure that when you fix the base stations with screws, the screws do not touch the PCB on the unit. Furthermore, avoid all contact with any high voltage lines.

Note: *RTX9431 is not designed for outdoor use.*

The images below illustrate the device installation characteristics for both RTX9431 *(Fig.11)* and RTX8663 *(Fig.12)* base stations.

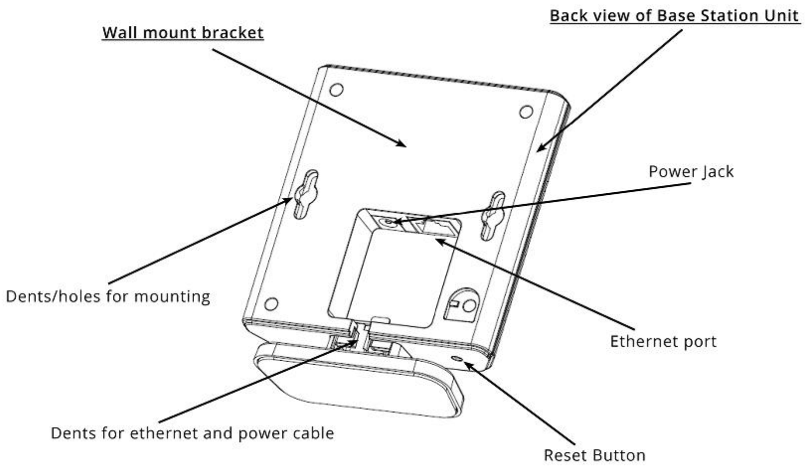


Figure 11 RTX9431 installation characteristics

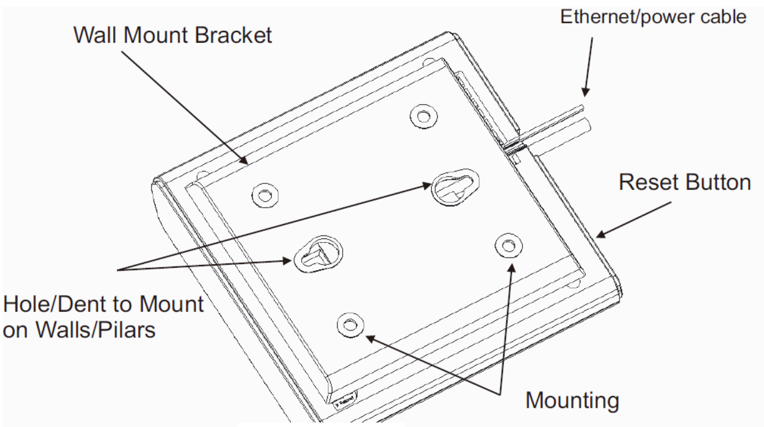


Figure 12 RTX8663 installation characteristics

5.1.1.1 RTX8663 base

The RTX8663 base station supports both horizontal and vertical signaling. This means that you should not follow any positioning guidelines since it can transmit both ways. The antenna radiation pattern can be seen on the figure below *(Fig.13)* where Antenna 0 is the green marking with max power 23,25 dBm and antenna 1 is the red pattern with max power 22.45 dBm.

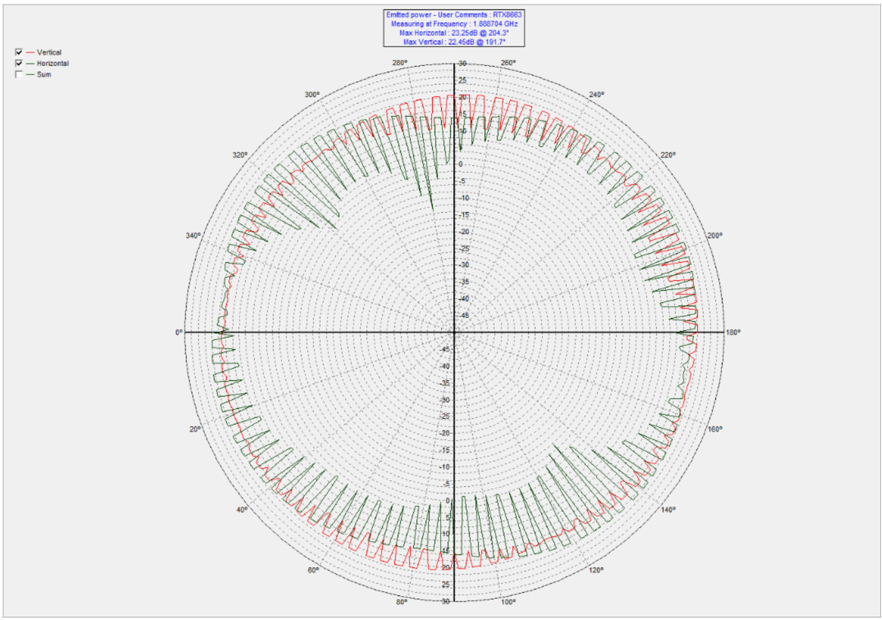


Figure 13 RTX8663 antenna pattern

5.1.1.2 RTX9431 base

The RTX9431 base station also supports both horizontal and vertical signaling, meaning that you should not follow any positioning guidelines (as it can transmit both ways). The antenna radiation pattern can be seen on the figure below *(Fig.14)* where Antenna 0 is the green marking with max power 23,19 dBm and antenna 1 is the red pattern with max power 22.92 dBm.

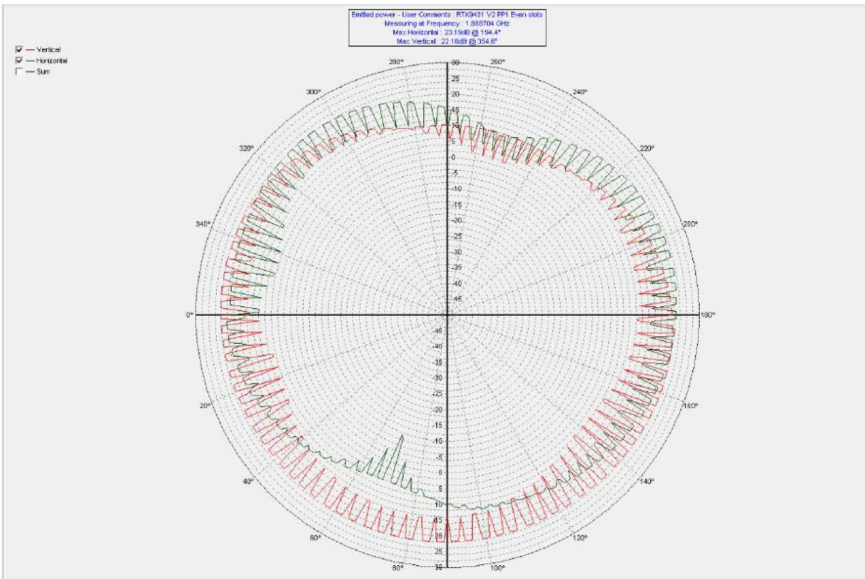


Figure 14 RTX9431 antenna pattern

Since the screws and anchors are not included in the RTX9431 packaging, you will have to provide your own two pieces of screws M3.5 x 31mm. The distance between them is 70mm *(Fig.15)*.

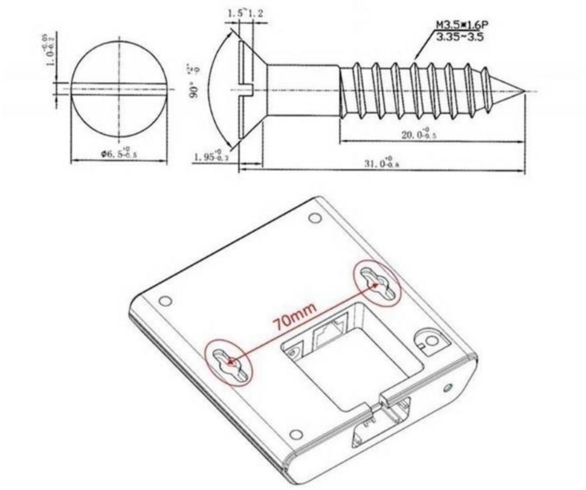


Figure 15 RTX9431 screws and anchors

5.2 Handsets

Before using the handset, you should be aware of the following precautions.

5.2.1 Handset battery

The handsets use Li-ion 3.7V 1100mAH 4.1Wh batteries to operate. Please, be aware of the following:

- Never dispose a battery in fire, otherwise it will explode. Instead, it must be delivered to a recycling company.
- Never replace the batteries in potentially explosive environments, e.g., close to inflammable liquids/gases.
- ONLY use approved batteries and chargers from the vendor or operator.
- Do not disassemble, customize, or short circuit the battery.

5.2.2 Handset charger

Each handset is charged using a handset charger. The charger is a compact desktop unit, designed to charge and automatically maintain the correct battery charge levels and voltage. The handset charger is powered by AC supply from 110-240VAC that supplies 5.5VDC at 600mA. When charging the battery for the first time, it is necessary to leave the handset in the charger for at least 10 hours before the battery is fully charged, and the handset is ready to be used.

For correct charging, ensure that the room temperature is between 0°C and 25°C / 32°F and 77°F. Do not place the handset in direct sunlight. The battery has a built-in heat sensor which will cause the battery to stop charging if its temperature is too high.

If the handset is powered off when placed in the charger, the handset will turn on and display (shortly) the charging message “Charger connected”. After a while, the display backlight will dim and then turn off. Anyway, the handset will be available for incoming calls, messages, etc.

If the handset is turned on when placed in the charger, the display will show (shortly) the charging message "Charger connected". After a while, the display backlight will first dim and then turn off, unless "Idle Screen Saver" is enabled.

Each handset comes with a battery which is already placed in the device. However, you need to activate the battery by removing the plastic foil. To access the battery slot, you first need to open the back cover. Depending on the handset type, the back cover can be opened in different ways *(Fig.16)*. Afterwards, you need to remove the battery followed by the plastic foil. ***Do not pull the plastic foil out without removing the battery, that might damage the charging pins.*** After that you can put the battery and the cover back in place and charge your handset. *(Fig.17)*.

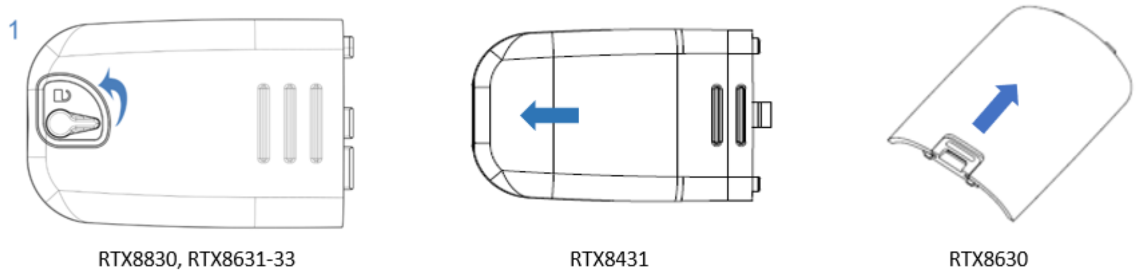


Figure 16 Open the back cover from the handsets.

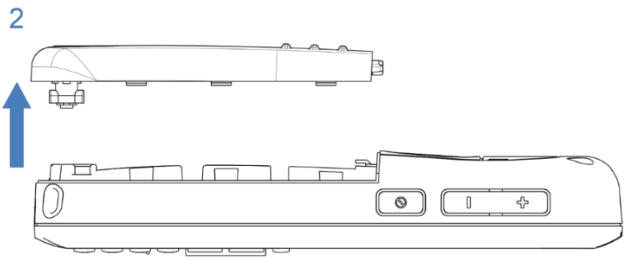


Figure 17 Removing the back cover from the handsets.

On the battery slot of each handset, you can find a label with the serial number (S/N) and IPEI/IPUI number of the device. This information can be also seen on the packaging label of the handsets.

5.3 Repeaters

Before installing the repeater, first determine the best location that will provide an optimal coverage, by considering the construction of the building, architecture, and choice of building materials.

The RTX4027 repeater is similar to the RTX9431 base station in terms of signaling and mounting. Therefore, for more details concerning the installation of the device, please see ***5.1.1.2 RTX9431 base***.

Note: *RTX4027 is not designed for outdoor use.*

6 Access to device

The following chapter aims to help you access the base station and handsets. The repeater is excluded from this section, as it does not have an interface.

6.1 Base station

6.1.1 Base station IP address

To enter the user interface of the base station, you first need to connect the base to a private network (via standard Ethernet cable CAT-5). Afterwards, you need to find the device's IP and type it in your internet browser. There are 2 ways of finding the IP address of the base – using the handset or the browser.

6.1.1.1 Using the handset

The handset has an "IP search" feature, which allows the user to view the IP address of the bases in range of the handset. To use this feature, please follow the steps below:

Step 1 Handset in idle
Step 2 Press the *Menu* key
Step 3 Type ***47***

The handset will now enter scanning mode and will listen for signals from the nearest base stations. Depending on the amount of powered on bases and the distance to them, it can take up to a few minutes to find a device. As soon as a base is found, the handset will display the MAC address of the base, together with its IP address *(Fig.18)*. You can use the navigation keys on the handset to scroll up and down along the list of found devices.

Figure 18 IP search menu

6.1.1.2 Using the browser

To find the IP address of the base via the browser, please follow the steps below:

Step 1 Open any standard browser and enter the below address, where <MAC-Address-Base-Station> is the MAC address of the device

http://ipdect<MAC-Address-Base-Station>

Example: http://ipdect00087B00AA10

By typing this information, the HTTP Web Server page will be retrieved from the base station with hardware address **<MAC-Address-Base-Station>** (in this case 00087B00AA10).

6.1.2 Login to base

To access the login window of the base, please follow the steps below:

Step 1 Connect the base station to a private network and find the IP address of the device (see previous section ***6.1.1 Base station IP address***)

a. If the handset method is used, open a browser and type in the IP address

Step 2 On the Login page, enter your authentication credentials (i.e. username and password) *(Fig.19)*. By default, the username and password are **admin/admin**. Press **OK.**

NOTE: *In case secure web server is enabled the server automatically redirects from http:// to* **Error! Hyperlink reference not valid.** *in the browser address line.*

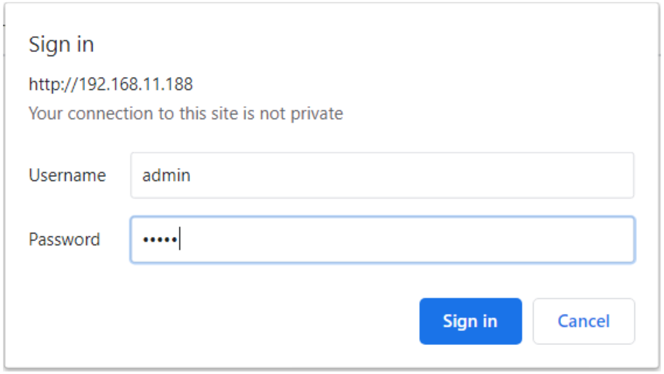


Figure 19 Login page of the base

Step 3 Once you have been authenticated, the browser will display the front end of the SME VoIP Configuration Interface *(Fig.6)*. The front page will show relevant information of the base station.

6.2 Handset

To get access to the handset, simply insert the battery and power on the device by long pressing the hook-on button. The display will light up and the handset will be ready for use.

7 Software characteristics

The following chapter will provide you with information about the features and different functionalities available in the software, for both the bases and the handsets. Since the repeater does not have any user interface and is not meant to be manually setup by the user, the device is not included in this chapter.

7.1 Base station

The SME VoIP Configuration Interface, also known as SME VoIP Administration page, is the main interface through which the system is managed and debugged. The SME VoIP Administration page is an in-built, HTTP Web Server service residing in each base station. It is a user-friendly interface, making it easy to handle even to a first-time user.

NOTE: *Enabling secure web is not possible. For secure configuration, use secure provisioning (refer to **7.1.22 Configuration**)*

This subchapter seeks to define various variables/parameters, by going through each of the available menus and settings on the base station. These settings can be configured in 2 ways – via the web interface or via provisioning (refer to ***7.1.22 Configuration*** for further details). Certain pages (Extensions, Repeaters, Location gateways and Multi/Dual cell) support an auto refresh feature, which allows the user to easily monitor the system. The pages are automatically re-freshed every 5 seconds.

IMPORTANT: The guide is based on the full feature set of RTX8663. Therefore, the settings, which are not supported by RTX9431, will be clearly stated (for ex. ***NOTE****: This feature is not supported by RTX9431*)

7.1.1 Web navigation

The screenshot below *(Fig.20)* illustrates the front page of the SME VoIP Administration Interface of the base station. The various menus, seen on the left-handed navigation panel, are briefly presented in the following table.

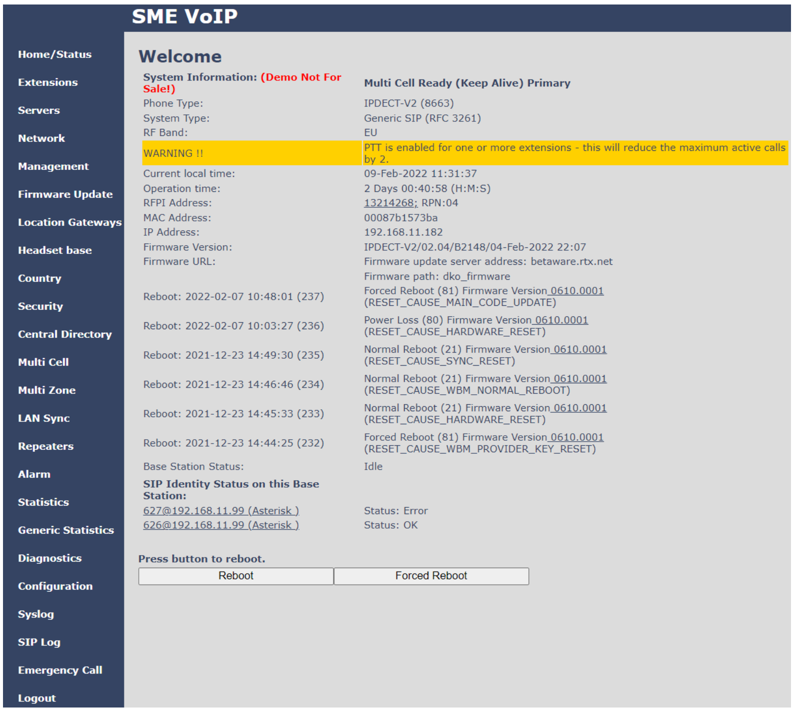


Figure 20 SME VoIP Administration Interface

FEATURE	DESCRIPTION
HOME/STATUS	This is the front end of the base station's HTTP web interface. This page shows a summary of the current operating condition and settings of the base station and handset(s).
EXTENSIONS	Administration of extensions and handsets in the system
SERVERS	On this page, the user can define which SIP/NAT server the network should connect to.
NETWORK	Provides the user with the main network settings, such as: **NAT settings**: allows configuration of features for resolving the Network Address Translation. These features enable interoperability with most types of routers. **IP settings:** allows setting dynamic (DHCP) or static IP address for your SME network **Virtual LAN:** allows specifying the Virtual LAN
MANAGEMENT	Defines additional settings such as the Configuration server address, Management transfer protocol, Text messaging and sizes of logs/traces that should be catalogued in the system.
FIRMWARE UPDATE	Remote firmware update (HTTP(s)/TFTP) settings of base stations and handsets.
LOCATION GATEWAYS	Administration of Location Gateways ***NOTE:** The Location gateway feature must be enabled via the Management page for this menu to appear on the navigation panel (refer to **7.1.7.8 Location gateway**)*
HEADSET BASE	Administration of headset bases ***NOTE:** The Headset base feature must be enabled via the Management page for this menu to appear on the navigation panel (refer to **7.1.7.9 Headset base**)*
COUNTRY	Allows specifying the country/territory where the SME network is located. Configuring these settings ensure that your system is working properly. ***NOTE:** The base language and country settings are independent of each other* **Time settings**: The user can configure the Time server to always have their base synched with the exact time. The time servers must deliver the time to conform to the Network Time Protocol (NTP). Handsets are synchronised to this time. Base units synchronise to the master using the Time server.
SECURITY	The users can administrate certificates and create account credentials with which they can log in or log out of the embedded HTTP web server.
CENTRAL DIRECTORY	Provides an interface to a common directory load of up to 3000 entries using *csv format, or to a configuration of an LDAP directory. ***NOTE:** LDAP and central directory cannot operate at the same time*
MULTI CELL	Enables the user to create a chain of base stations (a multi cell environment). Make sure the system ID for the relevant base stations is the same, otherwise the multi cell feature will not work. ***NOTE:** For RTX9431, this menu is displayed as Dual cell and allows you to create a chain of 2 bases*
MULTI ZONE	Administration and configuration of the zones in a Multi zone setup ***NOTE:** This feature is not supported by RTX9431*
LAN SYNC	Administration and configuration of network synchronization between the bases ***NOTE:** This feature is not supported by RTX9431*
REPEATERS	Administration and configuration of repeaters in the system
ALARM	Administration and configuration of the alarm settings on the system. This controls the settings for alarms that can be sent to the handsets. This feature is only available on certain types of handsets.

STATISTICS	Overview of the system and call statistics
GENERIC STATISTICS	Overview of general parameter statistics of the system
DIAGNOSTICS	Overview of base stations and extensions diagnostics
CONFIGURATION	Displays the detailed and complete SME system settings for the base station
SYSLOG	Reports overall network related events or logs (only live feed is shown)
SIP LOG	Reports SIP related logs
EMERGENCY CALL	Administration and configuration of emergency numbers
LOGOUT	Logout of the web interface

***NOTE:** Only one admin session is allowed at a time. If one admin is successfully logged in, the other admin session is automatically logged out.*

NOTE: After 10 failed logins within 5 min, all requests for authentication for the next 5minutes are rejected.

7.1.2 Home/Status

This section presents and describes the available parameters on the Home/Status page of the SME VoIP Administration Interface *(Fig.20)*.

PARAMETER	DESCRIPTION
SYSTEM INFORMATION	The current multi (dual) cell state of the specific base
PHONE TYPE	Always IPDECT
SYSTEM TYPE	The customer configuration for the specific base
RF BAND	The RF band setting for the specific base. The parameter is defined in production and relates to the radio approvals present on the label of the base.
CURRENT LOCAL TIME	The local time of the specific base
OPERATION TIME	Operation time for the base since the last reboot
RFPI-ADDRESS	The RFPI address of the specific base
MAC-ADDRESS	The MAC address of the specific base
IP-ADDRESS	The IP address of the specific base
FIRMWARE VERSION	The firmware version of the specific base
FIRMWARE URL	Firmware update server address and firmware path to the file on the server
REBOOT	Displays information about the last reboots of the base station
BASE STATION STATUS	Displays the status of the base **Idle**: When there are no calls on the base **In use:** When there are active calls on the base
SIP IDENTITY STATUS	List of extensions present on this base station. Format: “extension”@“this base’s IP address”(“server name”) followed by their status to the right. Below are listed possible status outputs: **OK:** Handset is ok **SIP Error:** SIP registration error
REBOOT	Reboot the base after all connections are terminated (such as active calls, directory access, active firmware update, etc.)
FORCED REBOOT	Reboot immediately.

7.1.3 Extensions

This section describes the different parameters available when creating extensions for the handsets. Depending on the base (RTX8663 or RTX9431), the **Extensions** page has a different view. The RTX8663 shares one page for both handsets and extensions *(Fig.21)*, whereas the RTX9431 provides 2 separate views for the extensions and devices via the **Extensions / Handset** links *(Fig.22)* This is because 9431 has the feature *Call Group* available, that is not present on 8663. However, there is no other significant differences in the settings between both bases.

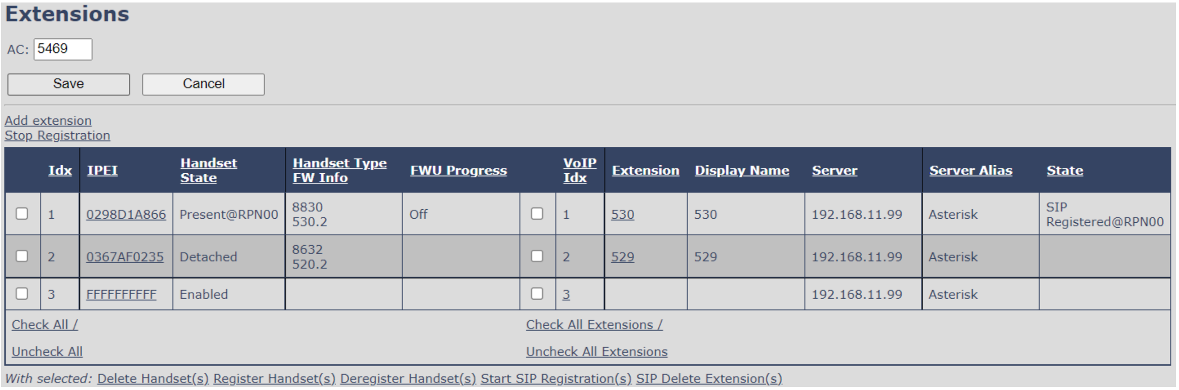


Figure 21 Extensions page view on RTX8663

Figure 22 Extensions page view on RTX9431

The **Extensions** page contains an extension/handset list and top/sub- menus. The extension/handset list is used to provide information about the existing devices and allow the control of paring and removing handsets from the system (DECT registration/de-registration). The top menu presents general operations, and the sub menu is always operating on selected handsets/extensions.

The table below defines the various options visible on the **Extensions** page.

PARAMETER	DESCRIPTION
AC	Access code used when registering a handset
LOCAL CALL GROUPS	Valid for RTX9431: Enable the feature to allow multiple handsets to be associated in a group ***NOTE:*** *When local call group mode is enabled, the maximum number of extensions is decreased to 30. However, the maximum number of handsets remains 20.*
ADD EXTENSION / ADD HANDSET	Provides access to the **Add extension** or **Add Handset** editor menu, where you can add new extensions and devices
STOP REGISTRATION	Manually stops DECT registration mode for the system. This prevents handsets to register to the system

DELETE HANDSET(S)	Valid for RTX9431: Delete the selected handset(s), but do not delete the extension(s). Valid for RTX8663: Delete the selected handset(s) and delete the extension(s).
REGISTER HANDSET(S)	Enable registration mode for the system to make it possible to register handsets at a specific extension (selected by checkbox)
DEREGISTER HANDSET(S)	Deregister the selected handset(s), but do not delete the extension(s).
START SIP REGISTRATION(S)	Manually start SIP registration for the selected handset(s).
SIP DELETE EXTENSION(S)	Deregister the selected handset(s) and delete the extension(s).
IDX	Index of handsets / extensions
IPEI	The IPEI of the handset. In general, this must not be changed manually. If IPEI is left at FFFFFFFFFF, any handset can register at this index. Else, only the handset with the specified IPEI number can register to the VoIP system ***NOTE:*** *If selected, the editor's menu will open*
HANDSET STATE	Displays the state of the given handset: **Present@RPNxx:** The handset is DECT located at the base with RPNxx **Detached:** The handset is detached from the system (e.g. powered off) **Located:** The handset is configured to locate on a specific base, but is has not been possible to do so (e.g. if the base is powered off) **Removed:** The handset has been out of sight for a specified amount of time (~one hour). **Enabled:** The slot is open for handset registration
HANDSET TYPE FW INFO	Name of the handset type Firmware version of the handset
FWU PROGRESS	Displays the available FWU progress states: **Off:** Means software version is specified to 0 = fwu is off **Initializing:** Means FWU is starting, and progress is 0%. **X%:** FWU ongoing **Verifying X%:** FWU writing is done and now verifying before swap **Waiting for charger** (handsets) / **Conn. term. wait** (repeater): All FWU is complete and is now waiting for a handset/repeater restart. **FWU complete:** Upgrade is complete for the handset/repeater: **Error:** Not able to fwu upgrade the device (e.g. file not found, file not valid etc.)
VOIP IDX	Index of the configured SIP extensions. Select/deselect to start SIP registration or delete extension.
EXTENSION	Given extension is displayed ***NOTE:*** *If selected, the editor's menu will open*
DISPLAY NAME	Given display name is displayed. If no name given, this field will be empty
SERVER	Server IP or URL
SERVER ALIAS	Given server alias is displayed. If no alias given, this field will be empty.
STATE	SIP registration state – if empty the handset is not SIP registered. Else, one the following states will appear: **SIP Registered:** Handset has successfully registered to the PBX **SIP Error:** No connection to the server, leading to unsuccessful SIP registration (due to wrong extension authentication, server setup, network, etc).

NOTE: *By powering off the handset, the device will SIP deregister from the PBX.*

A multi cell system can handle a maximum of 1000 extensions matching 1000 handsets which can be divided between the bases, whereas the dual cell system can handle a maximum of 40 extensions matching 20 handsets. As soon as the maximum limit is reached, it is not possible to add more extensions or devices. For more information related to number of calls and extensions per base, please refer to ***2.3.1 Base stations.*** To view a step-by-step guide on configuring extensions and handsets, please see chapter ***8. Setup guides.***

RTX8663 NOTE: *In a multi cell system with active multiline feature, the system can handle maximum 1000 handsets, whereas with 4 active lines in multiline, a maximum of 200 handsets can be active in the system.*

RTX9431 NOTE: *In a single/dual cell system with active multiline feature, the system can handle maximum 20/40 extensions, whereas with 5 active lines in multiline, a maximum of 4/8 handsets can be active in the system.*

IMPORTANT: Within a server or even with multi servers, extensions must always be unique. This means that the same extension number on server 1 cannot be re-used on server 2.

7.1.3.1 Add (edit) extension editor menu

The **Add (Edit) Extension** editor menu appears as soon as you select the *Add extension* link or select an already existing extension. The view output for each base is shown in the screenshots below *(Fig.23, 24)* and the displayed parameters are explained in the following table.

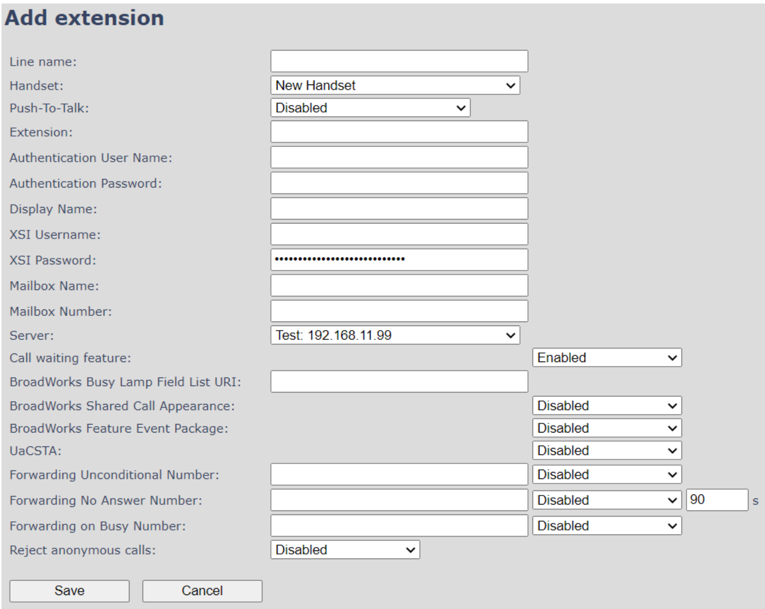


Figure 23 Add extension editor menu overview on RTX8663

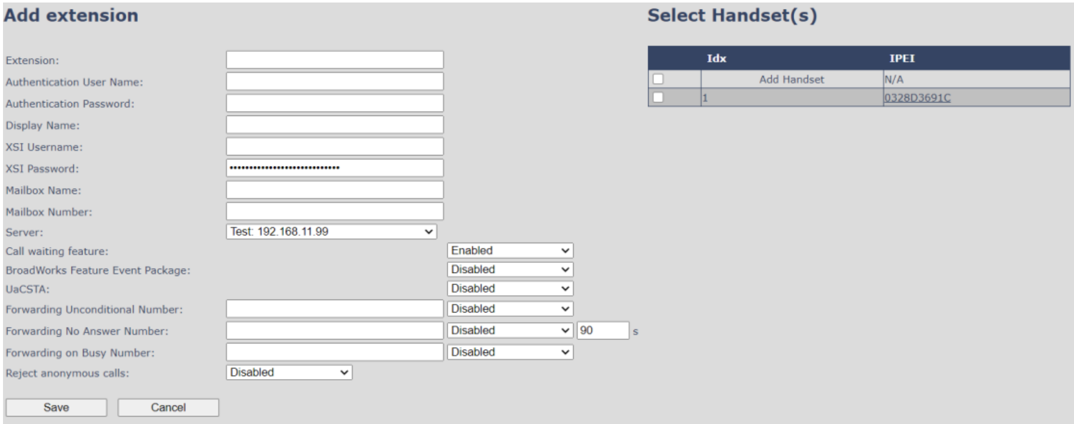


Figure 24 Add extension editor menu overview on RTX9431

PARAMETER	DEFAULT VALUE(S)	DESCRIPTION
LINE NAME	Empty	Used in the following 2 cases: Inform the user from which line an incoming call is received User selects from which line to make an outgoing call ***NOTE:*** *This option is not present in RTX9431*
HANDSET	New Handset	The extension must be associated to a handset. By default, a new handset can be configured. Alternatively, the user can select an already existing handset, by choosing one from the drop-down menu (Handset Idx x). ***NOTE:*** *For RTX9431 this option is located on the right-hand side of the page*
PUSH-TO-TALK (PTT)	Disabled	The feature allows having P2P walkie-talkie, meaning voice calls between handsets and headsets on the VoIP system. Enable the feature to broadcast a PTT conference call to all portable devices. ***NOTE:*** *For RTX9431 this parameter is located under the Handset menu*
EXTENSION	Empty	Handset phone number or SIP username depending on the setup. **Possible value(s):** 8-bit string length Example: 1024, etc. ***NOTE:*** *The extension must also be configured in the SIP server in order for this feature to function.*
AUTHENTICATION USERNAME	Empty	**Username:** SIP authentication username of the extension (pre-configured on the SIP server) **Permitted value(s):** 8-bit string length
AUTHENTICATION PASSWORD	Empty	**Password:** SIP authentication password of the extension (pre-configured on the SIP server) **Permitted value(s):** 8-bit string length
DISPLAY NAME	Empty	Human readable name used for the given extension **Permitted value(s):** 8-bit string length ***NOTE:*** *The name will not be displayed on handsets, using the modern MMI interface*
XSI USERNAME	Empty	To gain access to the XSI server, type in the SIP authentication username

		Permitted value(s): 8-bit string length
XSI PASSWORD	Empty	To gain access to the XSI server, type in the SIP authentication password **Permitted value(s):** 8-bit string length
MAILBOX NAME	Empty	Name of centralized system used to store phone voice messages that can be retrieved by the recipient later. **Valid Input(s):** 8-bit string Latin characters for the name
MAILBOX NUMBER	Empty	The number of the mailbox where voice messages are stored. This number is to be dialed by the user to hear the voice messages. This is done by long pressing key 1 on the handset. **Valid Input(s):** 0 – 9, *, # ***NOTE:*** *The Mailbox Number parameter is available only when it is enabled on the SIP server.*
SERVER	IP of Server 1	Select the FQDN or IP address of the SIP server (the SIP server should be pre-configured on the **Servers** page) Drop down menu to select between the various defined servers.
CALL WAITING FEATURE	Enabled	Used to enable/disable Call Waiting feature. If **disabled**, a second incoming call will be rejected. If **enabled**, the second incoming call will be presented as call waiting to be answered. The feature is only valid when there is an active call. ***NOTE:*** *An extension can only receive 1 alerting call. This means that during an alerting call that is not yet answered, the Call Waiting feature is not valid. Therefore, all other calls will be rejected and not queued.*
BROADWORKS BUSY LAMP FIELD LIST URI	Empty	The “BLF” feature on the handsets allows a specific extension to be monitored for state changes. BLF monitors the status (busy or idle) of extensions on the handset **Permitted value(s):** URL String ***NOTE:*** *This feature does not work with Group call. Therefore, Group call should be disabled.* ***NOTE:*** *This option is not supported by RTX9431*
BROADWORKS SHARED CALL APPEARANCE	Disabled	Enable Shared Call Appearance (SCA) to allow a group of SIP phones to receive inbound calls directed to a single destination (shared line); in this way, any phone from this group can answer the call, barge-in to the active call, or retrieve the call placed on hold. ***NOTE:*** *Must be supported by SIP server* ***NOTE:*** *This option is not supported by RTX9431*
BROADSOFT FEATURE EVENT PACKAGE	Disabled	If **enabled**, the given SIP extension subscribes for the Broadsoft Application Server Feature Event Package, and it becomes ready for reception of SIP NOTIFY with status on the following Broadsoft Server Services: **Do Not Disturb** **Call Forwarding (Always, Busy, No answer)** The received status will be displayed in the handset idle display.
UACSTA	Disabled	Enable/Disable uaCSTA support
	Empty	Number to which incoming calls must be re-routed, regardless of the handset’s state.

