SME VoIP System Guide for RTX8660

Installation & Configuration
Network Deployment
Operation & Management

Technical Reference Document
Version 4.5

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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 setup in the system.
APPENDIX – HOW-TO SETUP A MULTI CELL SYSTEM	Multi-Cell Setup & Management	Learn how to add servers and setup multiple bases into a multi-cell network
APPENDIX – ADDING EXTENSIONS	Registration Management - Handsets	Learn how to register handset and extensions to base stations

APPENDIX – FIRMWARE UPDATE	Firmware Upgrade/Downgrade Management	Provides the procedure of how to upgrade firmware to base stations and/or handsets and/or repeaters
APPENDIX - MULTILINE	Multiline	Allows the same handset to have more then one number/line
APPENDIX – SYSTEM OVERVIEW	System Functionality Overview	To gain detail knowledge about the system features.

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
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_v4.5
RTX8630 Handset_Manual_Operations_v4.5
RTX8631_Handset_Manual_Operations_v4.5
RTX8632_Handset_Manual_Operations_v4.5
RTX8633_Handset_Manual_Operations_v4.5
RTX8830_Handset_Manual_Operations_v4.5
How to Deploy SME VOIP System v1.4
Provisioning of SME VoIP System (23)

1.8 Document History

REVISION	AUTHOR	ISSUE DATE	COMMENTS
2.3	KMR	8-Sep-2014	Updated to V322 operation with RTX8830 handset
2.4	KMR	5-Jan-2015	Aligned with V323B14 operation
2.5	KMR	16-Feb-2015	Aligned with V324 operation
2.6	KMR	12-Jan-2016	Added V355 system size capabilities
4.0	HDJ	15-01-2018	Updated to V410 Firmware
4.4	HDJ	28-09-2018	Updated to V440 Firmware
4.5	HDJ	20-03-2019	Updated to V450 Firmware

1.9 What is new

What new features have been added.

VERSION	FEATURE
V420	uaCSTA
	LDAP over SSL
	SME VoIP handset – login(for GDPR)
V440	LLDP Support
V450	Firmware update warning
	8660 – 8663 Mixed mode

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

[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 administrator. These logs can be used to troubleshoot the system when the system faces unforeseen operational issues.

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

Quick Definitions

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.

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

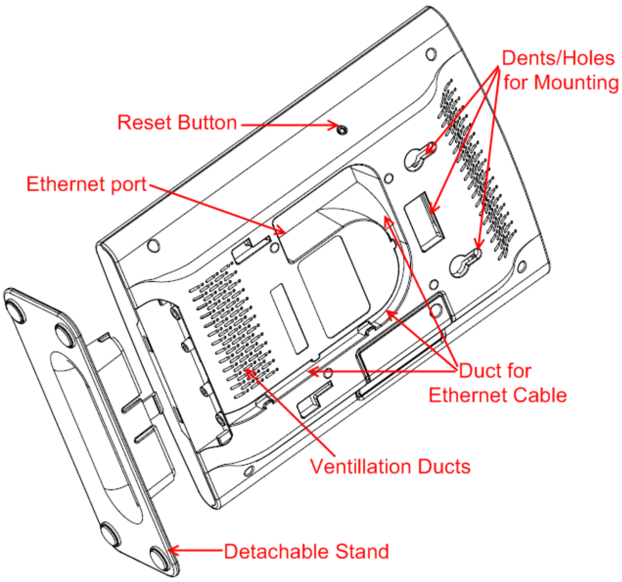
After planning the network, next is to determine the proper places or location the relevant base stations will be installed. Therefore, we briefly describe the how to install the base station in this chapter.

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 stand
- Base unit



Back view of Base Station Unit

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 it has been examined by the operator. 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
BLINKING GREEN	Initialization
SOLID RED	Factory reset warning or long press in BS reset button
BLINKING RED	Factory setting in progress
SOLID GREEN	Ethernet connection available (Normal operation)
BLINKING RED	Ethernet connect not available OR handset de/registration failed
SOLID RED	Critical error (can only be identified by RTX Engineers). Symptoms include no system/SIP debug logs are logged, etc.
ORANGE	Press reset button of base station.
BLINKING ORANGE	No IP address received

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.

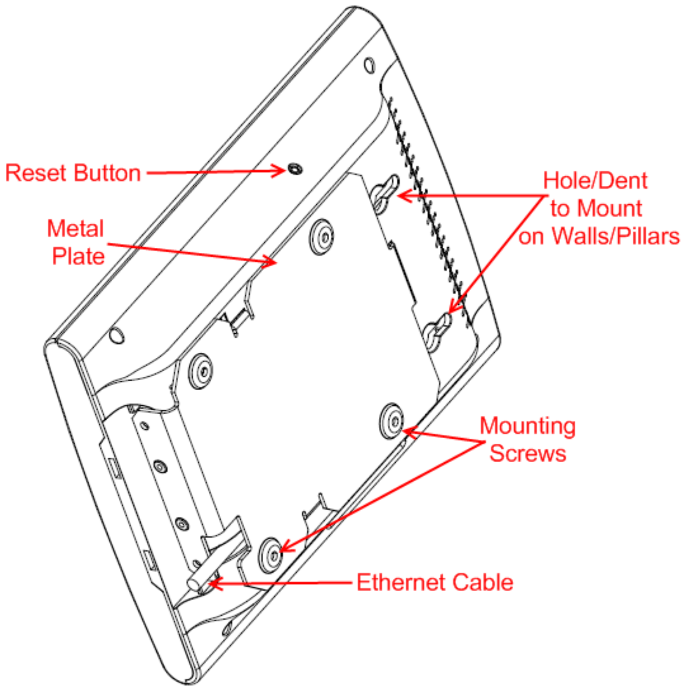
3.4 Installing the Base Station

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.

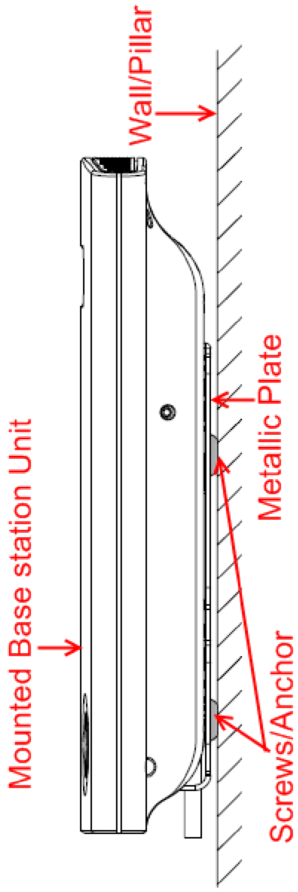
3.4.1 Mounting the Base Stations/Repeaters:

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.



Back view of Base Unit (No stand)



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

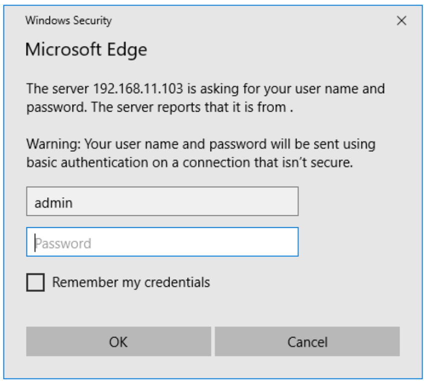
Open any standard browser and enter the address:
http://ipdect<MAC-Address-Base-Station>
for e.g. http://ipdect00087B00AA10. This will retrieve the HTTP Web Server page from the base station with hardware address **00087B00AA10**.
This feature requires an available DNS server.

3.6 Login to Base SME Configuration Interface

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

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.

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.



