SME VoIP System Guide for Rove B4

Installation & Configuration
Network Deployment
Operation & Management

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1   About This Document

This document describes the configuration, customization, management, operation, maintenance and troubleshooting of the SME VoIP System (RTX8660 base, RTX8630 handset, RTX8430 handset, RTX8830 ruggedized handset and RTX4024 Repeater) in RTX generic mode. For customer, specific modes refer to specific customer agreements, which describe the software operational deviations from this document.

1.1   Audience

Who should read this guide? First, this guide is intended for networking professionals responsible for designing and implementing RTX based enterprise networks.
Second, network administrators and IT support personnel that need to install, configure, maintain, and monitor elements in a "live" SME VoIP network will find this document helpful. Furthermore, anyone who wishes to gain knowledge on fundamental features in the Beatus system can also benefit from this material.

1.2   When Should I Read This Guide

Read this guide before you install the core network devices of VoIP SME System and when you are ready to setup or configure SIP server, NAT aware router, advanced VLAN settings, Base stations, and multi cell setup.

This manual will enable you to set up components in your network to communicate with each other and deploy a fully functionally VoIP SME System.

1.3   Important Assumptions

This document was written with the following assumptions in mind:
1) You understand network deployment in general
2) You have working knowledge of basic TCP/IP/SIP protocols, Network Address Translation, etc...
3) A proper site survey has been performed, and the administrator have access to these plans

1.4   What's Inside This Guide

We summarize the contents of this document in the table below:

WHERE IS IT?	CONTENT	PURPOSE
CHAPTER 2	Introduction to the SME VoIP Network	To gain knowledge about the different elements in a typical SME VoIP Network
CHAPTER 3	Installation of Base station/Repeater	Considerations to remember before unwrapping and installing base units and repeaters
CHAPTER 4	Making Handsets Ready	To determine precautions to take in preparing handsets for use in the system
CHAPTER 5	SME VoIP Administration Interface	To learn about the Configuration Interface and define full meaning of various parameters needed to be set up in the system.
CHAPTER 6	Multi-Cell Setup & Management	Learn how to add servers and setup multiple bases into a multi-cell network
CHAPTER 7	Registration Management - Handsets	Learn how to register handset and extensions to Base stations

CHAPTER 8	Firmware Upgrade/Downgrade Management	Provides the procedure of how to upgrade firmware to Base stations and/or handsets and/or repeaters
CHAPTER 9	Multiline Feature	Allows the same handset to have more than one number/line
CHAPTER 10	Functionality overview	To gain detail knowledge about the system features.
APPENDIX A	Server configuration	Basic understanding of a server configuration
APPENDIX B	VLAN Setup Management	Examines how to setup VLAN in the network
APPENDIX C	Local central directory file handling	Detailed description of central directory file format and upload.
APPENDIX D	Provisioning	How to do provisioning on the 9430 single cell base station.

1.5   What's Not in This guide

This guide provides overview material on network deployment, how-to procedures, and configuration examples that will enable you to begin configuring your VoIP SME System.

It is not intended as a comprehensive reference to all detail and specific steps on how to configure other vendor specific components/devices needed to make the SME VoIP System functional. For such a reference to vendor specific devices, please contact the respective vendor for documentation.

1.6   Abbreviations

For this document, the following abbreviations hold:

DHCP: Dynamic Host Configuration Protocol
DNS: Domain Name Server
DLC: Data Link Control
HTTP(S):   Hyper Text Transfer Protocol (Secure)
(T)FTP: (Trivial) File Transfer Protocol
IOS: Internetworking Operating System
PCMA: A-law Pulse Code Modulation
PCMU: mu-law Pulse Code Modulation
PoE: Power over Ethernet
RTP: Real-time Transport Protocol
RPORT: Response Port (Refer to RFC3581 for details)
SIP: Session Initiation Protocol
SME: Small and Medium scale Enterprise
VLAN: Virtual Local Access Network
TOS: Type of Service (policy based routing)
URL: Uniform Resource Locator
UA: User Agent

1.7   References/Related Documentation

RTX8430 Handset_Manual_Operations_v5.1
RTX8431 Handset_Manual_Operations_v5.1
RTX8630 Handset_Manual_Operations_v5.1
RTX8631_Handset_Manual_Operations_v5.1
RTX8632_Handset_Manual_Operations_v5.1
RTX8633_Handset_Manual_Operations_v5.1
RTX8830_Handset_Manual_Operations_v5.1
How to Deploy SME VOIP System v1.4
Provisioning of SME VoIP System (24)

1.8 Document History

REVISION	AUTHOR	ISSUE DATE	COMMENTS
2.3	KMR	8-Sep-2014	Updated to V322 operation with RTX8830 handset

1.9 What is new

What new features have been added.

VERSION	FEATURE
V420	uaCSTA
	LDAP over SSL
	SME VoIP handset – login(for GDPR)
V430	TLS 1.2
	Secure Syslog
V440	LLDP Support
V450	Firmware update warning
	New Generic statistics
	8660 – 8663 Mixed mode
	Diagnostics Logging
V460	RTX BTLE Beacon support
V480	OPUS codec
V490	Wireshark tracing
V500	IEEE1588 – Multi-level sync
	Handset picture upload
V510	BV32
	Main page notification when PTT is enabled
	Improved Statistics page

1.10 Documentation Feedback

We always strive to produce the best and we also value your comments and suggestions about 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 Introduction – System Overview

In a typical telephony system, the network setup is the interconnection between Base stations, "fat" routers, repeaters, portable parts, etc. The backbone of the network depends on the deployment scenario, but a ring or hub topology is used. The network has centralized monitoring and maintenance system.

The system is easy to scale up and supports from 1 to 249 bases in the same network. Further it can support up to 1000 registered handsets (RTX8630, RTX8830 and RTX8430). The Small and Medium Scale Enterprise (SME) VoIP system setup is illustrated below. 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 CAT-IQ repeater RTX4024 with support up to 5 channels simultaneous call sessions.

The following figure gives a graphical overview of the architecture of the SME VoIP System:

2.1 Hardware Setup

SME network hardware setup can be deployed as follows:
Base station(s) 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 are mounted on walls or lamp poles so that each base-station is separated from each other by up to 50m indoor[1] (300m outdoor). Radio coverage can be extended using repeaters that are installed with same distance to Base station(s). 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.

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[1] Measured with European DECT radio and depends on local building layout and material

2.2   Components of SME VoIP System

RTX SME VoIP system is made up of (but not limited to) the following components:
- At least one RTX Base station is connected over an IP network and using DECT as air-core interface.
- RTX IP DECT wireless Handset.
- RTX SME VoIP Configuration Interface; is a management interface for SME VoIP Wireless Solution. It runs on all IP DECT Base stations. Each Base station has its own unique settings.

2.2.1   RTX Base stations

The Base station converts IP protocol to DECT protocol and transmits the traffic to and from the end-nodes (i.e. wireless handsets) over a channel. It has 12 available channels.
In a multi-cell setup, each Base station has:
- 8 channels have associated DSP resources for media streams.
- The remaining 4 channels are reserved for control signaling between IP Base stations and the SIP/DECT end nodes (or phones).

Base stations are grouped into clusters. Within each Cluster, Base stations are synchronized to enable a seamless handover when a user moves from one Base station coverage to another. For synchronization purposes, it is not necessary for Base stations to communicate directly with each other in the system. E.g. 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 situation that one of the Base stations fails.
The 4 control signaling channels are used to carry bearer signals that enable a handset to initiate a handover process.

2.2.2   SME VoIP Administration Server/Software

This server is referred to as SME VoIP Configuration Interface.
The SME VoIP Configuration Interface is a web-based administration page used for configuration and programming of the Base station and relevant network end-nodes. E.g. handsets can be registered or de-registered from the system using this interface.
The configuration interface can be used as a setup tool for software or firmware download to Base stations, repeaters and handsets. Further, it is used to check relevant system logs that can be useful to the administrator. These logs can be used to troubleshoot the system when the system faces unforeseen operational issues.

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

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

2.2.3   RTX Wireless Handset

The handset is a lightweight, ergonomically, and portable unit compatible with Wideband Audio (G.722), DECT, GAP standard, CAT-iq audio compliant.
The handset includes color display with graphical user interface. It can also provide the subscriber with most of the features available for a wired phone, in addition to its roaming and handover capabilities. Refer to the relevant handset manuals for full details handset features.