FORWARDING UNCONDITIONAL NUMBER	Disabled	The feature must be enabled to function. ***NOTE:*** *Feature must also be enabled in the SIP server before it can function in the network*
FORWARDING NO ANSWER NUMBER	Empty	Number to which incoming calls must be re-routed when there is no response from the SIP end node.
	Disabled	The feature must be enabled to function.
	90	Specify the delay from a call to forward in seconds. ***NOTE:*** *Feature must also be enabled in the SIP server before it can function in the network*
FORWARDING ON BUSY NUMBER	Empty	Number to which incoming calls must be re-routed when SIP node is busy.
	Disabled	The feature must be enabled to function ***NOTE:*** *Feature must also be enabled in the SIP server before it can function in the network*
REJECT ANONYMOUS CALLS	Disabled	If **enabled**, calls from anonymous numbers will automatically be rejected.

NOTE: *Call forwarding can also be configured from the handset by the user. For more details, please see* ***7.3.4.15 Call Forward***

7.1.4 Handsets

For both bases (RTX9431 and RTX8663) the editor window of the handsets can be accessed by clicking on the IPEI link of the desired device from the **Extensions** page. In addition to this, as mentioned in ***7.1.3 Extensions***, the RTX9431 has also a separate menu (Handset) for displaying the devices. Entering this menu allows you to access the handset editor window by selecting the IPEI link of the desired handset or the *Add Handset* link. A screenshot of the editor window can be seen below (*Fig.25).*

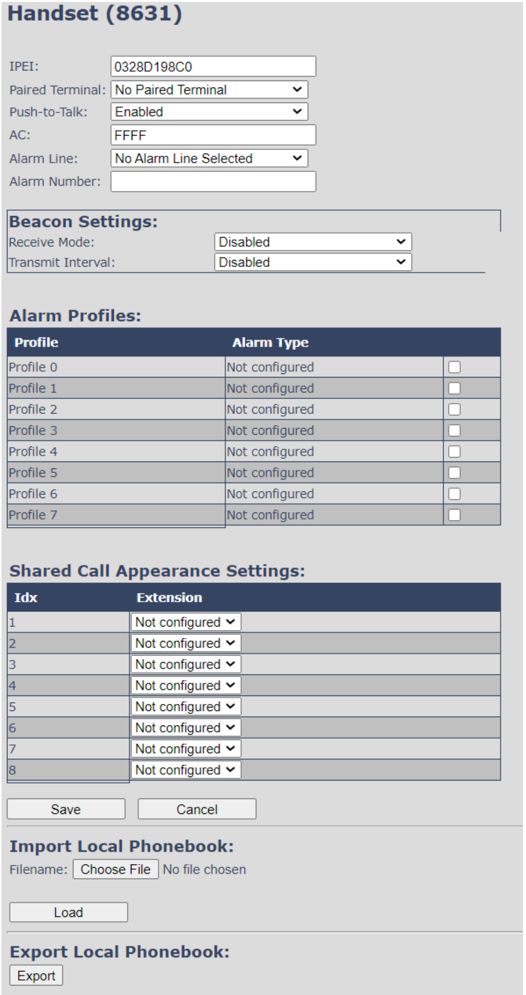


Figure 25 Handset editor window

PARAMETER	DEFAULT VALUE(S)	DESCRIPTION
IPEI	Handset IPEI	Displays the handset's IPEI. ***NOTE:*** *For an already registered handset, changing the IPEI will deregister the handset at the next location update.*
PAIRED TERMINAL	No Paired Terminal	You can pair the handset with an RTX8930 headset. For more details, please read the RTX8930 user guide. ***NOTE:*** *For RTX9431 this option is available only if the Local Call groups feature is disabled* ***NOTE:*** *When pairing a RTX725x headset the user must consider the following: When the headset is paired with the terminal idx of a handset, it assumes the extension data of the handset. If pairing from the Handset page and terminal idx of the headset is chosen, the handset assumes the extension details of the headset. For details refer to the* ***RTX 725x Headsets System Guide.***

PUSH-TO-TALK (PTT)	Disabled	The feature allows having P2P walkie-talkie, voice calls between handsets and headsets on the VoIP system. Enable the feature to broadcast a PTT conference call to all portable devices.
AC	FFFF	Displays the handset's Access code. The AC is used at handset registration. ***NOTE:*** *Changing the AC code for an already registered handset will have no effect.*
ALARM LINE	No Alarm Line Selected	The line of multiline to be used for alarm call feature
ALARM NUMBER	Empty	Number to be dialed in case of a triggered alarm. ***NOTE:*** *Alarms must be assigned to the handset in order for this feature to function*

7.1.4.1 Beacon settings

This section allows the administrator to enable the handsets to receive or transmit beacons. These settings are displayed only if the devices support the feature.

NOTE: *RTX8630, RTX8431and RTX8436 do not support this feature*

PARAMETER	DEFAULT VALUES	DESCRIPTION
RECEIVE MODE	Disabled	If **enabled,** the handset will list beacons in vicinity of the device and report it to an external server. ***NOTE:*** *When this feature is configured and an alarm is triggered, the strongest beacon will be included in the data sent to the message server.* Available options: **Enter/ Leave/ Enter or Leave proximity:** Report to the external server when the handset ***enters/ leaves/ enters or leaves*** vicinity of a beacon
RECEIVE SENSITIVITY	Low	Defines how close the handset should be to the beacon before detecting enter / leave Available options: **Low/ Medium/ High** - The handset should be in ***close/ medium/ long*** range to the beacon to detect enter / leave
RECEIVE SELECT	N/A	Select which beacon advertising packets should be listed by the handset Available options: **iBeacon/ Altbeacon/ Eddystone**: Listing for beacons transmitting an ***iBeacon/ Altbeacon/ Eddystone*** advertising packet
TRANSMIT INTERVAL	Disabled	If **enabled** the handset will transmit beacons. Define the interval between the beacon transmits Available options:

		Short/ Step x/ Long
TRANSMIT POWER	Low	Define the range of the transmit Available options: **Low/ Step x/ High**
TRANSMIT SELECT	N/A	Select which beacon advertising packets should be transmitted by the handset. Available options: **iBeacon/ Altbeacon/ Eddystone**

7.1.4.2 Alarm profiles

Alarm profiles are listed as soon as they are configured in the Alarm menu (refer to ***7.1.17 Alarm***). Check the box of the desired profile for the specific handset. After configuration, the handset must be rebooted.

NOTE: *The alarm feature is only available for some type of handsets (RTX8631, RTX8632, RTX8633 and RTX8830).*

NOTE: *RTX8630 cannot send alarms, but it can receive them*

7.1.4.3 Shared call appearance (SCA) settings

Each of the eight rows in the table represents an SCA status LED on the handset Idle screen. For each row it is possible to specify which shared line an LED should display the state of.

- Only shared lines can be selected. This means that only extensions, defined for the handset for which BroadWorks Shared Call Appearance is enabled, are included in the selector.
- A shared line can be reused for several LEDs. Each LED with the same shared line then corresponds to different appearance-indexes for that line (1 LED = appearance-index 1, 2 LEDs = appearance-indexes 1 and 2, and so on).

It is not necessary to select a shared line for all the LEDs. If an LED is not assigned to a line, its position on the screen is simply empty.

NOTE: *For RTX9431 this option is available only if the Local Call groups feature is disabled*

7.1.4.4 Import Local phonebook

This feature enables the administrator to import a phonebook with *.csv/.txt/.lo* file format to the specific handset. This feature is using a browse file approach. After file selection, the load button must be pressed to load the file. The system supports only the original *.csv* format. Please note, that some excel *.csv* formats are not the original *.csv* format.

For details on the contact list structure, please refer to ***9.1 Contact list structure.***

NOTE: *Depending on the handset type, a different number of entries can be saved:*

RTX843x – 50 entries

RTX863x, RTX8830 – 250 entries

RTX8630 – 100 entries

7.1.4.5 Export Local phonebook

The Export local phonebook feature makes it possible to retrieve all contracts from a specific handset to a *.csv* file format. The export button must be selected and then the file can be saved in a *.csv* format on the PC or server.

NOTE*: "Other" is not exported as this is considered private data.*

7.1.4.6 Menu Configuration

This submenu of Extensions allows the admin to restrain the user from manipulating device settings for the handsets.

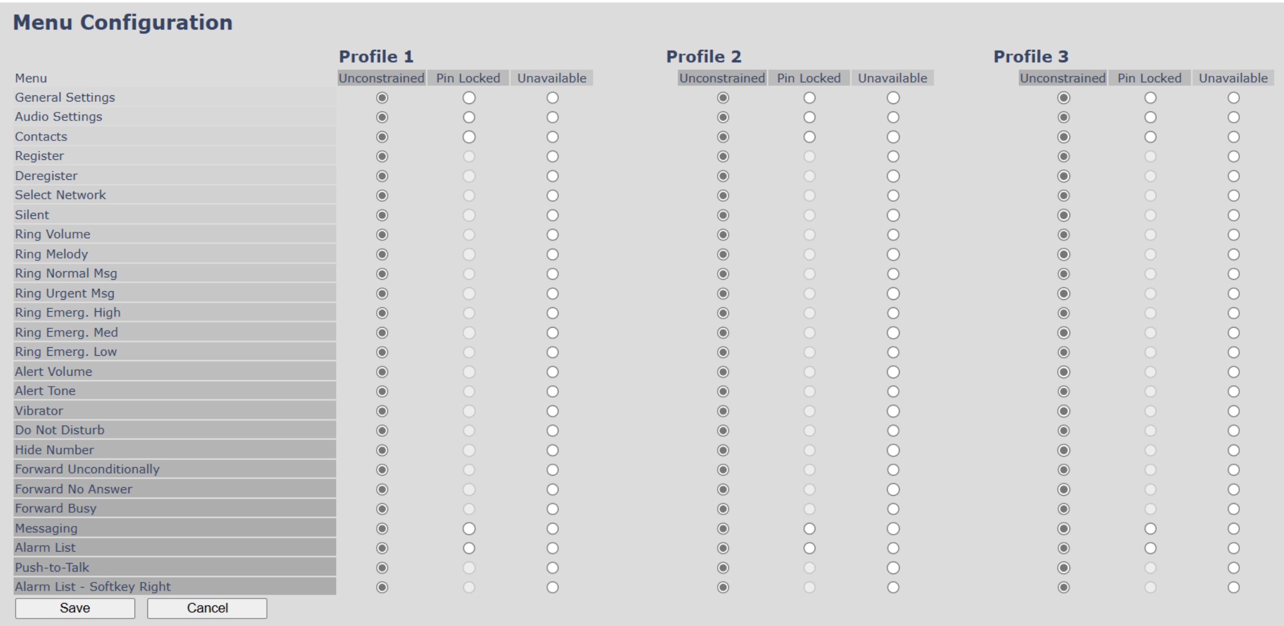


Figure 26 Extensions: Menu Configuration

7.1.5 Servers

This section presents and describes the different parameters available in the **Servers** configuration page *(Fig.26)*. Please note, that a maximum of 10 servers can be configured. Furthermore, within servers or even with multi servers, extensions must always be unique. This means that, the same extension number on server 1 cannot be re-used on server 2.

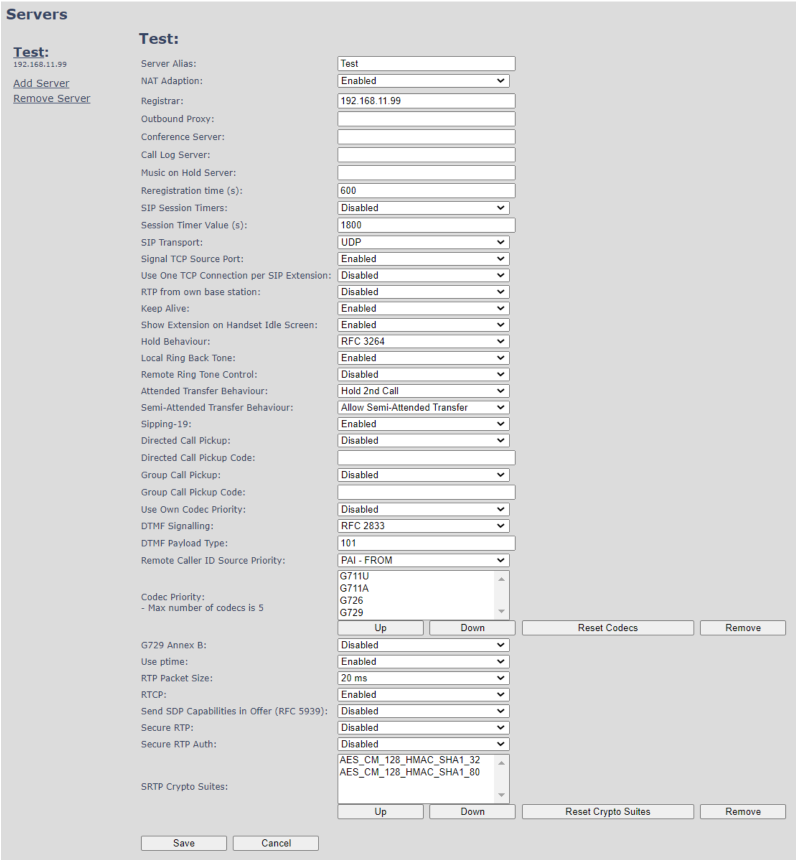


Figure 27 Servers page

PARAMETER	DEFAULT VALUE	DESCRIPTION
SERVER ALIAS	Empty	Displays the server name
NAT ADAPTION	Disabled	To ensure all SIP messages go directly to the NAT gateway in the SIP aware router. If **enabled** and the system receives a SIP response to a REGISTER request with a “Via” header that includes the “received” parameter (ex: “Via: SIP/2.0/UDP 10.1.1.1:4540;received=68.44.20.1”), the base will adapt its contact information to the IP address from the “received” parameter. Thus, the base will issue another REGISTER request with the updated contact information. If NAT Adaption is **disabled**, the “received” parameter is ignored.
REGISTRAR	Empty	SIP Server proxy DNS or IP address Permitted value(s): AAA.BBB.CCC.DDD:<Port-Number> or <URL>:<Port-Number> Example: “192.168.0.1”, “192.168.0.1:5062”,

		“nat.company.com” and “sip:nat@company.com:5065”. ***NOTE:*** *Specifying the port number is optional*.
OUTBOUND PROXY	Empty	This is a Session Border Controller DNS or IP address (Or SIP server outbound proxy address) Set the Outbound proxy to the address and port of the private NAT gateway, so that the SIP messages will be sent via the NAT gateway. Permitted value(s): AAA.BBB.CCC.DDD or <URL> or <URL>:<Port-Number> Examples: “192.168.0.1”, “192.168.0.1:5062”, “nat.company.com” and “sip:nat@company.com:5065”. If the value in the field is empty, then a call is made via Register.
CONFERENCE SERVER	Empty	Broadsoft conference feature. Set the IP address of the conference server. In case an IP is specified, pressing the conference option on the handset will establish a connection to the conference server. If the field is empty, the original 3-party local conference is used.
CALL LOG SERVER	Empty	Broadsoft call log feature. Set the IP address of the XSI call log server. If the field is empty, the local call log is used ***NOTE:*** *An internal call log cannot be accessed by the handset, if an external one is present*
MUSIC ON HOLD SERVER	Empty	Add the address of a server to ensure that music is on when call is on hold
RE-REGISTRATION TIME	600	The “expires” value in SIP REGISTER requests. This value indicates how long the current SIP registration is valid, and hence it specifies the maximum time between SIP registrations for the given SIP account. Permitted value(s): A value below 60 sec is not recommended, **Max value** 65636
SIP SESSION TIMERS:	Disabled	RFC 4028. A “keep-alive” mechanism for calls. The session timer value specifies the maximum time between “keep-alive” or more correctly session refresh signals. If no session refresh is received, as soon as the timer expires, the call will be terminated.

		Default value is 1800s according to the RFC. **Min:** 90s. **Max:** 65636s. If **disabled**, session timers will not be used.
SESSION TIMER VALUES (S):	1800	Default value is 1800s according to the RFC. If **disabled**, session timers will not be used. **Min**: 90s, **Max:** 65636s
SIP TRANSPORT	UDP	Select the protocol that is to be used during SIP calls (UDP, TCP or TLS 1.2)
SIGNAL TCP SOURCE PORT	Disabled	When SIP Transport is set to TCP or TLS, a TCP (or TLS) connection will be established for each SIP extension. The source port of the connection will be chosen by the TCP stack, and hence the local SIP port parameter, specified within the SIP/RTP Settings (*Error! Reference source not found. **SIP/RTP Settings***), will not be used. The "Signal TCP Source Port" parameter specifies if the used source port must be signaled explicitly in the SIP messages.
USE ONE TCP CONNECTION PER SIP EXTENSION:	Disabled	When using TCP as SIP transport, choose if a TCP/TLS connection should be established for each SIP extension, or if the base station should establish one connection which all SIP extensions use. Please note, that if TLS is used and SIP server requires client authentication (and requests a client certificate), this setting must be set to disabled. 0: **Disabled**. (Use one TCP/TLS connection for all SIP extensions) 1: **Enabled**. (Use one TCP/TLS connection per SIP extension).
RTP FROM OWN BASE STATION:	Disabled	If **disabled**, RTP stream will be send from the base where the handset is located. If **enabled**, the RTP stream will always be sent from the base, where the SIP registration is made.
KEEP ALIVE	Enabled	This directive defines the window period (30 sec.) to keep opening the port of relevant NAT-aware router(s), etc.
SHOW EXTENSION ON HANDSET IDLE SCREEN	Enabled	If **enabled**, the extension will be visible on handset idle screen. ***NOTE:*** *The name will not be displayed on handsets, using the modern MMI interface*
HOLD BEHAVIOUR	RFC 3264	Specify the hold behavior by the handset's "Hold" feature. RFC 3264: Hold is signaled according to RFC 3264, i.e. the connection information part of the SDP contains the IP

		address of the endpoint. Depending on the context, the direction attribute is sent only. RFC 2543: The ”old” way of signaling HOLD. The connection information part of the SDP is set to 0.0.0.0. Depending on the context, the direction attribute is sent only.
LOCAL RING BACK TONE	Enabled	In case the server does not play local ring back tone, the handset will do it.
REMOTE RING TONE CONTROL	Disabled	It enables the server to control the ring tone that is used on the handsets. When enabled, the ringtone selection will be available in Management page.
ATTENDED TRANSFER BEHAVIOUR	Hold 2nd Call	When there are two calls, and one call is on hold, it is possible to perform attended transfer. When the transfer soft key is pressed in this situation, the active call is placed on hold before the SIP REFER request is sent. However, it is experienced that some PBX’s do not expect that the 2nd call is put on hold, and therefore attended transfer fails on these PBX’s. The "Attended Transfer Behavior" feature defines whether the 2nd call should be put on hold before the REFER is sent. If **Hold 2nd Call** is selected, the 2nd call will be held before REFER is sent. If **Do Not Hold 2nd Call** is selected, the 2nd call will not be held before the REFER is sent
SEMI-ATTENDED TRANSFER BEHAVIOUR	Allow Semi-Attended Transfer	This feature can be enabled/disabled by selecting one of the following options: **Allow Semi-Attended Transfer** **Prohibit Semi-Attended Transfer**
SIPPING-19	Enabled	The setting controls Sipping-19 in a conference ***NOTE:*** *The conference package must be enabled by RTX in order to use the Sipping-19 feature*
DIRECT CALL PICKUP	Disabled	This is part of the BroadWorks SCA feature. If enabled, a direct call pickup code is sent to the handsets
DIRECT CALL PICKUP CODE	Empty	Code used to direct call pick up
GROUP CALL PICKUP	Disabled	Enable call group pickup
GROUP CALL PICKUP CODE	Empty	Code used to pick up a group call

USE OWN CODEC PRIORITY	Disabled	Disabled by default By enabling the system codec, priority during incoming call is used, instead of the calling party priority. E.g. If base has G722 as top codec and the calling party has a law on top and G722 is further down the list, the G722 will be chosen as a codec for the call.
DTMF SIGNALLING	RFC 2833	Conversion of decimal digits ('*' and '#') into sounds that share similar characteristics with voice to easily traverse networks designed for voice **SIP INFO:** Carries application-level data along SIP signaling path (e.g.: Carries DTMF digits generated during SIP session, or sending of DTMF tones via data packets in the same internet layer as the Voice Stream, etc.). **RFC 2833:** DTMF handling for gateways, end systems and RTP trunks (e.g.: Sending DTMF tones via data packets in different internet layers as the voice stream) **Both:** Enables SIP INFO and RFC 2833 modes.
DTMF PAYLOAD TYPE	101	This feature enables the user to specify a value for the DTMF payload type / telephone event (RFC2833).
REMOTE CALLER ID SOURCE PRIORITY	FROM	SIP information field used for Caller ID source **PAI - FROM** **FROM** **ALERT_INFO - PAI - FROM**
CODEC PRIORITY	G.711U G.711A G.726 G.722 G.729 OPUS BV32	Defines the codec priority that base stations, used for audio compression and transmission. Possible Option(s): G.711U, G.711A, G.726, G.722, G.729, OPUS, BV32 ***NOTE:** Modifications of the codec list must be followed by a "Reset Codecs" and "Reboot chain" from the multi cell page, to change and update handsets.* ***NOTE:** With G.722 as priority, the number of simultaneous calls per base station will be reduced from 10 to 5 calls. With G.722 in the list, the codec negotiation algorithm is active, causing the handset (phone) setup time to be slightly slower than if G.722 is removed from the list. Furthermore, it will reserve audio resources to be able to handle the possible G722 call and thus, the maximum number of possible narrowband calls will be reduced by one extra.*

		NOTE: *To use G.729 and OPUS, add on DSP module must be installed in all base stations. Contact your local dealer for price information.* ***NOTE:*** *If BV32 is the only codec used (the only one in the priority list), the user should use handsets that support BV32 (RTX8631-3), else a call will not be established.*
G729 Annex B		Enable/Disable Annex B of codec G729 ***NOTE:*** *Both parts must support it in order to avoid noise and any other kind of voice interruption*
USE PTIME	Enabled	Use the RTP Packet size, chosen in the below setting.
RTP PACKET SIZE	20ms	The packet size offered as preferred RTP packet size by 8630 when RTP packet size negotiation. Options available: **20ms, 40ms, 60ms, 80ms**
RTCP	Enabled	Enable/Disable RTCP
SEND SDP CAPABILITIES IN OFFER (RFC5939)	Disabled	Enable to support RFC 5939
SECURE RTP	Disabled	When enabled, RTP will be encrypted (AES-128) using the key negotiated via the SDP protocol at call setup.
SECURE RTP AUTH	Disabled	When enabled, secure RTP is using authentication of the RTP packages. ***NOTE:*** *With enabled SRTP authentication, maximum 10 concurrent calls are possible per base(RTX8663) in a single or multi cell system.*
SRTP CRYPTO SUITES	AES_CM_128_HMAX_SHA1_32 AES_CM_128_HMAX_SHA1_80	Field list of supported SRTP Crypto Suites. The device is born with two suites.

7.1.6 Network

This subchapter aims to present and describe the different parameters available in the Network configuration menu. Similar to the data representation on the *Network* page *(Fig.27),* the available settings are divided into sections and further explained in the following tables. All changes on this page require reboot to take effect.

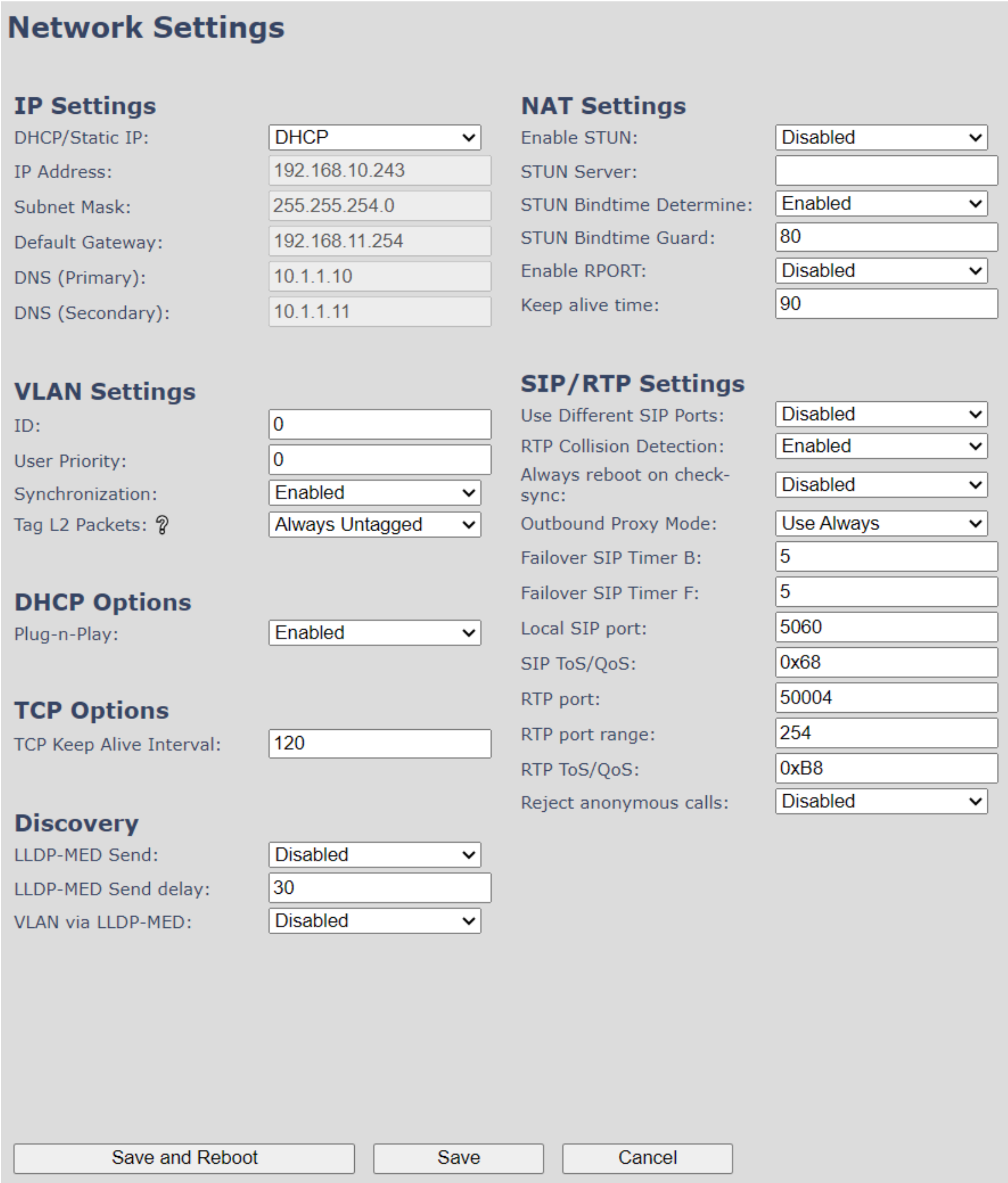


Figure 28 Network page

7.1.6.1 IP Settings

PARAMETER	DEFAULT VALUES	DESCRIPTION
DHCP/STATIC IP	DHCP	Select the technology that will be used to allocate an IP address to the base **DHCP:** IP addresses are allocated automatically from a pool of leased addresses. If the user selects the DHCP option, the other IP settings will not be editable. **Static IP:** The network administrator manually assigns IP addresses
IP ADDRESS	NA	32-bit IP address of the device (e.g. base station). 64-bit IP address will be supported in the future.

		Permitted value(s): AAA.BBB.CCC.DDD
SUBNET MASK	NA	The device's subnet mask. This is a 32-bit combination that describes which portion of the address identifies the network and which portion of the address identifies the node. **Permitted value(s):** AAA.BBB.CCC.DDD
DEFAULT GATEWAY	NA	Device's default network router/gateway (32-bit). The IP address of the network router that acts as an entrance to another network. This device provides a default route for TCP/IP hosts to use when communicating with other hosts on hosts networks. **Permitted value(s):** AAA.BBB.CCC.DDD
DNS (PRIMARY)	NA	Main server to which a device directs Domain Name System (DNS) queries. This is the IP address of a server that contains mappings of DNS domain names to various data, e.g. IP address, etc. You must specify this option when *static IP address* option is chosen. **Permitted value(s):** AAA.BBB.CCC.DDD or <URL>
DNS (SECONDARY)	NA	This is an alternate DNS server.
MDNS	Disabled	Enable to allow Multicast Domain Name system (MDNS)

7.1.6.2 VLAN settings

Enable users to define devices (e.g. base station, etc.) with different physical connection to communicate as if they are connected on a single network segment.

The VLAN settings can be used on a managed network with separate Virtual LANs (VLANs) for sending voice and data traffic. To work on these networks, the base stations can tag voice traffic, generated on a specific "voice VLAN", using the IEEE 802.1q specification.

PARAMETER	DEFAULT VALUES	DESCRIPTION
ID	0	A 12-bit identification of the 802.1Q VLAN. **Permitted value(s):** 0 to 4094 (only decimal values are accepted) A VLAN ID of 0 is used to identify priority frames and ID of 4095 is reserved. Null means no VLAN tagging or No VLAN discovery through DHCP.
USER PRIORITY	0	This is a 3-bit value that defines the user priority. **Permitted value(s):** 8 priority levels (i.e. 0 to 7) Values are from 0 (best effort) to 7 (highest); 1 represents the lowest priority. These values can be used to prioritize different classes of traffic (voice, video, data, etc.).
SYNCHRONIZATION	Disabled	When enabled, the VLAN ID is automatically synchronized between the bases in the chain. Bases will be automatically rebooted during the synchronization. ***NOTE:*** *If using different VLANs, the synchronization must be disabled. The setting must be changed on each base station as there is no synchronization between them.* ***NOTE:*** *This feature is not supported by RTX9431*
TAG L2 PACKETS	Always Untagged	This feature allows the user to configure the tagging of LLDP/CDP/EAP packets. It can either be set to **Always Untagged** or to **Follow VLAN ID.**

7.1.6.3 DHCP Options

PARAMETER	DEFAULT VALUES	DESCRIPTION

PLUG-N-PLAY	Enabled	DHCP option 66 is enabled by default to automatically provide PBX IP address to the base.

7.1.6.4 TCP Options

PARAMETER	DEFAULT VALUES	DESCRIPTION
TCP KEEP ALIVE INTERVAL	120s	Specifies the interval that the client waits for before sending a keep-alive message on a TCP connection.

7.1.6.5 Discovery

PARAMETER	DEFAULT VALUES	DESCRIPTION
LLDP-MED SEND	Disabled	If **enabled,** the base station will send 5 LLDP-MED messages when started
LLDP-MED SEND DELAY	30	Sends messages every 30 seconds to inform the network about its LLDP-MED data ***NOTE:*** *This option works only if the first parameter is enabled (LLDP-MED SEND)*
VLAN VIA LLDP-MED	Disabled	If **enabled**, the base station will try to retrieve a VLAN ID from the received LLDP-MED from a switch ***NOTE:*** *This feature is available only if the first parameter is enabled (LLDP-MED SEND)*
CDP SEND	DISABLED	Enable to send CDP messages ***NOTE:*** *This feature is not supported by RTX9431*
CDP SEND DELAY	60	Define the delay between messages in seconds ***NOTE:*** *This feature is not supported by RTX9431*

7.1.6.6 NAT settings

The following table defines some options available when NAT aware routers are enabled in the network.

PARAMETER	DEFAULT VALUES	DESCRIPTION
ENABLE STUN	Disabled	Enable to use STUN
STUN SERVER	Empty	**Permitted value(s):** AAA.BBB.CCC.DDD (Currently only Ipv4 is supported) or **URL** (e.g.: firmware.rtx.net).
STUN BINDTIME DETERMINE	Enabled	Enable/Disable the use of STUN Bindtime
STUN BINDTIME GUARD	80	**Permitted values:** Positive integer; default is 90; unit is in seconds
ENABLE RPORT	Disabled	Enable to use RPORT in SIP messages.
KEEP ALIVE TIME	90	Defines the frequency of how keep-alive are sent to maintain NAT bindings. **Permitted values:** Positive integer; default is 90; unit is in seconds

7.1.6.7 SIP/RTP settings

PARAMETER	DEFAULT VALUES	DESCRIPTION

USE DIFFERENT SIP PORTS	Disabled	If **disabled**, the Local SIP port parameter specifies the source port used for SIP signaling in the system. If **enabled**, the Local SIP Port parameter specifies the source port used for first user agent (UA) instance. Succeeding UA's will get succeeding ports.
RTP COLLISION DETECTION	Enabled	**Enabled**: If there are two sources with the same SSRC, then the following RTX is discarded. **Disabled**: No check – device will accept all sources.
ALWAYS REBOOT ON CHECK-SYNC	Disabled	Reboot base station when new configuration is loaded.
OUTBOUND PROXY MODE	Use Always	**Use Always:** All outbound calls are sent to the outbound proxy **Only Initial request:** Only use outbound proxy for initial SIP requests
FAILOVER SIP TIMER B	5	When the time expires and the corresponding SIP transaction fails, failover will be triggered
FAILOVER SIP TIMER F	5	When the time expires and the corresponding SIP transaction fails, failover will be triggered
LOCAL SIP PORT	5060	The source port used for SIP signaling **Permitted values:** Positive integer
SIP TOS/QOS	0x68	Priority of call control signaling traffic, based on both IP Layers of Type of Service (ToS) byte. ToS is referred to as Quality of Service (QoS) in packet-based networks. **Permitted values:** Positive integer, default is 0x68
RTP PORT	50004	The first RTP port to be used for RTP audio streaming. **Permitted values:** Positive integer
RTP PORT RANGE	40	The number of ports that can be used for RTP audio streaming. **Permitted values:** Positive integers, default is 40
RTP TOS/QOS	0xB8	Priority of RTP traffic based on the IP layer ToS (Type of Service) byte. ToS is referred to as Quality of Service (QoS) in packet-based networks. See RFC 1349 for details. "Cost bit" is not supported. o Bit 7..5 defines precedence. o Bit 4..2 defines Type of Service. o Bit 1..0 are ignored. Setting all three of bit 4..2 will be ignored. **Permitted values:** Positive integer, default is 0xB8
REJECT ANONYMOUS CALLS	Disabled	If **disabled**, all calls will be received. If **enabled**, anonymous calls will be automatically rejected at the PBX. This means that these calls will be rejected regardless of what is set on the individual extension page ***NOTE:*** *If you wish to enable this feature only for certain extensions, then the setting must be set individually (refer to **7.1.3.1 Add (edit) extension editor menu** for more information)*

7.1.7 Management

The Management menu allows the administrator to configure the base stations to perform specific functions, such as enabling text messaging, configuration file download, etc. The available settings are divided into sections, similar to the overview of the *Management* page *(Fig.28),* and further described in the following tables.