4. Once you have 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

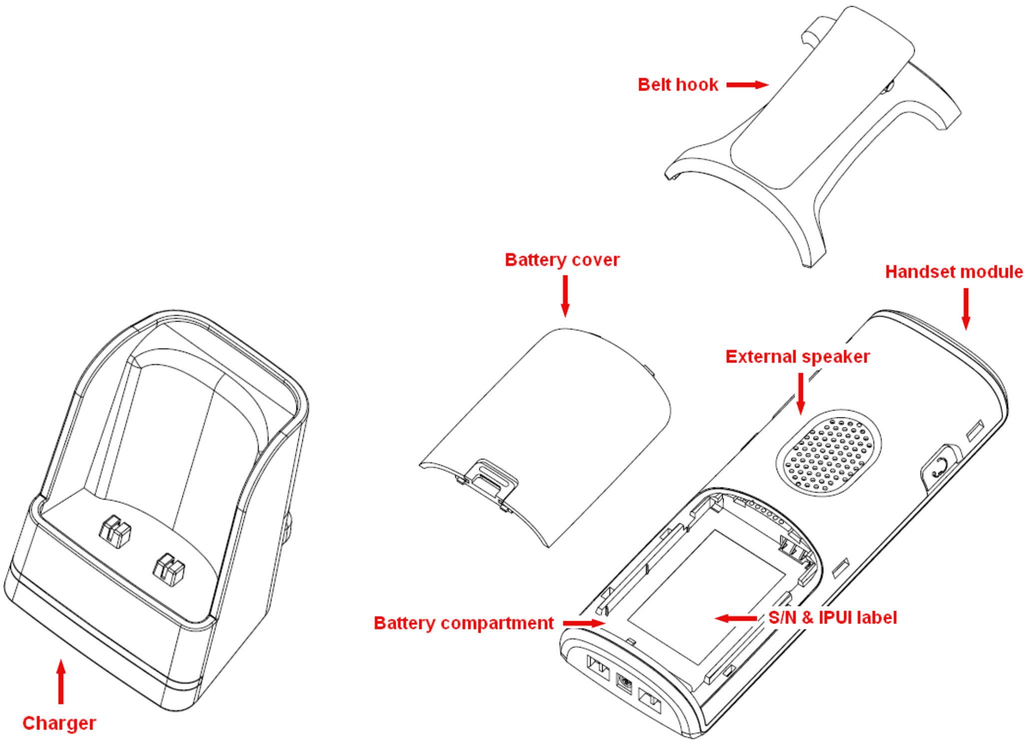
In this chapter, we briefly describe how to prepare the handset for use, install, insert and charge new batteries. Please refer to an accompanying Handset User Guide for more information of the features available in the Handset.

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

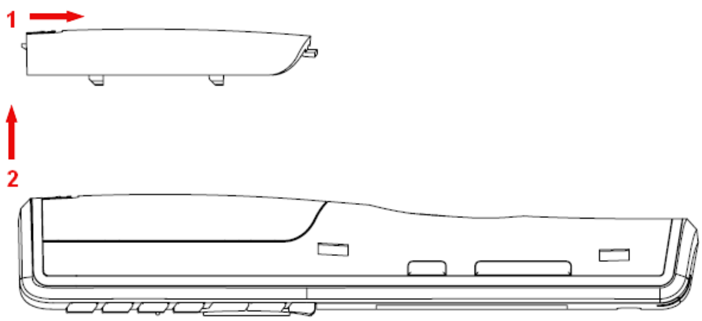
1. Never dispose battery in fires, otherwise it will explode.
2. Never replace the batteries in potentially explosive environments, e.g. close to inflammable liquids/ gases.
3. ONLY use approved batteries and chargers from the vendor or operator.
4. Do not disassemble, customize, or short circuit the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

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.

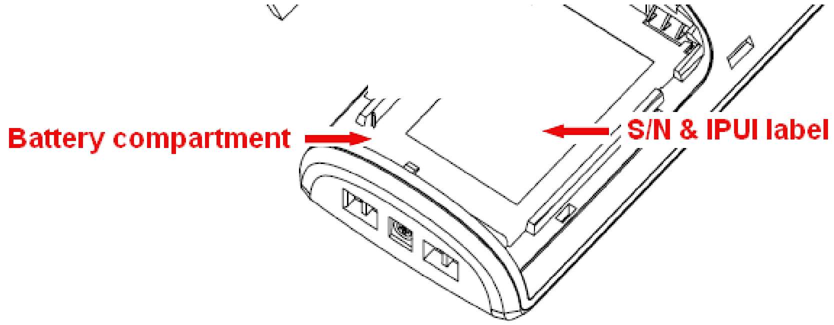
Handset in the Charger
For correct charging, ensure that the room temperature is between 5°C and 25°C/41°F and 77°F. Do not place the handset in direct sunlight. The battery has a built-in heat sensor which will stop charging if the battery temperature is too high.
If the handset is turned off when placed 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 Back Cover from 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 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.

This chapter seeks to define various variables/parameters available for configuration in the network.

5.1 Web navigation

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

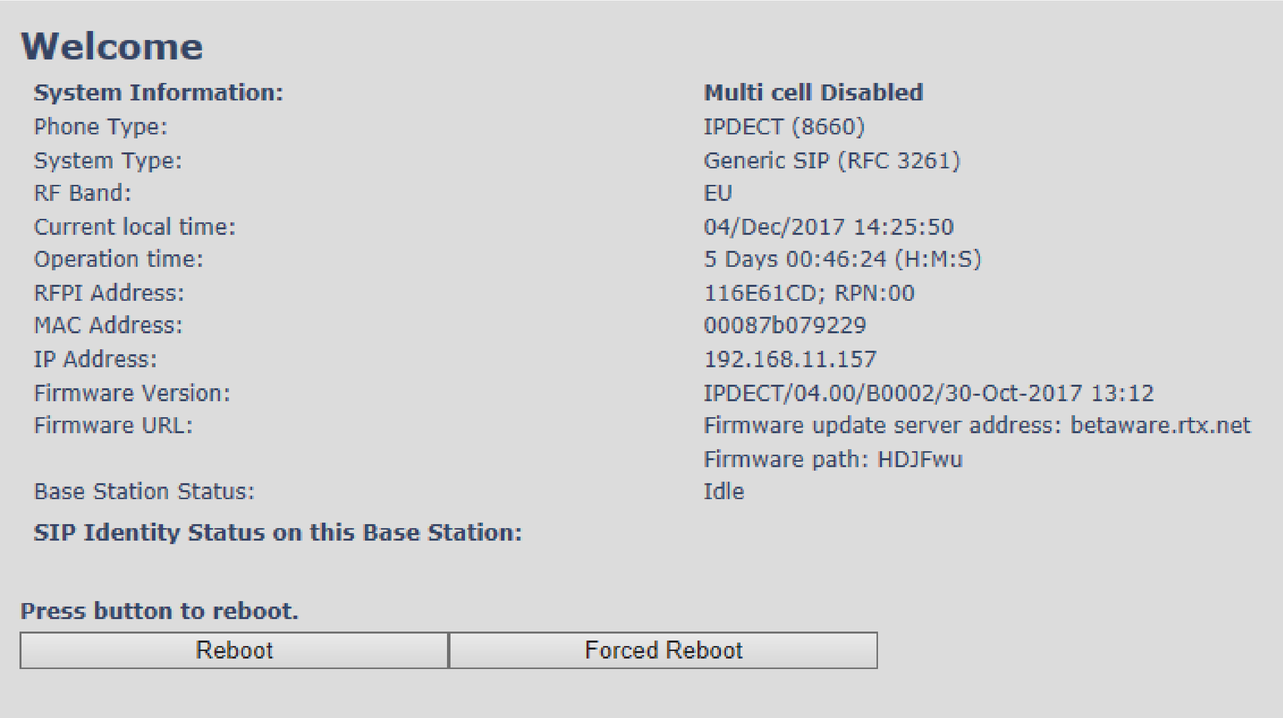
Screenshot

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.
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.
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, it is not possible to add extensions if no servers are defined. As well 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 same extension number on server 1 cannot be re-used on server 2.