2.3 Wireless Bands

The bands supported in the SME VoIP are summarized as follows:
Frequency bands:
1880 – 1930 MHz (DECT)
1880 – 1900 MHz (10 carriers) Europe/ETSI
1910 – 1930 MHz (10 carriers) LATAM
1920 – 1930 MHz (5 carriers) US

2.4 System Capacity (in Summary)

SME network capacity of relevant components can be summarized as follows:

DESCRIPTION	CAPACITY
Min ## of Bases Single Cell Setup	1
Max ## of Bases in Multi-cell Setup (configurable)	50/127/254
Single/Multi Cell Setup: Max ## of Repeaters	50 bases and 3 repeaters per Base 127 bases and 1 repeater per Base 254 bases and 0 repeaters
Multi-cell Setup: Total Max ## of Repeaters	100
Max ## of Users (SIP registrations) per Base	30
Max ## of Users per SME VoIP System	limited to 1000
Multi-cell Setup: Max ## of Synchronization levels	24
Single Cell Setup: Max ## Simultaneous Calls	10* per Base station
Multi-cell Setup: Max ## of Calls	8* per Base station
Total Max ## Simultaneous Calls (Multi-cell Setup)	Limited to 1000
Repeater: Max ## of Calls (Narrow band)	5
Repeater: Max ## of Calls (G722)	2

* If G722 is in the codec list, it will reserve audio resources to be able to handle the possible G722 call. This means, that the maximum number of possible narrowband calls is reduced by one extra.

Single Cell Setup: SME telephony network composed of one Base station
Multi-cell Setup: Telephony network that consists of more than one Base station
Synchronization Level: Is the air core interface between two Base stations.

NOTE: Please note that if there are over 150 base stations in a multicell, the "Multicast" data sync should be use

2.5 Advantages of SME VoIP System

They include (but not limited to):

1. Simplicity. Integrating functionalities leads to reduced maintenance and 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 customer requirement.

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 and reduces the call setup time and packet fragmentation and aggregation delays.

3   Installation of Base stations/Repeater

After planning the network, next is to determine the proper places or location where the relevant Base stations will be installed. Therefore, we briefly describe how to install the Base station in this chapter.

3.1   Package – Contents/Damage Inspection

Before Package Is Opened:
Examine the shipping package for evidence of physical damage or mishandling prior to opening. 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.

Contents of Package:
Make sure all relevant components are available in the package before proceeding to the next step.
Every shipped base unit package/box contains the following items:

- 2 x mounting screws and 2 x Anchors
- 1 x Plastic Wall mount
- Base unit

There is a possibility of using the device in a standalone mode via the power cable (without PoE).
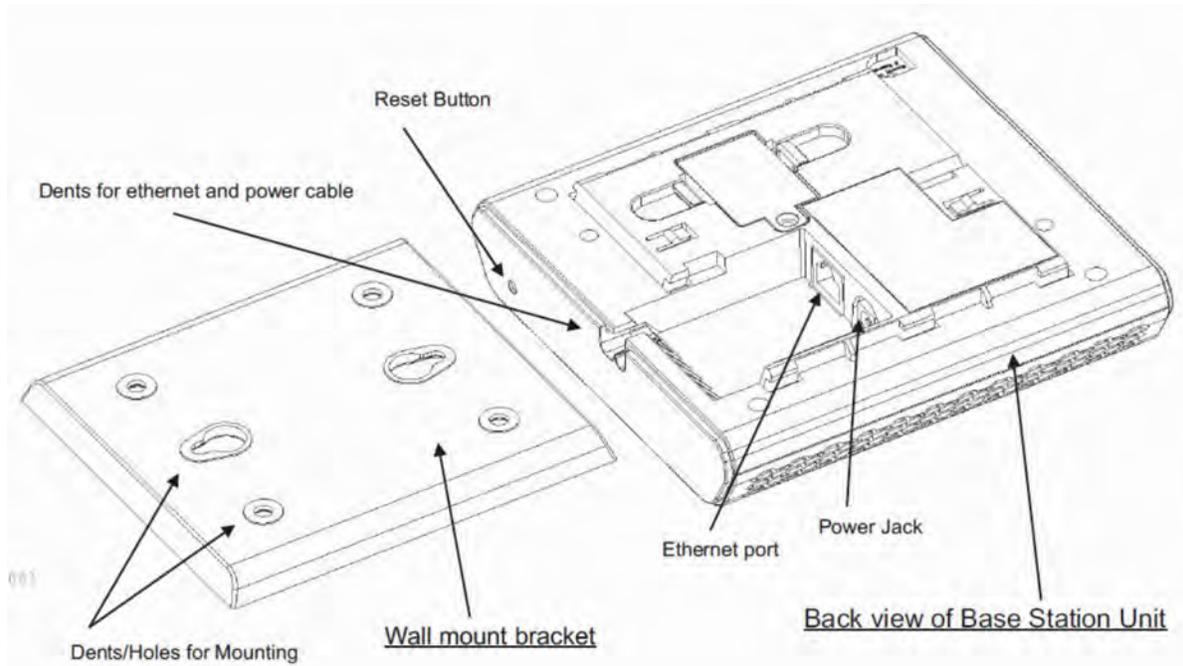
Damage Inspection:
The following are the recommended procedure for you to use for inspection:
1. Examine all relevant components for damage.
2. Make a "defective on arrival – DOA" report or RMA to the operator. Do not move the shipping carton until the operator has examined it. If possible, send pictures of the damage. The operator/regional representative will initiate the necessary procedure to process this RMA. They will guide the network administrator on how to return the damaged package if necessary.
3. If no damage is found, then unwrap all the components and dispose of empty package/carton(s) in accordance with country specific environmental regulations.

3.2   RTX Base station Mechanics

The Base station front end shows an LED indicator that signals different functional states of the base unit and occasionally of the overall network. The indicator is off when the base unit is not powered.
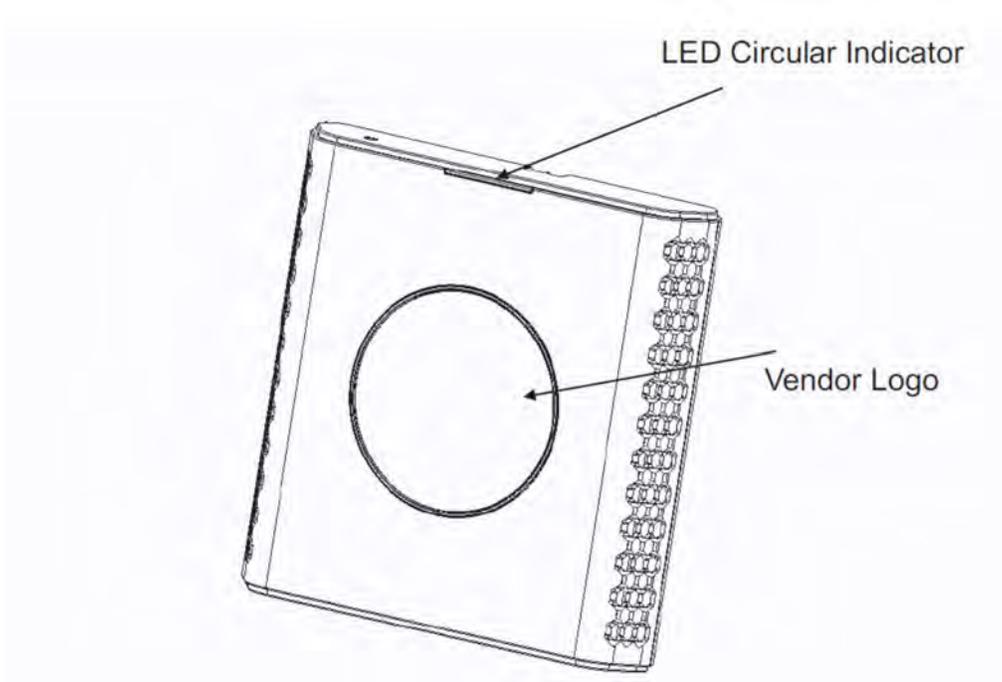
The table below summarizes the various LED states:

LED STATE	STATE
UNLIT	No power in unit
UNLIT/SOLID RED	Error condition; Unacceptable RSSI below -90dBm; Critical error
BLINKING ORANGE	Initialization, Searching for IP
BLINKING RED	Factory setting in progress, Firmware upgrade/downgrade in progress
SOLID GREEN	Ethernet connection available; Normal operation with good RSSI equal to or better than -75dBm)
SLOW BLINKING GREEN	Searching for base station(s)
BLINKING RED	Ethernet connect not available OR handset de/registration failed
SOLID ORANGE	Normal operation with pure RSSI between -75dBm and -90dBm

3.3   RTX Base Unit – Reset feature

It is possible to restart or reset the Base station unit by pressing a knob at the rear side of the unit. Alternatively, it can be reset from the SME Configuration Interface. We do not recommend this; but unplugging and plugging the Ethernet cable back to the PoE port of the Base station also resets the base unit.