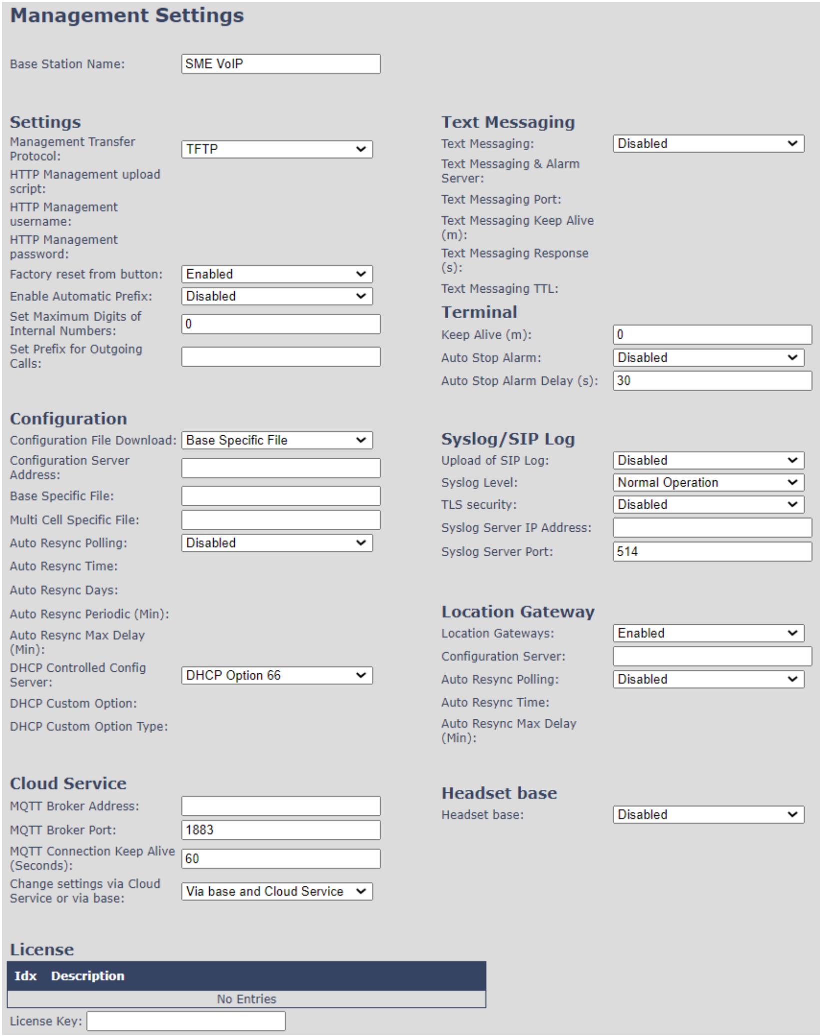


Figure 29 Management page

7.1.7.1 Settings

PARAMETER	Default value	Description
BASE STATION NAME	SME VoIP	It indicates the title that appears at the top window of the browser and is used in the *Multi cell* page **Maximum characters:** 35
MANAGEMENT TRANSFER PROTOCOL	TFTP	The protocol assigned for configuration file and central directory **Valid Input(s):** TFTP, HTTP, HTTPs
HTTP MANAGEMENT UPLOAD SCRIPT	Empty	The folder location or directory path that contains the configuration files of the Configuration server. The configuration upload script is a file

		located in, for example a TFTP server or Apache Server, which is also the configuration server. **Permitted value(s):** /<configuration-file-directory> *Example: /CfgUpload* ***NOTE:*** *Must begin with (/) slash character. Either / or \ can be used.*
HTTP MANAGEMENT USERNAME	Empty	Username should be entered to provide access to the configuration server. **Permitted value(s):** 8-bit string length
HTTP MANAGEMENT PASSWORD	Empty	Password should be entered to provide access to the configuration server. **Permitted value(s):** 8-bit string length
FACTORY RESET FROM BUTTON	Enabled	If **enabled**, a factory reset will be possible by pressing the button on the base station If **disabled**, no action will be present by pressing the button on the base station.
ENABLE AUTOMATIC PREFIX	Disabled	**Disabled**: Feature is turned off **Enabled**: The base will add the leading digit defined in "Set Prefix for Outgoing Calls". **Enabled + fall through on * and #:** The base will enable detection of * or **#** at the first digit of a dialed number, meaning that users can type any of these symbols before dialing a number. In addition to this, the leading digit defined in "Set Prefix for Outgoing Calls" is still valid, which means that it will be added by the PBX to the dialed number. *Examples:* 1: dialed number on handset *1234 - > dialed number to the PBX *1234 2: dialed number on handset #1234 - > dialed number to the PBX #1234 3: dialed number on handset 1234 - > dialed number to the PBX **0**1234
SET MAXIMUM DIGITS FOR INTERNAL NUMBERS	0	Used to detect internal numbers. In case of internal numbers, no prefix number will be added to the dialed number.
SET PREFIX FOR OUTGOING CALLS	Empty	Set the prefix for outgoing calls. Users need to dial this prefix to get an outside line.

7.1.7.2 Configuration

There are three ways of configuring the system:

- Manual configuration via the Web server (Web UI)
- Manual upload of a configuration file from a local drive, via the "Configuration" page on the Web server (Web UI) ***(see 7.1.21 Configuration)***
- Automatic download of configuration files from a configuration server (see the table below)

PARAMETER	Default value	Description
CONFIGURATION FILE DOWNLOAD	Disabled	**Base Specific file:** Used when configuring a single cell base **Base and Multi cell Specific File:** Used on out of factory bases to specify VLAN and settings / used when configuring a multi cell system **Multi cell Specific file:** Used when a multi cell system is present ***NOTE:*** *For RTX9431, the last two settings are displayed as "Dual Cell Specific file"*

CONFIGURATION SERVER ADDRESS	Empty	Server/device that stores the configuration file of the base station. **Type:** DNS or IP address **Permitted value(s):** AAA.BBB.CCC.DDD or <URL>
BASE SPECIFIC FILE	Empty	The file should be named after the MAC address of the base. An *example* is the file *00087b1573ba.cfg* In a multi/dual cell system, the base specific file contains the network settings (*for example*: vlan, static IP, MAC Address, etc.)
MULTI CELL SPECIFIC FILE	Empty	The file contains all data, except the specific network related settings (vlan, static IP, MAC address,etc). The file should be named after the chain ID number of the system **Permitted value(s):** Format of file is *chain ID.cfg* ***NOTE:*** *For RTX9431, this field is displayed as "Dual Cell Specific file"*
AUTO RESYNC POLLING	Disabled	Enable the base station to look for a new configuration file, within a predefined time interval
AUTO RESYNC TIME	00:00	A 24-hour setting Specify the time when the base station should load the configuration file
AUTO RESYNC DAYS	0	Number of days between Auto Resync
AUTO RESYNC PERIODIC (MIN)	0	Number of minutes between Auto Resync
AUTO RESYNC MAX DELAY (MIN)	15	Delay time in min, to prevent all base stations from asking for the configuration file at the same time
DHCP CONTROLLED CONFIG SERVER	Disabled	Provisioning server options. **DHCP Option 66:** Search for the provisioning file via the TFTP boot up server. **DHCP Custom Option:** Search for the provisioning file via the custom option **DHCP Custom Option & Option 66:** Search for the provisioning file, first via custom option and then via Option 66 From v460, the base station supports configuration files of up to 1 MB
DHCP CUSTOM OPTION	Empty	By default, it is option 160, but custom option can be defined. An option 160 URL defines the protocol and path information by using a fully qualified domain name for clients that can use DNS.
DHCP CUSTOM OPTION TYPE	Empty	**URL:** URL of server with path *Example of URL:* http://myconfigs.com:5060/configs **IP Address:** IP of server with path. Default configuration file on server must follow the name: *MAC.cfg*

7.1.7.3 Cloud Service

This section presents the available settings for configuring the RTX Cloud service.

NOTE: *This feature is not supported by RTX9431*

PARAMETER	Default value	Description
MQQT BROKER ADDRESS	Empty	Broker IP address or domain name

		NOTE: *The connection between the Cloud Service and the base is stopped if the broker URL is disabled.*
MQQT BROKER PORT	1883	Broker port (the port on the Broker URL) **Maximum value:** 65535 Port 1883 **–** MQQT connection is TCP Port 8883 – MQQT connection is TLS
MQQT CONNECTION KEEP ALIVE (SECONDS)	60	Defines how long the MQTT connection should be kept alive before data transferred is required. **Maximum value:** 255
CHANGE SETTINGS VIA CLOUD SERVICE OR VIA BASE	Via base and Cloud Service	It is possible to decline settings request from the Cloud Service. This field can also enable Cloud Control for 24 hours, meaning that it is possible to change the settings on the base for 24 hours.

7.1.7.4 License

PARAMETER	DEFAULT VALUE	DESCRIPTION
LICENSE	None	This feature allows the administrators to register RTX8930 generic headsets to the system. License key must be obtained from authorized resellers and only license matching the system's provider code will work. ***NOTE:*** *Changes to this feature require base station reboot.*

7.1.7.5 Text messaging

PARAMETER	DEFAULT VALUE	DESCRIPTION
TEXT MESSAGING	Disabled	**Enable** messaging with the use of a Message/Alarm server **Enable Without Server:** this allows a handset to send messages to other handsets in the same system configuration ***NOTE:*** *Changes to this feature require base station reboot.*
TEXT MESSAGING & ALARM SERVER	Empty	**Permitted value(s):** AAA.BBB.CCC.DDD or <URL> ***NOTE:*** *Text messaging server uses UDP and does not support TLS*
TEXT MESSAGING PORT	1300	Port number of the message server
TEXT MESSAGING KEEP ALIVE (M)	30	Defines the frequency of how keep-alive messages are sent **Permitted values:** Positive integer, unit is in minutes
TEXT MESSAGING RESPONSE (S)	30	Defines the frequency of response timeout **Permitted values:** Positive integer, unit is in seconds
TEXT MESSAGING TTL	0	Defines the text messaging time to live **Permitted values:** Positive integer, unit is in seconds
SELECT ALARM SORTING	Priority	Allows the user to choose parameter to sort alarm messages after. Could be by **Priority** or **Timestamp**.

7.1.7.6 Terminal

PARAMETER	DEFAULT VALUE	DESCRIPTION
KEEP ALIVE (M)	0	If different from "0", the handset sends an **emergencyLocationMsg** containing the RSSI measurements with interval "x" that is set. **Permitted values:** Positive integer, unit is in minutes
AUTO STOP ALARM	Disabled	Enable to activate "Auto Stop Alarm Delay"

AUTO STOP ALARM DELAY (S)	30	Handset automatically stops alarm announcement (emergencySms) after "x" sec.

7.1.7.7 Syslog/SIP log

PARAMETER	DEFAULT VALUE	DESCRIPTION
UPLOAD OF SIP LOG	Disabled	Enable this option to save low level SIP debug messages on the server. The SIP logs are saved in the file format: **<MAC_Address><Time_Stamp>SIP.log**
SYSLOG LEVEL	Normal Operation	**Off:** No data is saved on the syslog server **Normal Operation:** Normal operation events are logged, as well as incoming calls, outgoing calls, handset registration, DECT location, call lost (due to busy state), critical system errors and general system information. **System Analyze:** Contains the messages from Normal Operation, including handset roaming and handset firmware update status **Debug:** Information is used for debugging purposes by RTX. Should not be enabled during normal operation. Only when an issue occurs and debugging is required.
TLS SECURITY	Disabled	If enabled, it uses encrypted TCP, else - UDP
SYSLOG SERVER IP ADDRESS	Empty	Type in the IP address of the syslog server. **Permitted value(s):** AAA.BBB.CCC.DDD or <URL>
SYSLOG SERVER PORT	514	Port number of the syslog server.

7.1.7.8 Location Gateway

PARAMETER	DEFAULT VALUE	DESCRIPTION
LOCATION GATEWAYS:	Disabled	Enable to allow location gateways to register to the system. ***NOTE:*** *After enabling the feature, a new menu will be added (Location gateways) on the left-handed navigation panel.*
CONFIGURATION SERVER:	Empty	Server/device that stores the configuration file of the location gateway **Permitted value(s):** AAA.BBB.CCC.DDD or <URL>
AUTO RESYNC POLLING	Disabled	The poll enables the configuration server to start the provisioning process for the location gateway devices that it chooses. If **enabled**, it enables an automatic configuration synchronization poll for the location gateway(s)
AUTO RESYNC TIME	00:00	The time of the day that the automatic synchronization will occur (HH:MM).
AUTO RESYNC MAX DELAY (MIN)	15	To distribute the load as such, that all location gateways do not sync at the exact same time, the Auto Resync Max Delay can be set to delay the poll trigger with a given number of minutes. The delay will be a random number between 0 minutes and the value given in this parameter.

7.1.7.9 Headset base

The following table presents the available settings for enabling the *Headset base station.*

NOTE: *This feature is not supported by RTX9431*

PARAMETER	Default value	Description
HEADSET BASE	Disabled	Enable the setting to allow headset base registrations

		NOTE: *After enabling the feature, a new menu will be added (Headset base) on the left-handed navigation panel.*

7.1.7.10 Ring Tone Selection

*This menu is only present when **Remote ring tone control** is enabled in **Servers**.*

Allows the user to modify the melodies and alert info for Remote ring tone control. The Alert info string can have max length of 20 bytes. (NOTE: Special characters may use more than 1 byte.)

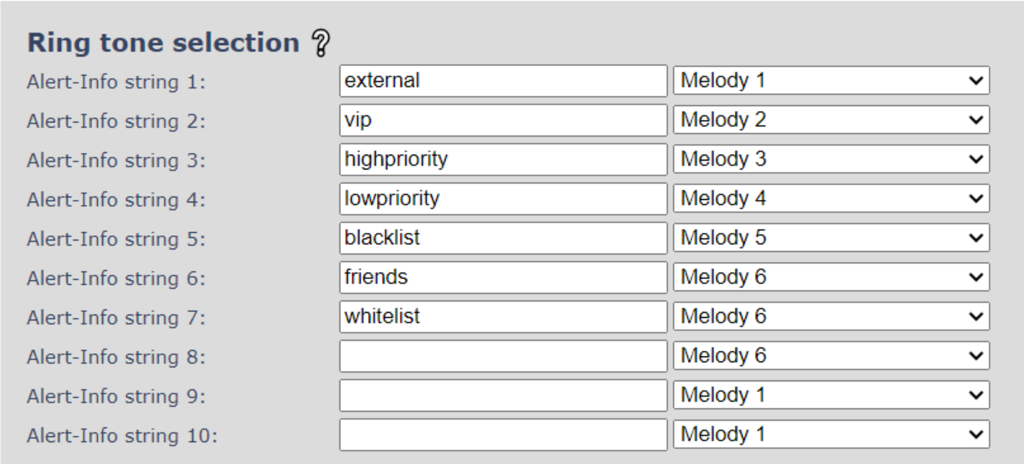


Figure 30 Ring tone selection

NOTE: *Something to be aware of when setting up things on the server side is that there needs to be an exact match between the received Alert-Info string from the SIP INVITE and the Alert-Info string otherwise default value is used.* ***Example:*** *Myfriends vs. MyFriends.*

7.1.8 Firmware update

The settings on this page allow the system administrator to set the desired firmware which the devices (base stations and SIP nodes) should upgrade/downgrade to and upload a startup/background picture to the handsets. The handset firmware/picture update status can be monitored on the *Extensions* page and the repeater firmware update status - on the *Repeaters* page. Base firmware update status can be found on the *Multi/Dual cell* or the *Home/Status* page. The screenshot below *(Fig.29)* illustrates the available options on the *Firmware* page, followed by a table with parameter definitions. For more details on how to upgrade/downgrade the firmware, or upload an image, please see ***8.4 Firmware Upgrade procedure.***

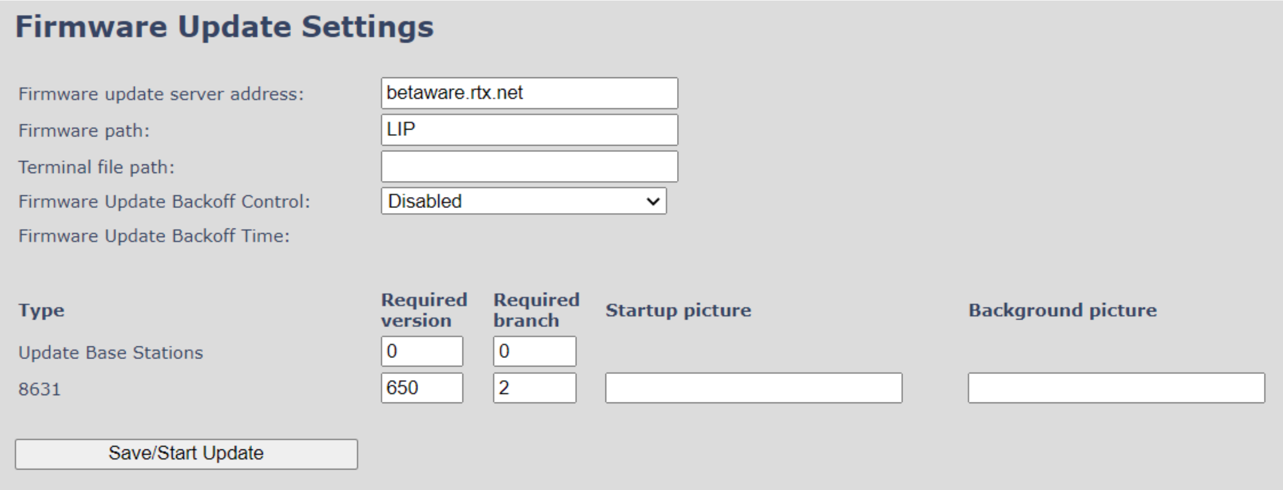


Figure 31 Firmware update page

PARAMETER	DEFAULT VALUE(S)	DESCRIPTION
FIRMWARE UPDATE SERVER ADDRESS	Empty	IP address or URL of the firmware server **Valid Inputs:** AAA.BBB.CCC.DDD or <URL>
FIRMWARE PATH	Empty	Location of the firmware files on the server (the folder where they are stored)
TERMINAL FILE PATH	Empty	Location of the image files (the folder where the background and start up image are stored, along with the voice prompt file for the headsets).
FIRMWARE UPDATE BACKOFF CONTROL	Disabled	In case of failed file download, this feature allows the user to set backoff control. That means the base station will retry downloading the firmware file again in the set time (depending on the mode chosen). There are 3 available modes: *Static, Linear* and *Exponential.*
FIRMWARE UPDATE BACKOFF TIME	Empty	When Backoff Update time enabled, we can enter the time interval for retry the firmware download.
REQUIRED VERSION	Empty	Version of the firmware to be upgraded (or downgraded) to on handset, repeater, base station, and headset. **Valid Input(s):** 8-bit string length. *E.g., 530* ***NOTE:*** *Value version "0" will disable the firmware upgrade* ***NOTE:*** *Two handset types will be serial firmware upgraded. For example, first type 8630, then type 8430.*
REQUIRED BRANCH	Empty	Branch of the firmware to be upgraded (or downgraded) to on handset, repeater, base station, and headset. **Valid Input(s):** 8-bit string length. *E.g., 1*
STARTUP PICTURE	Empty	Name of the startup picture you wish to have on the handsets when they are powered up. ***NOTE:*** *Images have the same resolution as the screen on the handset(s); Resolution info can be found in the handset datasheets. If the image does not have the same resolution as the screen, it will be placed in the top left corner. If it is too small, the rest of the screen will be black. If it is too large, only the left portion of the image will be shown.* ***NOTE:*** *Only .BMP is files are supported.*

BACKGROUND PICTURE	Empty	Name of the background picture you wish to have on the handset's screen ***NOTE:*** *Images have same resolution as the screen on the handset(s), Resolution info can be found in the handset datasheets.* *If the image does not have the same resolution as the screen, it will be placed in the top left corner. If it is too small, the rest of the screen will be black. If it is too large, only the left portion of the image will be shown* ***NOTE:*** *Only .BMP is files are supported.*
VOICE PROMPT	Empty	Name of the voice prompt file. This feature is only available for the headsets. ***NOTE:*** *When updating the voice prompt make sure that the correct Terminal path is given. It is also important to write the entire name of the file in the provided space (together with ".fwu").*

7.1.8.1 Firmware Update Backoff Control

This feature allows the user to set a backoff in case of failure to obtain the firmware update file from the server. The default value of the feature is *Disabled*. When clicked the arrow, we are given to choose from the dropdown menu between three modes that differ in the delay between each attempt.

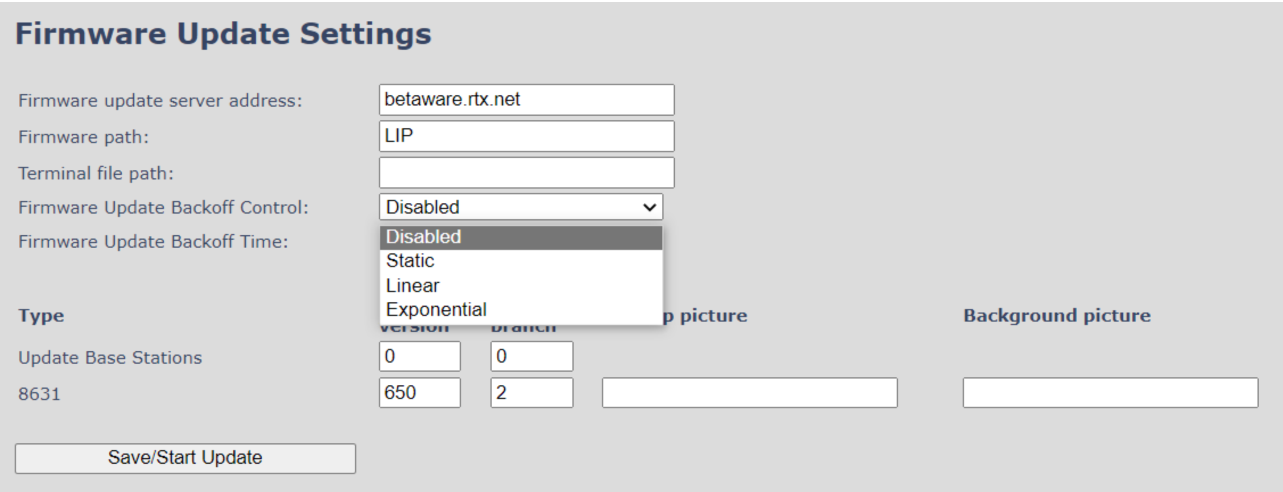


Figure 32 Firmware update Backoff Control

SETTING	DEFAULT VALUE	MAXUMUM VALUE
FIRMWARE UPDATE BACKOFF CONTROL	Empty	0. Disabled 1. Static 2. Linear 3. Exponential
FIRMWARE UPDATE BACKOFF TIME	1 (minutes)	360 (minutes) (6 hours total)

Static Backoff Control: When backoff control is set on *Static* the base station will wait for the chosen number of minutes before it reattempts the download.

Linear Backoff Control: When *Linear* backoff control is chosen the base station reattempts to download the firmware file in a linear manner. After failing to obtain the .fwu file the first time, it will try again in the set number of minutes (the value user has entered in *Firmware Update Backoff Time*). Every next attempt will be after the set Backoff Time multiplied with the number of attempts so far. That is limited to maximum of 5, meaning that after the 5th attempt it will continue trying after the same amount of time like the last one.

Backoff time cannot exceed 12 hours, meaning that if it reaches that maximum, it will just continue retrying every 12 hours.

Example: *If backoff time is set on 180 minutes (3 hours). The first attempt it will be 3 hours after failing, the next will be double that – 6 hours, then 9 hours and followed by another attempt in 12 hours. After that, as it had reached the maximum, system will keep trying to download the firmware on every 12 hours.*

Exponential Backoff Control: This setting allows the base to rise the time of attempts 5 times in exponential matter or until reaching the maximum of 12 hours. That means that the first time the base station will reattempt to download the firmware from the server after the time set by user multiplied by 2^X, where X is the number of attempts so far (from 1 to 5).

Example: *If the entered time by user is 2 minutes, the first attempt after the failure to download firmware will be on 2*2^1= 4 minutes. The next attempt will be after 2*2^2=8 minutes, then on 2*2^3=16 minutes, after that 2*2^4=32 and then it will rise the time to 2*2^5=64 minutes. Then the system will continue trying to download the firmware every 64 minutes.*

Backoff Control keeps running until the system succeeds downloading the firmware file or until it is disabled by the user.

NOTE: *The handsets have a limitation of only being allowed to attempt a download every 5 minutes, meaning if the timer is set to anything that isn't aligned with this, it will 'miss' its window and wait until the next one.*

NOTE: *If Backoff control is set to Static, Linear or Exponential, but the backoff time is set to 0, the base station will not reattempt the download until changes are made to the firmware version of the user starts the Firmware Update manually.*

NOTE: *In case Backoff control is enabled (either Static, Linear or Exponential) and backoff time is set to very high number (max 360 min), base station will keep attempting by scaling the time in the chosen method, but it will follow the rule of maximum 12 hours. That means, in the case of 360 minutes, base will attempt first after 6 hours and once it reaches the 12-hour limit, it will keep attempting every 12 hours*.

Firmware Update Backoff control can be provisioned via configuration file using the following variables: FWU_BACKOFF_MODE & FWU_BACKOFF_TIME. For more information on that refer to the latest ***Provisioning of SME VoIP System.pdf*** guide.

7.1.9 Location gateways

This menu is visible only if the "Location Gateways" feature is enabled on the *Management* page. Similar to the *Extensions* menu, this page provides information about the registered location gateway devices and allows the registration of new ones. A screenshot of the page can be seen below *(Fig.30)*, followed by a table with a description of the available options.

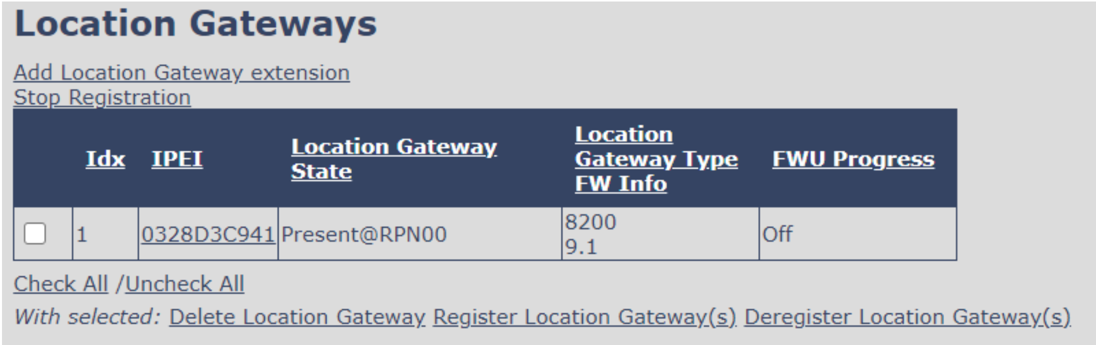


Figure 33 Location Gateways page

PARAMETER	DESCRIPTION
ADD LOCATION GATEWAY EXTENSION	Provides access to the Location Gateway editor menu, where the administrator can add new devices
STOP REGISTRATION	Manually stops DECT registration mode for the system. This prevents location gateways to register to the system
DELETE LOCATION GATEWAY	Deregisters and deletes the device from the VoIP base station
REGISTER LOCATION GATEWAY(S)	Enable registration mode for the selected location gateway entries.
DEREGISTER LOCATION GATEWAY(S)	Deregister the selected location gateway(s), but do not delete the entry
IDX	Index of the location gateway
IPEI	The IPEI of the location gateway. In general, this must not be changed manually. If IPEI is left at FFFFFFFFFF, any location gateway can register at this index. Else, only the location gateway with the specified IPEI number can register to the VoIP system ***NOTE:** If selected, the editor's menu will open*
LOCATION GATEWAY STATE	The state of the location gateway: **Present@RPNxx:** The location gateway is DECT located at the base with RPNxx **Enabled:** The slot is open for location gateway registration
LOCATION GATEWAY TYPE FW INFO	Name of the location gateway type Firmware version of the location gateway
FWU PROGRESS	Possible FWU progress states: **Off:** Means that the software version is specified to 0 = fwu is off **Initializing:** Means that the firmware is starting and the progress is 0%. **X%:** Firmware ongoing **Verifying X%:** Firmware writing is done and now it is verifying before the swap **FWU complete:** Upgrade is complete for the location gateway(s): **Error:** Not able to firmware upgrade the device (e.g. file not found, file not valid etc.)

7.1.10 Headset base

This menu is visible only if the "Headset base" feature is enabled on the *Management* page. Similar to the *Extensions* menu, this page provides information about the registered headset base devices and allows the registration of new ones. A screenshot of the page can be seen below *(Fig.31)*, followed by a table with a description of the available options.

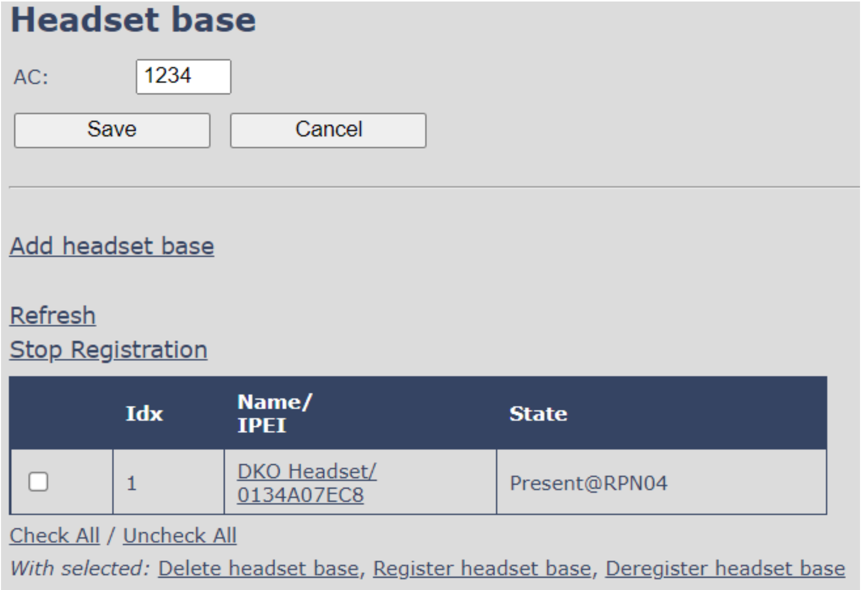


Figure 34 Headset base page

IMPORTANT: Both the headset base and the headset can be SIP registered to the VoIP System. However, depending on which device is SIP registered, the headset has a different behavior. If the headset base is registered to a SIP account, then the headset can be used in multiple environments – both in VoIP System and normal network. This means that a user can receive VoIP calls and at the same time use the headset for various PC applications, such as listening to music or talking via a softphone (Microsoft Teams). Moreover, these functionalities are available while moving around the VoIP system. However, if the headset is SIP registered as a standalone (without the headset base), then it can be used only for VoIP calls on the system. For more details on the Headset base and Wireless headsets, please refer to the relevant guide.

PARAMETER	DESCRIPTION
AC	Access code used when registering a headset base **Default value:** 0000
ADD HEADSET BASE	Provides access to the Headset base editor menu, where the administrator can add and edit devices
REFRESH	Refreshes the page
STOP REGISTRATION	Manually stops DECT registration mode for the system. This prevents headset bases to register to the system
CHECK ALL / UNCHECK ALL	Check/Uncheck all headset base entries in the table
DELETE HEADSET BASE	Deregisters and deletes the device from the VoIP base station
REGISTER HEADSET BASE	Enable registration mode for the selected headset base entries
DEREGISTER HEADSET BASE	Deregister the selected headset base(s), but do not delete the entry
IDX	Index of the headset base
NAME / IPEI	Headset base's name/IPEI. IPEI is a unique DECT identification number. ***NOTE:*** *If selected, the editor's menu will open*
STATE	The state of the headset base: **Present@RPNxx:** The headset base is DECT located at the VoIP base with RPNxx **Removed:** The headset base has been out of sight for a specified amount of time (~one hour). **Enabled:** The slot is open for headset base registration on this index

7.1.10.1 Headset base editor menu

The editor menu is presented to the administrator as soon as the "*Add headset base"* link is selected. The available settings are further explained in the following table.

PARAMETER	DEFAULT VALUES	DESCRIPTION
NAME	Empty	The name displayed on the headset base overview page
IPEI	FFFFFFFFFF	The IPEI of the headset base. In general, this must not be changed manually. If IPEI is left at FFFFFFFFFF, any headset base can register at this index. Else, only the headset base with the specified IPEI number can register to the VoIP system
TERMINAL	No Terminal	This setting enables the administrator to assign a SIP account to the headset base. The following options are available: **No Terminal:** No SIP account assigned **Terminal Idx x:** A free extension/SIP account slot which can be assigned to the headset base **New Terminal:** Creates a new terminal/extension slot on the ***Extensions*** page with the IPEI of the headset base. The administrator needs to edit the extension details in order to SIP register the headset base. ***NOTE:*** *By SIP registering the headset base, the user can use the headset both for VoIP calls and for other PC applications (such as music, softphones, etc.)*

7.1.11 Country

This menu allows the configuration of the country, language and time server for the base station *(Fig.32).*

The country setting controls the in-band tones used by the system.

The time server supplies the time used for data synchronisation in a multi/dual cell configuration. As such, it is mandatory to have the time server set for a multi/dual cell configuration, else the system will not work properly. Furthermore, the time server is used in the debug logs and SIP traces, along with determining the time to check for new configuration and firmware files.

NOTE: *It is not necessary to set the time server for standalone base stations (optional).*

NOTE: *When the time server parameters are modified/changed, synchronisation between base stations can take up to 15 minutes before all of them are synchronised (depending on the number of base stations in the system). Changing time settings will require a reboot of the system.*

NOTE: *By checked time zone and DST, the parameters in the Time section will be discarded.*

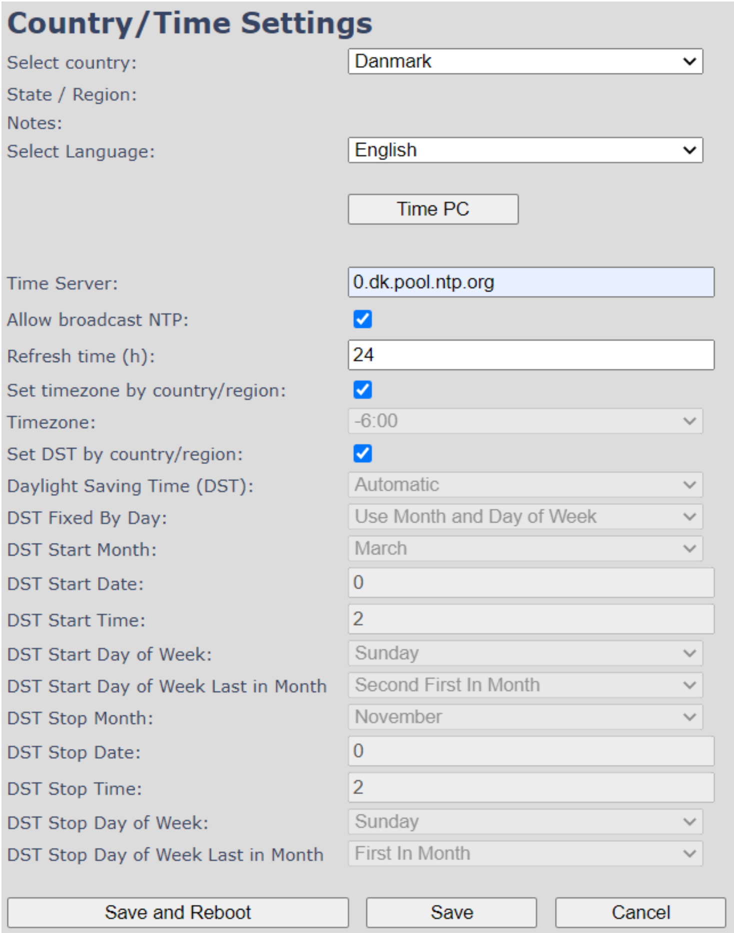


Figure 35 Country/Time settings page

PARAMETER	DEFAULT VALUES	DESCRIPTION
SELECT COUNTRY	US/Canada	Supported countries: Australia, Belgium, Brazil, Canada, Denmark, Germany, Greece, Spain, France, Ireland, Italy, Luxembourg, Netherlands, New Zealand, Norway, Portugal, Russia, Switzerland, South Africa, Finland, Sweden, Thailand, Turkey, United Kingdom, US, Austria, China
STATE / REGION	N/A	Displayed only when country selection is US, Canada, Australia or Brazil
NOTES	N/A	Displayed only when country selection is US, Canada or Russia
SELECT LANGUAGE	English	Web interface language. Available languages: English, Danish, Italian, Turkish, German, Portuguese, Croatian, Serbian, Slovenian, Dutch, French, Spanish, Russian, Polish.
TIME PC BUTTON	N/A	Use the current time from your PC as a time server
TIME SERVER	Empty	DNS name or IP address of an NTP server. Enter the IP/DNS address of the server that distributes reference clock information to its clients, including base stations, handsets, etc. **Valid Input(s):** AAA.BBB.CCC.DDD or URL (e.g. time.server.com) Currently, only Ipv4 address (32-bit) is supported.
ALLOW BROADCAST NTP	Checked	If checked, the time server is used.
REFRESH TIME (H)	24	The window time in hours in which the time server refreshes.