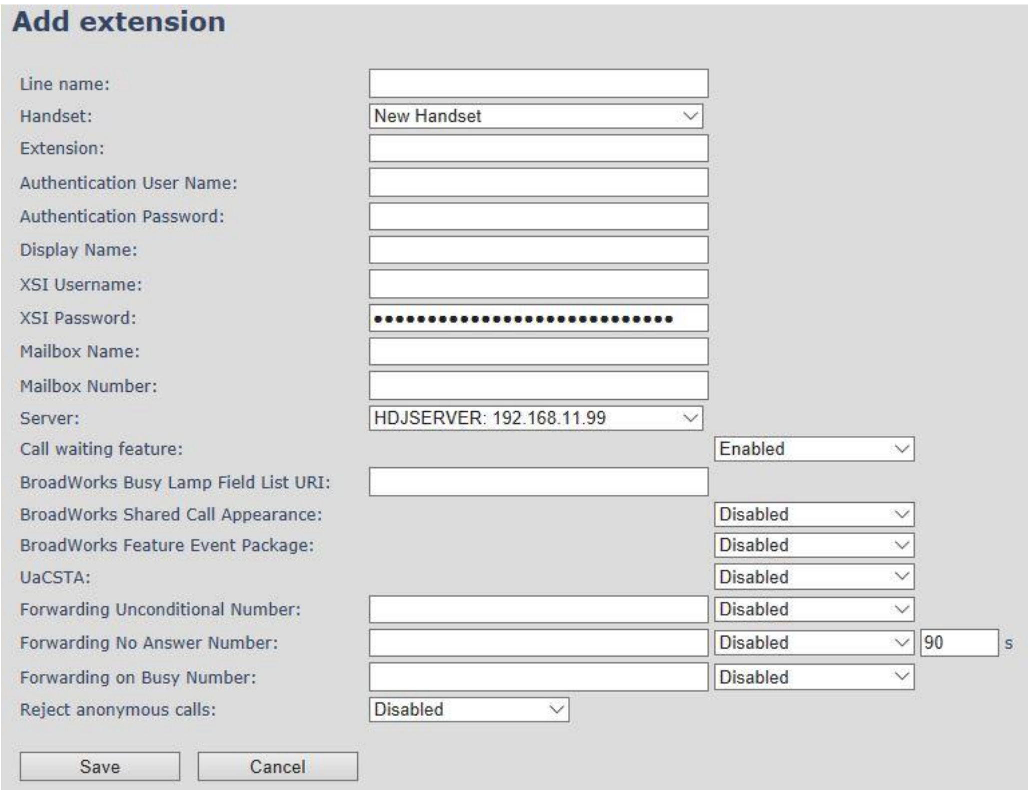
5.3.1 Add extension

Screenshot:

1. Click add extension

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
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

FORWARDING UNCONDITIONAL NUMBER	Empty Disabled	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. **Note:** Feature must be enabled in the SIP server before it can function in the network
FORWARDING NO ANSWER NUMBER	Empty Disabled 90	Number to which incoming calls must be re-routed to when there is no response from the SIP end node. 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 Specify delay from call to forward in seconds.
FORWARDING ON BUSY NUMBER	Empty Disabled	Number to which incoming calls must be re-routed to when SIP node is busy. 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).

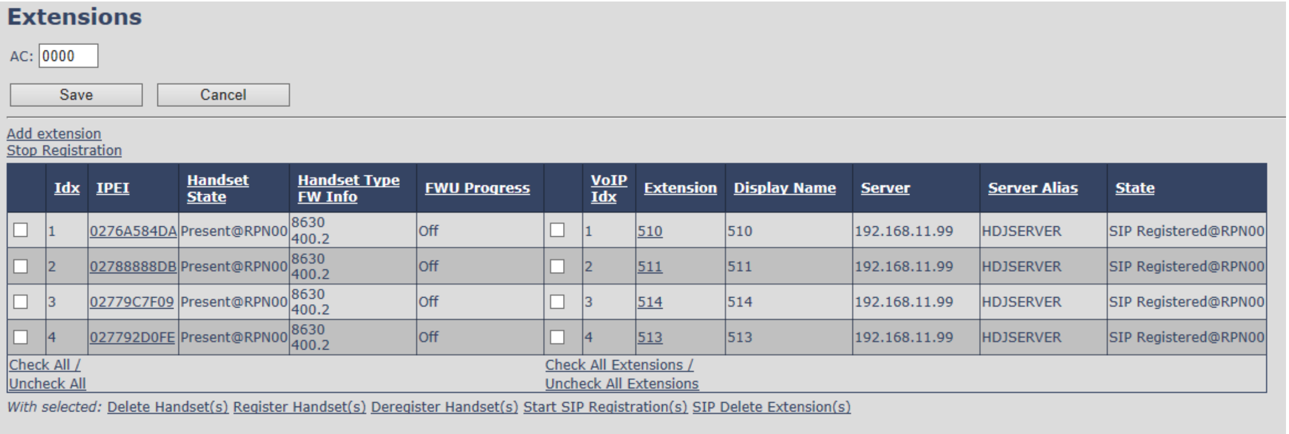
Screenshot

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



Extensions

AC: 0000

Save Cancel

Add extension
Stop Registration

	Idx	IPEI	Handset State	Handset Type FW Info	FWU Progress		VoIP Idx	Extension	Display Name	Server	Server Alias	State
☐	1	0276A584DA	Present@RPN00	8630 400.2	Off	☐	1	510	510	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	2	02788888DB	Present@RPN00	8630 400.2	Off	☐	2	511	511	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	3	02779C7F09	Present@RPN00	8630 400.2	Off	☐	3	514	514	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	4	027792D0FE	Present@RPN00	8630 400.2	Off	☐	4	513	513	192.168.11.99	HDJSERVER	SIP Registered@RPN00

Check All / Uncheck All   Check All Extensions / Uncheck All Extensions

With selected: Delete Handset(s) Register Handset(s) Deregister Handset(s) Start SIP Registration(s) SIP Delete Extension(s)

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 FW INFO	Name of the handset type 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

Add extension
Stop Registration

Check All / Uncheck All — Check All Extensions / Uncheck All Extensions

With selected: Delete Handset(s) Register Handset(s) Deregister Handset(s) Start SIP Registration(s) SIP Delete Extension(s)

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 Handset

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

Screenshot

Extensions

AC: 0000

Save Cancel

Add extension
Stop Registration

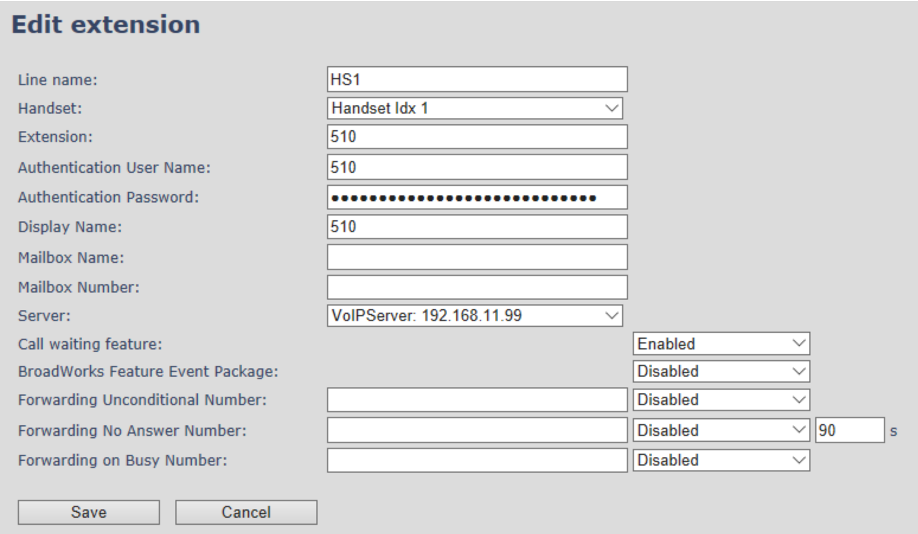
	Idx	IPEI	Handset State	Handset Type FW Info	FWU Progress		VoIP Idx	Extension	Display Name	Server	Server Alias	State
☐	1	0276A584DA	Present@RPN00	8630 400.2	Off	☐	1	510	510	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	2	02788888DB	Present@RPN00	8630 400.2	Off	☐	2	511	511	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	3	02779C7F09	Present@RPN00	8630 400.2	Off	☐	3	514	514	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	4	027792D0FE	Present@RPN00	8630 400.2	Off	☐	4	513	513	192.168.11.99	HDJSERVER	SIP Registered@RPN00