3.4   Installing the Base station

First determine the best location that will provide an optimal coverage taking account the construction of the building, architecture, and choice of building materials.

Next, mount the Base station on a wall to cover range between 50 – 300 meters (i.e. 164 to 984 feet), depending whether it's an indoor or outdoor installation.

3.4.1 Mounting the Base stations/Repeaters:

We recommend the Base station be mounted an angle other than vertical on both concrete/wood/plaster pillars and walls for optimal radio coverage. Avoid mounting the base unit's upside down as it significantly reduces radio coverage.

Mount the base unit as high as possible to clear all nearby objects (e.g. office cubicles and cabinets, etc.). Occasionally extend coverage to remote offices/halls with lower telephony users by installing Repeaters.
 Make sure that when you fix the Base stations with screws, the screws do not touch the PCB on the unit. Secondly, avoid all contacts with any high voltage lines.
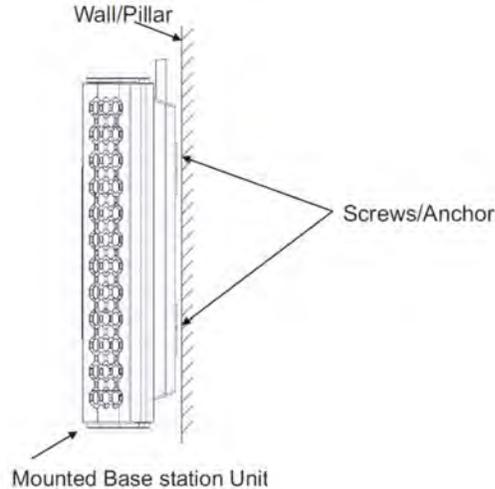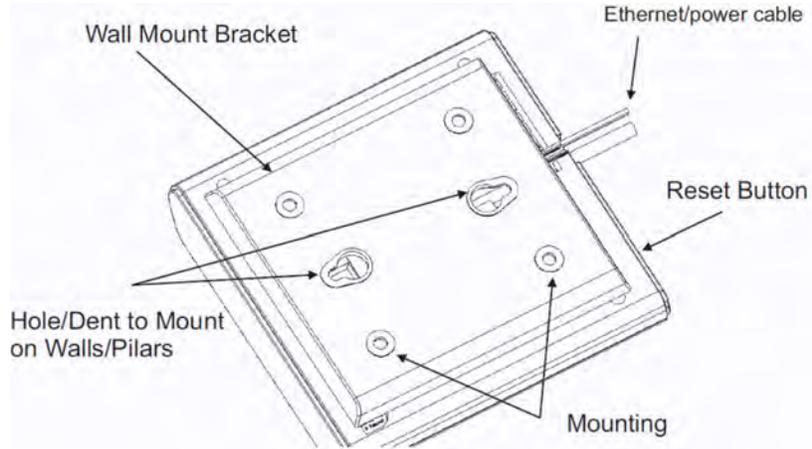
3.5 Find IP of Base station

To find IP of the installed Base station two methods can be used; Using handset Find IP feature or browser IPDECT feature.

3.5.1 Using handset Find IP feature

On the handset press "Menu" key followed by the keys: *47* to get the handset into find bases menu. The handset will now scan for 8660 bases. Depending on the amount of powered on bases with active radios and the distance to the base it can take up to minutes to find a base.

- Use the cursor down/up to select the base MAC address for the base
- The base IP address will be shown in the display

The feature is also used for deployment.

3.5.2 Using browser IPDECT

Open any standard browser and enter the address:
http://ipdect<MAC-Address-Base-Station>
for e.g. http://ipdect00087B00AA10. This will retrieve the HTTP Web Server page from the Base station with hardware address **00087B00AA10**.
This feature requires an available DNS server.

3.6 Login to Base SME Configuration Interface

STEP 1   Connect the Base station to a private network via standard Ethernet cable (CAT-5).

STEP 2   Use the IP find menu in the handset (Menu * 4 7 *) to determine the IP-address of the Base station by matching the MAC address on the back of the Base station with the MAC address list in the handset.

STEP 3   On the Login page, enter your authenticating credentials (i.e. username and password). By default, the username and password are **admin**. Click **OK** button.

STEP 4 Once you have been authenticated, the browser will display front end of the SME Configuration Interface. The front end will show relevant information of the Base station.

Screenshot:

4 Making Handset Ready

In this chapter, we briefly describe how to prepare the handset for use, install, insert and charge new batteries. Please refer to an accompanying Handset User Guide for more information of the features available in the Handset.

4.1 Package – Contents/Damage Inspection

Before Package Is Opened:
Examine the shipping package for evidence of physical damage or mishandling prior to opening. 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.

Contents of Package:
Make sure all relevant components are available in the package before proceeding to the next step.
Every shipped base unit package/box contains the following items:
- 2 x mounting screws and 2 x Anchors
- 1 x Handset hook
- 1 x A/C Adaptor
- 1 x Battery
- 1 x charger
- 1 x Handset Unit, 1 x Battery cover
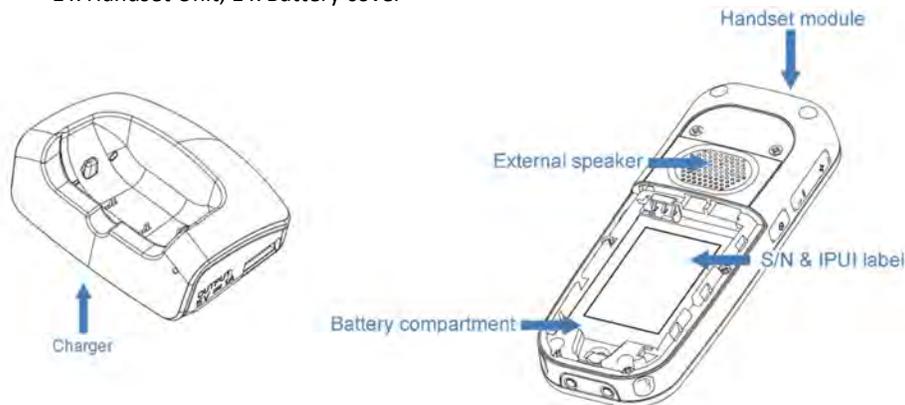
Damage Inspection:
The following are the recommended procedure for you to use for inspection:
1. Examine all relevant components for damage.
2. Make a "defective on arrival – DOA" report or RMA to the operator. Do not move the shipping carton until the operator has examined it. The operator/regional representative will initiate the necessary procedure to process this RMA. They will guide the network administrator on how to return the damaged package if necessary.
3. If no damage is found, then unwrap all the components and dispose of empty package/carton(s) in accordance with country specific environmental regulations.

4.2 Before Using the Phone

Here are the pre-cautions users should read before using the Handset:

Installing the Battery

1. Never dispose battery in fires, otherwise it will explode.
2. Never replace the batteries in potentially explosive environments, e.g. close to inflammable liquids/ gases.
3. ONLY use approved batteries and chargers from the vendor or operator.
4. Do not disassemble, customize, or short circuit the batter

Using the 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 charger Handset 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 ready for use.

Handset in the Charger
For correct charging, ensure that the room temperature is between 5°C and 25°C/41°F and 77°F. Do not place the handset in direct sunlight. The battery has a built-in heat sensor which will stop charging if the battery temperature is too high.
If the handset is turned off when placed in charger, only the LED indicates the charging. When handset is turned off, the LED flashes at a low frequency while charging and lights constantly when the charging is finished. There will be response for incoming calls.
If the handset is turned on when charging, the display shows the charging status.

Open Back Cover
1. Press down the back cover and slide it towards the bottom of the handset.
2. Remove the Back Cover from the handset
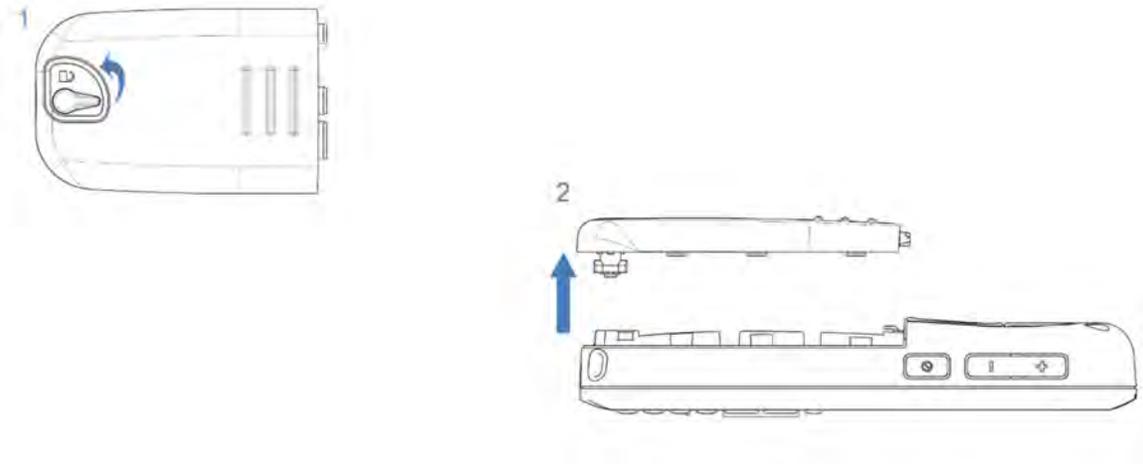
Handset Serial Number
The serial number (IPEI/IPUI number) of each handset is found either on a label, which is placed behind the battery, or on the packaging label. First, lift off handset back cover and lift the battery and read the serial number.
The serial number is needed to enable service to the handset. It must be programmed into the system database via the SME VoIP Configuration interface.