		Valid Inputs: positive integer
SET TIMEZONE BY COUNTRY/REGION	Checked	If checked, the country setting is used
TIMEZONE	0	Refers to local time in GMT or UTC format. **Min:** -12:00 **Max:** +13:00
SET DST BY COUNTRY/REGION	Checked	If checked, the country setting is used.
DAYLIGHT SAVING TIME (DST)	Disabled	The system administrator can **enable** or **disable** DST manually. **Automatic:** Enter the start and stop dates if this option is selected
DST FIXED BY DAY	Use Month and Day of week	You determine when DST changes. Choose the relevant date or day of the week, etc. from the drop-down menu.
DST START MONTH	March	The month that DST begins **Valid Input(s):** Gregorian months (e.g. January, February, etc.)
DST START DATE	0	Numerical day of the month DST comes to effect when DST is fixed to a specific date **Valid Inputs:** positive integer
DST START TIME	2	DST start time in the day **Valid Inputs:** positive integer
DST START DAY OF WEEK	Sunday	Day within the week DST starts
DST START DAY OF WEEK, LAST IN MONTH	Second First in Month	Specify the week for DST to start.
DST STOP MONTH	October	The month that DST stops.
DST STOP DATE	0	The numerical day of the month that DST turns off. **Valid Inputs:** positive integer (1 to 12)
DST STOP TIME	2	The time-of-day DST stops **Valid Inputs:** positive integer (1 to 12)
DST STOP DAY OF WEEK	Sunday	The day of the week DST stops
DST STOP DAY OF WEEK LAST IN MONTH	First in Month	The week within the month that DST will turn off.

7.1.12 Security

This menu takes into consideration the main security aspect of the system. It is used to load certificates, change web authentication and configure a secure web server *(Fig.33).* More details about the options can be found in the following table.

There are, however, some security settings which must be managed on other pages. For example, to setup secure firmware and configuration file download, the *Management* page must be used. Another example is securing the SIP and RTP, which are server dependent, meaning that the administrator must use the *Servers* page in order to configure them.

The Identity and Trusted certificates are preserved during upgrade. However, during a factory default, all certificates that have been installed via the web server, will be deleted. The base does not include any default trusted certificates.

***NOTE:** There is no expiration notification for the certificates.*

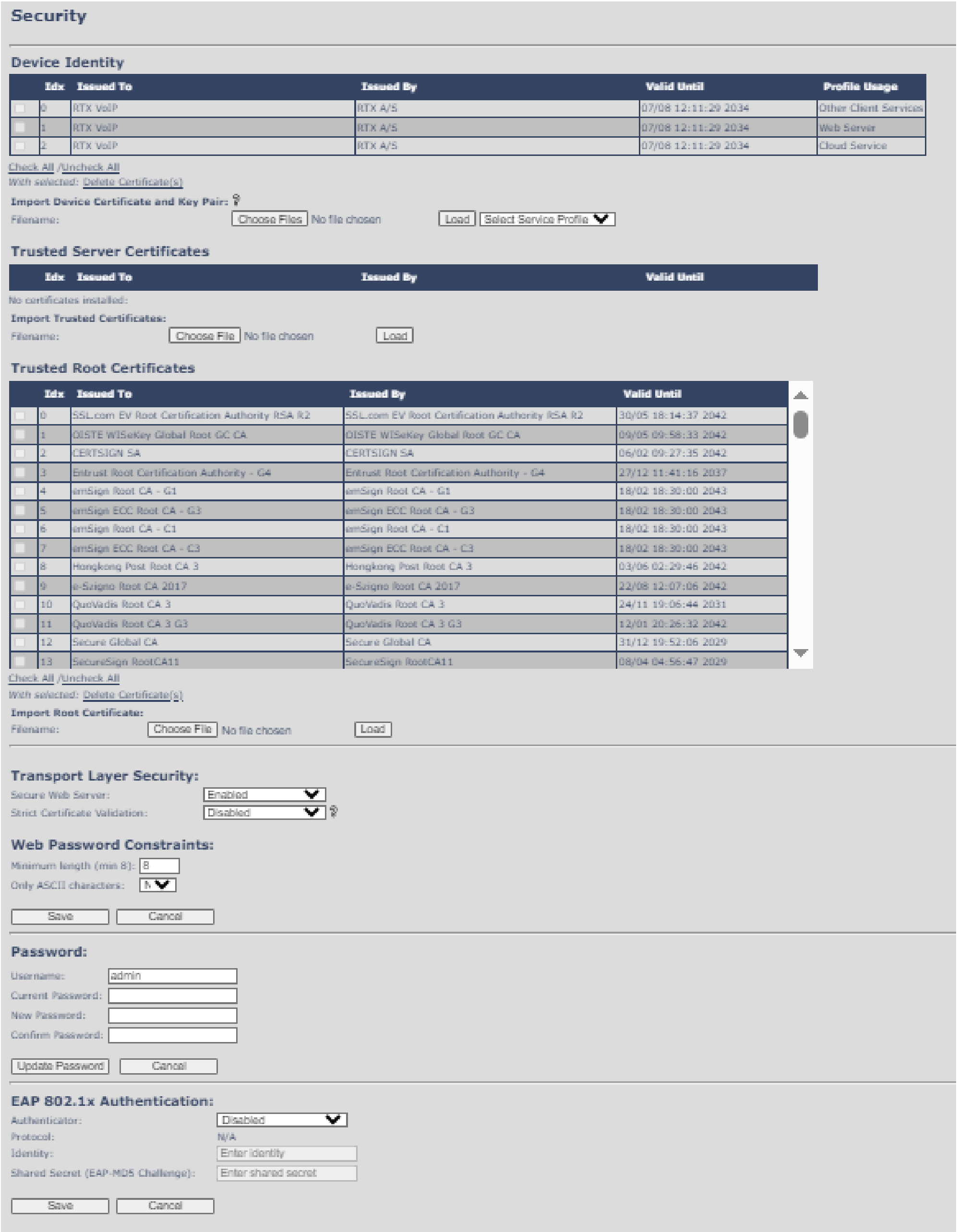


Figure 36 Security page

PARAMETER	DEFAULT VALUES	DESCRIPTION
DEVICE IDENTITY	N/A	The base stations support 3 different certificates 1: Other Client Service – The certificate needs to be pre-loaded during production and cant be changed. If no certificate is loaded during production, a dummy (self-signed) certificate generated on device and used as both client and server identity. 2: Web Server – This certificate is used for Secure Web Server access (HTTPS). 3: Cloud Service – Certificate need to access RTX Cloud service.

		NOTE: Certificate is not cleared during factory reset.
IMPORT DEVICE CERTIFICATE AND KEY PAIR	N/A	The certificate and private key used by the base when acting as a server or when the server requires client authentication in the TLS handshake procedure. When a Device Certificate is loaded, you need to select profile usage of certificated from dropdown. (Web Server or Cloud Service) The same certificate and key pair cannot be used for more than one profile.
IMPORT TRUSTED CERTIFICATES	N/A	Intermediate certificates (non-root certificates) trusted by the base. Used to validate a received certificate chain (or a chain of trust) in scenarios, where only the root certificate is sent by the server during the SSL handshake procedure.
IMPORT ROOT CERTIFICATE	N/A	Root certificates (self-signed) trusted by the base. Used to validate received root certificates sent by the server during the SSL handshake procedure.
SECURE WEB SERVER	Enabled	Allows the user to enable/disable the use of Secure Web Server.
STRICT CERTIFICATE VALIDATION	Disabled	If **enabled**, the certificate validity period will be checked. Therefore, the certificates which the base will receive from the server, must be valid and loaded into the system. If no valid matching certificate is found during the TLS connection establishment, the connection will fail. If the feature is **disabled**, all certificates received from the server will be accepted. ***NOTE:*** *It is important to use correct date and time of the system when using trusted certificates. In case of undefined time/date, the certificate validation can fail.*
WEB PASSWORD MINIMUM LENGTH	8	The minimum possible value is 8. User can extend that if desired up to 15.
ONLY ASCII CHARACTERS	No	Allows the user to define if the password can use only ASCII characters.
USERNAME	admin	Can be modified to any supported character and number. **Maximum characters:** 15
CURRENT PASSWORD	admin	Type in the current web-authentication password.
NEW PASSWORD*	Empty	Change to new password. **Maximum characters:** 15
CONFIRM PASSWORD	Empty	Confirm the password to ensure that there are no extra or less characters.
AUTHENTICATIOR	Disabled	When enabled it activates EAP 802.1x Authentication.
PROTOCOL	N/A	When the EAP Authentication is enabled, the protocol is automatically set. IT can be EAP-TLS Authentication or EAP-MD5 Challenge.
IDENTITY	Enter identity	This is the name the user chooses for the authentication.
SHARED SECRET (EAP-MD5 CHALLENGE)	Enter shared secret	Required when using EAP-MD5 Challenge protocol.

NOTE: *From v700 the user is forced to change their password on first login as well as after a password reset by the administrator.*

NOTE: In a chained setup of multi-cell or dual cell enabled bases, the bases may be provisioned using config files. If these config files <u>do not set the credentials</u>, then ONLY the synchronization <u>primary base</u> in the chain will be prompted for password change. It is therefore the responsibility of configurator to ensure credentials are correct either from synchronization primary base or via the configuration file! The other bases in the chain will be inaccessible until provisioned password is changed. Attempting to access with default credentials from a none-primary base will put following syslog: "Account locked due default credentials".

NOTE: *In v700 hashed passwords are introduced. Downgrade from this version is not recommended. However, if necessary, to downgrade from V700, the following guide should be followed:*

- Downgrade to V610B1-V610B3 requires to first downgrade to a FW higher than or equal to V610B4 AND lower than V700.

- Downgrade to V420-V530 requires first to downgrade to V530B10 or higher V530 FW versions.

Password valid special signs: @/|<>-_:.!?+#

*Password valid numbers: 0-9

*Password valid letters: a-z and A-Z

7.1.13 Central directory

The SME VOIP system supports three types of central directories - local central directory, LDAP and XML directory. Depending on the selected "Location" type, different configurable parameters will be displayed *(Fig.34)*. However, all directories' caller id look up is made with a match for 6 digits of the phone number.

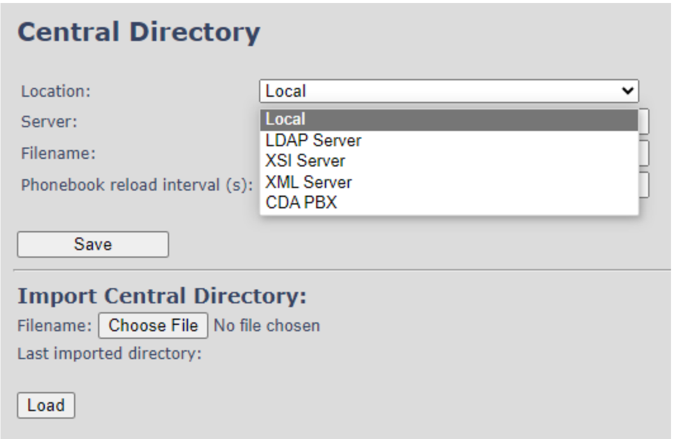


Figure 37 Central Directory

7.1.13.1 Local Central Directory

For details on the contact list structure, please refer to **9.1 Contact list structure.**

PARAMETER	DEFAULT VALUES	DESCRIPTION
LOCATION	Local	Drop-down menu to select between local central directory, LDAP based central directory and XML server ***NOTE:*** *Changes to this feature require base station reboot.*
SERVER	Empty	The parameter is used if the directory file is located on a server

		Valid inputs: aaa.bbb.ccc.ddd or <url>
FILENAME	Empty	The name of the file with the addresses. This parameter is used if the directory file is located on a server.
PHONEBOOK RELOAD INTERVAL (S)	0	This parameter is controlling the reload interface of a phonebook in seconds. The feature is for automatic reload of the base phonebook file from the server with intervals. It is recommended to specify a conservative value to avoid overload of the base station. With a default value setting 0, the reload feature is disabled. ***NOTE:*** *We recommend 86400 seconds (24 hours) as a minimum reload interval.*
IMPORT CENTRAL DIRECTORY	Empty	This feature enables the administrator to import a central directory file with *.csv/.txt/.xml* file format. This feature is using a browse file approach. After file selection, the load button must be pressed to load the file. The system supports only the original *.csv* format. Please note, that some excel *.csv* formats are not the original *.csv* format. The central directory feature can handle up to 3000 contacts (Max file size 100kb).

7.1.13.2 LDAP

To reach the LDAP configuration menu, select "LDAP server" from the drop-down menu *(Fig.33).*

PARAMETER	DEFAULT VALUES	DESCRIPTION
CENTRAL DIRECTORY LOCATION	LDAP Server	Drop-down menu to select between local central directory, LDAP based central directory and XML server. "LDAP Server" is displayed when LDAP server is selected.
SERVER	Empty	IP address of the LDAP server. **Valid Inputs:** AAA.BBB.CCC.DDD or <URL>
TLS SECURITY	Disabled	If enabled, it uses encrypted TCP, else – UDP ***Note:*** *In most cases LDAP over TLS is running on port 636*
SORT BY	commonName	Select the way the contacts are to be sorted. Choose between common name, given name and surname
PORT	Empty	The server port number that is open for LDAP connections ***Note:*** *In most cases LDAP over TLS is running on port 636*
SBASE	Empty	Search Base. The input depends on the configuration of the LDAP server. *Example* of the setting is *OU=umber,DC=umber,DC=loc*
LDAP FILTER	Empty	LDAP Filter is used as a search filter. For example, setting LDAP filter to *(|(givenName=%*)(sn=%*))* the base will use the filter when requesting entries from the LDAP server. % will be replaced with the entered prefix, e.g. searching on "J" will give the filter (|(givenName=J*)(sn=J*)) resulting in a search for given name starting with a "J" or surname starting with "J".
BIND	Empty	The username that will be used when the IP-DECT phone connects to the server. An example is the following: *CN=Dummy,OU=umber,DC=umber,DC=loc*
PASSWORD	Empty	The password for the LDAP Server
VIRTUAL LIST	Disabled	If enabled, virtual list searching will be possible

NAME	cn	The name can be used to specify if sn+givenName or cn (common name) is returned in the LDAP search results
WORK NUMBER	Empty	Work number is used to specify the LDAP attribute that will be mapped to the handset work number
HOME NUMBER	Empty	Home number is used to specify the LDAP attribute that will be mapped to the handset home number
MOBILE NUMBER	Empty	Mobile number is used to specify the LDAP attribute that will be mapped to the handset mobile number

7.1.13.3 XSI server

To be able to configure the XSI server, select this option from the drop-down menu. It allows the user to edit directory names and set priority of the directories shown on the handset when entering Central Directory.

PARAMETER	DEFAULT VALUES	DESCRIPTION
CENTRAL DIRECTORY LOCATION	XSI Server	Drop-down menu to select between local central directory, LDAP based central directory and XML server. “XSI Server” is displayed when XSI server is selected.
SERVER	Empty	IP address of the XML server. **Valid Inputs:** AAA.BBB.CCC.DDD or <URL>
ENTERPRISE	Enabled	Type of tag for the group/directory
ENTERPRISE COMMON	Enabled	Type of tag for the group/directory
GROUP	Enabled	Type of tag for the group/directory
GROUP COMMON	Enabled	Type of tag for the group/directory
PERSONAL	Disabled	Type of tag for the group/directory
DIRECTORY PRIORITY	Listed the inserted directory names	User can mark a directory name and edit its priority with the “Up” and “Down” button. It is possible to remove a name by pressing “Remove” and reset priority to the previous state by pressing “Reset Priority”

7.1.13.4 XML server

To reach the XML configuration menu, select “XML server” from the drop-down menu *(Fig.35).* That allows the user to add an XML server.

PARAMETER	DEFAULT VALUES	DESCRIPTION
CENTRAL DIRECTORY LOCATION	XML Server	Drop-down menu to select between local central directory, LDAP based central directory and XML server. “XML Server” is displayed when XML server is selected.
SERVER	Empty	IP address of the XML server. **Valid Inputs:** AAA.BBB.CCC.DDD or <URL>

NOTE: *In firmware version v720b200 and earlier XSI server and XML server are not separate features, and they can both be edited by selecting the “XML server” option in the drop-down menu.*

7.1.14 Dual/Multi cell

The following sub-chapter presents the available settings on the Dual/Multi cell page *(Fig.36)*. These are divided into several sections, which will be described as follows: *Settings for this unit, DECT System settings, Base station settings, Base station group* and *DECT chain.*

NOTE: *To join two or more base stations in a dual/multi cell system, you need to have one extension or handset added to the system. For further details and a Step-by-Step guide to a multi cell setup, please refer to* ***8.1 Multi cell system.***

NOTE: *RTX8663 base can be added to a multi cell with the old RTX8660 base station and thus, run in a mixed mode. For further details, please refer to* ***9.2 RTX8663-RTX8660 Mixed mode.***

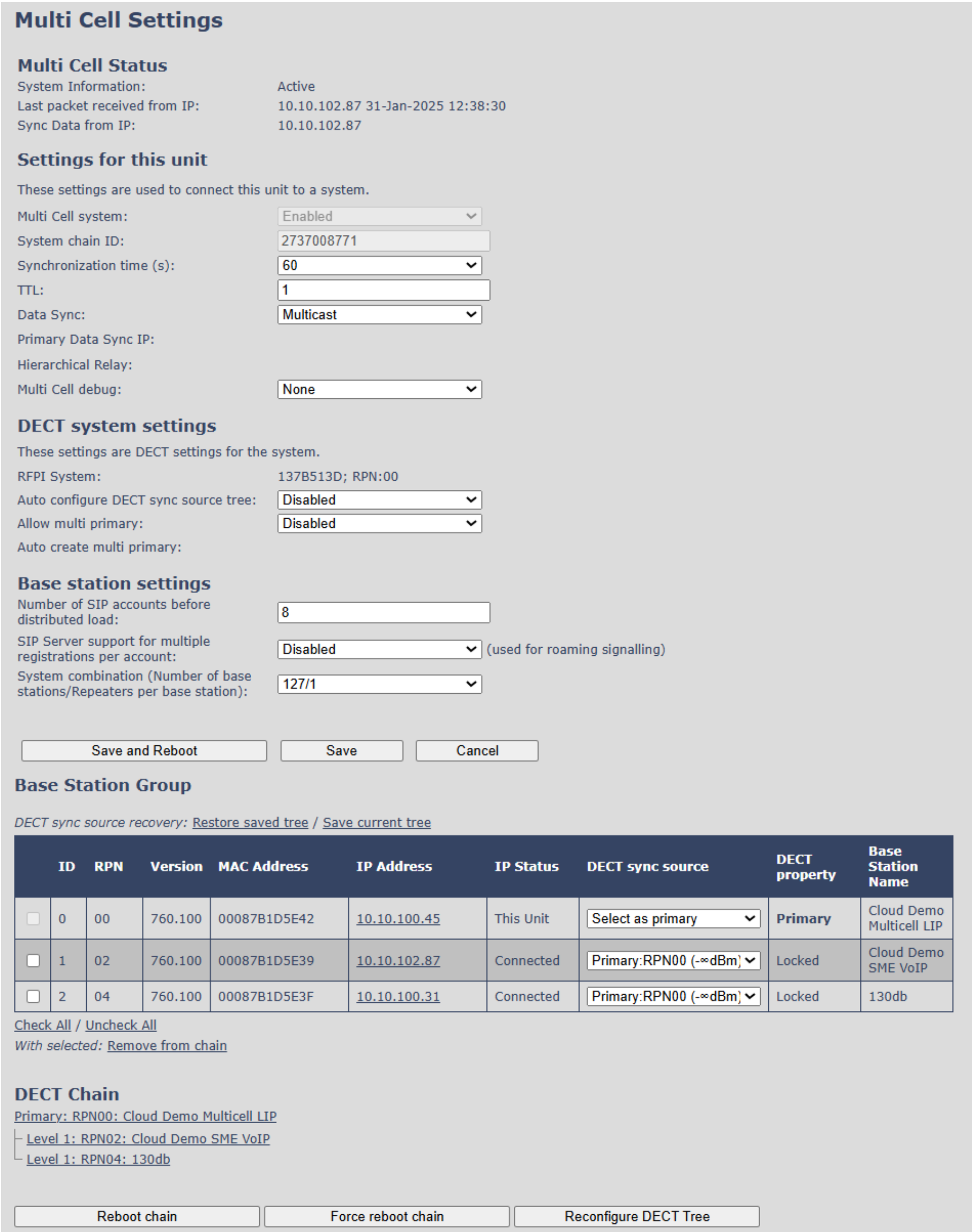


Figure 38 Multi / Dual cell settings page

The dual/multi cell data synchronization status is presented at the top of the page *(Fig.35)*, where "Keep-alive" means normal operation.

7.1.14.1 Settings for this unit

The following table presents the settings for the specific base station, used to connect it to a system.

PARAMETER	DEFAULT VALUES	DESCRIPTION
DUAL / MULTI CELL SYSTEM	Disabled	Enable this option to allow the base unit to be set in dual/multi cell mode **Valid Inputs:** Enable, Disable Must press **Save and Reboot** after changing from disabled to enabled. ***NOTE:*** *This feature is enabled by default for RTX9431*
SYSTEM CHAIN ID Dual cell Multi cell	2370683830 512	This is an identifier (in string format e.g. 2275) that is **unique** for a specific dual/multi cell system. It is a DECT cell identity, allocated to bridge several bases together in a chain. The Chain ID is used as a SIP account for check Sync. Therefore, the value MUST NOT be equal to a used SIP account. **Valid Input:** The Web site allows max 5 digits in this field. ***NOTE:*** *In a dual cell system (RTX9431), the Chain ID cannot be set or modified by the user. As soon as two bases are connected, the system automatically changes the default chain ID to a unique identifier.* ***NOTE:*** *In a multi cell system, the Chain ID is set by the user. The default value is 512, which means that extension 512 must not be used – unless the chain ID is modified. When there is a multi cell system up and running, the Chain ID can be modified by provisioning only.* ***NOTE:*** *There can be several multi cell systems with the same Chain ID in a SME network. Up to 24 levels of base station chains are permitted in a setup.*
SYNCHRONIZATION TIME (S)	60 sec	This specifies the period in seconds for a base station to synchronize to the primary base. If no keep-alive packets are received within a period of 2*Synchronization Time(s), the base will be indicated as lost in the multi cell configuration. The parameter is also used with “Auto create multi primary” feature from the following configuration settings ***NOTE:*** *This feature is not supported by RTX9431*
DATA SYNC:	Multicast	Select between “Multicast” or “Peer to Peer” data synchronization mode. The multicast feature uses UDP, along with the port range: 49200 – 49999 and IP range: 224.1.0.0 – 225.1.0.0. The used port range and IP addresses are calculated from the Chain ID. ***NOTE:*** *For multicast operation make sure that Multicast/IGMP is enabled on your switch(es), else use Peer-to-peer mode.*
PRIMARY DATA SYNC IP	Empty	IP of the base station data sync source – the base handling the data synchronization. When using multicast, this base IP is selected automatically. The data sync feature uses the port range: 49200 – 49999 When using Peer-to-Peer mode, the IP of the base used for data sync source MUST be defined.

		NOTE: *Using Peer-to-Peer mode with version below V306 limits the system's automatic recovery feature, meaning that there is* ***no*** *automatic recovery of the data sync. source in Peer-to Peer-mode.*
HIERARCHICAL RELAY	empty	This feature is available when **Peer-to-peer** data sync is chosen. The feature allows the user to distribute the data sync load between multicell base stations. **Recommended usage** based on the number of base stations in a system: • If the system has **less than 100** base stations, relay **should be set to 0** • If the system has **100 or more** base stations, relay **should be set to 3** ***NOTE: Currently other values than 0 and 3 SHOULD NOT BE USED!***
DUAL / MULTI CELL DEBUG	None	Enable this feature, if you want the system to catalogue low level dual/multi cell debug information or traces. Available Options: **Data Sync:** Writes header information for all packets, received and sent, to be used to debug any special issues. Generates a lot of syslog signaling and is only recommended to enable shortly when debugging. **Auto Tree:** Writes states and data related to the Auto Tree Configuration feature. **Both:** Both Data Sync and Auto Tree are enabled. ***NOTE:*** *Must only be used for debug purpose and not enabled on a normal running system*

7.1.14.2 DECT system settings

The following table presents the DECT settings for the whole system.

PARAMETER	DEFAULT VALUES	DESCRIPTION
DECT SYSTEM RFPI	Values may vary	This is a radio network identity accessed by all base units in a specific multi cell system. It is composed of five octets and has five different variables combined. **RFPI Format:** XX XX XX XX XX (where XX are HEX values)
AUTO CONFIGURE DECT SYNC SOURCE TREE	Enabled	Enable this to allow the system to automatically synchronize the dual/multi cell chain/tree. ***NOTE:*** *Must be enabled in order to allow a new primary to recover in case the original primary goes into faulty mode.*
ALLOW MULTI PRIMARY	Disabled	This feature is used for multi-location setups. Allows two or more primary bases in the same system. The two cells will be unsynchronized, and handover will not be possible. ***NOTE:*** *To run a system with two separate primaries in two locations "Allow multi primary" and "Auto configure DECT sync source tree" must be enabled. To add the second primary, the slave must manually be configured as primary. Alternatively, the "Auto create multi primary" must be enabled as well.* ***NOTE:*** *Not supported on Dual cell.*
AUTO CREATE MULTI PRIMARY	Disabled	This feature is used for multi-location setups. It automatically creates primary bases if they cannot connect to the main chain. For instance,

		if you have a site with an established multi cell system and would like to extend the system to another location, the added base will be automatically created as primary, if it cannot reach the chain from the first location. This means that both sites would be on the same system, but have different primaries, due to the range between them.

7.1.14.3 Base station settings

The following table lists the rest of the base station settings for the dual/multi cell system.

PARAMETER	DEFAULT VALUES	DESCRIPTION
NUMBER OF SIP ACCOUNTS BEFORE DISTRIBUTED LOAD	10	The maximum number of handsets or SIP end nodes, that are permitted to perform location registration on a specific base unit, before load is distributed to other base units. The parameter can be used to optimize the handset distribution among visible base stations. **Permitted Input:** Positive Integers (e.g. 6) ***NOTE:*** *A maximum of 10 simultaneous calls can be routed through each base unit in a multi cell setup.* ***NOTE:*** *This feature is not supported by RTX9431*
SIP SERVER SUPPORT FOR MULTIPLE REGISTRATIONS PER ACCOUNT	Disabled	Disable this option to make it possible to use the same extension (i.e. SIP Account) on multiple phones (SIP end nodes). These phones will ring simultaneously for all incoming calls. Available options: **Disabled:** No SIP de-registration will be made when a handset roams to another base station **Enabled:** The old SIP registration will be deleted with a SIP de-registration, when a handset roams to another base station
SYSTEM COMBINATION (NUMBER OF BASE STATIONS/REPEATERS PER BASE STATION):	50/3	Select between basic base configurations. 50/3: 50 bases and 3 repeaters 127/1: 127 bases and 1 repeater 254/0: 254 bases and 0 repeater ***NOTE:*** *If the system combination is set to 127/1 or 254/0, you can still register more than one repeater, but it will not get a DECT Sync source and will have no function.* ***NOTE:*** *This feature is not supported by RTX9431*

7.1.14.4 Base station group

The following table aims to introduce you to the “Base station group” table indications. The group table lists the base stations that are in a dual/multi cell system and provides information for each one of them.

PARAMETERS	DESCRIPTION
ID	Base unit identity in the chained network. **Permitted Output:** Positive integers
RPN	The Radio Fixed Part Number is an 8-bit DECT cell identity allocated by the installer. The allocated RPN within the SME must be geographically unique.

	Permitted Output: 0 to 255 (DEC) **OR** 0x00 to 0xFF (HEX)
VERSION	Base station's current firmware version. **Permitted Output:** positive Integers with dot (e.g. 480.1)
MAC ADDRESS	Contains the hardware Ethernet MAC address of the base station. It varies from base station to base station.
IP ADDRESS	Base station's current IP address
IP STATUS	Current base station behavior in the SME network. Possible Outputs **Connected:** The relevant base station(s) is online and connected to the network **Connection Loss:** Base station unexpectedly lost connection to the network **This Unit:** Indicates the current base station which http Web Interface is currently accessed
DECT SYNC SOURCE	With setting "Auto configure DECT sync source tree" set to "Enable", this tree will automatically be generated. If manually configured, the administrator should choose the relevant "multi cell chain" level he wants and organize the base units. Maximum number of "multi cell chain" levels is 24. **Format of the selection:** "AAAAAxx: RPNyy (-zz dBm)" AAAAA: indication of sync. source for the base. Can be "Primary" or "Level xx" xx: Sync. source base sync. level yy: Sync. source base RPN zz: RSSI level of sync. source base seen from the actual base "(Any) RPN": When a base is not synchronized to another base. State after reboot of chain.
DECT PROPERTY	Base station characteristics in connection to the current dual/multi cell network. Possible Output(s): **Primary:** Main base station to which all other nodes in the chain synchronize to. **Locked:** The base unit is currently synchronized and locked to the master base unit. **Searching:** Base unit in the process of locating to a Master/Slave as specified in DECT sync source **Free Running:** A locked base unit that suddenly lost synchronization to the Master. **Unknown:** No current connection information from specific base unit **Assisted lock:** Base has lost DECT sync. source and Ethernet is used for synchronization **Sync. Lost:** Handset has an active DECT connection with the base, but the base has lost DECT sync. source connection. The base will continue working as long as the call is active and will go into searching mode when call is stopped.
BASE STATION NAME	Name from management settings.

7.1.14.5 DECT chain

Below the "Base station Group" table is the DECT Chain tree. The DECT Chain tree is a graphical presentation of the "Base station Group" table levels and connections. If any repeaters are present, they are highlighted in green and if some units are part of the base group but not present in the tree, they are highlighted in red *(Fig.36)*.

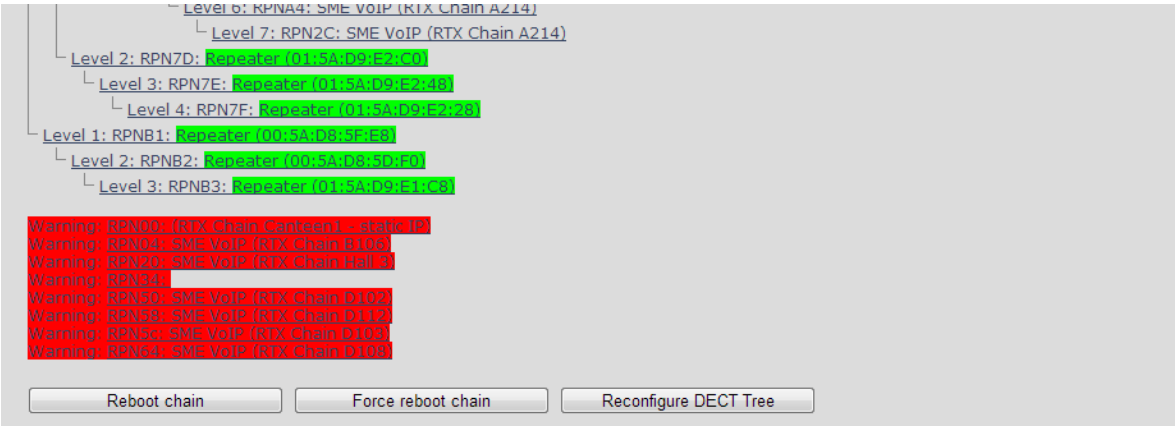


Figure 39 DECT chain visualization

7.1.15 Multi Zone

The RTX8663 base station supports a Multi Zone feature, which allows different multi cell systems to be interconnected via zones. This means that each zone is a separate multi cell system. Such setup allows the handsets to be used across all zones but configured only in one of them. The following sub-chapter presents the available settings on the page *(Fig.37)* These are divided into several sections, which will be described as follows – *Settings for this unit, DECT system settings and Multi zone System.* The settings can be configured both via the webpage and via provisioning. For a more detailed guide on configuring and using the feature, please refer to the ***RTX VoIP DECT System Traffic Planning and Deployment.pdf***.

NOTE: *This feature is not supported by RTX9431*

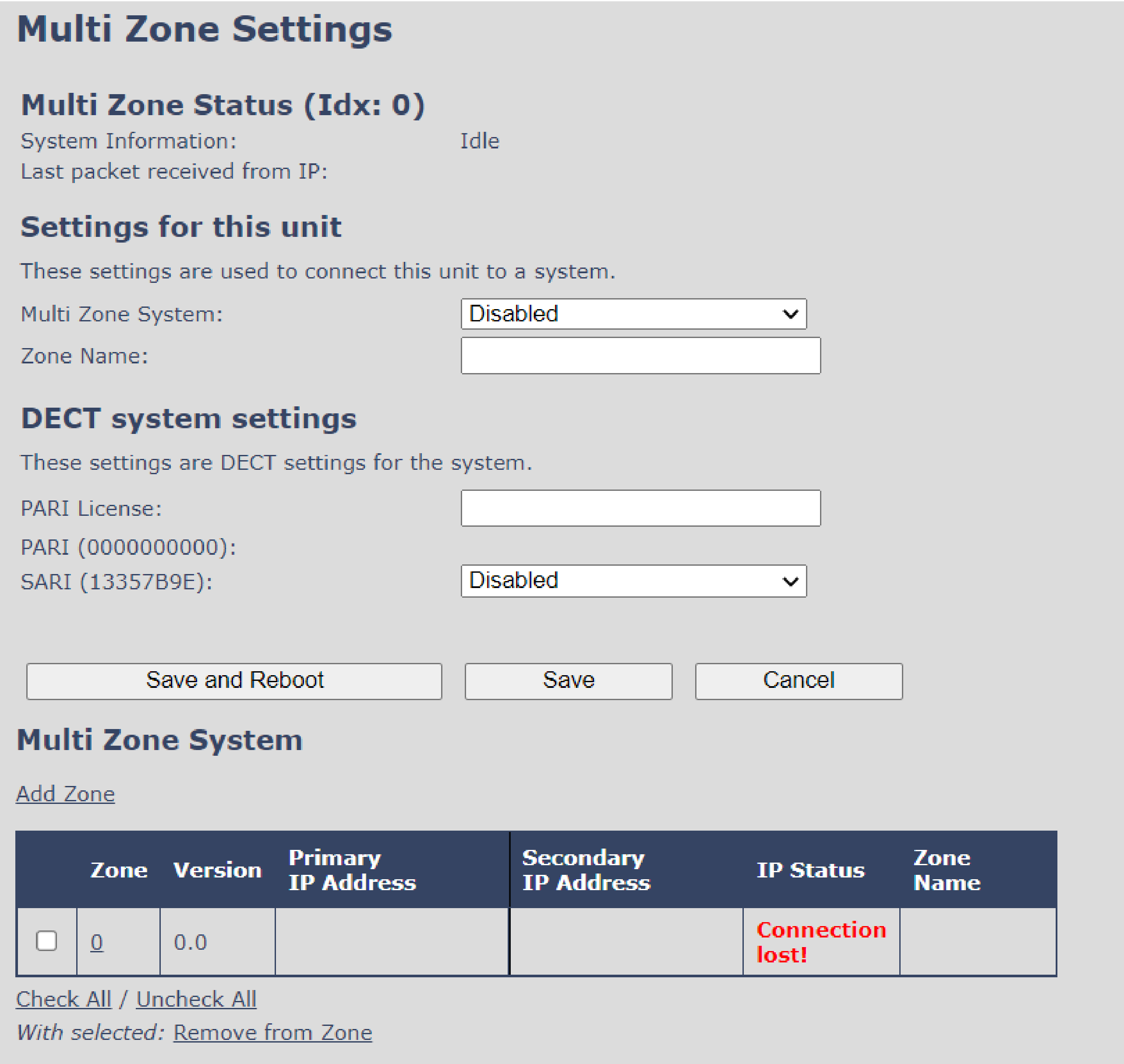


Figure 40 Multi Zone settings page

7.1.15.1 Settings for this unit

PARAMETER	DEFAULT VALUES	DESCRIPTION
MULTI ZONE SYSTEM	Disabled	An On/Off type of setting, which will enable/disable the Multi Zone feature
ZONE NAME	Empty	Type in the name of the Zone ***NOTE:*** *The setup should start with Zone 0*

7.1.15.2 DECT system settings

PARAMETER	DEFAULT VALUES	DESCRIPTION
PARI LICENSE	0000000000 (Empty)	The PARI value should be manually typed ONLY for Zone 0

		NOTE: *The PARI value should be purchased from RTX*
SARI	Disabled	The SARI value is automatically configured

7.1.15.3 Multi Zone System

The following list aims to introduce you to the Multi Zone System table indications.