Check All / Uncheck All — Check All Extensions / Uncheck All Extensions

With selected: Delete Handset(s) Register Handset(s) Deregister Handset(s) Start SIP Registration(s) SIP Delete Extension(s)

Then you will see the Edit extension page

Screenshot



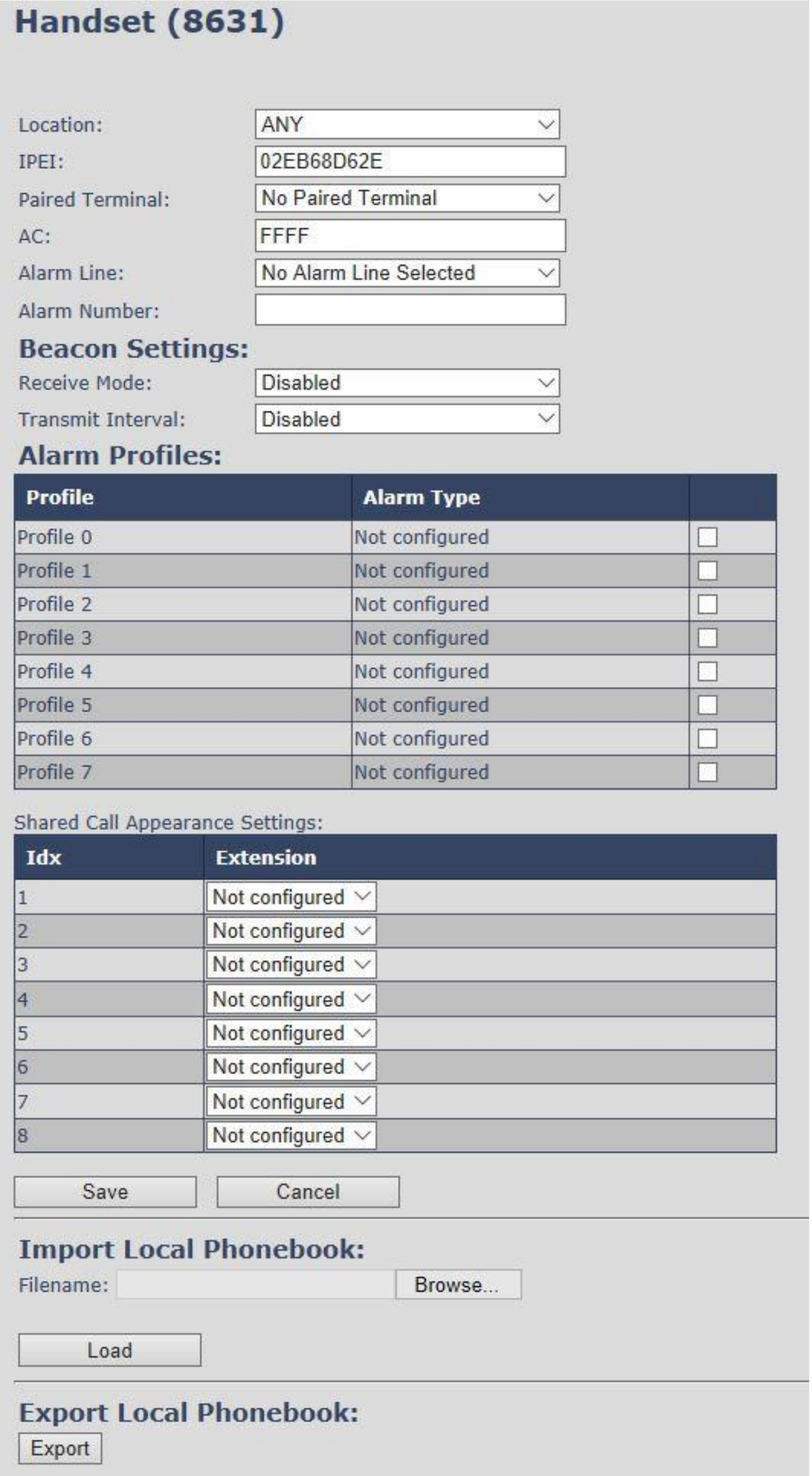
Now you can edit the needed information and save the changes.

For detailed description of each field please see section 5.3.1

5.3.3 Edit Handset

Use the mouse to click the handset IPEI link to open the handset edit window. In the handset edit view the handset SIP location can be fixed to either any or a specific base.

Screenshot

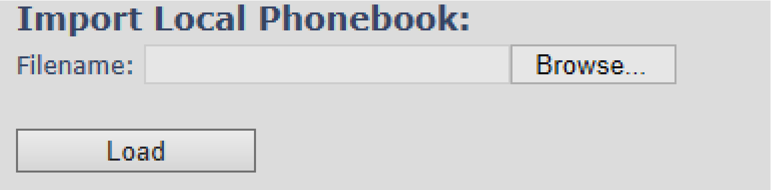


PARAMETER	DEFAULT VALUE(S)	DESCRIPTION	
Location	ANY	ANY = The handset can locate on any base station in the multi cell system. Dropdown to select specific base station. NB: Only allowing the handset to locate on a specific base station can affect the handover function. Handset reboot is necessary to relocate handset	
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	Enter Proximity: Leave Proximity: Enter or Leave Proximity:	FEATURE READY IN V460 FIRMWARE (Q2 2019)
TRANSMIT INTERVAL	Disabled	Short: Step1: Step2: Step3: Step4: Step5: Long:	FEATURE READY IN V460 FIRMWARE (Q2 2019)
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

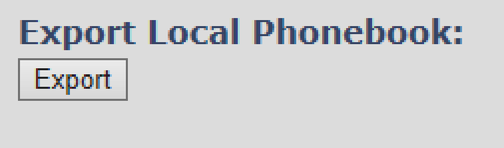


NOTE: The local phonebook can have 100 entries for RTX863x and RTX8830 and 50 entries for RTX8430.