Replace Battery
Remove Back Cover from Handset. Remove the old battery and replace with a new one.

4.3 Using the Handset

Please refer to the handset manual for detailed description of how to use the handset features

5   SME VoIP Administration Interface

The SME VoIP Administration Interface is also known as SME VoIP Configuration. It is the main interface through which the system is managed and debugged.

The SME VoIP Configuration Interface is an in-built HTTP Web Server service residing in each Base station. This interface is a user-friendly interface and easy to handle even to a first-time user.

NOTE: Enabling secure web is not possible. For secure configuration, use secure provisioning. From v460 the base station supports configuration files up to 1MB

This chapter seeks to define various variables/parameters for configuration in the network, by going through the available settings of the base station. Certain pages (Extensions, Repeaters, Location and Multi cell) support an auto refresh feature, which allows the user to easily monitor the system. The pages are automatically refreshed every 5 seconds.

5.1   Web navigation

We describe the left menu in the front end of the SME VoIP Administration Interface.

Screenshot

FEATURE	DESCRIPTION
HOME/STATUS	This is the front end of the Base station's HTTP web interface. This page shows the summary of 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	Typically, the user configures the Network settings from here. **NAT provisioning**: allows configuration of features for resolving of the NAT – Network Address Translation. These features enable interoperability with most types of routers. **DHCP:** allows changes in protocol for getting a dynamic IP address. **Virtual LAN:** specifies the Virtual LAN ID and the User priority. **IP Mode:** specifies using dynamic (DHCP) or static IP address for your SME network. **IP address:** if using DHCP leave it empty. Only write in, when you use static IP address. **Subnet mask:** if using DHCP, leave it empty. Only write in, when you use static IP address. **DNS server:** specify if using DHCP, leave it empty. Only write in the DNS server address of your Internet service provider, when you use static IP address. (DNS = Dynamic Name Server) **Default gateway**: if using DHCP, leave it empty. Write in the IP address of your router, when you use static IP address.
MANAGEMENT	Defines the Configuration server address, Management transfer protocol, sizes of logs/traces that should be catalogued in the system.
FIRMWARE UPDATE	Remote firmware updates (HTTP(s)/TFTP) settings of Base stations and handsets.
LOCATION GATEWAY	Administration of Location Gateways
COUNTRY	Specifying the country/territory where the SME network is located ensures that your phone connection functions properly. Note: The base language and country setting are independent of each other. Time settings: Here the user can configure the Time server. It should be used as time server in relevant country for exact time. The time servers have to 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	Interface to common directory load of up to 3000 entries using *csv format or configuration of LDAP directory. Note: LDAP and central directory cannot operate at the same time.
MULTI CELL	Specify to connect Base station or chain of Base stations to the network. Make sure the system ID for the relevant Base stations are the same otherwise the multi-cell feature will not work.
LAN SYNC	Allows Base stations to connect over LAN PTP Sync, this makes it possible to have greater distance between the Base stations, compared to Air Sync.
REPEATERS	Administration and configuration of repeaters of 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 system and call statistics for a system.
GENERIC STATISTICS	Overview of general parameter statistics of the system
DIAGNOSTICS	Overview of Base stations and Extensions diagnostics
CONFIGURATION	This shows detail and complete SME network settings for Base station(s), HTTP/DNS/DHCP/TFTP server, SIP server, etc.
SYSLOG	Overall network related events or logs are displayed here (only live feed is shown).

SIP LOG	SIP related logs can be retrieved from URL link. It is also possible to clear logs from this feature.
LOGOUT	Login out of the web interface.

5.2   Home/Status

We describe the parameters found in the Welcome front-end home/status of the SME VoIP Administration Interface.

Screenshot:

PARAMETER	DESCRIPTION
SYSTEM INFORMATION	This base current multi-cell state
PHONE TYPE	Always IPDECT
SYSTEM TYPE	This base customer configuration
RF BAND	This base RF band setting. The parameter is defined in production and relates to the radio approvals shown on the label of the base.
CURRENT LOCAL TIME	This base local time
OPERATION TIME	Operation is operation time for the base since last reboot
RFPI-ADDRESS	This base RFPI address
MAC-ADDRESS	This base MAC address
IP-ADDRESS	This base IP address
FIRMWARE VERSION	This base firmware version
FIRMWARE URL	Firmware update server address and firmware path on server
REBOOT	Shows the last reboots of the Base station and the reason for reboot
BASE STATION STATUS	"Idle": When no calls on base "In use": When active calls on base
SIP IDENTITY STATUS	List of extensions present at this Base station. Format: "extension"@"this base IP address"("server name") followed by status to the right. Below is listed possible status: OK: Handset is ok SIP Error: SIP registration error
REBOOT	Reboot after all connections is stopped on base. Connections are active calls, directory access, firmware update active
FORCED REBOOT	Reboot immediately.

5.3   Extensions

In this section, we describe the different parameters available whenever the administrator is creating extensions for handsets. Note that it is not possible to add extensions if no servers are defined (to add a server please see chapter *6.1.2 SIP server (or PBX Server) Setup*). Furthermore, the section describes the administration of extensions and handsets using the extension list and the extension list menu.

The system can handle maximum 1000 extensions matching 1000 handsets which can be divided between servers. When 1000 handsets are registered it is not possible to add more extensions. With active multiline feature, the system can handle maximum 1000 extensions. With 4 active lines in multiline maximum 200 handsets can be active in the system.

Note: 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.

Note: To view a step-by-step setup of Extensions and handsets, please see chapter *7 Appendix – Adding Extensions*

5.3.1 Add extension

STEP 1   Click add extension

Screenshot:

Note: The "AC" parameter below the **Extensions** header is the Access code used when registering a handset. For more information, please go to chapter *7 Adding Extensions*

STEP 2   Fill in the required information

Screenshot

PARAMETER	DEFAULT VALUE(S)	DESCRIPTION
LINE NAME	Empty	Name of line shown to be used to show from which line the incoming call is coming and used when user must select from which line to make outgoing call.
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 Idx.
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 SIP server in order for this feature to function.
AUTHENTICATION USER NAME	Empty	**Username:** SIP authentication username **Permitted value(s):** 8-bit string length
AUTHENTICATION PASSWORD	Empty	**Password:** SIP authentication password. **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
XSI USERNAME	Empty	**Username:** SIP authentication username **Permitted value(s):** 8-bit string length
XSI PASSWORD	Empty	**Password:** 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 recipient later. **Valid Input(s):** 8-bit string Latin characters for the Name
MAILBOX NUMBER	Empty	Dialed mail box number by long key press on key 1. Valid Input(s): 0 – 9, *, # **Note: Mailbox Number parameter** is available only when it's enabled from SIP server.
SERVER	Server 1 IP	FQDN or IP address of SIP server. Drop down menu to select between the defined Servers of SME VoIP Service provider.
CALL WAITING FEATURE:	Enabled	Used to enable/disable Call Waiting feature. When disabled a second incoming call will be rejected. If enabled a second call will be presented as call waiting.
BROADWORKS BUSY LAMP FIELD LIST URI	Empty	The "BLF" feature on the IP phones allows a specific extension to be monitored for state changes. BLF monitors the status (busy or idle) of extensions on the IP phone Permitted value(s): URL String **Note:** This feature does not work with Group call. Therefore, Group call should be disabled.
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); that 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
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

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FORWARDING UNCONDITIONAL NUMBER	Empty	Number to which incoming calls must be re-routed to irrespective of the current state of the handset. Forwarding Unconditional must be enabled to function.
	Disabled	**Note:** Feature must 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 to when there is no response from the SIP end node.
	Disabled	Forwarding No Answer Number must be enabled to function. **Note:** Feature must be enabled in the SIP server before it can function in the network
	90	Specify delay from call to forward in seconds.
FORWARDING ON BUSY NUMBER	Empty	Number to which incoming calls must be re-routed to when SIP node is busy.
	Disabled	Forwarding on Busy Number must be enabled to function. **Note:** Feature must be enabled in the SIP server before it can function in the network
REJECT ANONYMOUS CALLS	Disabled	Calls from anonymous numbers will automatically be rejected. Enable to rejects anonymous calls

NOTE: Call forwarding can as well be configured from the handset by the user (for operation refer to the handset guide).