PARAMETER	DESCRIPTION
ADD ZONE	Provides access to the editor menu, where you can add a new zone
ZONE	The zone identity in the system **Permitted input:** Positive integers
VERSION	The zone's current firmware version
PRIMARY IP ADDRESS	The IP address of the Primary base of the zone
SECONDARY IP ADDRESS	The IP address of the Secondary base (Primary Backup) of the zone
IP STATUS	The zone's current behavior on the system
ZONE NAME	The name of the zone
REMOVE FROM ZONE	Remove the selected zones from the multi zone system

7.1.15.4 Add zone editor menu

By clicking "Add zone", a separate editor page opens *(Fig.38)*, which parameters are further explained in the following table.

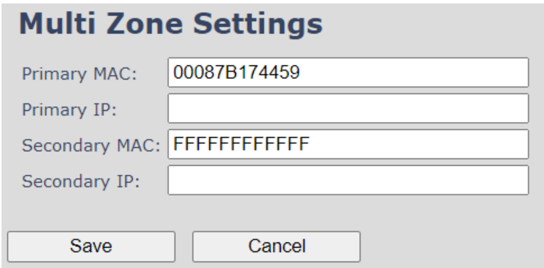


Figure 41 Add zone editor page

PARAMETER	DEFAULT VALUES	DESCRIPTION
PRIMARY MAC	Varies from device to device	The MAC address of the primary base station
PRIMARY IP	Empty	The IP address of the primary base station
SECONDARY MAC	FFFFFFFFFFFF	The MAC address of a secondary (backup) base station
SECONDARY IP	Empty	The IP address of the secondary (backup)base station

7.1.16 LAN Sync

The RTX8663 base station supports the LAN sync feature, which allows data synchronization over the network, instead of Over-the-air. This feature, specified by the IEEE1588 standard, has the advantage of providing a larger coverage installation option for places where the bases cannot "see" each other.

The following sub-chapter presents the available settings on the LAN sync page *(Fig.39).* These are divided into several sections, which will be described as follows – *IEEE1588 feature*, *Zone LAN Sync setup*, *External*

LAN Sync setup, *Base Station Group* and *This Unit Debug*. To secure a good user experience, settings are kept to a minimum and are all available to be configured both via the webpage and via provisioning. Before configuring the LAN sync, the administrator must be aware of the network requirements in order to minimize the impact from other devices on the network. Therefore, please refer to the ***RTX VoIP DECT System Traffic Planning and Deployment .pdf*** for more details on the LAN sync configuration and usability.

NOTE: *This feature is not present in RTX9431*

Figure 42 LAN Sync page

7.1.16.1 IEEE1588 feature

The initial setting of the page provides the option to enable / disable the LAN Sync feature for the device.

PARAMETERS	DEFAULT VALUES	DESCRIPTION
IEEE1588	Disabled	An On/Off type of setting, which will enable/disable the LAN synchronization option

7.1.16.2 Zone LAN sync setup

These settings cover the global configuration of the synchronization in a zone.

PARAMETERS	DEFAULT VALUES	DESCRIPTION
MULTICAST IP ADDRESS	224.0.1.129	This setting defines the IP address where to listen for IEEE1588 PTP packages (IP address of the multicast group). The IP must be the same for all devices in the specific zone. ***NOTE:*** *This should only be changed in case other IEEE1588 equipment is on the network using this specific IP address.* ***NOTE:*** *The IP address must start with 224.0.xx.xx and this cannot be changed.* ***NOTE:*** *Before setup, make sure no other devices are using the given IP. This means that each zone should have a different IP!*
MULTICAST PORT	319	Define the port which the system will communicate on (where to listen for the IEEE1588 PTP messages). The port must be the same for all devices in the specific zone. ***NOTE:*** *This should only be changed in case other IEEE1588 equipment on the network is using this specific port.* ***NOTE:*** *Each zone must have a different port*
DOMAIN NUMBER	0	This setting selects the domain number, which is the preferred method to divide the IEEE1588 PTP messages into zones in IEEE 1588-2008. **Valid input:** 0-127
ALTERNATIVE DOMAIN NUMBER	64	Alternative domain is only used in case the primary sync source from the main domain fails. If so, the base station will sync to the alternative domain. ***NOTE:*** *It must NOT have the same value as the domain number.* **Valid input:** 0-127
MULTI CELL DEBUG MODE	None	This feature should be enabled if one wants the system to catalogue low level multi cell debug information or traces. ***NOTE:*** *Must only be used for debug purpose and not enabled on a normal running system* Available Options: **Data Sync:** Writes header information for all sent and received packets to be used to debug specific issues. Generates a lot of Syslog signaling and is only recommended to enable shortly when debugging. **Auto Tree:** Writes states and data related to the Auto Tree Configuration feature. **Both:** Both Data Sync and Auto Tree are enabled. **IEEE1588 Debug:** Writes IEEE1588 debug information to the syslog

7.1.16.3 External LAN sync setup

The "External LAN Sync setup" covers the configuration of an external (multi-level) synchronization. This means that, in order to support more than 250 Base stations in a system, it is necessary to use multi-level synchronization. This feature is used to define the Primary zone to which all the other zones will sync.

PARAMETERS	DEFAULT VALUES	DESCRIPTION
EXTERNAL SYNC SOURCE	Disabled	Enable the feature by choosing one of the options below: **Primary zone configuration** **Secondary zone configuration** In a default setup, Zone 0 is set as Primary zone and the rest of the zones are configured as Secondary zone.
MULTICAST IP ADDRESS	224.0.1.129	To listen for IEEE1588 PTP packages, the IP address should be defined ***NOTE:*** *This should only be changed in case other IEEE1588 equipment is on the network using this specific IP address.* ***NOTE:*** *The IP address must start with 224.0.xx.xx and this cannot be changed.* ***NOTE:*** *Before setup, make sure no other devices are using the given IP.*
MULTICAST PORT	319	Define the port on which the system will listen to IEEE1588 PTP messages ***NOTE:*** *This should only be changed in case other IEEE1588 equipment is on the network using this specific port.*
DOMAIN NUMBER	1	This setting selects the domain number, which is the preferred method to divide the IEEE1588 PTP messages into zones in IEEE 1588-2008. **Valid input:** 0-127 ***NOTE:*** *The input must NOT be the same as the one used in the previous feature "Zone LAN Sync setup"*
ALTERNATIVE DOMAIN NUMBER	65	Alternative domain is only used in case the primary sync source from the main domain fails. If so, the base station will sync to the alternative domain. ***NOTE:*** *The input must NOT have the same value as the "Domain number" from the previous parameter and must NOT be used in the "Zone LAN Sync setup" feature.*

7.1.16.4 Base station group

The Base station group lists various parameter settings for the base stations and allows the administrator to check the status information for the whole system.

PARAMETERS	DESCRIPTION
ID	Base unit identity in the chained network.

	Permitted Output: Positive Integers
STATUS	Base station characteristics in connection to the current multi cell network. Possible Output(s): **Connecting:** The base unit is in the process of locating a Master/Slave as specified in "Sync source" **Locked:** The base unit is currently synchronized and locked to the master base unit **Free Running:** IEEE master is found and is DECT synchronizing
PREFERED ROLE	**Disabled:** Disable the feature **Primary:** The base station that is used for main sync; only one primary is allowed to the system ***NOTE:** It is recommended to use base stations that are closer to the backbone as primary* **Secondary:** Base stations that will never be selected as primary. They become slaves **Automatic:** System finds primary sync source – it allows the system to decide the role of the base **Alt. Primary:** Backup for primary base station in case it fails; only one redundant sync. master is allowed in the system
CURRENT ROLE	The current role of the base station Possible Output(s): **Disabled:** Dispalyed when LAN Sync is disabled for the device **Primary:** Main base station which all other nodes in the chain synchronize to **Alternate:** Fallback base station for redundancy, which will be used in case the primary base is offline **Secondary:** Secondary base stations which are synchronized to the primary or alternate base **Initializing:** Initial state
SYNC SOURCE	Shows to which base station this specific device is synchronized and indicates if it is via **LAN** or **DECT** Indications: **ID:** the device with ID from the current list **RPN:** the DECT ID used from the Over-the-air symbolization.
ALT. SYNC SOURCE	Alternative sync source in case main sync source fails
NETWORK JITTER [NS]	Measures how the IEEE1588 packets are received, the lower the Jitter is the better **Min:** Displays the minimum jitter average between primary and slave

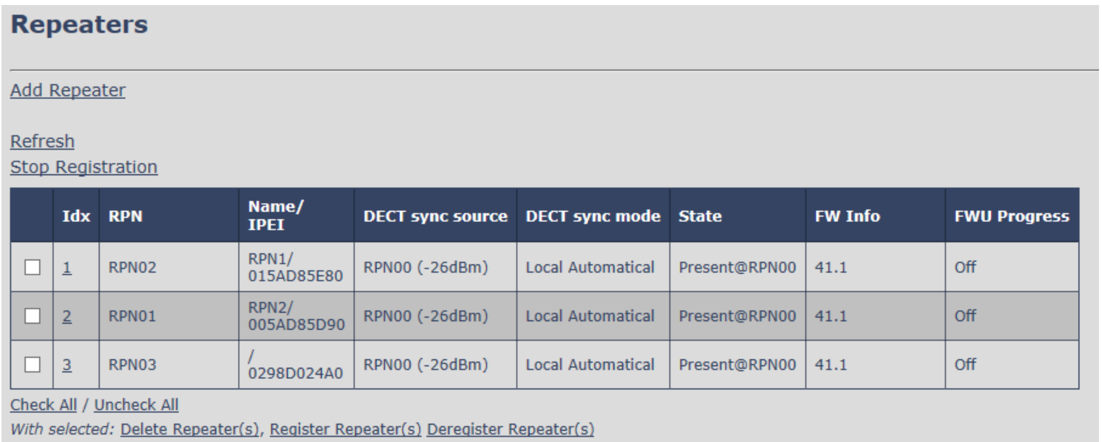
(MIN/AVG/MAX)	**Avg:** Displays the average jitter between primary and slave **Max:** Displays the maximum jitter average between primary and slave
NETWORK DELAY [NS] (MIN/AVG/MAX)	Measures the time it takes an IEEE packet to travel from Primary to Slave base station in ns. **Min:** Displays the minimum average delay between primary and slave **Avg:** Displays the average delay between primary and slave **Max:** Displays the maximum average delay between primary and slave
IP STATUS	Current base station behavior in the SME network. Possible Output(s): **Connected:** The relevant base station(s) is online and connected to the network **Connection Loss:** Base station unexpectedly lost connection to the network **This Unit:** Current base station which http Web Interface is currently being accessed
BASE STATION NAME	Name from the *Management* page

7.1.16.5 This unit debug

This section displays debug information, which is used only by RTX to debug LAN Sync network issues. In case debug is needed, send this information to RTX support team. The debug data is not synced between the bases.

7.1.17 Repeaters

This section presents and describes the different parameters available in the *Repeaters* configuration page *(Fig.40)*.


Repeaters

Add Repeater

Refresh
Stop Registration

	Idx	RPN	Name/ IPEI	DECT sync source	DECT sync mode	State	FW Info	FWU Progress
☐	1	RPN02	RPN1/ 015AD85E80	RPN00 (-26dBm)	Local Automatical	Present@RPN00	41.1	Off
☐	2	RPN01	RPN2/ 005AD85D90	RPN00 (-26dBm)	Local Automatical	Present@RPN00	41.1	Off
☐	3	RPN03	/ 0298D024A0	RPN00 (-26dBm)	Local Automatical	Present@RPN00	41.1	Off

Check All / Uncheck All
With selected: Delete Repeater(s), Register Repeater(s) Deregister Repeater(s)

Figure 43 Repeaters page

Depending on the base station type and system, the number of supported repeaters can vary (see ***2.3.1 Base stations***). For a multi cell system with RTX8663 base stations, the number of repeaters, allowed on each base station, is defined on the *Multi cell* page ***(7.1.14 Dual/Multi cell).***

PARAMETERS	DESCRIPTION
ADD REPEATER	Provides access to the "Add Repeater" editor menu, where new devices can be added
REFRESH	Refreshes the page
STOP REGISTRATION	Manually stops DECT registration mode of the system. This prevents repeaters to register to the system
DELETE REPEATER(S)	Deregisters and deletes the device from the VoIP base station
REGISTER REPEATER(S)	Enable registration mode for the selected repeater entries.
DEREGISTER REPEATER(S)	Deregister the selected repeater(s), but keep the entry
IDX	Repeater unit identity in the chained network. **Permitted Output:** Positive Integers ***NOTE:*** *If selected, the editor's menu will open*
RPN	The Radio Fixed Part Number (RPN) is an 8-bit DECT cell identity allocated by the installer. The allocated RPN within the SME must be geographically unique. **Permitted Output:** 0 to 255 (DEC) **OR** 0x00 to 0xFF (HEX) Allocation of the repeaters to the bases by RPN is done the following way: The base stations have an RPN identity which is incremented by 2^2 in hex, meaning that the first base is RPN00, second base is RPN04, etc. The repeaters' RPN identification is addressed in between the RPN of the bases. For example: Base stations are named RPN00 – RPN04 – RPN08, etc. - jumping 4 numbers each time (HEX numbers) Repeaters, which connect to base station RPN00, will be named after RPN01 – RPN02 – RPN03 (HEX numbers) Repeaters, which connect to base station RPN04 will be called RPN05 – RPN06 – RPN07 (HEX numbers). Etc. An exception of this rule is when using a multi cell (RTX8663) with a system combination 127/1. In this case, since there is only 1 repeater allowed per base, the RPN would increase only by 2. For example: Base stations are named RPN00 – RPN02 – RPN04, etc. - jumping 2 numbers each time (HEX numbers) Repeater, which connects to base station RPN00 will be named after RPN01 (HEX numbers) Repeater, which connects to base station RPN02 will be named after RPN03 (HEX numbers) Etc.
NAME/IPEI	Contains the name and the unique DECT serial number of the repeater. If no name is given, the field will be empty.
DECT SYNC SOURCE	The chain connection to the specific base/repeater unit. Maximum number of chain levels is 12. **Sync. source format:** "RPN**yy** (-**zz** dBm)" **yy**: RPN of source **zz:** RSSI level seen from the actual repeater
DECT SYNC MODE	Synchronization type:

	Manually: User-controlled by manually assigning the repeater's "RPN" and "DECT sync source" RPN **Local Automatical:** Auto -controlled by detecting the base station with the best signal and auto assigning an RPN.
STATE	**Enabled:** The slot is open for repeater registration on this index **Present@unit:** The repeater is DECT located at the VoIP base with RPNyy **Removed:** The repeater has been out of sight for a specified amount of time (~one hour).
FW INFO	Firmware version of the repeater
FWU PROGRESS	Possible FWU progress states: **Off:** FWU version is set to 0 - FWU is off **Initializing:** FWU is starting and progress is 0%. **X% :** Ongoing FWU progress **Verifying X%:** FWU writing is done and verifying before the swap "**Conn. term. wait**" (Repeater): All FWU is complete and is waiting for connections to stop before the repeater restarts. **Complete:** FWU complete **Error:** Not able to FWU upgrade. For example, due to file not found, not valid, etc.

NOTE: *If there are unsupported repeaters registered to the base station, the system shows the following warnings: "**System is configured with unsupported repeaters.**" and "Reboot is needed for changes to take effect".*

7.1.17.1 Add repeater editor menu

By selecting the *"Add repeater"* link, a separate editor page opens which parameters are further explained in the following table.

PARAMETERS	DESCRIPTION
NAME	Repeater's name. If no name is specified, the field will be empty
DECT SYNC MODE	**Manually:** User-controlled by manually assigning the repeater's "RPN" and "DECT sync source" RPN **Local Automatic:** Auto -controlled by detecting the base station with the best signal and auto assigning an RPN. ***NOTE:*** *With "Local automatic" mode repeater on repeater (chain) is not supported.*
RPN	The identity of the repeater ***NOTE:*** *This option is visible only if "DECT sync mode" is set to "Manually"*
DECT SYNC SOURCE	The device (base/other repeater) to which the repeater will connect to ***NOTE:*** *This option is visible only if "DECT sync mode" is set to "Manually"*

NOTE: *A good practice when adding repeaters to a Dual Cell system (RTX9431) is to use a manual registration, because then you can control to which base station the repeater(s) connects to. In this way, the repeater will also have a static RPN whereas if the repeater is set to automatic sync mode, the RPN may vary from one registration to another. This is dependent on the repeater which will register to the base first, after for example a base reboot. If at least 1 repeater is set to manual sync mode, it is strongly recommended to have* *all repeaters in the setup configured the same way, in order to make sure that the repeaters connect to the desired RPN.*

7.1.18 Alarm

The Alarm page allows the administrator to enable different types of alarms for the handsets *(Fig.41).* Up to eight alarm profiles can be configured on this page, which later can be assigned to the handsets. The following section aims to describe the available settings. For further details about the alarms and use case examples, please refer to Appendix ***9.4 Alarm characteristics***. For an alarm configuration guide, please refer to ***8.5 Alarms.***

Alarm

Idx	Profile Alias	Alarm Type	Alarm Signal	Stop Alarm from Handset	Trigger Delay	Stop Pre-Alarm from Handset	Pre-Alarm Delay	Howling	Alarm Priority
0		Disabled	Call	Enabled	0	Enabled	0	Disabled	0
1		Disabled	Call	Enabled	0	Enabled	0	Disabled	0
2		Disabled	Call	Enabled	0	Enabled	0	Disabled	0
3		Disabled	Call	Enabled	0	Enabled	0	Disabled	0
4		Disabled	Call	Enabled	0	Enabled	0	Disabled	0
5		Disabled	Call	Enabled	0	Enabled	0	Disabled	0
6		Disabled	Call	Enabled	0	Enabled	0	Disabled	0
7		Disabled	Call	Enabled	0	Enabled	0	Disabled	0

Save Cancel

Figure 44 Alarm page

PARAMETERS	DESCRIPTION
IDX	Indicates the index number of a specific alarm.
PROFILE ALIAS	A user-friendly name to help identify the different alarm profiles
ALARM TYPE	The alarms are dependent on the type of handsets being used. **RTX8631/RTX8632/RTX8633:** Alarm button **RTX8830:** Alarm button Man Down No Movement Running Pull Cord ***NOTE:*** *RTX8430, RTX8431 and RTX8436 do not support the alarm feature* ***NOTE:*** *RTX8630 cannot send alarms, but it can receive them*
ALARM SIGNAL	The way the alarm is signaled when received on the handset. **Message:** A text message via an alarm server. ***NOTE:*** *If a handset supports Bluetooth and Receiver mode is enabled, the beacon data will be included in the message. For more details, please go to* ***7.1.4 Handsets.*** **Call:** An outgoing call to the specified emergency number (see ***7.1.4 Handsets***).
STOP ALARM FROM HANDSET	Enable/Disable the possibility to stop/cancel the alarm from the handset.

TRIGGER DELAY	The possibility to delay an alarm in a specific number of seconds. This is the period from when the alarm has been fired until the handset displays a pre-alarm warning. If this setting is set to 0, there will be no pre-alarm warning, and the alarm will be signaled immediately. The alarm algorithm typically needs about 6 sec. to detect, for example, Man down, etc.
STOP PRE-ALARM FROM HANDSET	Enable/Disable the possibility to stop/cancel the pre-alarm from the handset.
PRE-ALARM DELAY	The period from displaying the pre-alarm warning message until the actual alarm is signaled. The maximum value is 255.
HOWLING	Enable/Disable howling on the handset when the alarm is signaled. If disabled, only the configured signal is sent (call or message).
ALARM PRIORITY	Define the priority of the alarm

7.1.18.1 Use of Emergency alarms

As described above, it can be configured if it should be possible to stop an alarm from the handset. If the possibility to stop an alarm from the handset is disabled, it is ensured that an alarm is not stopped before someone at e.g., an emergency center, has received the alarm and reacted upon it.

The behavior of a handset, when an alarm "is sent", depends on the configured Alarm Signal:

- **Call:** When the Alarm Signal is configured as "Call", the handset will make a call to the specified emergency number, and the alarm will terminate when the call is ended. If it is not allowed to stop the alarm from the handset, it will not be possible to terminate the call from the handset, and the alarm will be considered as stopped only when the remote end (e.g. the emergency center) terminates the call.
- **Message:** When the Alarm Signal is configured as "Message", the handset will send an alarm message to the specified alarm server and enable auto answer mode. This means that if the receiver responds to the alarm message by calling the alarm initiator, the handset will immediately answer the call and allow the person in danger to talk to the other party.
  If Howling is enabled on the sender part, the handset will also start the Howling tone. An alarm is considered as stopped, depending on its configuration (for ex. after confirming that the alarm has been received)

All type of alarms have the same priority. This means that once an alarm is active, it cannot be overruled by another alarm until the first one has been stopped. However, if the alarm is not yet active, i.e. if it is in "pre-alarm" state and an alarm configured with no pre-alarm is fired, then the new alarm will become active and stop the pending alarm.

Alarms with no pre-alarm are considered important, and there is no possibility to cancel them before they are sent. Therefore, alarms with no pre-alarm are given higher priority than alarms in pre-alarm state.

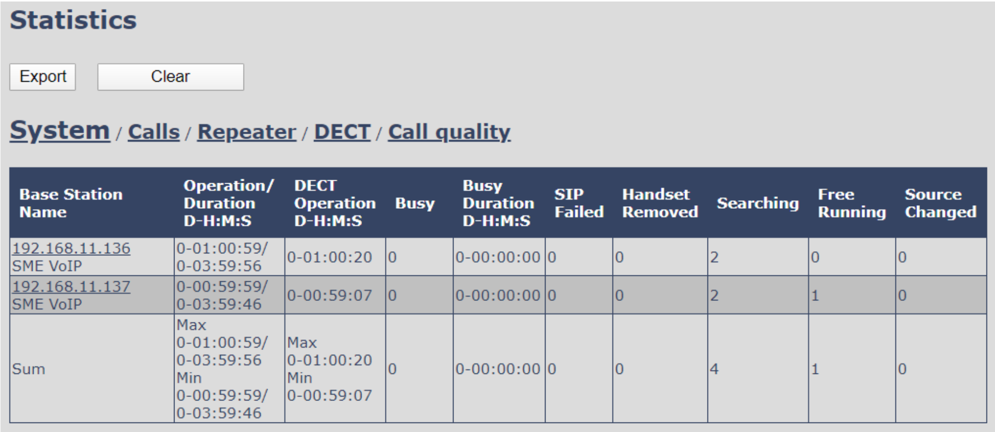
7.1.19 Statistics

This menu provides different kind of system information *(Fig.42).* It includes the following five administrative pages, which are described accordingly.

1. System
2. Calls
3. Repeater

4. DECT data
5. Call quality *(**NOTE:** RTX9431 does not support this feature)*

All five views have an embedded export function, which allows the administrator to export all data to common separated file. By pressing the clear button, all data in the system is cleared.



Statistics

Export Clear

System / Calls / Repeater / DECT / Call quality

Base Station Name	Operation/ Duration D-H:M:S	DECT Operation D-H:M:S	Busy	Busy Duration D-H:M:S	SIP Failed	Handset Removed	Searching	Free Running	Source Changed
192.168.11.136 SME VoIP	0-01:00:59/ 0-03:59:56	0-01:00:20	0	0-00:00:00	0	0	2	0	0
192.168.11.137 SME VoIP	0-00:59:59/ 0-03:59:46	0-00:59:07	0	0-00:00:00	0	0	2	1	0
Sum	Max 0-01:00:59/ 0-03:59:56 Min 0-00:59:59/ 0-03:59:46	Max 0-01:00:20 Min 0-00:59:07	0	0-00:00:00	0	0	4	1	0

Figure 45 Statistics page

7.1.19.1 System

The table below describes the available parameters on this page.

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base name (name can be edited in the *Management* page)
OPERATION/DURATION D-H:M:S	**Operation:** Operation time for the base since last reboot. **Duration:** Operation time for the base since the last reset of statistics, or firmware upgrade.
DECT OPERATION D-H:M:S	Displays information about Days, Hours, Minutes and Seconds that the base station has been running
BUSY	Displays the number of times the base has been busy.
BUSY DURATION D-H:M:S	Displays the total time a base has been busy for speech (8 or more calls active).
SIP FAILED	Displays the number of times a SIP registration has failed
HANDSET REMOVED	Displays the number of times a handset has been marked as removed
SEARCHING	Displays the number of times a base has been searching for its sync source ***NOTE:*** *This is not supported by RTX9431*
FREE RUNNING	Displays the number of times a base has been free running (explained below) ***NOTE:*** *This is not supported by RTX9431*
SOURCE CHANGED	Displays the number of times a base has changed sync source ***NOTE:*** *This is not supported by RTX9431*
IEEE1588 SYNC LOST	Counts how many times a base has lost sync with both "Sync Source" and "Alternative Sync Source" (the primary and alternative primary base) ***NOTE:*** *This parameter is visible only if IEEE1588 is enabled*
IEEE1588 PRIMARY LOST	Counts how many times a base has lost sync with the "Sync Source" (the primary base). ***NOTE:*** *This parameter is visible only if IEEE1588 is enabled*

7.1.19.1.1 Free running

The state *Free running* is NOT an error state, but a simple trigger state, indicating that some changes must be made to ensure continuous DECT synchronization.

The state *Free running* informs the application that the base has not received any synchronization data from its synchronization source in the last 10 seconds. There could be several reasons for this:

1. The two bases are using the same DECT slots and therefore cannot see each other.
2. Many simultaneous voice or data calls.
3. A sudden change of the environment (e.g., closing a fire door)
4. Distortion of DECT frequency (around 1.8MHz) either by other DECT systems, or other equipment.

When the *Free running* state is triggered, several recovery mechanisms can be activated:

1. Move the DECT slot to avoid using the same base state as its synchronization source
2. Use information from all other base stations to monitor how they are seeing the current device in the DECT air. This can be marked by changing to state *Assisted lock*

The state *Assisted lock* can be stabile for a long time and usually can change to state *Locked* again. The state *Free Running* can also change back to state *Locked* again.

If the base is in state *Free running* and the synchronization source base is not seen and no data is available for the assisted lock mechanism, the base station will change to a new state after two minutes:

1. If the base station does NOT have any active calls, the base will change to state *Searching*.
2. If the base station has an active call, this base will change to state *Sync lost*. After the call is released, the state will change to state *Searching.*

7.1.19.2 Calls

The table below describes the available parameters on this page.

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base name (name can be edited in the *Management* page)
OPERATION TIME/DURATION	**Operation:** Operation time for the base since last reboot. **Duration:** Operation time for the base since the last reset of statistics, or firmware upgrade.
COUNT	Counts the number of calls on the base.
DROPPED	Displays the number of active calls that were dropped. E.g., if a user has an active call and walks out of range, the call will be counted as a dropped call. An entry is stored in the syslog when a call is dropped.
EMERGENCY CALLS	Counts the number of emergency calls on the base
CALL DROPS DUE TO EMERGENCY CALL	Displays the number of active calls that were dropped due to an emergency call
EMERGENCY CALLS REJECTED	Displays the number of emergency calls that were rejected
NO RESPONSE	Displays the number of calls that have no response. E.g., if an external user tries to make a call to a handset that is out of range, the call is counted as no response. An entry is stored in the syslog when a call is in no response.
DURATION	Displays the total time that calls are active on the base.

ACTIVE	Displays the number of active calls on the base station (not active DECT calls, but active calls). The max number of active calls per base/system can be seen in ***2.3.1 Base stations***
MAX ACTIVE	Displays the maximum number of calls that have been active at the same time.
CODEC	Logging and count of used codec types on each call.
HANDOVER ATTEMPT SUCCESS	Counts the number of successful handovers.
HANDOVER ATTEMPT ABORTED	Counts the number of failed handovers.
AUDIO PACKET LOSS	Counts the number of times where audio connection was not established.

7.1.19.3 Repeater

The table below describes the available parameters on this page.

PARAMETERS	DESCRIPTION
IDX/NAME	Base IP address and base name (name can be edited in the *Management* page)
OPERATION D-H:M:S	Total operation time for the repeater since last reboot
BUSY	Displays the number of times the repeater has been busy.
BUSY DURATION D-H:M:S	Displays the total time a repeater has been busy for speech (5 or more calls active).
MAX ACTIVE	Displays the maximum number of calls that have been active at the same time.
SEARCHING	Displays the number of times a repeater has been searching for its sync source
RECOVERY	Displays the number of times a repeater has been in recovery mode. (In case the sync source is not present anymore, the repeater will lock on to another base or repeater, and display recovery mode)
SOURCE CHANGED	Displays the number of times a repeater has changed its sync source
WIDE BAND	Displays the number count of wideband calls on the repeater
NARROW BAND	Displays the number count of narrow band calls on the repeater

7.1.19.4 DECT

The table below describes the available parameters on this page.

PARAMETERS	DESCRIPTION
FREQUENCY	Displays the number of the DECT slot frequency
SLOTX	Displays the number of connections that have been active on each frequency

7.1.19.5 Call quality

The table below describes the available parameters on this page.

NOTE: *RTX9431 does not support this feature*

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base name (name can be edited in the *Management* page)
TYPE	**Call:** Amount of calls on the base, measured in RTP connections. **Relay conn:** Amount of calls created on one base, and then handed over to another. This results in the initial base having a relay connection.
CALL COUNT	Displays the number of calls

LOCAL/REMOTE SIDE	These two parameters indicate the call quality. **Local:** Quality of the call from this end (this base) **Remote:** Quality of the call from "the other end" – The base the other handset is connected to
JITTER[MS]	Measures how the RTP packets are received. The lower the Jitter, the better.
ROUND TRIP LATENCY [MS]	Measures the time it takes for RTP packets to reach its destination.
PACKET LOSS [%]	Displays the percentages of lost packets
R-VALUE	A way to measure call quality, from 0-120. Refer to the user satisfaction rate table below
MOS-VALUE	Measures subjective call quality for a call. MOS scores range from one, unacceptable, to 5 - excellent. Refer to the user satisfaction rate table below VOIP calls often are in the 3.5 to 4.2 range

USER SATISFACTION LEVEL	MOS	R-value
MAXIMUM USING G.711	4.4	93
VERY SATISFIED	4.3-5.0	90-100
SATISFIED	4.0-4.3	80-90
SOME USERS SATISFIED	3.6-4.0	70-80
MANY USERS DISSATISFIED	3.1-3.6	60-70
NEARLY ALL USERS DISSATISFIED	2.6-3.1	50-60
NOT RECOMMENDED	1.0-2.6	Less than 50

7.1.20 Generic statistics

This page provides further system information, distributed among five categories in the form of a table *(Fig.43)*. The categories and the related display output are described accordingly. The statistics are base-related.

The menu has an embedded "Export all fields" function, which allows the administrator to see the statistics hour by hour. By pressing the "Reset all statistics" button, all data in the system is cleared.

TABLE PARAMETER	DEFAULT VALUE	DESCRIPTION
PARAMETER	Varies	Headline of the different category statistics
VALUE	Varies	Varies from point to point
24 HR DATA	Varies	Data from the last 24 hours

Figure 46 Generic Statistics page

7.1.20.1 DECT Statistics

The table below describes the available parameters under this category

PARAMETERS	DESCRIPTION
TOTAL NUMBER OF DLC INSTANCE	The lifetime total count of instantiated DLC instances.
MAX CONCURRENT DLC INSTANCES	The lifetime highest concurrent count of instantiated DLC instances.
CURRENT NUMBER OF DLC INSTANCES	The current count of instantiate DLC instances.
TOTAL NUMBER OF TIMES IN MAX DLC INSTANCES IN USE	The number of times we reach the currently highest count of DLC instances.
TOTAL TIME SPEND IN MAX DLC INSTANCES IN USE	The time we have spent in the highest concurrent number of instantiated DLC instances.
AVERAGE FREQUENCY X USAGE THIS HOUR (MAX 100 PER SLOT)	The average use of frequency number X. The value is 100 if the frequency is fully used by a slot in the measured time frame.
AVERAGE EVEN SLOT USAGE THIS HOUR (MAX 100 PER SLOT)	The average use of even numbered slots.
AVERAGE ODD SLOT USAGE THIS HOUR (MAX 100 PER SLOT)	The average use of odd numbered slots.
PERCENTUAL TIME OF X SLOTS USED THIS HOUR	The percentual time that X number of DECT slots are used during the given hour (compared to other slot counts).
TOTAL CHO SUCCESS	The number of times connection handover is successfully made.
TOTAL NUMBER OF FORCED PP MOVES	The lifetime total count that this base forces PP moves.

7.1.20.2 DECT Synchronization statistics

The table below describes the available parameters under this category.

PARAMETERS	DESCRIPTION
CURRENT SYNCHRONISATION STATE	The current DECT sync state (e.g. Master, Searching, Free Running, etc).
CURRENT SYNCHRONISATION CHAIN	The current DECT sync source (base station ID) of this device
TIMESTAMP FOR LAST CHANGED SYNCHRONISATION CHAIN	Timestamp of the last time this base changed DECT sync source.
HOURLY NUMBER OF SYNCHRONISATION CHAIN CHANGES	The number of times this base changed DECT sync source in the current hour.
TOTAL NUMBER OF SYNCHRONISATION CHAIN CHANGES	The lifetime total count of times this base changed DECT sync source.
TOTAL TIME IN SYNC STATE: X	The total time that the base has been in one of the following states: Master/ Locked/ Free Running/ Locked Assisted/ Sync Lost/ Searching/ Unknown
LAST REPORTED SYNC INFORMATION	The time when the base received the last sync information

7.1.20.3 RTP Statistics

The table below describes the available parameters under this category

PARAMETERS	DESCRIPTION
TOTAL RTP CONNECTIONS (INCLUDING CONNECTION TYPE INFORMATION, E.G. EXTERNAL, RELAY, RECORDING)	The lifetime total count of instantiated RTP streams.
MAX CONCURRENT RTP CONNECTIONS (INCLUDING CONNECTION TYPE INFORMATION, E.G. EXTERNAL, RELAY, RECORDING)	The lifetime highest concurrent count of instantiated RTP streams.
TOTAL TIME SPEND IN MAX RTP CONNECTIONS IN USE	The time spent in the highest concurrent count of instantiated RTP streams.
CURRENT RTP CONNECTIONS (INCLUDING CONNECTION TYPE INFORMATION, E.G. EXTERNAL, RELAY, RECORDING)	The current count of instantiated RTP streams.
CURRENT LOCAL RTP CONNECTIONS	
CURRENT LOCAL RELAY RTP CONNECTIONS	
CURRENT REMOTE RELAY RTP CONNECTIONS	
CURRENT RECORDING RTP CONNECTIONS	
CURRENT BLACKFIN DSP STATUS	Data is only available if DSP module is installed ***NOTE:*** *This feature is not present in RTX9431*

TOTAL NUMBER OF BLACKFIN DSP RESTARTS	Data is only available if DSP module is installed ***NOTE:*** *This feature is not present in RTX9431*

7.1.20.4 IP-Stack statistics

The table below describes the available parameters under this category:

PARAMETERS	DESCRIPTION
TOTAL CONNECTIONS OPEN	The lifetime total count of used sockets.
MAX CONCURRENT CONNECTIONS OPEN	The lifetime highest concurrent count of used sockets.
CURRENT CONNECTIONS OPEN	The current count of used sockets.
TOTAL NUMBER OF TX MESSAGES	The lifetime total count of transmitted IP packets.
TOTAL NUMBER OF RX MESSAGES	The lifetime total count of received IP packets.
TOTAL NUMBER OF TX ERRORS	The lifetime total count of errors occurred during IP packet transmission.