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

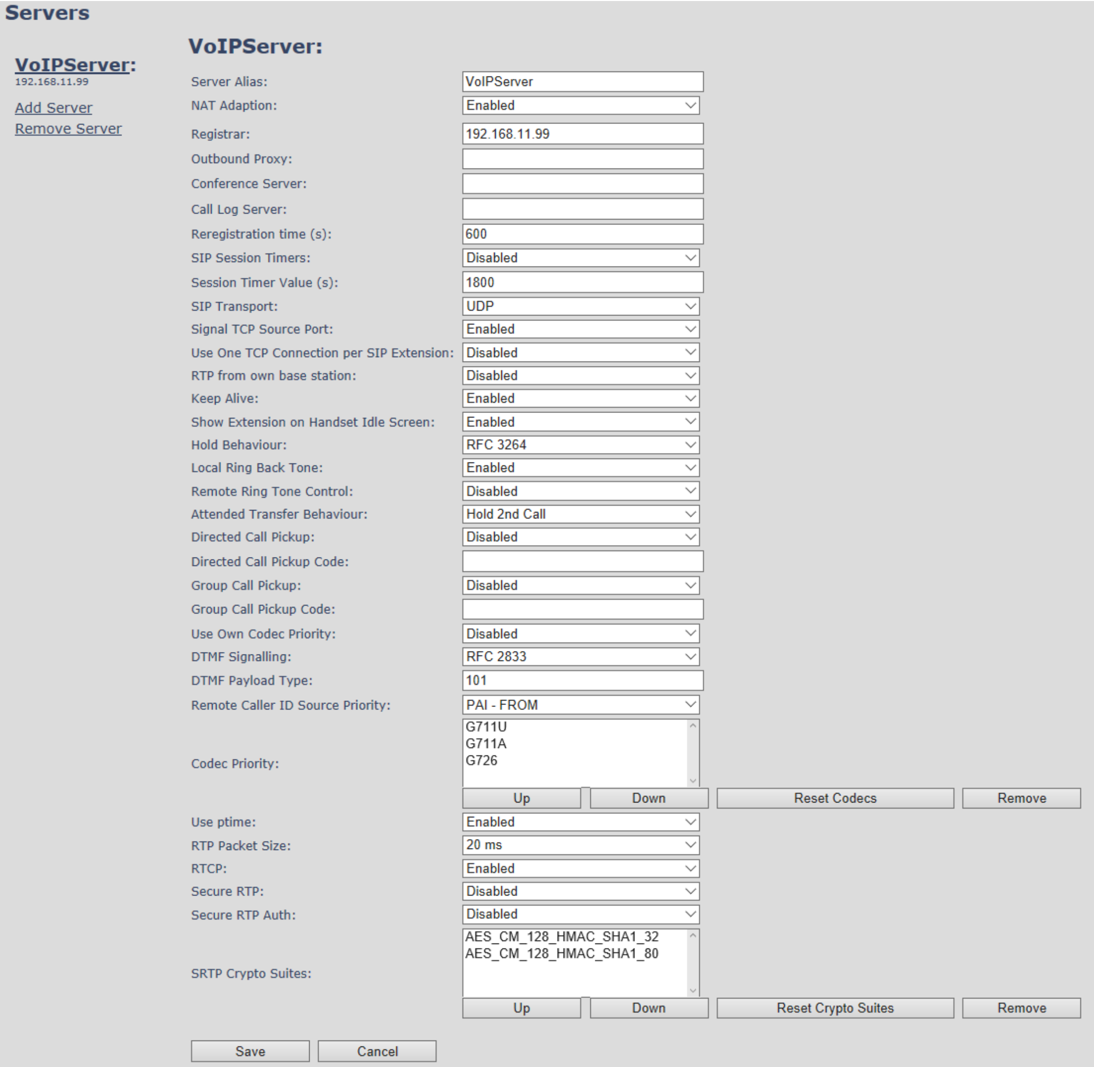


Press the export button and save the .CSV file on you PC or Server.

5.4 Servers

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

Screenshot



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
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 1.1
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 5.5.5) 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/TLS CONNECTION PER SIP EXTENSION:	Disabled	When using TCP or TLS 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. This setting is typically enabled for operation with Cisco.
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 send 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 send only, recvonly or inactive dependent of the context
LOCAL RING BACK TONE	Enabled	In case the server don’t play local ring back tone the handset will do it.
REMOTE RING TONE CONTROL	Enabled	Sometimes call distinguished ringing. It enable 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 PBXes do not expect that the 2nd call is put on hold, and therefore attended transfer fails on these PBXes. 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 pickup a group call
USE OWN CODEC PRIORITY	Disabled	Default disabled. By enable the system codec priority during incoming call is used instead of the calling party priority.

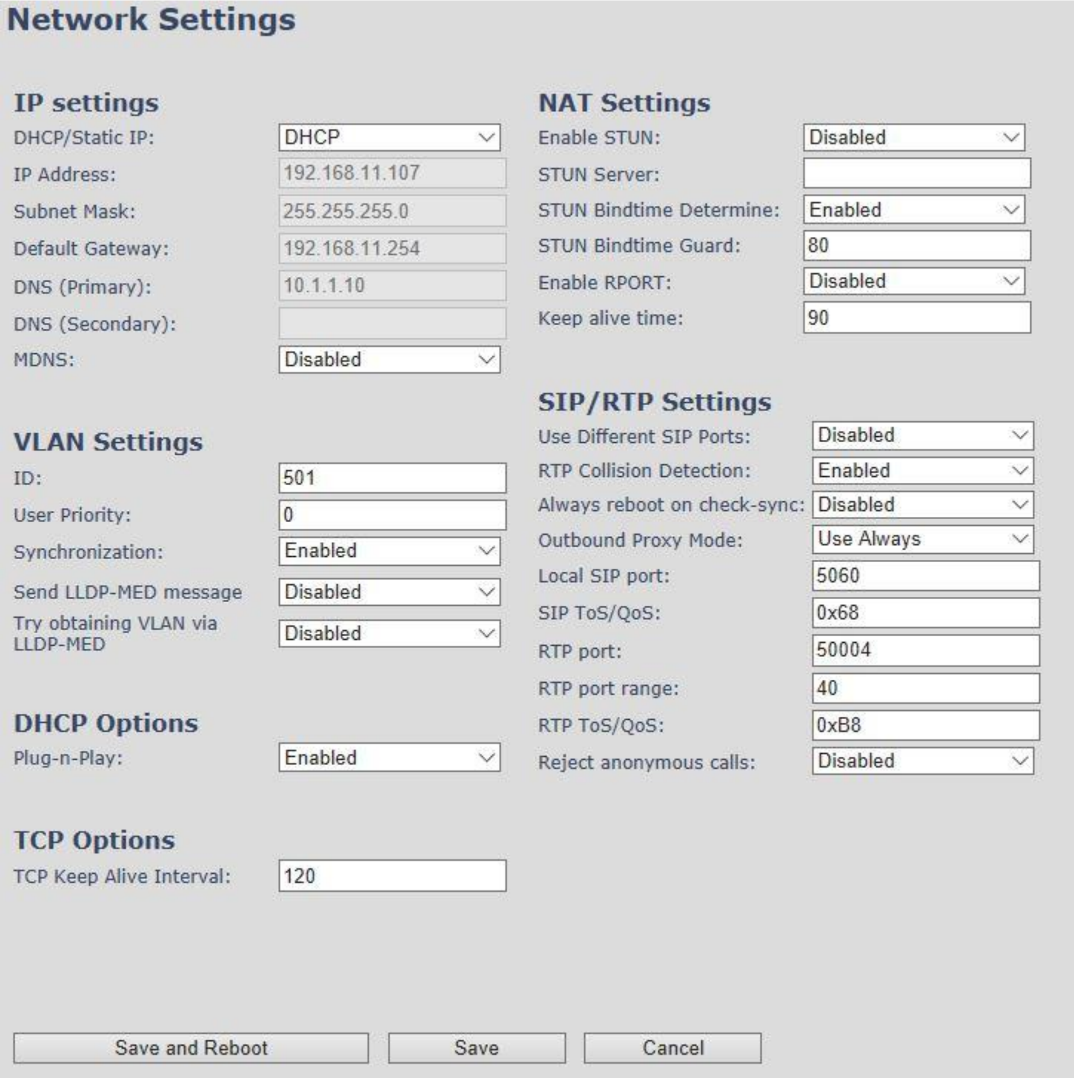
		E.g. If base has G722 as top codec and the calling party has Alaw on top and G722 further down the list, the G722 will be chosen as codec for the call.
DTMF SIGNALLING	RFC 2833	Conversion of decimal digits (and '*' and '#') into sounds that share similar characteristics with voice to easily traverse 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	Defines the codec priority that base stations use for audio compression and transmission. Possible Option(s): G.711U,G.711A, G.726, G.729, G.722. 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 first 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. To use G.729, add on DSP module must be installed in all base stations. Contact your local dealer for price information.
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
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	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.