Screenshot

5.3.1.1   Extension list
The added extensions will be shown in the extension lists.
The list can be sorted by any of the top headlines, by mouse click on the headline link.

Screenshot

PARAMETER	DESCRIPTION
IDX	Index of handsets
IPEI	Handset IPEI. IPEI is unique DECT identification number.
HANDSET STATE	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).
HANDSET TYPE	Name of the handset type
FW INFO	Firmware version of handset
FWU PROGRESS	Possible FWU progress states: **Off:** Means sw 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**" (HS) / "**Conn. term. wait**" (Repeater): All FWU is complete and is now waiting for handset/repeater restart. Complete HS/repeater: FWU complete **Error:** Not able to fwu 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
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.

5.3.1.2 Handset and extension list top/sub-menus

The handset extension list menu is used to control paring or deletion of handset to the system (DECT registration/de-registrations) and to control SIP registration/de-registrations to the system.
Above and below the list are found commands for making operations on handsets/and extensions. The top menu is general operations, and the sub menu is always operating on selected handsets/extensions.

Screenshot

In the below table, each command is described.

ACTIONS	DESCRIPTION
ADD EXTENSION	Access to the "Add extension" sub menu
STOP REGISTRATION	Manually stop DECT registration mode of the system. This prevents any handset from registering to the system
DELETE HANDSET(S)	Deregister selected handset(s), but do not delete the extension(s).
REGISTER HANDSET(S)	Enable registration mode for the system making it possible to register at a specific extension (selected by checkbox)

DEREGISTER HANDSET(S)	Deregister the selected handset(s) and delete the extension(s).
START SIP REGISTRATION(S)	Manually start SIP registration for selected handset(s).
DELETE SIP EXTENSION(S)	Deregister the selected handset(s) and delete the extension(s).

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

5.3.2   Edit Extension

To edit an extension simply click the extension number that you want to edit.

Screenshot

Then you will see the Edit extension page

Screenshot

Now you can edit the needed information and save the changes.

For detailed description of each field please see section *5.3.1 Add extension*

5.3.3 Edit Handset

Use the mouse to click the handset IPEI link to open the handset editor window.

Screenshot

PARAMETER	DEFAULT VALUE(S)	DESCRIPTION
IPEI	Handset IPEI	Shows the handset IPEI. For an already registered handset changing the IPEI will deregister the handset at next handset location update.
PAIRED TERMINAL	No Paired Terminal	You can pair 2 terminals, e.g. RTX handset with RTX8930 headset For detailed information see RTX8930 user guide
AC	Handset AC code	Shows the handset AC code. AC code is used at handset registration. 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 handset alarm key is pressed (Long keypress > 3 seconds on navigation center key)
RECEIVE MODE	Disabled	**NOTE:** This feature is only shown if Handset have BTLE. (RTX8630 and RTX8430 is not supported)
TRANSMIT INTERVAL	Disabled	**NOTE:** This feature is only shown if Handset has BTLE. (RTX8630 and RTX8430 are not supported)
ALARM PROFILES	Not configured	Check the wanted alarm profiles for the particular handset.
SHARED CALL APPEARANCE SETTINGS	Not configured	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, that is, 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 a line, its position on the screen is simply empty.
IMPORT LOCAL PHONEBOOK		Import phonebook from csv file to this specific extension
EXPORT LOCAL PHONEBOOK		Exports this extensions phonebook as csv file NB: Home is not exported as this is considered private data.

5.3.3.1 Import local phonebook

The import local phonebook feature is using a browse file approach. After file selection press the load button 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.

Screenshot

NOTE: The local phonebook can have 100 entries for RTX863x and RTX8830 and 50 entries for RTX8430.

5.3.3.2   Export local phonebook

The Export local phonebook feature makes it possible to retrieve all contracts from a specific phone to a .CSV file.

Screenshot

Export Local Phonebook:

Export

Press the export button and save the .CSV file on you PC or Server.

5.4   Servers

In this section, we describe the different parameters available in the Servers configurations menu.
Maximum 10 servers can be configured.

Screenshot

Servers

VoIPServer:

VoIPServer:	
192.168.11.99	
Add Server	
Remove Server	

Server Alias:	VoIPServer
NAT Adaption:	Enabled
Registrar:	192.168.11.99
Outbound Proxy:	
Conference Server:	
Call Log Server:	
Reregistration time (s):	600
SIP Session Timers:	Disabled
Session Timer Value (s):	1800
SIP Transport:	UDP
Signal TCP Source Port:	Enabled
Use One TCP Connection per SIP Extension:	Disabled
RTP from own base station:	Disabled
Keep Alive:	Enabled
Show Extension on Handset Idle Screen:	Enabled
Hold Behaviour:	RFC 3264
Local Ring Back Tone:	Enabled
Remote Ring Tone Control:	Disabled
Attended Transfer Behaviour:	Hold 2nd Call
Directed Call Pickup:	Disabled
Directed Call Pickup Code:	
Group Call Pickup:	Disabled
Group Call Pickup Code:	
Use Own Codec Priority:	Disabled
DTMF Signalling:	RFC 2833
DTMF Payload Type:	101
Remote Caller ID Source Priority:	PAI - FROM
Codec Priority:	G711U / G711A / G726

Up   Down   Reset Codecs   Remove

Use ptime:	Enabled
RTP Packet Size:	20 ms
RTCP:	Enabled
Secure RTP:	Disabled
Secure RTP Auth:	Disabled
SRTP Crypto Suites:	AES_CM_128_HMAC_SHA1_32 / AES_CM_128_HMAC_SHA1_80

Up   Down   Reset Crypto Suites   Remove

Save   Cancel

PARAMETER	DEFAULT VALUE	DESCRIPTION
SERVER ALIAS	Empty	Parameter for server alias
NAT ADAPTION	Disabled	To ensure all SIP messages goes directly to the NAT gateway in the SIP aware router. If 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> **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 private NAT gateway so that SIP messages 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 empty 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 handset, conference will establish a connection to the conference server. If the field is empty, the original 3-party local conference on 8630 is used.
CALL LOG SERVER	Empty	Broadsoft call log feature. Set the IP address of the XSI call log server. In case an IP is specified pressing handset will use the call log server. If the field is empty, the local call log is used
MUSIC ON HOLD SERVER	Empty	Add the address of a server for ensuring 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 is specifies the maximum time between SIP registrations for the given SIP account. Permitted value(s): A value below 60 sec is not recommended, Maximum 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 when the timer expires the call will be terminated. Default value is 1800 s according to the RFC. Min: 90 s. Max: 65636. 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. Permitted value(s): Minimum value 90, Maximum 65636
SIP TRANSPORT	UDP	Select UDP, TCP, 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 (see 5.5.7) will not be used. The "Signal TCP Source Port" parameter specifies if the used source port shall be signaled explicitly in the SIP messages.
USE ONE TCP CONNECTION PER SIP EXTENSION:	Disabled	When using TCP as SIP transport, choose if a TCL/TLS connection shall be established for each SIP extension or if the Base station shall 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 extensions).
RTP FROM OWN BASE STATION:	Disabled	If disabled RTP stream will be send from the base, where the handset is located. By enable the RTP stream will always be send 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 extension will be shown on handset idle screen.
HOLD BEHAVIOUR	RFC 3264	Specify the hold behavior by handset 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, and the direction attribute is sent only, recvonly or inactive dependent of the context RFC 2543: The "old" way of signaling HOLD. The connection information part of the SDP is set to 0.0.0.0, and the direction attribute is sent only, recvonly or inactive dependent of the context
LOCAL RING BACK TONE	Enabled	In case the server doesn't play local ring back tone the handset will do it.
REMOTE RING TONE CONTROL	Enabled	Sometimes call distinguished ringing. It enables the server to control what ring tone that is used on the handsets.
ATTENDED TRANSFER BEHAVIOUR	Hold 2nd Call	When we have 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, we have traditionally also put the active call on hold before the SIP REFER request is sent. However, we have 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 shall 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
DIRECT CALL PICKUP	Disabled	This is Part of BroadWorks SCA feature. 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	Default disabled. 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 further down the list, the G722 will be chosen as codec for the call.
DTMF SIGNALLING	RFC 2833	Conversion of decimal digits (and '*' 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 layer 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 use 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 codes" and "Reboot chain" on the multipage to change and update handsets. **Note:** With G.722 as priority, the number of simultaneous calls per Base station will be reduced from 10 (8) to 4 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. Selections 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	With enable RTP will be encrypted (AES-128) using the key negotiated via the SDP protocol at call setup.
SECURE RTP AUTH	Disabled	With enable secure RTP is using authentication of the RTP packages. Note: with enabled SRTP authentication maximum 4 concurrent calls are possible per base in a single or multicell system.
SRTP CRYPTO SUITES	mT	Field list of supported SRTP Crypto Suites. The device is born with two suites.