7.1.20.5 System statistics

The table below describes the available parameters under this category:

PARAMETERS	DESCRIPTION
UP TIME	The time the base has been running consecutively.
CURRENT CPU LOAD	The current load percentage of CPU. This is refreshed once every 5 seconds.
MAIL QUEUE ROS_SYSLOG	Size of internal mail queue for syslogs.
MAIL QUEUE ROS_X	Size of internal mail queue.

7.1.20.6 Heap Statistics

The table below describes the available parameters under this category:

PARAMETERS	DESCRIPTION
HEAP SIZE	Size of the heap memory in Bytes
HEAP USED	The current use of heap in Bytes.
HEAP FREE	Number of free Bytes.
HEAP FREE (%)	Free heap memory in %.
LARGEST FREE BLOCK	Size of largest free block in Bytes.

7.1.21 Diagnostics

This menu provides information about the Ethernet connection to each base station and extension *(Fig.44).* It includes the following five administrative pages, which are described accordingly.

1. Base stations

2. Extensions
3. Logging
4. Heap Monitoring
5. Load Reporting

Diagnostics

Base stations / Extensions / Logging / Heap Monitoring / Load Reporting

Base Station Name	Active Dect Ext (Mm/Ciss/CcOut/CcIn)	Active Dect Rep (Mm/Ciss/CcOut/CcIn)	Active Dect Location Gateway (Mm/Ciss/CcOut/CcIn)	Active RTP (Lcl/Rx BC)	Active Relay RTP (Lcl/Remote)	Latency [ms] (Avg.Min/Average/Avg.Max)
Sum	0/0/0/0	0/0/0/0	0/0/0/0	0/0	0/0	NA

Figure 47 Diagnostics.

7.1.21.1 Base stations

The table below describes the available parameters on this page.

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base name (name can be edited in the *Management* page)
ACTIVE DECT EXT (MM/CISS/CCOUT/CCIN)	Number of active DECT MAC connections to extensions on the different base stations. Type of connection is mm/Ciss/CcOut/CcIn
ACTIVE DECT REP (MM/CISS/CCOUT/CCIN)	Number of active DECT MAC connections to repeaters on the different base stations. Type of connection is mm/Ciss/CcOut/CcIn
AACTIVE DECT LOCATION GATEWAY (MM/CISS/CCOUT/CCIN)	Number of active DECT MAC connections to location gateways on the different base stations. Type of connection is mm/Ciss/CcOut/CcIn
ACTIVE RTP (LCL/RX BC)	Number of active RTP Streams used. Types of streams are Local RTP stream/Broadcast Receive RTP stream
ACTIVE RELAY RTP (LCL/REMOTE)	Number of active RTP Relay Streams used. Types of streams are Local RTP Relay stream/Remote RTP Relay stream
LATENCY [MS] (AVG.MIN/AVERAGE/AVG.MAX)	Ping latency between base station performed by base index 0. Average Minimum delay/Average/Average Maximum delay)

7.1.21.2 Extensions

Information on this page is visible only if there is a handset crash. The table below describes the available parameters.

PARAMETERS	DESCRIPTION
IDX	Extension index number
NO OF HS RESTARTS	Displays the number of times the handset has restarted
LAST HS RESTART (DD/MM/YYYY HH:MM:SS)	Displays the date and time of the last handset restart

7.1.21.3 Logging

This page allows the administrator to collect system diagnostics information into a zip file. The zip file includes all type of information, such as RSX trace, Syslog, SIP Log, Config file(s), etc.

There are two types of tracing which can be enabled – RSX and PCAP internal tracing. After configuring the type of traces that need to be collected, the changes must be saved. Next, after collecting the data, the administrator can choose from which base stations to download the traces. If it is a multi/dual cell system, the data can be downloaded from all base stations in one file. Else, the system diagnostics can be downloaded from the current machine. The available trace settings are described in the following sub-sections.

7.1.21.3.1 RSX internal tracing

RSX internal tracing can be either enabled or disabled. If enabled, the traced data should be sent to RTX, since it can only be investigated by RTX engineers.

7.1.21.3.2 PCAP internal tracing

This feature allows the user to choose which trace to investigate by selecting the desired parameter.

PARAMETERS	DESCRIPTION
TRACE PACKETS TO/FROM THIS BASE (EXCEPT AUDIO)	If selected, all Ethernet packets sent to/from the base station's MAC address will be traced. Broadcast packets sent from the base are also being traced.
TRACE AUDIO PACKETS TO/FROM THIS BASE	If selected, RTP streams to/from the base station will be traced. Audio packets are filtered by the port number used for RTP packets (configured on the Network page)
TRACE RECEIVED BROADCAST PACKETS	If selected, all broadcast packets, received by the base station, will be traced.
TRACE RECEIVED IPV4 MULTICAST PACKETS	If selected, all received IPv4 multicast packets will be traced
TRACE RECEIVED PACKET WITH DESTINATION MAC BETWEEN	If selected, each byte of the received destination MAC is checked to verify if it is in the trace range
TRACE RECEIVED ETHERTYPE	If selected, the user can select up to 3 received Ethertypes to trace
TRACE RECEIVED IPV4 PROTOCOL	If selected, the user can select up to 3 received IPv4 protocols to trace
TRACE RECEIVED TCP/UDP PORT	If selected, the user can select up to 3 received TCP/UDP ports to trace.

7.1.21.4 Heap Monitoring

This feature allows the user to define the values of heap usage parameters that will be tracked in syslog and under ***Generic Statistics*** page.

PARAMETERS	DESCRIPTION
HEAP STATISTICS	Default value 60 sec. Defines the distance between data points in usage curves.
HEAP USAGE IN SYSLOG	Default value 60 sec. Defines how often is heap usage written to syslog.
HEAP ANALYSIS	**Time window:** Default value 30 min. Blocks older than defined time window will be ignored when running the analysis. **Analysis in syslog. Interval:** Default value 0 min. Defines how often is heap analysis logged to the syslog. Result:

NOTE: *This feature is disabled by default to save energy and alleviate performance. User can enable it as needed but should have in mind that it uses resources and therefore will slow down the performance of the base station.*

7.1.21.5 Load Reporting

This feature allows to change the parameters that define the reporting of the system load of the base station. Those results can be seen in the syslog.

PARAMETERS	DESCRIPTION
ENABLE LOAD REPORTING	Default value: Disabled. When enabled allows to collect information about the system load of the base station.
SYNCRONIZATION OF LOAD REPORTING INTERVAL	Default value: Disabled. This parameter controls if the sample interval shall be synchronized between all bases in a multi cell setup, or if it applies only to the current base on which it is changed.
LOAD REPORTING INTERVAL	Default value: 20 sec. Defines the interval of time where data is sampled and presented in the syslog.

***NOTE:** This feature is disabled by default to save energy and alleviate performance. User can enable it as needed but should have in mind that it uses resources and therefore will slow down the performance of the base station.*

7.1.22 Configuration

This page provides a non-editable information of the entire SME VoIP parameter settings *(Fig.45).* They represent the “background” of the user-friendly interface. These settings can be exported, edited, and then uploaded to the base again. The file handling, which contains these parameters, is called a *configuration (provisioning) file*, whereas the settings application process is called *provisioning*.

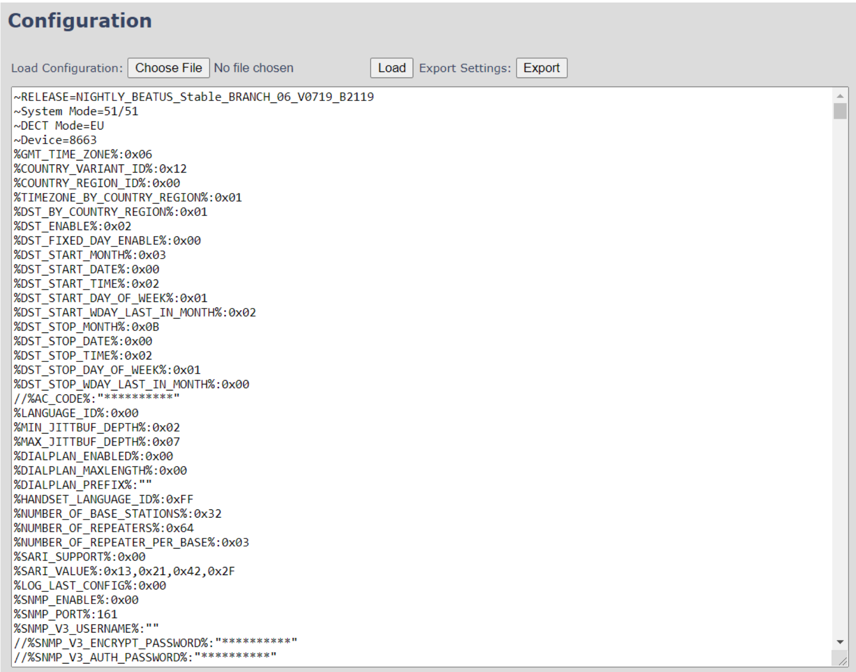


Figure 48 Configuration page

There are three ways to edit and apply the parameter settings.

- **Using the SME VoIP Web interface** – This method includes a manual configuration of the base, by using the web interface. The parameters from the configuration file represent the settings on every page. Therefore, if any setting is modified via the web interface, it will also take effect on the parameters from the Configuration page (file).

- **Using the Configuration page of the VoIP SME web interface (provisioning)** – This method includes a manual upload of the configuration file directly to the base. An example of using this method is to export the configuration file and modify the relevant parameters. Afterwards, save the changes and upload the file to the base station via the *Configuration* page.
- **Retrieving the relevant configuration file from the TFTP server (provisioning)**– This method includes a server on which the configuration file is stored and respectively downloaded by the base. An example of using this configuration method is to export the configuration file from the base station and rename it to the MAC address of the device (*<MAC_Address>.cfg*). Then, modify the settings and save the configuration file in the folder */Config* on the TFTP server. Navigate the base to the server (via the *Management* page). The base station will retrieve the file and apply the changes. Afterwards, any changes made to the file on the server, will be applied to the base.

***NOTE:** From v460 the base stations support configuration files up to 1MB*

***NOTE:** Passwords/Credentials are not exported for security reasons.*

For a detailed description on how to use provisioning, please refer to the ***Provisioning of SME VoIP System (24).pdf*** guide.

7.1.23 Syslog

This page displays live feed of system level messages of the current base station *(Fig.46).* The displayed messages depend on the configured settings on the *Management* page. The logs can either show only *Boot Log* or *All logs,* meaning all system and boot logs. The Debug log is saved in the file format **<Time_Stamp>b.log** in a relevant location on the TFTP server, as specified in the upload script.

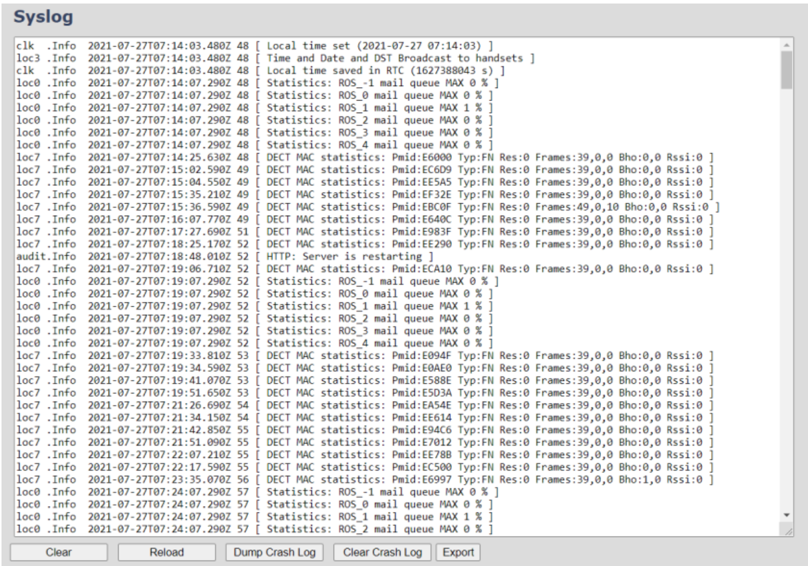


Figure 49 Syslog page

There are four buttons available, allowing the administrator to:

- **Clear** – delete the current logs.

- **Reload** – refresh the logs
- **Dump Crash Log** – import the crash errors, if any, into the log. An example is an *Unexpected Reboot* of the base
- **Clear Crash log** – delete the crash errors from the log
- **Export** – download the logs (syslog file)

7.1.24 SIP log

This page displays SIP server related messages that are logged during the operation of the SME system *(Fig.47).* The SIP log is saved in the file format **<MAC_Address><Time_Stamp>SIP.log** on the relevant TFTP server.

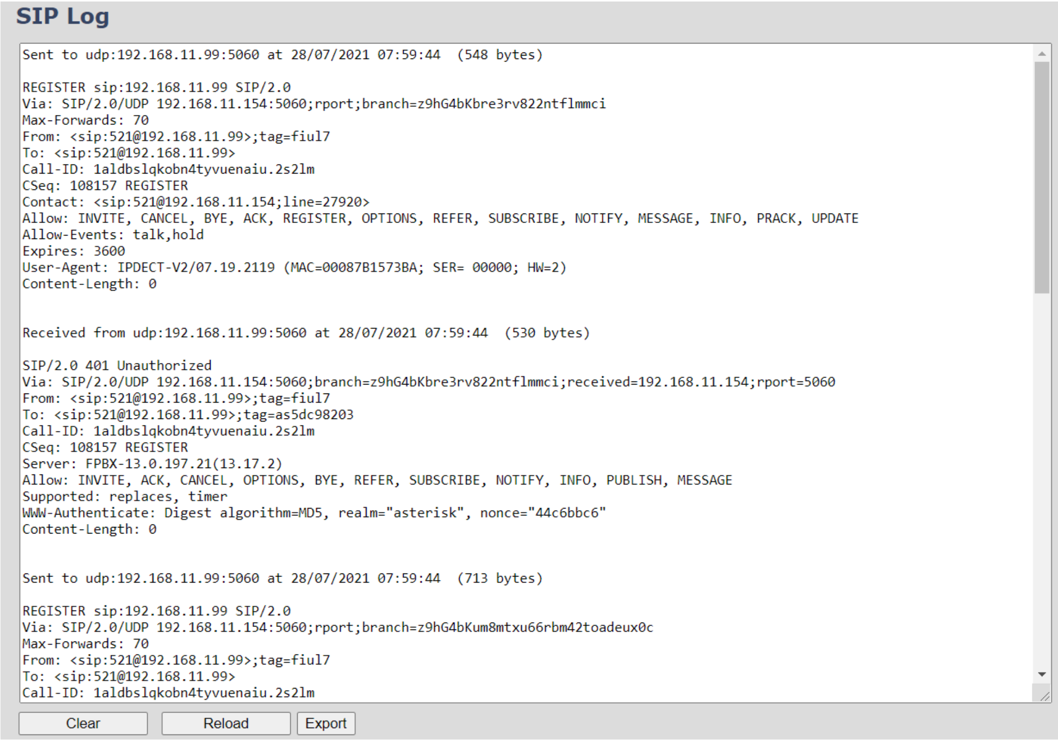


Figure 50 SIP log page

There are four buttons available, allowing the user to:

- **Clear** – delete the current logs.
- **Reload** – refresh the logs
- **Export** – download the logs (siplog file)

7.1.25 Emergency call

The base station allows the user to define emergency numbers, which can be dialed during key lock *(Fig.48).* They have the highest priority during a call, meaning that if needed, the base will release non-emergency calls to free resources to handle the emergency ones. However, in case the base is fully occupied with emergency calls, then priority is given to the established calls and new incoming emergency calls are rejected.

If no numbers are configured on the base station, the handset can still dial national emergency numbers (refer to ***9.2.1 National emergency numbers***). However, if any number has been configured, the national emergency numbers become unavailable, unless they are also added to the configuration.

Further details on the ELIN and HELD support can be found in ***9.3 ELIN and HELD support***.

NOTE: *When updating the setting on a multi/dual cell system, a reboot of the entire multi/dual cell or handsets is required. It is needed for the handsets to receive the setting update and bypass any key-lock /security functionality implemented on the device.*

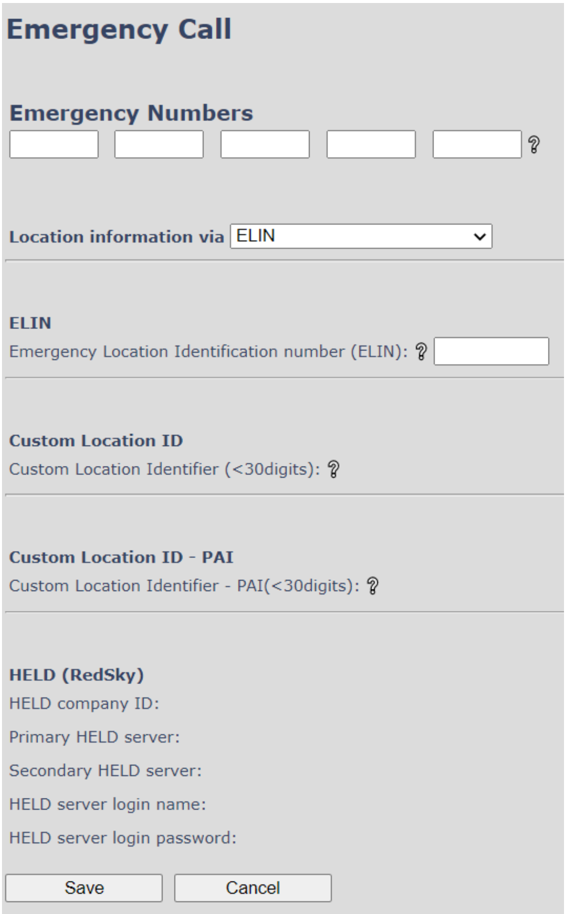


Figure 51 Emergency Call page

PARAMETER	Default value	Description
EMERGENCY NUMBERS	Empty	Configuration of the emergency numbers. **Permitted values:** 0-9 **Maximum digits:** 7
LOCATION INFORMATIO N VIA	ELIN	Select the desired method for providing location **Available options:** ELIN/ Custom ID/ Custom ID-PAI/ HELD-RedSky
ELIN	Empty	Configuration of the unique ELIN number **Permitted values:** 0-9 **Maximum digits:** 10
CUSTOM LOCATION ID	Empty	Configuration of the Custom Location ID

		Permitted values: ABCDEFGHIJKLMNOPQRSTUVXYZabcdefghijklmnopqrstuvxyz01234567890-.!%*_+~ **Maximum characters:** 29 ***NOTE:*** *No spaces are allowed as part of the string*
CUSTOM LOCATION ID PAI	Empty	Configuration of the Custom Location ID-PAI **Permitted values:** ABCDEFGHIJKLMNOPQRSTUVXYZabcdefghijklmnopqrstuvxyz01234567890-.!%*_+~ **Maximum characters:** 29 ***NOTE:*** *No spaces are allowed as part of the string*
HELD COMPANY ID	Empty	The HELD Company ID is the account in RedSky which the administrator's company obtains. **Maximum length:** 48
PRIMARY HELD SERVER	https://primelab.e911cloud.com/	Configuration of the Primary HELD Server **Maximum length:** 69
SECONDARY HELD SERVER	https://primelab.e911cloud.com/	Configuration of the Secondary HELD Server **Maximum length:** 69 ***NOTE:*** *This setting is optional. However, it is advisable to have it present, in case a server exists.*
HELD SERVER LOGIN NAME	Empty	Configuration of the HELD Server Login name **Maximum length:** 19 ***NOTE:*** *Currently, login credentials are not mandatory. However, they may become if RedSky's authentication requirements change*
HELD SERVER LOGIN PASSWORD	Empty	Configuration of the HELD Server Login password **Maximum length:** 49 ***NOTE:*** *Currently, login credentials are not mandatory. However, they may become if RedSky's authentication requirements change*

7.1.26 Logout

This feature allows the user to logout from the system.

7.2 Handsets

All available functions and settings are displayed as icons and text on the handset screen. Even though there are 2 types of user interfaces (Classic and Modern), the icons are similar and the functionality is the same (refer to ***2.3.2 Handsets*** for more details).

7.2.1 General overview

The screenshot below *(Fig.49)* illustrates a Modern user interface of an RTX8631 handset. This interface is used in throughout chapter to identify the available icons, menus and symbols present in each of RTX's handset devices. Any visible difference in the layout between the various handset models, is additionally mentioned via a note.

The idle screen represents the standard layout of a handset, which consists of a Softkey bar, an Active area and a Status bar. The bottom of the screen is referred to as the Softkey bar, which provides you both with fast access to some menus (Contacts, Central Directory and Calls) and possibility to select various functions (Select, View, More, etc.).

The middle part of the screen is referred to as the Active area of the display. Apart from displaying the time and date to the user (in idle), it provides the view to all menus, functions, and pop-up dialog text boxes (such as alarms, missed calls, etc.).

NOTE: *The Classic interface can also display the following in idle: extension name, which can be set on the web interface of the base station; the registration number; and the handset name, which can be set locally on the handset.*

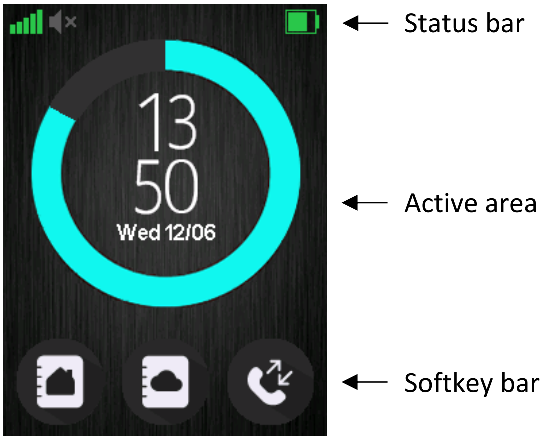


Figure 50 General UI overview

The top of the display, which is visible on all screens (independent on the menus), is referred to as the Status bar. The icons listed below can be visible on the Status bar and provide different information to the user.

ICON	TYPE	DESCRIPTION
	Signal strength	Always visible in the upper left corner. The number of green staples is displayed according to the strength of the signal.
	Battery	Always visible in the upper right corner. The icon displays the battery status. 00 - 20% **empty** battery symbol 21 - 32% **red** battery symbol 33 - 44% **yellow** battery symbol 45 - 68% **partly green** battery symbol 69 - 100% **full green** battery symbol
	Missed call	Indicates that there has been a missed call. The call can be seen in the *Call log* menu.
	Voice mail message	Indicates that there are voice mails that have not been listened to. This icon remains visible until the voice mail has been opened.
	Message	Indicates that there are messages on the handset that have not been read.
	Alarm clock	Indicates that an alarm clock has been set on the handset.
	Silent Alarm clock	This icon is visible when an alarm clock is enabled and the handset is either in silent mode, or the volume is muted
	Sound off/Silent	Indicates that silent mode is enabled on the handset

	Keylock	Indicates when the keypad is locked. To unlock it, the user needs to longpress the * -key to be able to use the handset again.
	Do not disturb	Indicates that the feature “Do not disturb” is enabled. In this case, the handset does not react to any incoming calls.
	Headset attached	Indicates that a wired headset has been inserted and detected
	High Definition	This icon is displayed during a call and indicates if the call quality is High definition and thus, using the 722 codec

7.2.2 Call Screen

To make a call from Idle simply enter the phone number from the keypad and press the off-hook key.

To answer a call simply press the hook-off key.

When the handset is in a call it shows on the screen the number we are connected to. In some models it shows in the top right corner the line we are using to connect to that call (that is in the case of having multiple lines used on a single device. To see how to setup a multiline look at section 8.3.1.1). There can also be present (depending on the model of the handset) small icons showing incoming call, outgoing call and if the call is currently on hold.

7.3 Menu navigation

All menus on the handset can be accessed by pressing the *Menu* key on the device. During navigation, the highlighted icon is moving to indicate the user where the navigation key is. The table below provides further information about the different menus available on the handsets.

MENU	DESCRIPTION
CONTACTS	Contains all names/numbers in the local phonebook.
CALLS	Contains the call log and provides information about the calls
CONNECTIVITY	Contains network options, including registering/deregistering a handset to/from the base.
SETTINGS	Contains handset settings such as time/date, language, handset name, etc.
MESSAGING	Contains sent and received messages, along with the available messaging settings.
ALARMS	Contains alarm clock settings
F-KEYS	Contains personal handset settings for the F-Keys
ALARM LIST	Contains alarms received on the handset.
CENTRAL DIRECTORY	Contains all names/numbers in the central directory of the system. The central directory can be set up on the base (refer to ***7.1.13 Central Directory*** for more information)

Some of the menus can also be accessed via the three softkeys in idle *(Fig.49).* These are the *Contacts* menu (left softkey), the *Central Directory* menu (middle softkey) and the *Calls* menu (right softkey). These fast access menus are pre-fixed and cannot be changed.

The following subsections will introduce each menu of the handset and its available options.

7.3.1 Contacts

This menu displays the available local contacts on the handset *(Fig.50)*. If the contact list has entries, there will be three available options assigned to the softkeys – *Call* (to call the selected contact), *Edit* (to edit the selected contact) and *More* (other available options, further described in the following table). If the contact has more than one number available (work, home, mobile, etc.) a small arrow appears to the right of the contact. Pressing the navigation key to the right lets us toggle between the different numbers.

Figure 51 Contacts menu overview

When *Central Directory* is enabled, the user is presented with additional menu where they can choose between Contacts and Central Directory. Upon choosing Contacts handset will access the Contacts list. If User chooses Central Directory; handset will access the central directory contacts list.(refer to ***7.3.9 Central Directory*** for more details).

PARAMETER	DESCRIPTION
SEND MESSAGE	Provides the possibility to send a text message to the selected contact
ADD CONTACT	Provides the possibility to create a new contact in the list by adding the name, work/mobile/home/ other number of the contact and assigning a specific ringtone.
EDIT BEFORE CALL	Provides the possibility to edit the contact's number before calling it
EDIT CONTACT	Provides the possibility to edit the contact's information, such as name, number or ringtone.
DELETE CONTACT	Provides the possibility to delete the selected contact
DELETE ALL CONTACTS	Provides the possibility to delete all contacts from the list
SPEED DIAL	The feature allows the user to add a contact to a speed dial number (2 – 9). This enables the user to directly call a contact from idle, by long pressing the key (2-9) to which the contact is assigned to. A contact can only be assigned to one speed dial number at a time. The first defined number for the contact will be dialed when the speed dial key is pressed. This means that if all numbers (work, mobile, home, and other) are defined for the contact, then the work number will be dialed first. If the work number is not defined, then the mobile number is dialed, and etc. ***NOTE:*** *The '1' key is reserved for voicemail. Long key press on '0' is used for starting a normal dial string with '+'.* ***NOTE:*** *The speed dial list stores both contacts from the Contact and F-Keys list* ***NOTE:*** *RTX8431/RTX8436 has 4 extra keys (F1,F2,F3 and F4) to which a contact can be assigned as a speed dial key*

7.3.2 Calls

This is one of the menus that can be accessed via multiple ways – (1) the main menu, (2) the right softkey in idle and (3) the green off-hook button from the keypad *(Fig.51).*

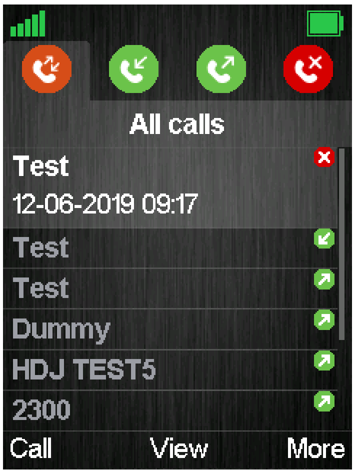


Figure 52 Calls menu overview

The menu provides lists of all the calls that have been made to and from the handset. The lists are organized in tabs, which can be accessed by using the left/right navigation keys, and include the following:

- **All calls –** This list contains all the different types of calls (including both outgoing and incoming).
- **Answered calls –** This list contains all the incoming calls that have been answered.
- **Dialed calls –** This list contains all the calls that have been dialed from the handset.
- **Missed calls –** This list contains all the incoming calls that have not been answered.

NOTE: *When there is a missed call on the handset, a red icon will be displayed in the status bar until the entry has been viewed in the missed call list.*

NOTE: *The RTX8431 and RTX8436 do not support the tab-menu feature. All calls are listed in one menu*

In every list, there are three available options assigned to the softkeys – *Call* (to call the highlighted entry), *View* (displays call details of the highlighted entry) and *More* (other available options, further described in the following table).

PARAMETER	DESCRIPTION
VIEW DETAILS	Displays call details, such as number, line, status (answered, missed, dialed), duration and time stamp. It has the same output as the middle softkey option “View”
SAVE AS CONTACT	Provides the possibility to save the number as a contact
APPEND TO CONTACT	Provides the possibility to add the number to an existing contact
SEND MESSAGE	Provides the possibility to send a message to the highlighted entry
EDIT BEFORE CALL	Provides the possibility to edit the number before calling it
DELETE ITEM	Provides the possibility to delete the highlighted entry
DELETE ALL ITEMS	Provides the possibility to delete all entries from the specific call list

7.3.3 Connectivity

This menu allows the user to register/deregister a handset to/from a certain base system *(Fig.52)*. The following sections represent the available features and their related options.

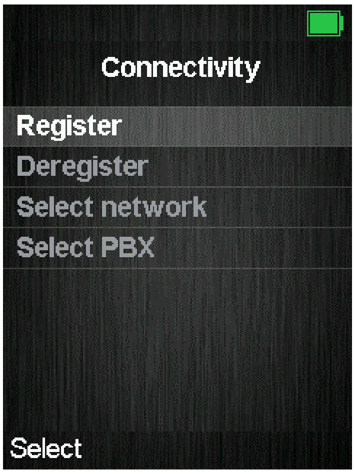


Figure 53 Connectivity menu overview

7.3.3.1 Register

This feature allows the user to register a handset to a base station (VoIP system). Depending on the type of handset, the device can register up to 10 different systems that are out of range from each other. To register the handset, the user must select an empty slot and type in an access code (base registration PIN). By default, the access code is “0000” and can only be changed via the base station (refer to ***7.1.3 Extensions*** for more information).

It is also possible to overwrite an existing registration by selecting the desired entry, instead of using an empty slot.

NOTE: *For the handset to function on different base systems, the device must be completely out of range of system 1 to connect to system 2.*

NOTE: *RTX8431/RTX8436 can only register to 1 VoIP system.*

7.3.3.2 Deregister

This option allows the user to deregister the handset from a base station (VoIP system). To deregister the device, a PIN code (base deregistration PIN) must be typed by the user. By default, the code is “0000” and can be changed via the handset (refer to ***7.3.4.6 Security*** for more information). Afterwards, the registration, which should be removed, must be selected and confirmed.

7.3.3.3 Select Network

If the handset is registered to different VoIP base systems (networks), this menu allows the user to select the preferred network. In addition to this, it allows them to configure the handset to automatically select a network. By doing so, the device will automatically choose a network in range.

NOTE: *This feature is not available for RTX8431/RTX8436 handset*

7.3.3.4 Select PBX

This menu provides the user with the possibility to select another PBX, which is to be used by the handset. This means that, the handset can be connected to a different PBX than the one used by the base station system. Hence, the handset can function on a PBX without being connected to a system of base stations.

7.3.3.5 Easy registration

This feature enables the user to add new handsets/repeaters without having to open the base webpage. This means, that the user can physically enable the registration mode by pressing the reset button on the base station, or by using the "Easy registration" menu on the handset. For further information on how to use the feature, please refer to ***8.3.4 Easy registration.***

NOTE: *This feature is only available for RTX8431/RTX8436 handset*

7.3.4 Settings

This menu allows the user to configure various types of handset settings *(Fig.53)*. The following sections present the sub-menus along with the related features.

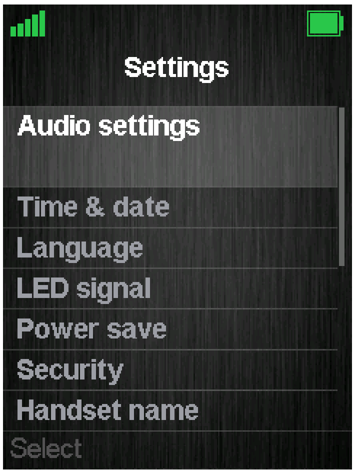


Figure 54 Settings menu overview

7.3.4.1 Audio settings

This sub-menu contains different settings related to the audio/sounds on the handset. The table below displays the available features and their configuration possibilities.

PARAMETER	DESCRIPTION
SILENT	Enable/Disable silent mode on the handset If **enabled:** - Only melodies for emergency alarm messages will be played on the device. Other sounds will be muted. - Only the following sub-menus can be accessed: "Ring emerg. high / med / low", Vibrator" and "Noise reduction" - The "Sound off" icon will be displayed on the status bar.
RING VOLUME	Select the desired ring volume level for calls/messages on the handset. The currently applied ring melody is played when adjusting the ring volume, thus the volume level can be heard before being selected. The volume can be adjusted from mute (no sound) to level 6 (highest sound). ***NOTE:*** *If the volume is set to mute, no melodies will be played when receiving an incoming call or message. The "Sound off" icon will be displayed on the status bar.*
RING MELODY	Contains a list of melodies from which the user can browse through. The chosen melody will be played during an incoming call. The melodies can be heard before being selected.

RING NORMAL MSG	Contains a list of melodies from which the user can browse through. The chosen melody will be played when messages, prioritized as normal, are received on the handset. ***NOTE:*** *The message ring melodies are played with the same ring volume as the setup for incoming calls.*
RING URGENT MSG	Contains a list of melodies from which the user can browse through. The chosen melody will be played when messages, prioritized as urgent, are received on the handset. ***NOTE:*** *The message ring melodies are played with the same ring volume as the setup for incoming calls.*
RING EMERGENCY HIGH /MED /LOW	Contains a list of melodies from which the user can browse through. The chosen melody will be played when alarm messages are received on the handset. Depending on the priority of the alarm (high/med/low), a different melody can be selected
ALERT VOLUME	Select the desired ring volume level for the alarm clock. The currently applied ring melody is played when adjusting the ring volume, thus the volume level can be heard before being selected. The volume can be adjusted from mute (no sound) to level 6 (highest sound). ***NOTE:*** *If the volume is set to mute, no melody will be played when the alarm occurs.* ***NOTE:*** *If an alarm clock is configured and the volume level is set to mute, a "Sound off alarm clock" icon will be displayed on the status bar.*
ALERT TONE	Contains a list of melodies from which the user can browse through. The chosen melody will be played when an alarm clock ring occurs. The melodies can be heard before being selected.
VIBRATOR	Provides different vibrator options for the handset: **Off:** Vibrator is disabled. The handset will not vibrate during alerts (calls/messages) **Vibrate then ring:** When an alert occurs (call/message), the handset will vibrate 4 times. Afterwards, the vibration will stop, and the handset will ring instead. **Vibrate only:** The handset will vibrate only, without ringing (even if the ringing signal is enabled on a certain level) **Vibrate and ring:** The handset will both vibrate and ring at the same time, when an alert occurs (call/message)
KEY SOUND	Controls if a sound is played when keys are pressed on the handset. The following options are available: **Silent:** No key sound will be played **Click:** A "click" key sound will be played **Tone:** A "tone" key sound will be played ***NOTE:*** *Even if one of the options (Click/Tone) is enabled, no key sound will be generated when pressing the keys during "key lock state"*
CONFIRMATION SOUND	Enable/Disable confirmation sounds on the handset If **enabled**, an audible indication is given when an event succeeds or fails. For example, a positive confirmation sound is generated when a new setting is set, or an entry is saved successfully in the phonebook. A negative confirmation sound is generated if the action fails
COVERAGE WARNING	Enable/Disable a coverage warning sound to occur when the handset is out of range of a base station in the system. If **enabled**, an audible indication is given in the earpiece of the handset when the user is close to the maximum range of the nearest base station in the system. Thus, the

	user knows that the call is at the edge of the possible coverage and must move back in coverage to keep the call active. ***NOTE:*** *This feature is only working when the handset is in call. Not when in idle.*
CHARGER WARNING	Enable/Disable the handset to generate a sound, when put in the charger, to indicate that the handset is charging.
SPEAKERPHONE PROFILES	Customize the tonality of the speakerphone mode. The following profiles are available: **Balanced:** controls the pleasant tonality of the speech **Loud:** prioritizes the loudness of the speech ***NOTE:*** *If the user selects "Balanced", the sound quality will become more pleasant, but also low.* ***NOTE:*** *This feature is only supported by RTX8631-32-33*
NOISE REDUCTION	Enable/Disable noise reduction. It is a function used when the handset is in a call. An algorithm is used to reduce the noise from the background. In this way, the user should sound clearer to the counterpart since the noise is reduced to a certain level. The following level options are available: **Low:** The noise is reduced by approx. 6dB **High:** The noise is reduced by approx. 12dB ***NOTE:*** *This feature is not supported by RTX8630 and RTX8830*
NOISE CANCELING	Enable/Disable noise canceling. It is a function used when the handset is in a call. This feature is more advanced than *Noise reduction*, as it uses 2 microphone solutions to reduce the background noise to a minimum. ***NOTE:*** *This feature is only supported by RTX8830*

7.3.4.2 Time & date

This sub-menu allows the user to configure the time and date on the handset. The table below displays the available features and their configuration possibilities.