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

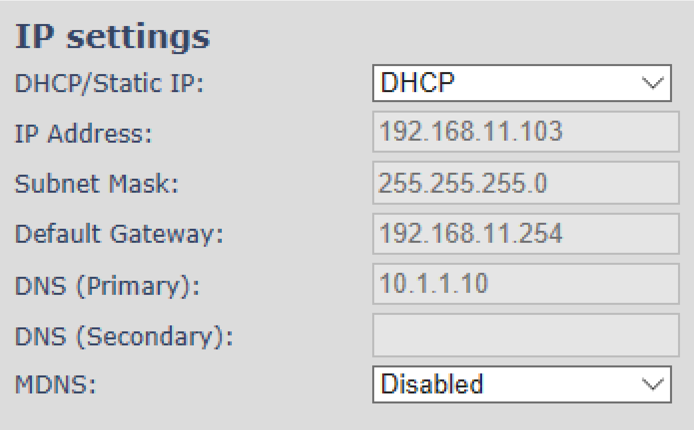
In this section, we describe the different parameters available in the network configurations menu.

Screenshot



5.5.1 IP Settings

Screenshot



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.

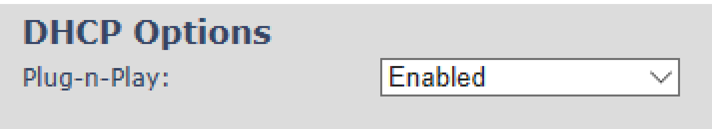
Screenshot

PARAMETER	DEFAULT VALUES	DESCRIPTION
VLAN 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.
VLAN 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)
VLAN 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.
SEND LLDP-MED MESSAGE	Disabled	Disable/Enable
TRY OBTAINING VLAN VIA LLDP-MED	Disabled	Disable/Enable

For further help on VLAN configuration refer to Appendix.

5.5.3 DHCP Options

Screenshot

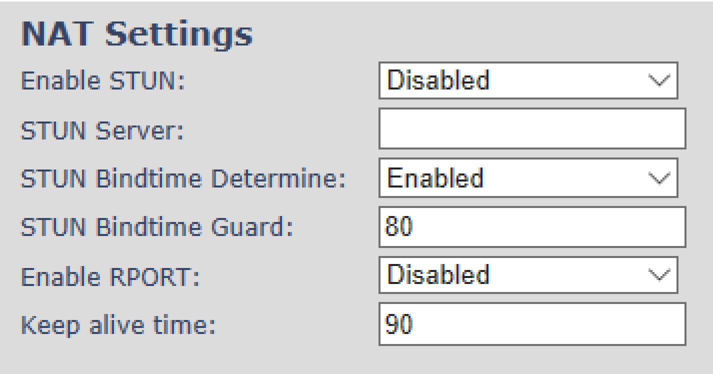


PARAMETER	DEFAULT VALUES	DESCRIPTION
PLUG-N-PLAY	Enabled	Enabled: DHCP option 66 to automatically provide PBX IP address to base.

5.5.4 NAT Settings

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

Screenshot

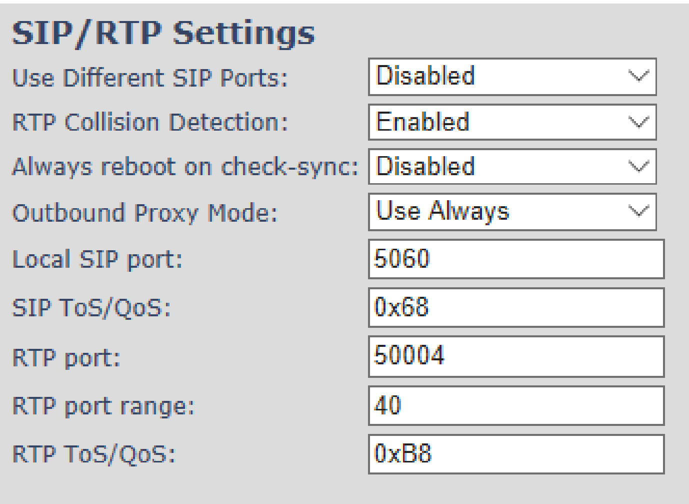


PARAMETER	DEFAULT VALUES	DESCRIPTION
ENABLE STUN	Disabled	Enable to use STUN
STUN SERVER	NA	**Permitted value(s): AAA.BBB.CCC.DDD** (Currently only Ipv4 are supported) or **URL** (e.g.: firmware.rtx.net).
STUN BINDTIME DETERMINE	Enabled	
STUN BINDTIME GUARD	80	**Permitted values:** Positive integer default 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

5.5.5 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
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