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

5.5   Network

In this section, we describe the different parameters available in the network configurations menu.

Screenshot

5.5.1 IP Settings

Screenshot

IP settings

DHCP/Static IP:	DHCP
IP Address:	192.168.11.103
Subnet Mask:	255.255.255.0
Default Gateway:	192.168.11.254
DNS (Primary):	10.1.1.10
DNS (Secondary):	
MDNS:	Disabled

PARAMETER	DEFAULT VALUES	DESCRIPTION
DHCP/STATIC IP	DHCP	If DHCP is enabled, the device automatically obtains TCP/IP parameters. Possible value(s): Static, DHCP **DHCP:** IP addresses are allocated automatically from a pool of leased address. **Static IP:** the network administrator manually assigns IP addresses. If the user chooses DHCP option, the other IP settings or options are not available.
IP ADDRESS	NA	32-bit IP address of 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	Is device subnet mask. Permitted value(s): AAA.BBB.CCC.DDD This is a 32-bit combination used to describe which portion an IP address refers to the subnet and which part refers to the host. A network mask helps users know which portion of the address identifies the network and which portion of the address identifies the node.
DEFAULT GATEWAY	NA	Device's default network router/gateway (32-bit). Permitted value(s): AAA.BBB.CCC.DDD e.g. 192.168.50.0 IP address of network router that acts as entrance to another network. This device provides a default route for TCP/IP hosts to use when communicating with other hosts on hosts networks.
DNS (PRIMARY)	NA	Main server to which a device directs Domain Name System (DNS) queries. Permitted value(s): AAA.BBB.CCC.DDD or <URL> This is the IP address of server that contains mappings of DNS domain names to various data, e.g. IP address, etc. The user needs to specify this option when static IP address option is chosen.
DNS (SECONDARY)	NA	This is an alternate DNS server.
MDNS	Disabled	Enable to allow Multicast Domain Name system (MDNS)

5.5.2 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 it generates on a specific "voice VLAN" using the IEEE 802.1q specification.

Screenshot

PARAMETER	DEFAULT VALUES	DESCRIPTION
ID	0	Is 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 (i.e. FFF) 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. 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.). **Permitted value(s):** 8 priority levels (i.e. 0 to 7)
SYNCHRONIZATION	Disabled	Default disabled. By enabled the VLAN ID is automatic synchronized between the bases in the chain. Bases will be automatic rebooted during the synchronization. **Note:** If using different VLANs, the synchronization should be disabled. The setting must be changed on each Base station due to non-synchronization between them

For further help on VLAN configuration refer to Appendix.

5.5.3 DHCP Options

Screenshot

PARAMETER	DEFAULT VALUES	DESCRIPTION
PLUG-N-PLAY	Enabled	Enabled: DHCP option 66 to automatically provide PBX IP address to base.

5.5.4 TCP Options

Screenshot

TCP Options

TCP Keep Alive Interval: 120

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

5.5.5 Discovery

Screenshot

Discovery

LLDP-MED Send: Disabled
LLDP-MED Send delay: 30
VLAN via LLDP-MED: Disabled
CDP Send: Disabled
CDP Send delay: 60

PARAMETER	DEFAULT VALUES	DESCRIPTION
LLDP-MED SEND	Disabled	If "Enabled", the BS 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 BS 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
CDP SEND DELAY	60	Define the delay between messages in seconds

5.5.6 NAT Settings

We define some options available when NAT aware routers are enabled in the network.

Screenshot

NAT Settings

Enable STUN: Disabled
STUN Server:
STUN Bindtime Determine: Enabled
STUN Bindtime Guard: 80
Enable RPORT: Disabled
Keep alive time: 90

PARAMETER	DEFAULT VALUES	DESCRIPTION
ENABLE STUN	Disabled	Enable to use STUN
STUN SERVER	NA	**Permitted value(s): AAA.BBB.CCC.DDD** (Currently only Ipv4 are supported) or **URL** (e.g.: firmware.rtx.net).
STUN BINDTIME DETERMINE	Enabled	
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	This defines the frequency of how keep-alive are sent to maintain NAT bindings. **Permitted values:** Positive integer default is 90, unit is in seconds

5.5.7   SIP/RTP Settings

These are some definitions of SIP/RTP settings:

Screenshot

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	Enable: If two sources with same SSRC, 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 I loaded.
OUTBOUND PROXY MODE	Use Always	Use Always: All outbound calls are sent to 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:** Port number default 5060.
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 use for RTP audio streaming. **Permitted values:** Port number default 50004 (depending on the setup).
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, calls not registered will be automatically rejected

5.6   Management Settings Definitions

The administrator can configure Base stations to perform some specific functions such as configuration of file transfers, firmware up/downgrades, password management, and SIP/debug logs.

Screenshot

5.6.1   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 multicell 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 e.g. 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 that should be entered in order to have access to the configuration server. **Permitted value(s):** 8-bit string length
HTTP MANAGEMENT PASSWORD	Empty	Password that should be entered in order to have 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 BS If disabled, no action will be present by pressing the button on the BS
ENABLE AUTOMATIC PREFIX	Disabled	**Disabled**: Feature off. **Enabled**: The base will add the leading digit defined in "Set Prefix for Outgoing Calls". **Enabled + fall through on * and #:** Will enable detection of * or # at the first digit of a dialed number. In case of detection the base will not complete the dialed number with a leading 0. Examples: 1: dialed number on handset * 1234 - > dialed number to the pabx *1234 2: dialed number on handset #1234 - > dialed number to the pabx #1234 3: dialed number on handset 1234 - > dialed number to the pabx 01234
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.

5.6.2 Configuration:

PARAMETER	Default value	Description
CONFIGURATION FILE DOWNLOAD	Disabled	Base Specific file: Used when configuring a single cell base Base and Multicell Specific File: Used on out of factory bases to specify VLAN and settings.
CONFIGURATION SERVER ADDRESS	Empty	Server/device that provides configuration file to Base station. **Type:** DNS or IP address **Permitted value(s): AAA.BBB.CCC.DDD or <URL>**
BASE SPECIFIC FILE	Empty	Base configuration file
MULTI CELL SPECIFIC FILE	Empty	The file name must be the chain id of the system. E.g. 00087b0a00b3.cfg **Permitted value(s): Format of file is chain ID.cfg**
AUTO RESYNC POLLING	Disabled	Enable to have the Base station look for new configuration file, with a predefined time interval
AUTO RESYNC TIME	00:00	Time when the Base station shall load the configuration file 24 hour setting
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 sec, to prevent all Base station asking for configuration fin at the same time.
DHCP CONTROLLED CONFIG SERVER	Disabled	Provisioning server options. DHCP Option 66: Look for provision file by TFTP boot up server. DHCP Custom Option: Look for provision file by custom option DHCP Custom Option & Option 66: Look for provision file by first custom option and then option 66. From v460, the Base station supports configuration files of up to 1 MB
DHCP CUSTOM OPTION	Empty	By default, 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 Default configuration file on server must follow the name: MAC.cfg IP Address: IP of server with path.

5.6.3 Text messaging:

PARAMETER	DEFAULT VALUE	DESCRIPTION
TEXT MESSAGING	Disabled	Disable/enable messaging using a Message/Alarm server Enable Without Server. With this setting handset can send messages to other handsets, which support messaging.
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 message server.
TEXT MESSAGING KEEP ALIVE (M)	30	This defines the frequency of how keep-alive are sent **Permitted values:** Positive integer, unit is in minutes
TEXT MESSAGING RESPONSE (S)	30	This defines the frequency of how response timeout **Permitted values:** Positive integer, unit is in seconds
TEXT MESSAGING TTL	0	This defines the text messaging time to live **Permitted values:** Positive integer, unit is in seconds

5.6.4 Terminal:

PARAMETER	DEFAULT VALUE	DESCRIPTION
KEEP ALIVE (M)	0	If different from "0" the handset sends a (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.