PARAMETER	DESCRIPTION
TIME	Set the time and choose the desired format: **24h** **12h** (**am/pm)**
DATE	Set the date and choose the desired format: **dd-mm-yyyy** **mm/dd/yyyy** and etc.

7.3.4.3 Language

This sub-menu lists the available languages to which the handset can be configured. The list can be seen in the table below.

English	Español	Deutsch	Français	Italiano	Nederlands
Português	Dansk	Svenska	Turkçe	Polski	Norsk
Russian – 1	Russian – 2	Hrvatski	Srpski	Slovenian	

7.3.4.4 LED signal

This sub-menu represents the configuration of the LED, located at the upper left corner of the handset. The LED has 3 colors (green, yellow (purple), red), which can respectively be configured to indicate the user with a certain event.

PARAMETER	DESCRIPTION
GREEN /YELLOW (PURPLE) /RED LED	For every color, select one of the following events which it should correspond to. **Off:** The LED signal is disabled, and does not blink in any situation **Missed call:** The LED signal blinks in the selected color when there is a missed call **Voice message:** The LED signal blinks in the selected color when there is a voice mail message that has not been listened to. **Low battery:** The LED signal blinks in the selected color when the battery is low (below 20% battery on the handset). ***NOTE:*** *The purple color is only available for RTX8830*

7.3.4.5 Power save

This sub-menu allows the user to minimize the battery consumption by configuring the display to turn off after a certain amount of time.

SECONDS	5s	10s	20s	30s	45s	60s

7.3.4.6 Security

This sub-menu provides the user with the possibility to configure and control the user's access to the device and its features. The table below illustrates the available options.

PARAMETER	DESCRIPTION
PHONE LOCK	Enable/Disable the possibility to lock the phone while in standby mode. If **enabled**, a code must be entered to unlock the device. Use the * key to access the code pop-up message. **Change PIN:** Allows the user to change the PIN code of the phone lock. ***NOTE:*** *The default code is 0000* ***NOTE:*** *To apply the saved settings, the handset needs to be rebooted*.
AUTOMATIC KEYLOCK	Enable/Disable this handset option to automatically lock the keypad after a certain amount of time that the device has not been active. This means that the handset keys cannot be used until the user unlocks the device (by long pressing *). The following time intervals can be selected: **Off** **15s / 30s / 45s** **1 min / 2 min / 3 min / 4 min / 5 min**

	NOTE: *Emergency numbers (e.g., 911 / 112) can be called without the need to unlock the keypad*
CHANGE PIN	Change the handset's security PIN code. The code is used when deregistering the handset from the base and when resetting the handset's settings. ***NOTE:*** *The current PIN code is 0000*

7.3.4.7 Alarm settings

This sub-menu allows the user to customize the following alarm settings:

PARAMETER	DESCRIPTION
ALARM IN CALL	ON/OFF When enabled, and an alarm arrives during a call, it will play the call waiting tone and show the alarm on the screen. If it is disabled, it will not play anything until the call is over.
CALLBACK CONFIRM	ON/OFF When enabled, and alarm has been answered using a call back option, a confirmation message will be sent to the Alarm server, once the call has been established with the specified callback number. If disabled, no alarm confirmation message will be sent to the alarm server. ***NOTE:*** *This confirmation functionality is currently not compatible with voicemail options from the PBX, as this connects the call and there currently is no way to distinguish it from the call being answered on the remote end.*

7.3.4.8 Handset name

This sub-menu allows the user to edit the name of the handset.

HANDSET NAME	Type in a name for the handset. The device's name will be displayed on the third line of the idle screen. ***NOTE:*** *If the Modern MMI is used, then the name will not be visible (this is not valid for the modern interface of the RTX8663 handset)*

7.3.4.9 Line

This sub-menu allows the user to select the preferred line of the handset, if the device is registered to multiple systems/SIP accounts.

LINE	Select the desired line from the list.

7.3.4.10 Reset settings

This sub-menu provides the possibility to reset the handset's settings to default.

RESET SETTINGS	Reset the settings to default by typing in the PIN code (refer to ***7.3.4.6 Security*** for more details on the PIN code) ***NOTE:*** *After the reset, the handset is still registered to the base.* ***NOTE:*** *When selecting this feature, the base is not reset to default*

7.3.4.11 Status

This sub-menu provides useful information about the handset and the base station it is registered to.

STATUS	The following information is displayed for the devices: **Base station status:** - Software version (the current software version of the base station) - Hardware version (the hardware version of the base station) - IP address (the IP address of the base station) - MAC address (the MAC address of the base station) - System name (the name of the system of base stations that the handset is registered to) **Handset status:** **-** Software version (the current software version of the handset) - Hardware version (the hardware version of the handset) - DECT band (the DECT band that the handset is using, e.g., EU or US) - Battery level (the current battery level of the handset) - IPEI (the IPEI/IPUI number of the handset is a static number identifying the specific handset) - Bluetooth (the Bluetooth MAC address of the handset) ***NOTE:*** *The Bluetooth MAC address is displayed only for devices supporting the Bluetooth feature*

7.3.4.12 Auto answer

This sub-menu allows the user to configure the handset to answer incoming calls in various ways.

AUTO ANSWER	Select one of the below conditions to configure the desired mode **Normal:** The user must manually answer the incoming calls by pressing the green "Off-hook" button. ***NOTE:*** *The call is not answered by removing the handset from the charger.* **Any Key:** The user can press any key from the keypad to answer the incoming call. ***NOTE:*** *This does not include the red "On-hook" button and the three softkeys, which have their different functions.* **Automatic:** Incoming calls are automatically answered after 5 seconds.

7.3.4.13 Silent charging

This sub-menu enables the handset to be configured to ring in silent mode while charging.

SILENT CHARGING	Enable/Disable the handset to ring in silent mode while charging, by selecting one of the options below: **Off:** Audio alert is played during an incoming call **Silent:** Incoming calls are received, but no audio is played ***NOTE:*** *The handset must be placed in the charger for this feature to work.*

7.3.4.14 Do not disturb

This sub-menu allows the user to set up a "Do not disturb" mode, meaning that incoming calls will not be received on the handset.

DO NOT DIRTURB	Enable/Disable the incoming call activity. If **enabled**, incoming calls will be rejected or redirected to a voicemail (if voicemail feature is set). This means that, the base will not send any ringing indications. Else, it will be the PBX that is sending the ringing before contacting the base. ***NOTE:*** *A "Do not disturb" icon will be displayed on the status bar if the feature is enabled* ***NOTE****: If* ***BroadSoft event feature package*** *enabled the icon will not be showing. When BroadSoft server is used, DND feature is controlled by the PBX and therefore, icon on the idle screen won't show. In that case DND will appear as a text on idle screen instead.*

7.3.4.15 Hide Number

This sub-menu provides the user with the opportunity to hide their number during an outgoing call. By enabling this feature, the counterpart will not see the caller ID.

HIDE NUMBER	Enable/Disable the device to hide its number by selecting one of the options below: **Off:** The caller ID of the handset is displayed to the counterpart of the call **On for next call:** The caller ID of the handset is hidden to the counterpart only for the next outgoing call. Afterwards, the following calls will display the number. **Always on:** The caller ID of the handset is always hidden to the counterpart for all outgoing calls, until the feature is disabled again. ***NOTE:*** *When this feature is enabled, the base sends a SIP header "Privacy: id" to address sip:anonymous@anonymous.invalid* ***NOTE:*** *The PBX must support anonymous calls. Else, if the feature is enabled, the call will be rejected. Thus, the handset will not be able to make an outgoing call.*

7.3.4.16 Call forward

Depending on the scenario, this sub-menu enables the user to forward all incoming calls to a selected number.

PARAMETER	DESCRIPTION
FORWARD UNCONDITIONALLY	Enable/Disable the possibility to forward all incoming calls to a pre-defined number **Forward To:** Type in the number to which the calls should be forwarded

	NOTE: The handset will not receive any incoming calls, as all of them will be directly forwarded to the provided number from above.
FORWARD NO ANSWER	Enable/Disable the possibility to forward all incoming calls that are not answered. **Forward To:** Type in the number to which the calls should be forwarded **Forward After (s):** Type in the timeframe which the user can makes use of to answer the call. If this period ends and the call has not been answered, the call will be forwarded to the provided number.
FORWARD BUSY	Enable/Disable the possibility to forward all incoming calls while the handset is busy (e.g., user is in another call) or when the user presses "Reject" on the call. **Forward To:** Type in the number to which the calls should be forwarded

7.3.4.17 Idle screensaver

This sub-menu enables the handset to dim the light of the screen while in the charger.

IDLE SCREENSAVER	Enable/Disable the screen light saving, while in the charger, by selecting one of the following options: **Off:** Feature is disabled. No lights will be dimmed while the handset is in the charger. **Idle clock (dimmed):** The idle screen will dim the lights as soon as the handset enters power save mode while in the charger. **Startup image (dimmed):** The startup image will dim the lights as soon as the handset enters power save mode while in the charger. ***NOTE:** If the feature is enabled, the handset screen will dim the light, but never turn off*

7.3.4.18 Idle clock

This sub-menu allows the user to control the visibility of the clock on the idle screen.

IDLE CLOCK	Enable/Disable the appearance of the clock on the idle screen ***NOTE:** A "Do not disturb" icon will be displayed on the status bar if the feature is enabled* ***NOTE:** This feature is not supported by RTX8631 and RTX8632*

7.3.4.19 Bluetooth

This sub-menu enables the handset to connect to a Bluetooth headset. When this connection is present, the audio on all calls is routed through the Bluetooth headset. This means that, when receiving an incoming call, alerting is heard both from the handset and the paired Bluetooth headset.

***NOTE:** This feature is not supported by RTX8431, RTX8436 and RTX8630*

The following types of headsets are on the whitelist for RTX8830 Handset:

- Jabra Talk
- Jabra Stone
- Jabra Supreme
- Nokia BH-904

NOTE: *The Jabra Supreme headset has two features - Jabra Voice Guidance and Jabra Voice Control. These two features guide the user on how to use the headset and make it possible to control the hands-free by using voice control. These Jabra Voice features are both supported by the RTX8830 handset.*

The table below summarizes the available options in the Bluetooth sub-menu.

PARAMETER	DESCRIPTION
ENABLE/DISABLE	Enable/Disable the possibility to pair the handset to a Bluetooth headset. ***NOTE:*** *Pairing a handset to a Bluetooth device is possible only when Bluetooth is* ***enabled****.* ***NOTE:*** *When the feature is* ***enabled****, a Bluetooth icon is displayed on the status bar of the handset. Depending on the paired mode, the icon appears in a different style and color:* - *Gray – if handset is* ***not*** *paired to another device* - *Blue – if handset is paired to another device*
SEARCH FOR DEVICES	Select this option to search for other Bluetooth devices to which the handset can connect. The detected devices (in pairing mode) will be listed by their IDs / names on the handset display. ***NOTE:*** *Bluetooth must be enabled on the handset in order to search for other Bluetooth devices. Else, as soon as a search is initiated, the Bluetooth feature will automatically be enabled.* ***NOTE:*** *The Bluetooth devices, that are to be found by the handset, need to be in pairing mode.* ***NOTE:*** *If a PIN code is requested when pairing the devices, it is usually 0000.*
PAIRED DEVICES	Provides a list with all paired devices. The handset can be paired with up to four different Bluetooth headsets. The user has the opportunity to **Disconnect** a Bluetooth device from the handset or explore the following **More** options: **Change name:** Allows the user to change the name of the paired Bluetooth device **Delete:** Allows the user to delete the selected paired device. If deleted, the handset will no longer be paired with this device **Delete all:** Allows the user to delete all paired devices. If deleted, the handset will no longer be paired with any of these devices. **Info:** Provides details about the paired device, such as *Status*, *Headset Name* and *Address*.

7.3.4.20 Push To Talk

This sub-menu enables the handset to behave like a Walkie Talkie. This is done by pressing and holding the mute button on the left side of the handset when in Idle state. As long as the mute button is pressed, it is possible to talk to the other devices on the system that have the Push To Talk feature enabled. The received voice is heard via the loudspeaker of the handset. To end the transmitting mode and allow the counterpart to talk, the mute button must be released.

NOTE: *This feature is not supported by RTX8431/RTX8436 handset*

IMPORTANT: When enabled, this feature uses 2 channels to allow communication between the devices. Therefore, even if the feature is enabled only on one device, the system has already reserved 2 channels for the Push To Talk feature. This means that, instead of having the opportunity to establish 10 simultaneous calls, the base system will be limited to 8 (***see 2.3.1 Base stations*** for more info on the call capacity). In this way, it is possible to have both 8 simultaneous calls and Push To Talk communication in the DECT environment.

PUSH TO TALK	Enable/Disable the "Walkie Talkie" function ***NOTE:*** *The feature must be enabled on at least two handsets in order for it work.* ***NOTE:*** *Only one person at a time can talk on the Push To Talk system.* ***NOTE:*** *Pressing and holding the mute button, while another user is talking, will only mute the loudspeaker.*

7.3.4.21 Function keys

As previously mentioned, the RTX8431/RTX8436 has 4 extra keys below the original keypad, referred to as Function key X (F1, F2, F3 and F4) from left to the right-hand side. They can be configured by the user to execute a certain action when being pressed. Selecting an action type for a function key can be done by the following ways:

- Pressing a function key where no action has been assigned yet
- Navigating to the *Function keys* menu item.

NOTE: *A function key will only perform its programmed action in idle state*

NOTE: *By default, no action is configured on the function keys*

NOTE: *This feature is only supported by RTX8431/RTX8436 handset*

The table below summarizes the list of programmable options in the *Function keys* sub-menu.

PARAMETER	DESCRIPTION
IGNORE KEY PRESS	Enable/Disable the key press event If **enabled,** any type of key press on the selected function key will be ignored
SPEED DIAL	Allows the user to assign a contact from the local *Contacts* or *F-Keys* list to a function key. The handset will call the assigned contact when the key is pressed. The function keys are also visible in the *Speed dial* menu and can be found as F1, F2, F3 and F4 (refer to ***7.3.1 Contacts*** for more details) ***NOTE:*** *If a function key is assigned to a speed dial through the Speed dial menu, the Function key menu will reflect the current function key state.*
SHORTCUTS	Opens a certain application, meaning that the handset will navigate to the specified shortcut location and display the corresponding menu.

	The following menus are supported as shortcuts: - Send Message - Inbox - Sent Messages - Settings - Time and Date - Audio settings - Ringtone - Ring Volume - Status - Connectivity ***NOTE:** Dismissing the shortcut will return to idle screen*
Direct Call	Lets the user set a number, that is called when the function key is pressed. The Direct Call feature resembles a speed dial, however, while a speed dial requires a contact in the phone book, the direct call is simply a number. One Direct Call number per function key can be defined. To disable the Direct Call function 'Ignore key press' should be set to 'On'. This will not clear the number. NOTE: This feature is not available for all handset models.

7.3.5 Messaging

This menu enables the user to send text messages to other handsets in the system and read the received ones *(Fig.54)*. It contains the following four sub-menus, which are organized in tabs and can be accessed by using the left/right navigation keys:

- **Inbox –** Contains a list of all text messages that have been received on the handset
- **Drafts –** Enables new messages to be created and contains a list of unsent messages (drafts)
- **Sent Messages –** Contains a list of all text messages that have been sent from the handset
- **Settings –** Provides various message configuration options

The sub-menus and their related options are further described in the following sections.

***NOTE:** To enable the messaging feature, the base station must be configured accordingly (refer to **7.1.7.5 Text Messaging**).*

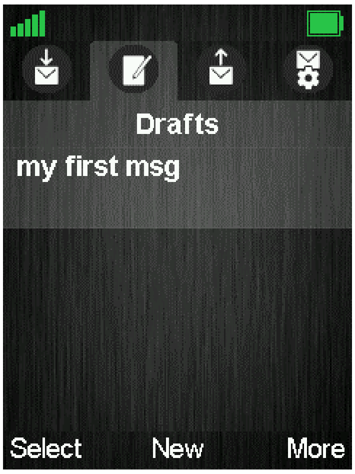


Figure 55 Messaging menu overview

Depending on the type of handset there is a difference in the amount of text messages that can be stored on the device. The table below displays the maximum number of messages per handset. The number includes both messages in *Inbox, Drafts* and *Sent messages*.

HANDSET TYPE	MAXIMUM NUMBER OF TEXT MESSAGES
RTX8431/RTX8436	3
RTX8630	70
RTX8631/RTX8632/RTX8633 RTX8830	300

7.3.5.1 Inbox

This sub-menu stores and displays all the received text messages on the handset. When messages are present, there are three available options assigned to the softkeys – *Select* (opens the highlighted message), *Reply* (enables the user to reply to the highlighted message) and *More* (other available options, further described in the following table).

PARAMETER	DESCRIPTION
REPLY	Provides the possibility to reply to the received message
FORWARD	Provides the possibility to edit and forward (send) the message to another contact (number)
SAVE AS CONTACT	Provides the possibility to save the counterparts number, from whom the message was received, to the local phonebook
APPEND TO CONTACT	Provides the possibility to add the counterpart's number to an existing contact
DELETE MESSAGE	Provides the possibility to delete the highlighted message
DELETE ALL MESSAGES	Provides the possibility to delete all messages from the *Inbox*

7.3.5.2 Drafts

This sub-menu enables the user to create a new text message and save it as a draft if the message is not sent. Draft messages can be edited and sent to the counterpart. Therefore, this sub-menu provides both a list of saved drafts and the opportunity to write a new message.

When messages are present in the list, there are three available options assigned to the softkeys – *Select* (opens the highlighted message), *New* (enables the user to write a new message) and *More* (other available options, further described in the following table). Else, only the middle softkey option *New* is visible to the user.

PARAMETER	DESCRIPTION
SEND MESSAGE	Provides the possibility to send the written message.
DELETE MESSAGE	Provides the possibility to delete the highlighted message
DELETE ALL MESSAGES	Provides the possibility to delete all messages from the *Drafts* sub-menu

NOTE: *When sending a message on some models, user is provided with the choice to either send it as a Normal message or as an* ***Urgent message****. When sending an Urgent Message bear in mind that the handset receiving the message will continue ringing (not just a short beep as with normal message) until the user takes action and acknowledges or declines the message. If the handset receiving the message has silent mode enabled there is no indication that the text message is urgent.*

7.3.5.3 Sent messages

This sub-menu enables the user to review all the messages that have been sent from the handset. When messages are present, there are three available options assigned to the softkeys – *Select* (open the highlighted message), *Forward* (edit and send the message to another contact) and *More* (other available options, further described in the following table).

PARAMETER	DESCRIPTION
SAVE AS CONTACT	Provides the possibility to save the counterparts number, to whom the message was sent, to the local phonebook
APPEND TO CONTACT	Provides the possibility to add the counterpart's number to an existing contact
DELETE MESSAGE	Provides the possibility to delete the highlighted message
DELETE ALL MESSAGES	Provides the possibility to delete all messages from the *Sent messages* sub-menu

7.3.5.4 Settings

This sub-menu enables the user to configure some of the messaging settings. The available options can be seen in the table below:

PARAMETER	DESCRIPTION
OVERWRITE OLD	Enable/Disable the option to overwrite old messages if the maximum number of messages is reached (refer to the table from ***7.3.5 Messaging*** for storage details) If the feature is **disabled** and the number of **x** messages is reached, then no new messages will be received by the handset. The user must manually delete messages in order to receive new ones.
SENT MESSAGES	Enable/Disable the option to save sent messages
DISPLAY	Enable/Disable the option to display the content of the received message in idle state.

7.3.6 Alarms

This menu allows the user to set up an alarm clock to occur at a specific time *(Fig.55)*. The sub-menus and their related options are further described in the following table.

NOTE: *This feature is not supported by RTX8431/RTX8436*

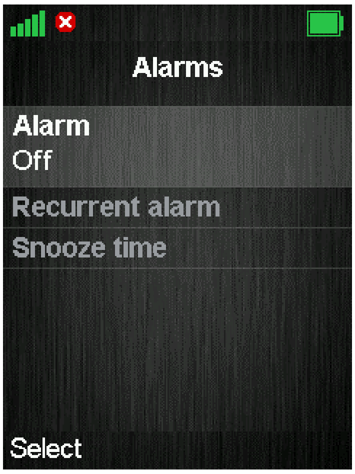


Figure 56 Alarm clock menu overview

PARAMETER	DESCRIPTION

ALARM	Schedule a one-time alarm clock. The time can be set by both the numeric keypad and the up/down navigation keys. ***NOTE:*** *The alarm will occur at the specified time.*
RECURRENT ALARM	Schedule an alarm clock for a specific time and day of the week. The alarm will then occur at the set time frame. The time can be set by both the numeric keypad and the up/down navigation keys. ***NOTE:*** *This alarm clock is different from the first one. This means that the user can set 2 different alarms – 1x recurrent and 1x one-time.*
SNOOZE TIME	Set the time for suspending the alarm after it goes off. ***NOTE:*** *The snooze time takes effect only if the Snooze button is pressed when the alarm occurs.*

7.3.7 F-keys

This menu allows the user to add certain functional numbers to the list, for example a user's mailbox or a reception desk phone number. It represents another list for contacts used to ease the user in dialing specific numbers *(Fig.56)*.

Figure 52 F-keys menu overview

When entries are present, there are three available options assigned to the softkeys – *Call* (call the highlighted entry), *Edit* (edit the entry's details) and *More* (other available options, further described in the following table).

PARAMETER	DESCRIPTION
ADD F-KEY	Provides the possibility to add a new F-key in the list by adding the name and number of the entry
EDIT F-KEY	Provides the possibility to edit the highlighted entry's details
DELETE F-KEY	Provides the possibility to delete the highlighted entry
DELETE ALL F-KEYS	Provides the possibility to delete all entries from the *F-key* list
SPEED DIAL	Provides the possibility to add an F-key to a speed dial number (2 – 9).

	NOTE: *The speed dial list stores both contacts from the Contact and F-Keys list. For additional information, please refer to* ***7.3.1 Contacts*** ***NOTE:*** *The '1' key is reserved for voicemail. Long key press on '0' is used for starting a normal dial string with '+'.* ***NOTE:*** *RTX8431/RTX8436 has 4 extra keys (F1,F2,F3 and F4) to which an F-key can be assigned as a speed dial key*

7.3.8 Alarm list

This menu stores all the received emergency alarms on the handset and lists them to the user *(Fig.57)*. The different alarm types are defined on the base station (refer to ***7.1.18 Alarm*** for further details).

NOTE: *This menu is not supported by RTX8430, RTX8431 and RTX8436*

Figure 58 Alarm list

The displayed alarms can be identified by the time they were received (date/time) and their alarm priority (high, medium and low). If selected, the detailed content of the alarm is displayed. For further details about the alarms and use case examples, please refer to Appendix ***9.4 Alarm characteristics***. For an alarm configuration guide, please refer to ***8.5 Alarms.***

NOTE: *Due to limited space on the handsets (max of 50 alarm messages), old alarms will be overwritten. This means that as soon as the handset's alarm storage is full, the new upcoming alarm will overwrite the oldest one.*

The alarm list is sorted by the following criteria:

- Read / unread (unread are displayed first)

* Priority (highest displayed first)
* Timestamp (newest are displayed first)

Depending on the alarm status there are several available options assigned to the softkeys of the highlighted item – *Select* (open the alarm), Delete (delete the alarm), OK (acknowledge the alarm) and *Reject* (reject the alarm).

STATUS ALARMS	LEFT SOFTKEY	CENTER KEY	RIGHT SOFTKEY
ACCEPTED	Select	-	Delete
NOT ACCEPTED	Select	OK	Reject

NOTE: *If the handset receives an alarm during call, a beep tone can be heard multiple times, indicating for the alarm. Then the user can decide upon action for that alarm.*

7.3.9 Central directory

This sub-menu contains the shared directory of contacts among the handsets in the system *(Fig.58)*. When enabled it can be found under Contacts in the Manu. Depending on the Central Directory mode, LDAP or Local, the contact list can be uploaded in a different way. In the case of LDAP, the list is retrieved only from a server, whereas if Local mode is used, then the list with contacts can either be retrieved from a server or uploaded by the administrator via the base (refer to ***7.1.13 Central Directory*** for more details).

Figure 59 Central directory overview

No matter the mode, the Central Directory menu displays very similar screens. The difference can be noticed when opening the menu itself:

* If LDAP is used, a search screen is displayed. The user can either search for a contact in the list, by typing a letter or the name of the contact, or continue with an "empty" search, by pressing the middle softkey "Search" without any input (to display all the contacts from the directory)
* If Local mode is enabled, the list with contacts is displayed, from which the user can browse through. The search mode can be accessed manually by pressing the middle softkey "Search".

NOTE: *If a blue/green arrow is visible on the right-hand side of the highlighted contact, this means that there is more than one number added to the contact. It is possible to browse through the different numbers that are assigned to a contact by using the right/left navigation keys.*

Depending on the sub-menu that is opened – *Search* or *List with contacts*, there are several available options assigned to the softkeys.

SUB-MENU	SOFTKEY OPTIONS	DESCRIPTION
SEARCH	Symbol	Choose among various symbols that are not present on the keypad of the handset
	Search	Search for a specific contact by typing their name or first letters
	Clear	Delete the typed data from the search field
LIST WITH CONTACTS	Line	Select the desired line from which to call the contact. The call will then be routed via that line. ***NOTE:*** *A handset should be assigned to different extensions in order to have a list with line options (refer to* ***7.3.4.8 Line*** *for more details)*
	Search	Enter the search field
	Back	Return to the search field. ***NOTE:*** *This option is visible only in LDAP mode*

8 Setup guides

This chapter aims to present various setup scenarios, such as configuring a dual/multi cell system, alarms, multizone, etc., by providing a detailed step-by-step guide to the configuration process. The table below summarizes the available setup guides in this section.

NOTE: *The guides provide the general setup overview which aim to build the user's understanding of setting up the systems and using the related features. It does not provide customer specific setups.*

GUIDE	DESCRIPTION
MULTI CELL	Guided steps on how to connect the bases in a multi cell chain
DUAL CELL	Guided steps on how to connect the bases in a dual cell chain
CREATING EXTENSIONS AND ADDING HANDSETS	Guided steps on how to create extensions and attach handsets to them
EASY REGISTRATION	Guided steps on how to use the Easy registration feature
ALARMS	Guided steps on how to create, assign and initiate alarms from a handset
MULTIZONE	Guided steps on how to connect multi cell systems in zones
FIRMWARE UPGRADE (BASE, HANDSETS,REPEATERS)	Guided steps on how to upgrade the firmware of the devices
PROVISIONING	Guided steps on how to provision the base – via the server / via the user interface
CENTRAL DIRECTORY (LOCAL, LDAP AND XML)	Guided steps on how to configure the shared (central) directory with contacts among the handsets
MULTILINE	Guided steps on how to set multiple lines on one handset
DEBUGGING	Guided steps on how to debug the devices and gather debug information for RTX analysis.

8.1 Multi cell system

This sub-chapter describes how to setup a multi cell environment, by connecting bases into one system. Please, follow the steps below to create your own network.

NOTE: *It is possible to have RTX8660 and RTX8663 in the same chain, but the features of the system will be reduced to the ones of RTX8660. This means that if a user has a multi cell system with 50x RTX8663 and adds 1x RTX8660, the system will run on the RTX8660 features and exclude the extra ones from RTX8663.*

Step 1 Connect the base stations to a private network via the standard Ethernet cable (CAT 5).

Step 2 Access the first base by determining the IP address and then logging into the user interface of the device (for more details refer to ***6.1 Base station***)

Once you have been authenticated, the browser will display the front end of the SME Configuration Interface of the base (for more details refer to ***7.1.1 Web navigation***)

Step 3 To ensure a stable synchronization between the bases in the system, you must configure the time zone. For this purpose, navigate to the **Country** page and configure the *Country* and *Time server* settings (for more details refer to ***7.1.11 Country***). Press *Save and Reboot*.

TIP: To set the time server, you can press the *Time PC* button or use an internet time server (i.e. search for *pool.ntp.org* in the web and use the specific pool zone for your country. For example, for DK it is *0.dk.pool.ntp.org*)

NOTE: *Make sure that the Time server is set on the base, else the dual/multi cell feature will not work. This can be done by verifying that the Time Server IP address is still present in the field, after the reboot.*

Step 4 Navigate to the **Servers** page and click on *Add Server* to add the PBX (SIP) server information. This is the server that contains the extensions. As a generic setup, fill in the *Server Alias* and *Registrar* fields. The rest of the parameters' configuration depend on your desired setup and SIP server support (for more details refer to ***7.1.5 Servers***). Press *Save*.

NOTE: *Every service provider/customer should refer to their SIP server vendor on how to setup the PBX (SIP) server itself*

NOTE: *Disable the NAT adaption parameter if the NAT function of the SIP aware router is not enabled.*

Step 5 Navigate to the **Extensions** page and click on *Add Extension*. As a generic setup, fill in the *Line name, Extension, Authentication Username, Authentication password and Display name* fields (for more details refer to ***7.1.3 Extensions***). The rest of the parameters' configuration depend on your desired setup and the SIP server's configuration itself. Press *Save*.

NOTE: *To configure a dual/multi cell environment, it is important to create at least one extension on the base station to define the primary base. Thus, registering a handset at this point is not a requirement. Registration of devices is further explained in this guide (**8. 3 Handset registration process**), as soon as the dual/multi cell system is configured.*

Step 6 After creating an extension, you can now configure the base in multi cell mode. To do so, navigate to the **Multi cell** page. Enable the *Multi cell system* parameter and type in a value for the *System chain ID* (for more details refer to ***7.1.14 Dual/Multi cell***). Press *Save and Reboot.*

NOTE: *After reboot, on the front page of the UI will be displayed the following System information status:*

"Unchained Allowed to Join as Primary"

Step 7 Login to the base station that you wish to connect to the multi cell chain.

Step 8 Navigate to the **Multi cell** page, enable the *Multi cell system* parameter and enter the *System chain ID* that you previously typed for the primary base station (refer to **step 6**). Press *Save and Reboot.*

After reboot, the base will try to find and synchronize to the primary base station.

NOTE: *It takes up to 5 minutes (synchronization time) to add a new base station to a multi cell system.*

NOTE: *Depending on the base's role, the System information status will display **Primary** or **Secondary** on the front page of the UI, as soon as the multi cell system is ready.*

To add more base stations, repeat **Step 7-8.**

Step 9 Register your handsets by following the steps from section ***8. 3 Handset registration process***

8.2 Dual cell system

This sub-chapter describes how to setup a dual cell environment by connecting two bases into one system. Please, follow the steps below to create your own network.

Step 1 Set up the primary base by following **step 1-5** from the previous section ***8.1 Multi cell system.***

Step 2 The RTX9431 bases follow the Plug-n-Play concept, meaning that the dual cell feature is enabled by default and allows the second base to connect without any extra human effort. (To verify this, you can navigate to the **Dual cell** page and ensure that the *Dual cell system* parameter is enabled).

To add the secondary base to the primary, simply connect the second base station to a private network via the standard Ethernet cable (CAT 5). After a while, it will automatically connect to the primary base.

NOTE: *It may take up to 5 minutes (synchronization time) to add the base station to the dual cell system*

NOTE: *Depending on the base's role, the System information status will display* ***Primary*** *or* ***Secondary*** *on the front page of the UI, as soon as the dual cell system is ready.*

Step 3 Register your handsets by following the steps from section ***8.3 Handset registration process***.

8.3 Replacing Primary Base station in a Dual cell system

Here is a step-by-step description of how to replace the Primary Base Station in your system in the cases when that is necessary.

Step 1 Make sure to wait long enough for the Dual cell to reopen for registration.

Normally connection between bases is verified every 30 seconds. If verification fails, the status on the Dual cell tab on the web UI will show the message: "***Connection lost!***".

When the connection has been down for a defined amount of time (default is 15 minutes, but it can be configured. See ***8.3.1. Timeout configuration***.) the base will switch to *Ready for replacement* mode. Then the following message appears in the web UI of the functioning base, under System Information: *"**Replace the other base**"*. At this point the Dual cell is open for new registrations, and we can replace the missing base.

NOTE: *The replacement base needs to be in factory mode. If you are using a base that has been registered on another system before, make sure to factory reset it beforehand. (Refer to 4.3.2 for more information on resetting base stations.)*

NOTE: *All handset registrations are locked to the data primary base, meaning that if we remove it all handsets will need to be reregistered. If the primary base is the one failing to connect, after a predefined amount of time all registrations will be transferred to the secondary base and the system will reopen for the possibility of a new base joining.*

NOTE: *In cases when specific parameters are different from the default ones, those has to configured before adding the base to the chain. This might be network parameters, for example if static IP or VLAN is used. In case the peer-to-peer Data Sync feature is used, it is necessary to enable that on the replacement base beforehand and configure the IP of the remaining base.*

Step 2 Connect the new base to the network and wait for it to join the chain.

The default behavior of bases in factory mode is to attempt to join any open system, therefore the new base should connect to the reopened for registrations Dual cell system automatically. When the new base registers successfully to the system it should be visible in the Dual cell tab on the web UI. It is advisable to make sure the MAC address matches the new replacement base.

Once the base has been added to the chain, the user can choose which base to be assigned as primary and which as secondary. The primary base will take over all the handset registrations.

NOTE: *The system remembers the MAC address of the primary base. That is why it is very important when replacing a primary to replace it with a new base (with another mac address). If for some reason user tries to reset the primary base and add it again to the system as primary, that will create conflicts and will disturb the system.*

8.3.1 Timeout configuration

The time that the system waits before it reopens for registrations can be defined from the Dual cell tab in the web UI of the base. Under the field ***Base registration timeout***, you can type a value between 15 and 255 (in minutes). After that press **Save** so the system can save the changes.

8.4 Handset registration process

The following section provides a detailed guide on how to register your handset to the base station. The process consists of first, enabling handset registration mode on the base, and then registering the device itself. The UI of the **Extensions** page is different between the bases. Therefore, depending on the type of base that is used, the handset registration steps vary. Please follow the steps accordingly:

8.4.1 Enable registration mode for RTX8663.

Step 1 Login to the base and navigate to the **Extensions** page.
Step 2 Create an extension by following **Step 5** from the previous section ***8.1 Multi cell system.***

NOTE: *If extensions are already present, you can skip this step*

Step 3 Select the check box of the extension which you wish to register the device to, and then click on *Register Handset (s).*

The base now displays **Enabled** in the *Handset State* column and is open for handset registration for 5 min.

TIP: If you wish to register more handsets, click on **Check All** and then **Register Handset (s)**. This will enable registration mode for all extensions, meaning that you will only need to follow the steps from ***8.3.3 Register a handset*** to add the handsets to the base system.