5.5.6 TCP Options

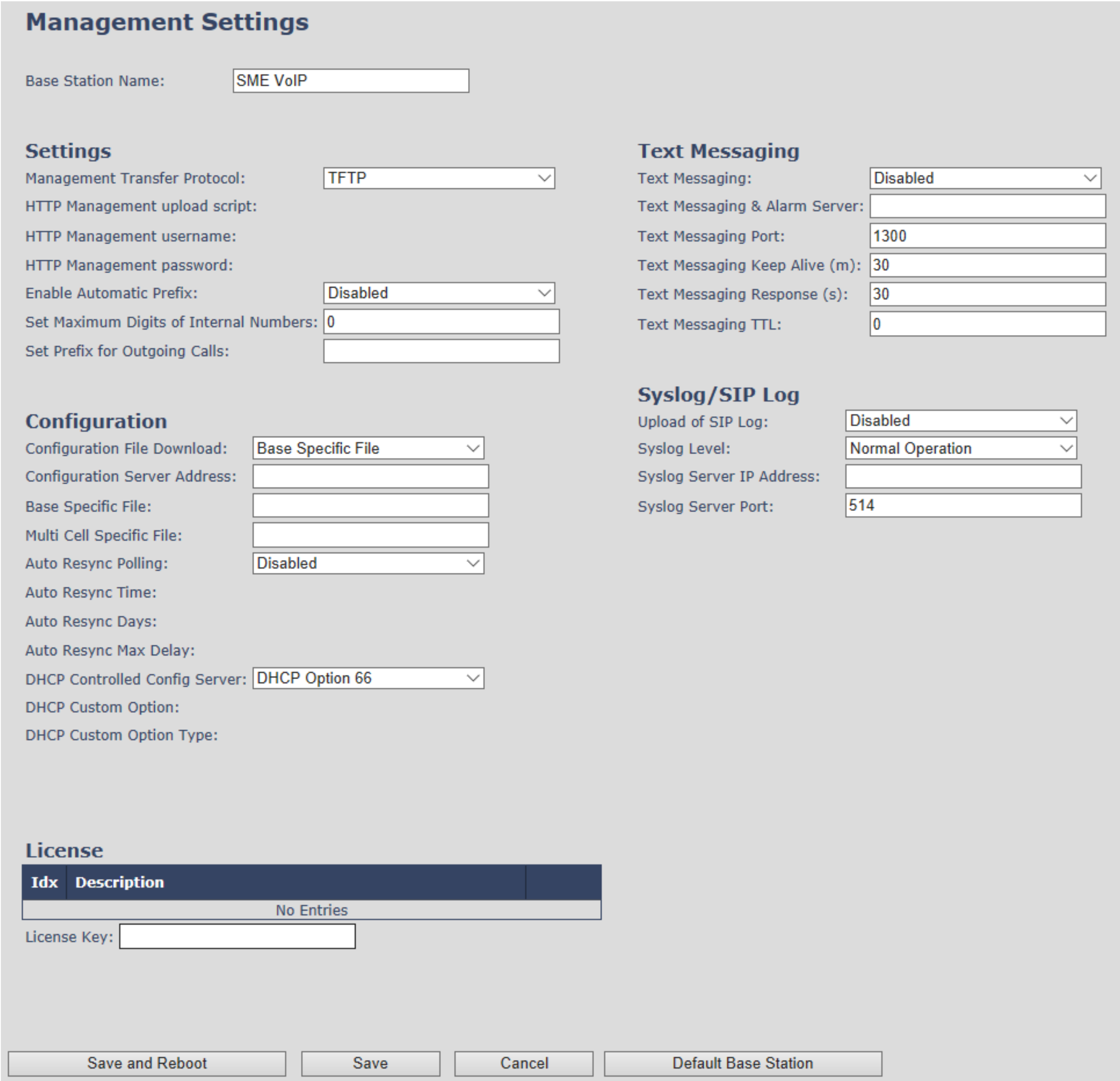
Screenshot

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.

5.6 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
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 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 DELAY	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.
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>**
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 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.
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.

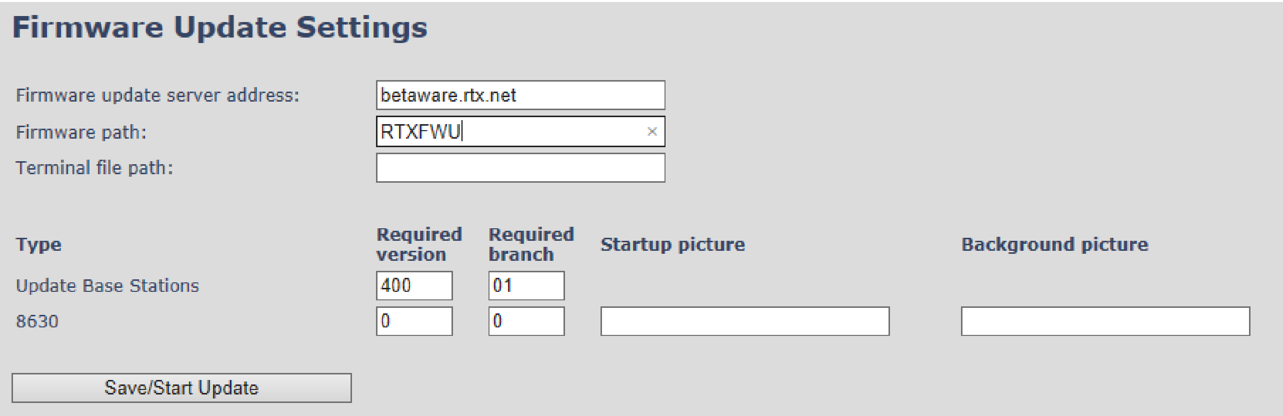
5.6.5 License:

PARAMETER	DEFAULT VALUE	DESCRIPTION
LICENSE	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

In this page, the system administrator can configure how base stations and SIP nodes upgrade/downgrade to the relevant firmware. Handset firmware update status can be found in the extensions page and repeater firmware update status in the repeater page. Base firmware update status is found in the multicell page.

Screenshot



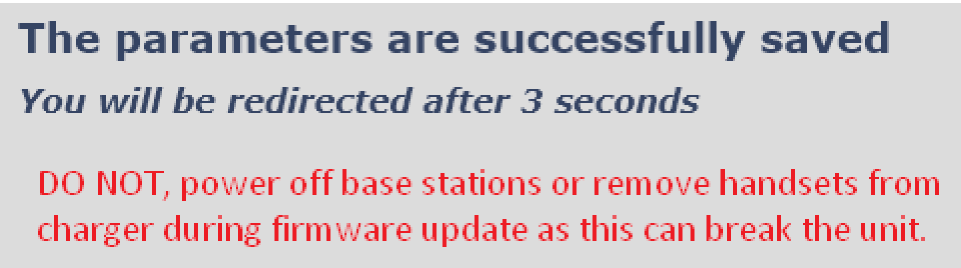
PARAMETER	DEFAULT VALUE(S)	DESCRIPTION
FIRMWARE UPDATE SERVER ADDRESS	Empty	IP address or DNS of firmware update files source **Valid Inputs:** AAA.BBB.CCC.DDD or <URL> **Example:** firmware.rtx.net or 10.10.104.41
FIRMWARE PATH	Empty	Location of firmware on server (or firmware update server path where 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:** Image 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. To small the rest of the screen will be black. To large only the left portion of the image will be shown. **NOTE:** Only .BMP is files are supported. **NOTE:** Changing startup picture is not available for new GUI (RTX8631/RTX8632 and RTX8633)
BACKGROUND PICTURE	Empty	Name of the background picture you want on the handsets when they are powered up. **NOTE:** Image 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. To small the rest of the screen will be black. To large only the left portion of the image will be shown **NOTE:** Only .BMP is files are supported. **NOTE:** Changing background picture is not available for new GUI (RTX8631/RTX8632 and RTX8633)
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.

Screenshot



5.8 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.

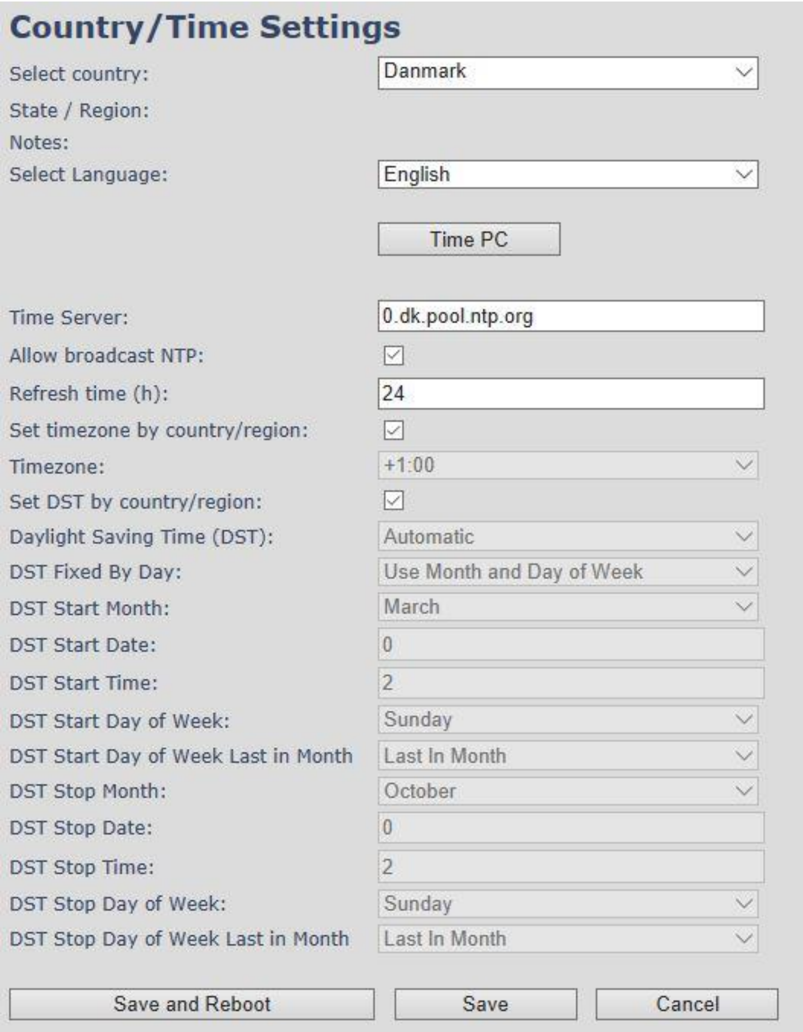
As well the time server is used in the debug logs and for SIP traces information pages and used to determine when to check for new configuration and firmware files.