5.6.5 Syslog/SIP Log:

PARAMETER	DEFAULT VALUE	DESCRIPTION
UPLOAD OF SIP LOG	Disabled	Enable this option to save low level SIP debug messages to 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 syslog server Normal Operation: Normal operation events are logged, incoming call, outgoing calls, handset registration, DECT location, and call lost due to busy, critical system errors, general system information. System Analyze: Handset roaming, handset firmware updates status. The system analyze level also contains the messages from normal operation. Debug: Used by RTX for debug. Should not be enabled during normal operation.
TLS SECURITY	Disabled	If enabled, it uses encrypted TCP, else - UDP
SYSLOG SERVER IP ADDRESS	Empty	**Permitted value(s): AAA.BBB.CCC.DDD or <URL>**
SYSLOG SERVER PORT	514	Port number of syslog server.

There are three ways of configuring the system.

1. Manual configuration by use of the Web server in the Base station(s)
2. By use of configuration files that are uploaded from a disk via the "Configuration" page on the Web server.
3. By use of configuration files which the Base station(s) download(s) from a configuration server.

5.6.6 Location Gateway

PARAMETER	DEFAULT VALUE	DESCRIPTION
LOCATION GATEWAYS	Disabled	Enable to allow Location Gateways onto the system. When enabled "Location Gateway" menu will be shown on main menu on the left.
CONFIGURATION SERVER	Empty	**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		The time of the day that the automatic synchronization will occur (HH:MM).
AUTO RESYNC MAX DELAY (MIN)		To distribute load such that all bases 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.

5.6.7 License

PARAMETER	DEFAULT VALUE	DESCRIPTION
LICENSE KEY	None	This feature allows administrators to register RTX8930 genetic headsets to the system. License key must be obtained from authorized resellers and only license matching the systems provider code will work.

5.7 Firmware Update Definitions

On this page, the system administrator can configure how Base stations and SIP nodes upgrade/downgrade to the relevant firmware and upload startup/background picture to the handsets. Handset firmware update status can be found in the **Extensions** page and repeater firmware update status in the **Repeaters** page. Base firmware update status is found in the **Multi cell** page. For more details on how to upgrade/downgrade the firmware, or upload an image, please see *8. Appendix – Firmware Upgrade procedure*.

Screenshot

PARAMETER	DEFAULT VALUE(S)	DESCRIPTION
FIRMWARE UPDATE SERVER ADDRESS	Empty	IP address or DNS of firmware update files source **Valid Inputs:** AAA.BBB.CCC.DDD or <URL> **Example:** firmware.rtx.net or 10.10.104.41
FIRMWARE PATH	Empty	Location of firmware on server (or firmware update server path where firmware update files are located). Example: RTXFWU
TERMINAL FILE PATH	Empty	Location of image (folder where background and start up image are located). Example: Images
REQUIRED VERSION	Empty	Version of firmware to be upgraded (or downgraded) on handset, repeater, or Base station. **Valid Input(s):** 8-bit string length. E.g. 400 **Note: Value version 0** will disable firmware upgrade **Note:** Two handset types will be serial firmware upgraded. First type 8630 then type 8430.
REQUIRED BRANCH	Empty	Branch of firmware to be upgraded (or downgraded) handset, repeater or Base station. **Valid Input(s):** 8-bit string length. E.g. 01

STARTUP PICTURE	Empty	Name of the startup picture you want 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 want on the handsets when they are powered up. **NOTE:** Images have same resolution as the screen on the handset(s), this 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 Only possible for headsets.

5.7.1 Warning message when firmware upgrading

A warning message will be displayed when starting firmware upgrade.

5.8 Location Gateways

This section aims to show the user how to connect the Location Gateway to the system. For more details of the RTX8200 device, please ask for the RTX8200 Location Gateway user manual.

5.8.1 Register Location gateway

STEP 1 Allow Location Gateways on the system by enabling the "Location Gateway" parameter on the management page (Please go to 5.6.6). Press **Save**

STEP 2   Select **Add Location Gateway extension**

STEP 3   Press save and leave the IPEI: FFFFFFFFF

STEP 4   Check the box of the Location gateways that you want to add and select Register Location Gateway

STEP 5   Power on the Location Gateway and after a few seconds the device will be registered

5.9   Country/Time Settings

In this section, we describe the different parameters available in the Time Server menu.

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-cell configuration. As such it is mandatory for a multi-cell configuration. The system will not work without a time server configured.

As well the time server is used in the debug logs and for SIP traces information pages and used to determine when to check for new configuration and firmware files.

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

Press the "Time PC" button to grab the current PC time and use in the time server fields.

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

PARAMETER	DEFAULT VALUES	DESCRIPTION
SELECT COUNTRY	US/Canada	Supported countries: Australia, Belgium, Brazil, Denmark, Germany, Spain, France, Ireland, Italia, Luxembourg, Nederland, New Zealand, Norway, Portugal, Swiss, Finland, Sweden, Turkey, United Kingdom, US/Canada, Austria
STATE / REGION	NA	Only shown by country selection US/Canada, Australia, Brazil
SELECT LANGUAGE	English	Web interface language. Number of available languages: English, Dansk, Italiano, Trike, Deutsch, Portuguese, Hrvatski, Srpski, Slovenian, Nederland's, Francaise, Espanyol, Russian, Polski.
TIME SERVER	Empty	DNS name or IP address of 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) nomenclature is supported.
ALLOW BROADCAST NTP	Checked	By checked time server is used.
REFRESH TIME (H)	24	The window time in hours within which time server refreshes. **Valid Inputs:** positive integer
SET TIME ZONE BY COUNTRY/REGION	Checked	By checked country setting is used (refer to country web page).
TIME ZONE	0	Refers to local time in GMT or UTC format. **Min:** -12:00 **Max:** +13:00
SET DST BY COUNTRY/REGION	Checked	By checked country setting is used (refer to country web page).
DAYLIGHT SAVING TIME (DST)	Disabled	The system administrator can Enable or Disable DST manually. Automatic: Enter the start and stop dates if you select Automatic.
DST FIXED BY DAY	Use Month and Day of week	You determine when DST actually changes. Choose the relevant date or day of the week, etc. from the drop-down menu.
DST START MONTH	March	Month that DST begins **Valid Input(s):** Gregorian months (e.g. January, February, etc.)
DST START DATE	0	Numerical day of 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 begins
DST START DAY OF WEEK, LAST IN MONTH	Second First In Month	Specify the week that DST will actually start.
DST STOP MONTH	October	The month that DST actually stops.
DST STOP DATE	0	The numerical day of 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 week DST stops
DST STOP DAY OF WEEK LAST IN MONTH	First in Month	The week within the month that DST will turn off.

NOTE: By checked time zone and DST the parameters in web page Time will be discarded.

The following types of in-band tones are supported:
- Dial tone
- Busy tone
- Ring Back tone
- Call Waiting tone
- Re-order tone

5.10 Security

The security section is used for loading certificates, changing web authentication and configuring a secure web server.
To setup secure fwu and configuration file download, go to the **Management** menu and select HTTPs for the "Management Transfer Protocol".
SIP and RTP security are server dependent, therefore the user must use the menu option **Servers** in order to configure them.

5.10.1 Device identity

The certificate and personal key used by the base when acting as a server or when the server requires client authentication in the SSL handshake procedure.

Screenshot:

5.10.2 Trusted Server Certificates

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.

Screenshot:

5.10.3 Trusted Root Certificates

Root certificates (self-signed) trusted by the base. Used to validate received root certificates sent by the server during the SSL handshake procedure.

Screenshot:

By enabling "Use Only Trusted Certificates", 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. When "Use Only Trusted Certificates" 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.

5.10.4 Password

In the below the authentication parameters are defined.

PARAMETER	Default Values	Description
USERNAME	Admin	Can be modified to any supported character and number Maximum characters: 15
CURRENT PASSWORD	Admin	Can be modified to any supported character and number
NEW PASSWORD	Empty	Change to new password Maximum characters: 15
CONFIRM PASSWORD	Empty	Confirm password to reduce accidently wrong changes of passwords

Password valid special signs: @/|<>-_:.!?*+#
Password valid numbers: 0-9
Password valid letters: a-z and A-Z

5.10.5 Secure Web Server

This setting allows all communication with the Web Server to be encrypted.

Screenshot

PARAMETER	DEFAULT VALUES	DESCRIPTION
HTTPS	Disabled	Enable to use HTTPS for Web Server Communication.

5.11 Central Directory and LDAP

The SME VOIP system supports three types of central directories - local central directory, LDAP and XML directory.
For all directories' caller id look up is made with match for 6 digits of the phone number.

5.11.1 Local Central Directory

Select "Local" and save for local central directory.