8.4.1.1 Setting up multiline

Step 1 Login to the base and navigate to **Extensions** page.
Step 2 Create an extension by following **Step 5** from the previous section ***8.1 Multi cell system.***

NOTE: *If extensions are already present, you can skip this step*

Step 3 Navigate to the **Extensions** page and click on *Add Extension*. As a generic setup, fill in the *Line name, Extension, Authentication Username, Authentication password and Display name* fields (for more details refer to ***7.1.3 Extensions***). In the field *Handset*, instead of *New Handset* use the drop-down menu to choose the existing handset you want to assign the new line to. The rest of the parameters' configuration depend on your desired setup and the SIP server's configuration itself. Press *Save*.

Step 4 The extension will now show in the extension list with the same Idx and IPEI as the handset selected.

NOTE: *If the handset has been already registered, it has to be rebooted before the changes to take effect.*

8.4.2 Enable registration mode for RTX9431.

To enable registration mode on the RTX9431, you first need to create slots for the handsets and then enable handset registration.

8.4.2.1 Create handset slots.

Step 1 Login to the base and navigate to the **Extensions** page**.**

There are two ways which you can use to create handset slots.

a. Via the **Extensions** page (while adding an extension)
   i. Create an extension by following **Step 5** from the previous section ***8.1 Multi cell system.***

NOTE: *If an extension is already present and you wish to assign a handset to it, click on the extension to open the editor's window.*

   ii. On the right-hand side select the check box for adding a new handset. Press *Save*

b. Via the **Handset** tab page
   i. Click on the **Handset** tab to enter the **Handset** page
   ii. Click on **Add Handset (s)**. A new window will open. Click on **Save**

8.4.2.2 Setting up multiline

Setting up a multiline means assigning more than one extension to a handset. To register a handset to a system with an active multiline feature follow the steps below:

Step 1 Register handset by following the steps described in

Step 2 Add a multiline to a handset by creating a new extension, but instead of **Add Handset** in the **Select Handset(s)** menu, select the existing handset that you want to add a multiline to.

Step 3 The extension will now show in the **Extension** list with a new Idx but the same IPEI. In the **Handset tab** we can see the Idx will be the same and the **Extension** column will show the available extensions for this handset.

NOTE: *If the handset has been already registered, it has to be rebooted before the changes to take effect.*

The handset will now have two numbers. When making a call the user can chose which **Line** to call from. The way that happens depends on the software version of your handset.

One handset will be able to support up to 4 lines, but only 2 call appearances. The limitation of maximum 20 handsets in the Single cell system is maintained, and the maximum number of SIP registrations that the base station can handle, is maintained. With 4 lines per handset maximum number of devices registered in the system is 5. With 1 line per handset maximum number of devices registered in a single cell system is 20.

8.4.2.3 *Enable handset registration.*

Follow the steps below to enable handset registration mode:

Step 1 On the **Extensions** page, click on the **Handset** tab to enter the **Handset** page.
Step 2 Select the check box of the handset slot which you wish to register the device to, and then click on *Register Handset (s).*

The base now displays **Enabled** in the *Handset State* column and is open for handset registration for 5 min.

TIP: If you wish to register more handsets, click on **Check All** and then **Register Handset (s)**. This will enable registration mode for all handset slots, meaning that you will only need to follow the steps from ***8.3.3 Register a handset*** to add the handsets to the base system.

8.4.3 Register a handset.

After enabling the handset registration mode, it is now possible to register the device.

Step 1 Enter the **menu** of the handset – **Connectivity – Register – Select** and type in the *Access Code* of the base.

NOTE: The Access code (AC) is used to allow the handset to register to the base station. By default, the value is 0000, but the administrator can change the AC to another numeric value. This can be done via the **Extensions** page on the base station (See ***7.1.3 Extensions*** for more details).

The handset indicates that it is trying to register to the base by displaying **Registering** in idle screen.

Step 2 As soon as the handset registers, the extension number will appear on the handset and the base station will display the IPEI number of the device on the **Extensions/Handset** page.

NOTE: *To verify that the handset has been registered to the base, you need to refresh the* ***Extensions / Handset*** *page, as it will not update automatically.*

Step 3 Verify that the handset is SIP registered by finding a **SIP Registered** state under the *State* column.

8.4.4 Easy registration

The RTX9431 device is currently the only base station that supports the new feature “Easy registration”. It enables the user to add new handsets and repeaters without having to open the base webpage. This means, that the user can physically enable the registration mode by rebooting the base station via the UI or by pressing the button (until orange LED is shown), or by using the “Easy registration” menu on the RTX843x handset.

NOTE: *To enable this feature, please contact RTX.*

NOTE: *Access code must be 0000 for the feature to work.*

8.4.4.1 First registration overview

When the administrator first unpacks the system and powers on the base(s), the base(s) will open for registration for 120 seconds. If the registration process times out, the user can easily restart the process by rebooting the base or by unplugging and plugging the power connection. When the first device has been added, the system will close for registration, and the administrator needs to follow the normal registration flow, as described in the following sections.

NOTE: *The base must be provisioned with extensions/SIP registrations to be able to add a device*

8.4.4.2 Enabling registration mode

Enabling “Easy registration” to add a new device can be done by the following two ways. In either case, remember that this will require the base to already be provisioned with extension/SIP registration data for the allowed handsets.

- **Pressing the reset button** on the base for two seconds, which will reset the base. This is only if you have not DECT registered the first handset.
- **Using the Easy Registration menu option** in the RTX843x handset. This option is found in the “Connectivity” menu of the handset.

For every successfully registered device (besides the first one), if the user acknowledges the registration success, the base will count 2 extra minutes to allow the user to add more devices. Else, if not acknowledged, the base will close the registration “window”.

Please have in mind that the “Easy registration” feature can be enabled only by devices supporting the feature and namely RTX9431 and RTX843x. Once enabled, different models of handsets can be registered.

8.4.4.3 Registering devices

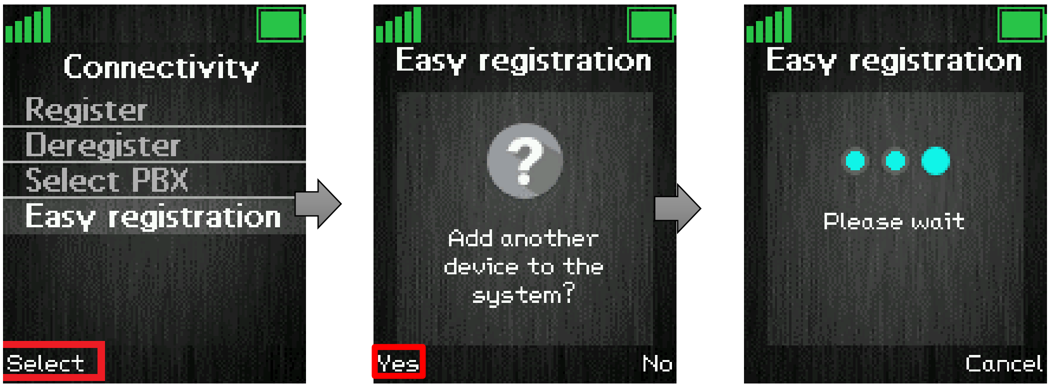
8.4.4.3.1 Handsets

As previously mentioned, once the base station is unpacked and powered on, it enters a registration mode for 120 seconds. In order to establish a connection, the base station must be already provisioned with extension/SIP registration data for the upcoming handsets/repeaters. While the base is open for registration, the LED will blink to indicate the “open” state. Meanwhile, the administrator must power on an RTX8431 handset which has no current registration. A “registering” screen will be displayed after booting of the handset is completed. The handset will automatically register to the base station unless the user presses “Cancel” to stop the registration process. If the registration process of 120 seconds has timed out, the user can easily restart the process by rebooting the base or by unplugging/plugging the power connection. As soon as the first handset has been registered, the user may continue adding handsets following the normal procedure mentioned below.

After having registered handset number one to the base station, the system is now up and running. There are two ways of registering a second handset:

1. **Brand new RTX843x -** In order to connect a second unpacked RTX843x handset, the administrator should go to the “Connectivity” menu of the already registered RTX843x device and select the “Easy registration” feature (please see the screenshots below). The system will enter the registration mode and the administrator can then power on the new handset number two. The

handset enters immediately registration mode and after a moment, it will be registered to the base station. The administrator will see a confirmation screen on both handsets number one and two, where the user can compare if the IPEI match. After pressing “Okay”, the system opens again the registration window for 120 seconds where the user can add another handset.



2. **Other handsets –** The administrator must go to the “Connectivity” menu of the already registered RTX843x device and select the “Easy registration” feature (please see the screenshots above). The system will then enter registration mode.
   To register other handset models, that do not support the “Easy registration” feature, the administrator must go to the “Connectivity” menu of the handset that must be registered, select “Register” and type in the “Access code”. After a while, the second handset will be registered.

NOTE: *If a new device is not added successfully to the system, a popup error message will be displayed, containing one of the following errors:*

- “Registration failed.”

- An error text to serve as hint to the user for solving the issue

- An error code to serve the specialized technicians with extra debug information

8.4.4.3.2 Repeaters

The feature also allows the registration of repeaters to the system. To register a repeater, enter the “Easy registration” menu from the RTX843x handset and allow new devices to the system (for more details, please see the previous section, point 2). Power on the repeater. As soon as the handset’s screen has displayed success, the user may compare the IPEI from the screen of the device with the IPEI on the repeater’s label. By doing so, the user will make sure that the correct device has been added.

NOTE: *To be able to register a repeater, it should be already provisioned/added in the base station as a device.*

8.5 Firmware/Picture upgrade procedure

All devices can be upgraded via the user interface (Firmware update page), or via provisioning. In addition to this, both HTTP and TFTP can be used to handle the upgrade process. While upgrading the handset with a new firmware/picture, the progress can be monitored on the *Extensions* page under the *FWU Progress* column. The progress is displayed in % and as soon as the file upload is done, the column will display a message “*Waiting for charger”*. The user must then place the handset in the cradle to overwrite the old firmware/picture with the new one. The handset is ready to be used as soon as it reboots.

NOTE: *The handset must not be removed from the charger until the **FWU Progress** column on the **Extensions** page displays “Done” or “Off”.*

NOTE: *If the new image is not present on the handset after the reboot, the user must reset the settings of the device (Settings – Reset settings)*

For a detailed firmware upgrade procedure, please refer to RTX’s document ***Firmware update guide. (9430-8660-HS).pdf***

8.6 Alarms

The following sub-chapter describes how to setup alarms on the handsets. Before proceeding to the configuration, it is important to note that when an alarm is generated, it can be delivered in 2 ways - either by call, or by message. Depending on the desired transmission, the configuration is different. The next sections present a step-by-step guide for setting up both scenarios.

NOTE: *Not all handsets can generate/receive alarms. Therefore, carefully review which handsets support the feature (refer to **7.1.18 Alarm** for more details)*

8.6.1 Via a message

To be able to send and receive alarms in a message format, an alarm server is needed. Therefore, before configuring the alarms on the DECT system, please make sure that an alarm server is present.

NOTE: *This guide does not cover alarm server configuration. It is up to the customer to decide which server solution to use for handling the alarms.*

As soon as an alarm server is set up, you can proceed to the alarm configuration on the DECT system. Please, follow the steps below.

Step 1 Enter the base station’s web UI and go to the **Alarm** page

Step 2 Create the desired alarms by configuring the settings on the page (for further details on the settings, please refer to ***7.1.18 Alarm***). Press **Save**

Step 3 Navigate to the **Extensions** page and select the IPEI of the handset, which you wish to assign an alarm to. The handset editor’s menu will open.

Step 4 Navigate to the *Alarm Profiles* section where the available alarms are listed (the ones you configured in **step 2**). Select the checkbox of the alarm which you wish to assign to the given handset and press **Save.**

Repeat **steps 3-4** to assign alarms to other handsets

Step 5 Go to the **Management** page and navigate to the *Text messaging* section. Enable the *Text messaging* setting, type in the alarm server address in the *Text messaging & Alarm Server* field and edit the rest of the settings (refer to ***7.1.7.5 Text messaging*** for further details on the fields). Press **Save**

Step 6 Generate an alarm to verify that it will be distributed and received by the counterpart. In case the alarms do not work, please reboot the system/devices and try again.

8.6.2 Via a call

It is possible to send and receive alarms in a call format. This means that all communication is handled by the DECT base and no message server is needed.

Step 1 Follow **steps 1-3** from the previous section (***8.5.1 Via a message***)

Step 2 Edit the *Alarm line* and *Alarm Number* fields (refer to ***7.1.4 Handsets*** for further details on the settings)

Step 3 Navigate to the *Alarm Profiles* section where the available alarms are listed (the ones you configured previously). Select the checkbox of the alarm which you wish to assign to the given handset and press **Save.**

Step 4 Generate an alarm to verify that it will be distributed and received by the counterpart. In case alarms do not work, please reboot the system/devices and try again.

9 Appendix

9.1 Contact list structure

As previously mentioned, the administrator can upload a contact list either to a selected handset or share the list as a local central directory. The supported file formats are **.csv**, **.txt** and **.xml**. This section provides details on the structure of the contact list, both in text *(Fig.59)* and xml format *(Fig. 60)*.

NOTE: *Contact name must not contain more than 23 characters and contact number must not contain more than 21 digits.*

NOTE: *When using a server to import the file (via the Central Directory page – Local), remember to set the transfer protocol via the Management page – Settings - Management Transfer Protocol*

```
File  Edit  Format  View  Help
Dennis Iversen,+4596322382
Torsten Krogh Elgaard,2381
Rune Thor Jensen,2445
Maija-Liisa Knudsen,2377
Jesper Jensen,2346
Kristian Kjaer,2447
Gitte Dyhr Petersen,2470
Sukesh Reddy,2749
Morten Fredegod,4726
Annemarie Dahl,2861
Hans Back,2721
Henrik Olsen,2733
Jens Martin Jensen,2782
Kenneth Skiveren,2363
Lars Christensen (RTX),2433
```

Figure 60 .csv or .txt format

```
File  Edit  Format  View  Help
<IPPhoneDirectory>
<DirectoryEntry>
<Name>Mark Ross</Name>
<Telephone>100</Telephone>
<Office>+4501234567894</Office>
<Mobile>+4511234567894</Mobile>
<Fax>+4521234567894</Fax>
</DirectoryEntry>
</IPPhoneDirectory>
```

Figure 53 .xml format

.txt file limitations:

- Contact name must NOT contain "**,**"
- Contact number digits must be: +0123456789
- Contact number does not support SIP-URI
- Spaces between name section "**,**" and number section are not supported.

9.2 RTX8663-RTX8660 Mixed mode

RTX8663 can be added to existing systems using RTX8660 base station. Even though the two base stations will be able to co-exist in the same multi cell setup, the system will be set to some limitations. This means that the multi cell will disable the features of RTX8663, that are not supported by RTX8660, and run-on mixed mode but with limited to RTX8660 features.

NOTE: *LAN SYNC will not work in mixed mode.*

NOTE: *RTX8660 cannot be added to an existing RTX8663 Multi-cell. Only RTX8663 can join an RTX8660 system.*

The system will display a warning message on the Home/Status page *(Fig.61)*

Figure 54 Warning mixed mode message.

9.3 ELIN and HELD support

9.3.1 National emergency numbers

COUNTRY	DEFAULT NATIONAL EMERGENCY NUMBERS
Australia	000
Austria	112, 122, 133, 144
Belarus	01, 02, 03
Belgium	112, 100, 101
Bosnia	124, 92, 93, 94
Brazil	190, 192, 193
Bulgaria	112, 150, 160, 166
Canada	911
China	110, 112, 119, 120, 122
Croatia	112, 92, 93, 94
Cyprus	112, 999
Czech	112, 150, 155, 158
Denmark	112
England	112, 999
Estonia	112, 110
Finland	112, 10022
France	112,15,17,18
Germany	112,110
Greece	112,100,166,199
Holland	112
Hong Kong	999
Hungary	112, 104, 105, 107

Iceland	112
Ireland	112,999
Israel	100, 101, 102
Italy	112,113,115,118
Kazakhstan	03
Latvia	112, 01, 02, 03
Lithuania	112, 01, 02, 03
Luxemburg	112,113
Malta	112, 191, 196, 199
New Zealand	111
Norway	112, 110, 113
Poland	112, 997, 998, 999
Portugal	112
Romania	112, 961, 962, 955, 981
Russia	112
Serbia	92, 93, 94
Slovakia	112, 150, 155, 158
Slovenia	112, 113
South Africa	107, 10111, 10177
Spain	112, 061, 062, 080, 091
Sweden	112
Switzerland	117, 118, 144
Taiwan	110, 119
Thailand	191, 199, 112, 1669, 1155, 1554, 1860
Turkey	112, 110, 155
Ukraine	118, 01, 02, 03
United States Of America	911

9.3.2 Prerequisites for ELIN & HELD support

The E911 makes use of location information extracted from LLDP for HELD data and it is used when creating HELD requests towards the RedSky server. Therefore, the network setup must support LLDP and be configured on RedSky.

ELIN must be provided from the local exchange carrier (LEC) and if an SBC is used, this must be configured for emergency calls with ELIN numbers. Whether reuse of the same ELIN should be used throughout the Enterprise is for the local IT/Security policy to decide. Having only 1 ELIN restricts the amount of simultaneous emergency call(s) to 1.

Administrator needs to populate the Location Information Server (LIS) database with a network wire map, which maps the Enterprise's network elements to civic addresses.

NOTE: *The bases do not support ELIN information obtained via LLDP*

NOTE: *The bases do not support Location by Value*

9.3.3 Planning considerations for E911 ELIN/Custom UUID

The following planning considerations must be considered when configuring E911 ELIN/Custom UUID:

- Verify the local or national regulations with regards to the number of ERLs and the maximum area in square meters allowed per Emergency Response Location (ERL).
- Ensure that the location is sufficiently specific to allow dispatched emergency services to rapidly locate the 911 caller.
- Consider assigning up to five ELINs per ERL.
- Request your phone service provider to assign you X amount of phone numbers that you will designate as ELINs.
- Ensure to maintain an updated LIS database. For example, if you change or create a new network element.
- As a general rule, subnets that resolve to corporate virtual private network (VPN) connections should be excluded from the LIS. This is because typically there is not a civic address that can be associated to the subnets that correlate to where the users are located.

9.4 Alarm characteristics

An alarm consists of the following characteristics:

- Priority
- Confirmation type
- Number of rings
- Reference number
- Text
- Status

An incoming alarm has 9 priorities, from 1-9, where Priority 1 is considered as the most important and Priority 9, the least important. The ***priority*** is first used to determine which message to display to the user, if there are more than one unread alarms present on the handset. When an alarm is displayed to the user, the alarm is grouped in 3 groups. Priority 1-3 is considered high priority, 4-6 considered medium priority and 7-9 is considered low priority. Alarms are presented to the user depending on which of the 3 groups their priority is. This means that the played melody and the used colour in the display (both configurable by the user on the handset) depend on the group.

Confirmation type is a number between 0 and 2. It indicates how much feedback communication there is from the base / handset to the message server. Confirmation type 0 confirms that the message is delivered to the base (one message back to the message server). Confirmation type 1 confirms that the message is delivered to the base + the message is delivered to the handset (two messages back to the message server). Finally, Confirmation type 2, three messages are delivered back to the server.

An alarm can be defined by the ***number of rings***, which is a number between 0 and 9. It indicates how many times the handset should play the melody that corresponds to the alarm message presented to the user. 0 means that the melody is infinity number of times.

Reference number is a unique number for each alarm and is used to identify the alarm. If an alarm should be replaced or deleted by the message server, then this number is used to identify the specific alarm.

Text is the message string presented to the user. It is a string between 1 and 160 characters.

Status is a number that indicates what to do with the received message. If this number is 10, then an already received alarm message (identified by its Reference Number) should be deleted. Otherwise, it is treated as a new message or a message to be replaced.

9.4.1 Receive alarms.

An alarm is received and stored in the handset. If the user presses the on-hook (red) button, the alarm is considered as “unread”. Else, if the user presses one of the softkeys related to the alarm (Select, OK, Reject, Delete) the alarm is considered as “read”. The alarm with highest priority, which has not been read, is displayed to the user. The number of rings is specified by the message server. If there are more unread alarms to come with the same priority, then the most recently received of them will be displayed. When the current alarm has been accepted/rejected, and there are more unhandled alarms, the handset will display next the most recent alarm with the highest priority.

The priority of different scenarios is set on the server. The colour of the alarm represents what priority the alarm has. For priority 1,2 and 3, the top bar (Alarms) is red. For priority 4,5 and 6, the top bar is yellow. For priority 7,8 and 9 (or higher), the top bar is green.

ALARM COLOR	ALARM PRIORITY
RED	High
YELLOW	Medium
GREEN	Low

NOTE: *RTX handset supports 3 different colors, but different servers might support more priorities*

When an alarm is displayed, the softkeys have the following functions:

CONFIRMATION TYPE	LEFT SOFTKEY	CENTER KEY	RIGHT SOFTKEY
0,1 (UNREAD)	Call	OK	Delete
2 (UNDREAD)	Call	OK	Reject
0,1,2 (READ)	Call	OK	Delete

If the message server has sent a number to call back, then the left softkey will display “Call”. If the left softkey is pressed, then the number is dialed.

9.4.2 Use case: Receive two alarms (priority 1, then priority 2)

An alarm of priority 1 is received and no action is performed by the user – neither OK / Delete (confirmation 0 and 1) or OK / Reject (confirmation 2) softkeys are pressed. The current state of the display might be in-call, showing the alarm or in idle.

An alarm with priority 2 is received.

The received priority 1 alarm is displayed again because it has higher priority than any other unread alarm. The number of rings is specified by the message server for the priority 1 alarm. If priority 1 alarm is already displayed and alerting, it will continue to be displayed / alerting. However, it can be heard that priority 2 alarm is received.

9.4.3 Use case: Receive two alarms (priority 2, then priority 1)

An alarm of priority 2 is received and no action is performed by the user – neither OK / Delete (confirmation 0 and 1) or OK / Reject (confirmation 2) softkeys are pressed. The current state of the display might be in-call, showing the alarm or in idle.

Afterwards, a priority 1 alarm is received.

The received priority 1 alarm is displayed, because it has higher priority than any other unread alarm. The number of rings is specified by the message server for the priority 1 alarm.

9.4.4 Use case: Receive two alarms with same priority

An alarm of any priority is received, and no action is performed by the user – neither OK / Delete (confirmation 0 and 1) or OK / Reject (confirmation 2) softkeys are pressed. The current state of the display might be in-call, showing the alarm or in idle.

Afterwards, another alarm of the same priority is received.

In this case, the last received alarm will be displayed. The number of rings is specified by the message server for the last received alarm.

9.4.5 Delete alarms

The following sections represent the various ways of deleting alarms.

9.4.6 Upon server request

The following scenario is taken into consideration:

An alarm is present, and the message server requests the alarm to be deleted. The request to delete an alarm is received. Thereafter, the alarm is removed. This means that if the alarm, which needs to be removed, is currently being displayed / alerting, it will not be displayed anymore, and it will stop ringing.

9.4.7 From the alarm list

If the user enters the Alarm list menu and deletes an alarm, the alarm will be deleted from the list and the user will remain on the menu.

It is possible to delete all types of alarms (confirmation type 0, 1 and 2) at any time.

The message server does not receive any information regarding the successful/unsuccessful delete of the alarms.

9.4.8 Replace alarms

The following scenario is taken into consideration:

An alarm is present, and the message server requests the alarm to be replaced. The old alarm is removed, and the replacing alarm is treated as if it is a new alarm. If the alarm to be replaced is the one currently alerting, then the alerting will stop. The unread alarm with the highest priority (lowest number) will start alerting – this might be the same alarm or a different one.

9.4.9 Receiving an alarm while handset is not in idle

If an alarm is received during a call, a silent beep will be heard, which will not interrupt the conversation.

If the handset is not in idle and receives an alarm, the device will treat the alarm as if the handset is in idle. This means that the alarm will pop-up on the screen. For example, if the user is composing a normal SMS, then the SMS will be stored as a draft before the alarm is handled.

9.5 Provisioning

For a quick guide on the provisioning process, please refer to ***7.1.22 Configuration***.

9.6 Calls Operations

In this section the different types of calls and call handling will be described.

9.6.1 Incoming Call

As the handset receives an incoming call an alerting screen will be displayed. This will be accompanied by a ring signal and/or a vibrating handset according to the respective settings on the handset.

The calling party's handset number will be shown in the middle of the display, and an additional name is displayed if the calling party's handset number is stored in the local phonebook.

When in active call some handset models show small icons right under the *Status bar* that visualize if the call is incoming, outgoing or on hold.

NOTE: *In case of multiline feature being used some handset models display the line used in the top right corner. The same feature also shows in different tabs the calls on hold and the active call and allows the user to switch between the tabs. Because of the difference in screen size between the models, that feature is not available on all models.*

9.6.1.1 Answering an incoming call

To answer an incoming call, press the green "Off-hook" button, or press the "Loudspeaker" key to answer the call in a speaker mode.

NOTE: *When "Auto answer" is set to "Automatic" an incoming call is answered automatically after one ring signal. If "Auto answer" is set to "Any key" an incoming call can be answered by pressing any key on the keypad, and not just the green "Off-hook" button.*

9.6.1.2 Reject an incoming call

It is possible to reject an incoming call by pressing the right softkey "Reject" when receiving an incoming call. This way the calling party will receive a busy tone, and "User busy" displayed on the screen.

NOTE: *An incoming call can also be rejected by pressing the red "On-hook" key.*

9.6.1.3 Silencing an incoming call
When the handset is alerting it is possible to press the left softkey "Silent" whereas the handset will continue to alert, but without any heard ringing.

9.6.1.4 Ending a call
To end an ongoing call, press the red "On-hook" key, and the call will be terminated. The duration of the call is shown in the display. Furthermore, this duration time can also be viewed from the call list where the call is stored.

NOTE: *If one call is active and another call is on hold, then the active call will be terminated when pressing the red "On-hook" button, and the handset automatically returns to the call on hold, i.e., the call on hold is now active, and the user must press the red "On-hook" button again to terminate this call.*

9.6.2 Initiating a call
A call can be initiated from different locations on the handset:

- By entering the number directly from idle
- From the "Contacts" menu
- From the "Calls" menu
- From the "Central Directory"

How to initiate a call from these different directories is explained in the following sub sections.

9.6.2.1 Initiating a call from Idle
When initiating a call, it must be done according to the following:

Step 1 Enter the number of the person you want to call when the handset is in idle.
Step 2 Press the green "Off-hook" button to call the number.
Step 3 The handset will show "Calling" on the screen and the number we are calling to.
Step 4 When the counterpart answers the call, the screen will show "Connected", and the two parts are now in call.

9.6.2.2 Line call
If the handset is used in a multiline environment, it is possible to make a call from different lines by pressing the middle softkey "Line" and choose the line according to which system the call should be made from.

9.6.2.3 Initiating a call from Contacts
To initiate a call from the contact list, there must be contacts added to the local phonebook. When this prerequisite is met a local contact can be called by highlighting the contact in the contact list, and then pressing the left softkey "Call", or pressing the green "Off-hook" button.

NOTE: *The list of local contacts can be reached from the menu "Contacts" or directly from idle by pressing the left softkey when in idle.*

9.6.2.4 Initiating a call from Calls

The number of an outgoing, incoming, or missed call can be called directly from the call list by highlighting the number and pressing the left softkey “Call” or the green “Off-hook” button to call the entry.

NOTE: *The “Calls” list can be reached from the menu “Calls” or directly from idle by pressing either the green “Off-hook” button or the right softkey when in idle.*

9.6.3 Call Holding/Retrieving

It is possible to hold a call which means that the one party of the call is put on hold and cannot hear the other party but can only hear a repeated short tone on hold until the call is retrieved.

Step 1 To put the counterpart of the call on hold press the left softkey “Hold”, and this party is put on hold.

Step 2 To retrieve the call, and thereby continue the conversation, press left softkey “Retrieve” which is only showed on the screen when the other party of the call is put on hold.

NOTE 1: *If one call is active and another call is on hold, then the active call will be terminated when pressing the red “On-hook” button, and the handsets automatically returns to the call on hold, i.e. the call on hold is now active, and the user must press the red “On-hook” button again to terminate this call.*
NOTE 2: *When the counterpart is put on hold it is also possible establish a second call and make a conference, swap or transfer which will be explained in the following sections.*

9.6.4 Attended and Unattended Transfer of a Call

When in an ongoing call, it is possible for either party to transfer the call, either attended or unattended, to another handset.

9.6.4.1 Attended call transfer.

In an attended transfer the party that call is transferred to, is called before the transfer is made. The procedure to do this is as follows.

Step 1 Initiate a call with a counterpart, so that the handset displays “Connected” as shown below.

Step 2 Press the right softkey “Transfer” on either of the handsets in call, and the counterpart of the call is put on hold.

Step 3 Enter the number that the transfer should be made to and press the green “Off-hook” button.

Step 4 A ring back tone is heard, and a third party is called.

Step 5 As the third party answers the call the handset user initiating the transfer will show “Connected” on the screen.

Step 6 To make the transfer after the third party has answered the call, press the right softkey “Transfer”, and the call is transferred so that the second and third party are now in call, and “Call ended” is displayed on the handset’s screen.

NOTE: *It is also possible to make an attended transfer by pressing either the left softkey “Hold”, or the middle softkey “Conf.” instead of pressing the right softkey “Transfer” in step 2.*

9.6.4.2 Unattended call transfer

When making an unattended transfer, the call will be transferred directly to a third party without the party initiating the transfer needing to call this third party before the transfer.

1. Initiate a call with a counterpart so that the handset displays “Connected”.
2. Press the right softkey “Transfer” on either of the handsets in call, and the counterpart of the call is put on hold.
3. Enter the number that the transfer should be made to, and press transfer.
4. This will end the call on the first handset party (screen will show “Call ended” message), and the second party will stop being on hold and hear a ring back tone instead which indicates that this handset is now calling the third party.
5. When the third party answers the call, an unattended transfer has been made and the second and third party are now in call.

NOTE: *As with attended transfer, it is also possible to make an unattended transfer by pressing either the left softkey “Hold”, or the middle softkey “Conf.” instead of pressing the right softkey “Transfer” in step 2.*

9.6.5 Swap between two calls

It is possible to swap between two calls. In this case, one party will be in ongoing call, and the second party will be on hold. i.e., “Swap” means that it is possible to toggle between these two parties.

Step 1 Initiate a call with a counterpart so that the handset displays “Connected”.

Step 2 Put this call on hold by pressing the left softkey “Hold”.

Step 3 Enter the number of a second party and press the green “Off-hook” button.

Step 4 A ring back tone is now heard in the handset as the second party is alerting. As the second party answers the call a “Connected” screen as above is displayed.

Step 5 By pressing the left softkey “Swap” the ongoing call is now between the first party whereas the second party is now put on hold.

Step 6 From this point, it is possible to toggle between the two parties by pressing “Swap”. For each time that “Swap” is pressed a toggle is made between the two external parties, and the “Connected” screen shows the phone numbers to indicate if it is the first or second call that is ongoing.

9.6.6 Call waiting

When two parties are in call it is possible to receive a call waiting on one of the handsets which means that a third-party calls one of the two parties already in call. During a call waiting two short beeps are heard every three seconds in the handset receiving the call waiting, and “Alerting” is displayed on the screen together with the number or name of the contact (in the case when this contact is present in the local phonebook).

It is possible to reject the call waiting with the right softkey “Reject”, and hereby continuing the ongoing call with the first party.

If the call party wants to answer the call waiting the green off-hook key should be pressed. On the screen it will be displayed “Connected” together with the number or name of contact.

NOTE: *In case of multiline feature being used some handset models display the line used in the top right corner. The same feature also shows in different tabs the calls on hold and the active call and allows the user to switch between the tabs. Because of the difference in screen size between the models, that feature is not available on all models.*

9.6.7 Conference Call

In a conference call there are always at least three parties in the same conversation. Hereby, this feature enables the user to have an ongoing call with more than one person – the parties in a conference call are called participants.

9.6.7.1 Establishing a conference call

In the following, it is described how to establish a conference call between three parties.

Step 1 Initiate a call with a counterpart so that the handset displays “Connected” on the screen.
Step 2 Press the middle softkey “Conf.” to start a conference call.
Step 3 The second party is put on hold and will hear a tone on hold.
Step 4 Enter the number of the party to invite to the conference and press the green “Off-hook” button to call this participant.
Step 5 When the called party answers the call the second party is still on hold, and the conversation is only between the third party and the first party, i.e., initiator of the conference.
Step 6 To start conversation between the three parties – i.e., a conference – press the middle softkey “Conf.”
Step 7 The “Conference” screen as seen above will then be displayed, and the three participants will be in a conference call. Thereby the conversation is between all three parties.

NOTE 1: *When using a Conference server, the procedure is slightly different from the steps above. If a Conference server is used, you need to start the conference with the second party before you press “Hold” and call the third party. If you follow the standard procedure by immediately pressing “Hold” and calling the third party without initiating first the network-based conference, the Conf key will not be present on the display.*

NOTE 2: *It is possible to invite more participants to the conference by repeating the steps 2-6 for one of the participants in the conference that has not yet invited a party to the conference. Once a participant has invited a new party to the conference it is not possible for this participant to invite new parties to the conference.*

NOTE 3: *The maximal number of participants to participate in a conference call is set according to the number of bases in a system. In a system with two bases, it should be possible to have 10 participants in a conference call.*

9.6.7.2 Terminating a conference call

When a conference call is terminated it is dependent of which participant that terminates it, if the whole conference is terminated, or if the conference is ongoing between the two participants left in the conference.

Option 1

1. The user who has started the conference presses the red “On-hook” key to terminate the conference call.

2. The whole conference is terminated, and none of the participants continue to be in call.

Option 2

1. One of the participants that have been invited to the conference presses the red "On-hook" key to terminate the conference call.
2. The call is only terminated for the participant that terminated the call
3. There is still an ongoing call between the initiator of the conference, and the participants that have not ended the call.

NOTE: *Different configurations on the PBX that the base system uses may result in different terminations of a conference call. E.g., the configuration of some PBX's could cause that the whole conference call is not terminated if the initiator of the conference presses the red "On-hook" key to terminate.*

9.7 Call Capacity

DESCRIPTION	SINGLE CELL		DUAL CELL		MULTI CELL
	9431	8663	9431	8663	8663
Number of SIP extensions per system	40	40	40	80(40+40)	1000
Number of DECT registrations per system	20	40	20	80(40+40)	1000
Voice channels per device	10	10	10	10	10
DECT slots per device	12	12	12	12	12
Number of NB calls per device	10	10	10	10	10
Number of NB calls per device with G722 in codec list (last is outgoing call)	9	9	9	9	9
Max of NB per system	10	10	20	20	2000
G722 per device	5	5	5	5	5
G722 per system	5	5	10	10	1000
G729 internal calls per system	4	4	8	8	1000
G729 BF(external DSP) calls per system	N/A	10	N/A	20	2000/2000
BV32 BF(external DSP) calls per system	N/A	8	N/A	16	2000
Opus (Silk only, 16kHz, 20ms packets. TOC-byte=0x48) BF (external DSP) calls per system	2	4	4	8	1000
Number of repeaters supported	6	3	12	6	100 (max 100 repeaters per system)

Number of calls repeater NB/WB	5/2	5/2	5/2	5/2	5/2
Number of calls with chaining repeaters	4/2	4/2	4/2	4/2	4/2

NOTE: *No matter which mode (single-, dual- or multi cell):*

RTX8663 supports up to:

- 3 repeaters pr. base station (depending on the System Combination setting on the base station (Number of base stations/repeaters per base station), which can be set to 50/3, 127/1 or 254/0)

- 3 repeaters in one daisy chain (also depending on the System Combination setting)

Any daisy chaining combination within these two limits is possible.

RTX9431 supports up to:

- 6 repeaters pr. base station (The System Combination setting does not exist for M400)

- 3 repeaters in one daisy chain

Any daisy chaining combination within these two limits is possible, i.e.:

- *6 chains of 1 repeater*
- *3 chains of 2 repeaters*
- *2 chains of 3 repeaters*
- *1 chain with 3 repeaters, 1 chain with 2 repeaters and one chain with only 1 repeater.*
- *Etc.*

NOTE: *When* ***Call waiting*** *feature is enabled, it reserves resources, and the number of available slots is reduced. That will affect the possible call capacity.*