Screenshot

PARAMETER	DEFAULT VALUES	DESCRIPTION
LOCATION	Local	Drop down menu to select between local central directory, LDAP based central directory and XML server
SERVER	Empty	The parameter is used if directory file is located on a server **Valid inputs:** aaa.bbb.ccc.ddd or <url> Refer to appendix for further details.
FILENAME	Empty	The parameter is used if directory file is located on server. Refer to appendix for further details
PHONEBOOK RELOAD INTERVAL (S)	0	The parameter is controlling the reload interface of phonebook in seconds. The feature is for automatic reload 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 default value setting 0 the reload feature is disabled.

5.11.2 Import Central Directory

The import central directory feature is using a browse file approach. After file selection press the load button 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). For further details of the central directory feature refer to appendix.

5.11.3 LDAP

Select "LDAP Server" from the "Location" parameter and wait a few seconds for the new configuration menu. Fill in the empty fields with the needed data and press **Save**.

PARAMETER	DEFAULT VALUES	DESCRIPTION
CENTRAL DIRECTORY LOCATION	LDAP Server	Drop down menu to select between local central directory and LDAP based central directory. 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	Disable	If enabled, it uses encrypted TCP, else – UDP **Note:** In most cases LDAP over TLS is running on port 636
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 criteria depend on the configuration of the LDAP server. Example of the setting is CN=Users, DC=umber, DC=loc
LDAP FILTER	Empty	LDAP Filter is used to as a search filter, e.g. setting LDAP filter to (|(givenName=%*)(sn=%*)) the IP-DECT will use this 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	Bind is the username that will be used when the IP-DECT phone connects to the server
PASSWORD	Empty	Password is the password for the LDAP Server
VIRTUAL LIST	Disabled	By enable, virtual list searching is possible
NAME	Empty	The name can be used to specify if sn+givenName or cn (common name) is return in the LDAP search results
WORK NUMBER	Empty	Work number is used to specify that LDAP attribute that will be mapped to the handset work number
HOME NUMBER	Empty	Home number is used to specify that LDAP attribute that will be mapped to the handset home number
MOBILE NUMBER	Empty	Mobile number is used to specify that LDAP attribute that will be mapped to the handset mobile number

5.11.4 Characters supported

The below table shows which characters are supported in the communication between Rove B4 and handset.

5.11.5 XML Server

Select "XML Server" server from the drop-down menu and fill in the empty fields. All types of directories are supported. Please note that only directories that are enabled are shown on the handset. If one directory is enabled, then the handset will only enter the chosen directory. If multiple directories are enabled, the option "All Search" is shown.

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 configuration is displayed when the "XML server" parameter is chosen.
SERVER	Empty	IP address of the LDAP 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

5.12 Multi-cell Parameter Definitions

NOTE: To join 2 or more Base stations in a Multi Cell system, you need to have one handset added to the system. For further details and Step-by-Step guide to a Multi Cell setup, please see *6 Appendix How-To setup a Multi cell System.*

In this section, we describe the different parameters available in the Multi-cell configurations menu.

5.12.1 Settings for this unit

Description of Settings for specific base units is as follows:

Screenshot

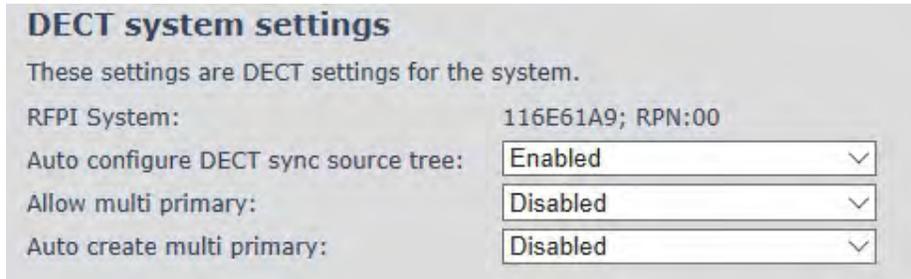Multicell status covers status of data synchronization. The status "Keep-alive" means normal operation.

PARAMETER	DEFAULT VALUES	DESCRIPTION
MULTI CELL SYSTEM	Disabled	Enable this option to allow the Base unit to be set in multi-cell mode **Valid Inputs:** Enable, Disable Must **Save and Reboot** after change from disabled to enable.
SYSTEM CHAIN ID	Empty	This is an identifier (in string format e.g. 2275) that is **unique** for a specific multi-cell system. The Chain ID value MUST NOT be equal to a used SIP account. **NOTE:** Chain ID is used as SIP account for check Sync. Default value is 512, which means 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 in SME network. Up to 24 levels of Base stations chains are permitted in a setup. **Valid Input:** The Web site allow max 5 digits in this field.
SYNCHRONIZATION TIME (S)	60 sec	This specifies the period in seconds when elements/nodes (e.g. Base units) in a specific Multi-cell will synchronize to each other. If no keep-alive packets are received within a period of 2*NETWORK_SYNC_TIME, 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 next section **DECT system settings**.
DATA SYNC:	Multicast	Select between "Multicast" or "Peer to Peer" data synchronization mode. The multicast port range and IP addresses used is calculated from the chain id. **NOTE:** Please note that if there are over 150 base stations in a multicell, the "Multicast" data sync should be used The multicast feature uses the port range: 49200 – 49999 The multicast feature IP range: 224.1.0.0 – 225.1.0.0 Multicast uses UDP. **For multi-cast 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 Base station data sync source – the base handling the data synchronization.

		Using multicast this base IP is selected automatically. The data sync feature uses the port range: 49200 – 49999 **NOTE:** 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 – as there is **no** automatic recovery of the data sync. source in Peer to Peer mode.
MULTI CELL DEBUG	None	Enable this feature, if you want the system to catalogue low level multi-cell debug information or traces. Options: **Data Sync:** Writes header information for all packets received and sent to be used to debug any special issues. Generates LOTS 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

5.12.2 DECT System Settings

Description of DECT Settings for Specific Base units is as follows:

Screenshot

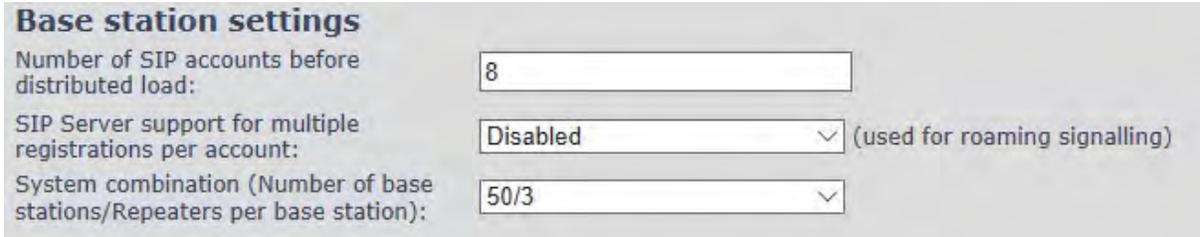| PARAMETER	DEFAULT VALUES	DESCRIPTION
DECT SYSTEM RFPI	May vary	This is a radio network identity accessed by all Base units in a specific multi-cell system. It is composed of 5 octets and has 5 different variables combined. **RFPI Format:** XX XX XX XX XX (where XX are HEX values)
AUTO CONFIGURE DECT SYNC SOURCE TREE	Enabled	Enable this to allow the system to automatically synchronize the 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 in the same system. The two cells will be unsynchronized, and handover will not be possible. "Auto Configure DECT sync source tree" must be enabled for this feature to also be enabled
AUTO CREATE MULTI PRIMARY:	Disabled	When enabled, the system will generate cells in case a base goes into faulty mode. Two cells will only be generated in case no radio connection between the two cells is present. In order to recover the full system after establishing of the faulty base, the system must be rebooted. "Allow multi primary" must also be enabled for this feature to work.

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.

5.12.3 Base station settings

Description of SIP Settings for specific Base units is as follows:

Screenshot

Parameter	Default Values	Description
NUMBER OF SIP ACCOUNTS BEFORE DISTRIBUTED LOAD	8	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. **Note:** A maximum of 8 simultaneous calls can be routed through each Base unit in a multi-cell setup. **Permitted Input:** Positive Integers (e.g. 6)
SIP SERVER SUPPORT FOR MULTIPLE REGISTRATIONS PER ACCOUNT	Disabled	Disable this option so it is 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. When a phone (from a SIP account group) initiates a handover from Base X to Base Y, this phone will de-register from Base X, and register to Base Y after a call. Permitted Input: 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 Deregistration, 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 The configuration cannot be modified after a system is established. The configuration must be set during first multicell configuration.

5.12.4 Base station Group

The Base station group list various parameter settings for Base stations including chain level information.

Screenshot:

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 network **This Unit:** Current Base station whose http Web Interface is currently being 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 multi cell network. Possible Output(s) **Primary:** Main Base station to which all other nodes in the chain synchronizes to. **Locked:** The Base unit is currently synchronized and locked to the master Base unit.