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.

Screenshot



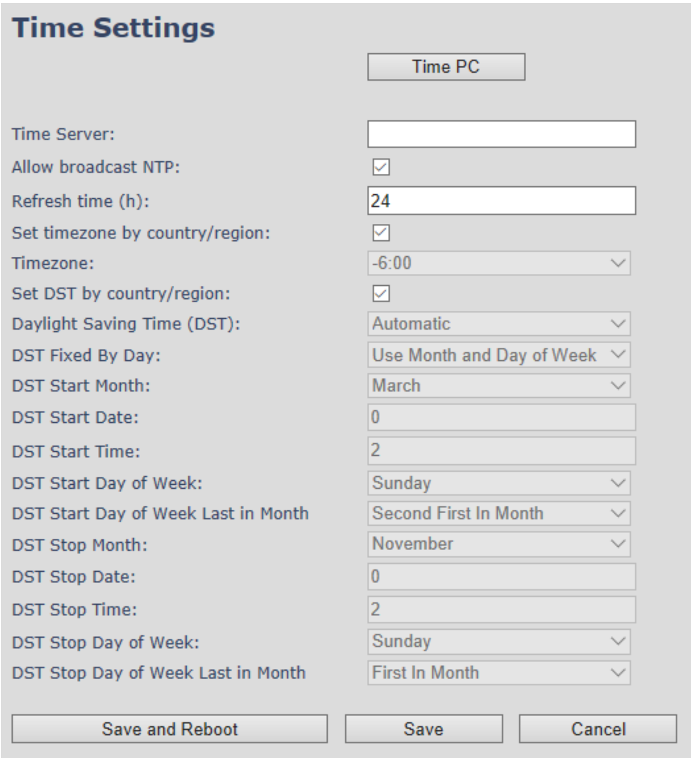
PARAMETER	DEFAULT VALUES	DESCRIPTION
SELECT COUNTRY	US/Canada	Supported countries: Australia, Belgium, Brazil, Denmark, Germany, Spain, France, Ireland, Italia, Luxembourg, Nederland, New Zealand, Norway, Portugal, Swiss, Finland, Sweden, Turkey, United Kingdom, US/Canada, Austria
STATE / REGION	NA	Only 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.
SET TIMEZONE BY COUNTRY/REGION	checked	When checked time zone will follow country/region
SET DST BY COUNTRY/REGION	checked	When checked DST will follow country/region
NOTES	Empty	Only showing notes to time setting for countries: US/Canada, Brazil

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

Screenshot



PARAMETER	DEFAULT VALUES	DESCRIPTION
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.

5.9 Security

The security section is used for loading of certificates and for selecting if only trusted certificates are used. Furthermore, web password can be configured.
The Security web is divided into three sections: Certificates (trusted), SIP Client Certificates (and keys) and Password administration.

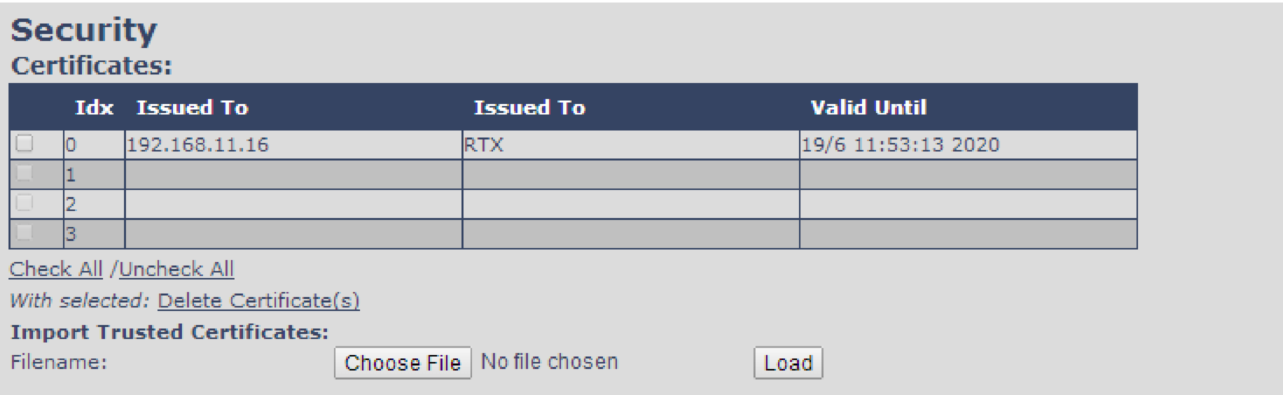
To setup secure fwu and configuration file download select HTTPs for the Management Transfer Protocol (refer to management web).

SIP and RTP security are server dependent and to configure user must use the web option Servers (refer to server's web).

5.9.1 Certificates

The certificates list contains the list of loaded certificates for the system. Using the left column check mark, it is possible to check and delete certificates. To import a new certificate, use the mouse "select file" and browse to the selected file. When file is selected, use the "Load" bottom to load the certificate.
The certificate format supported is DER encoded binary X.509 (.cer).

Screenshot



5.9.2 Certificates list

PARAMETER	DEFAULT VALUES	DESCRIPTION
IDX	Fixed indexes	Index number
ISSUED TO	Empty	IP address – which is part of the certificate file
ISSUED TO	Empty	Organization, Company – which is part of the certificate file
VALID UNTIL	Empty	Date Time Year – which is part of the certificate file

Screenshot

By enabling Use Only Trusted Certificates, the certificates 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.

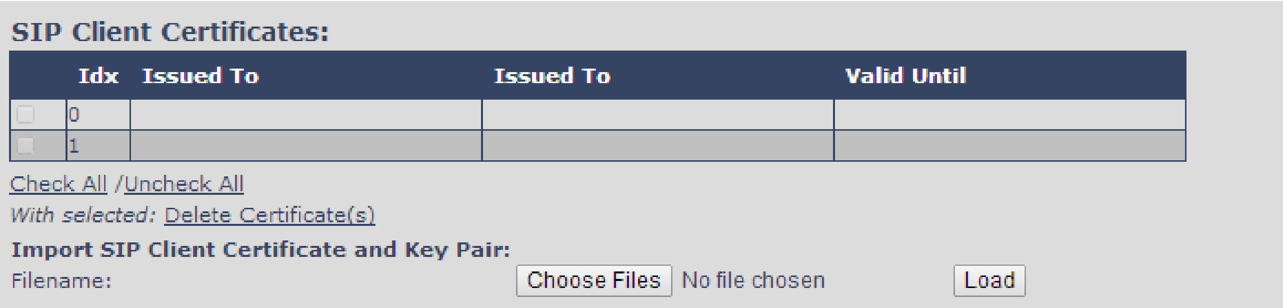
NOTE: It is important to use correct date and time of the system when using trusted certificates. In case of time/date not defined the certificate, validation can fail.

5.9.3 SIP Client Certificates

To be able to establish a TLS connection in scenarios, where the server requests a client certificate, a certificate/key pair must be loaded into the base. This is currently supported only for SIP.
To load a client certificate/key pair, both files must be selected at the same time, and it is done by pressing “select files” under “Import SIP Client Certificate and Key Pair” and then select the certificate file as well as the key file at the same time. Afterwards, press load.
The certificate must be provided as a DER encoded binary X.509 (.cer) file, and the key must be provided as a binary PKCS#8 file.

NOTE: Use Chrome for loading SIP Client Certificates

Screenshot



5.9.4 Device identity

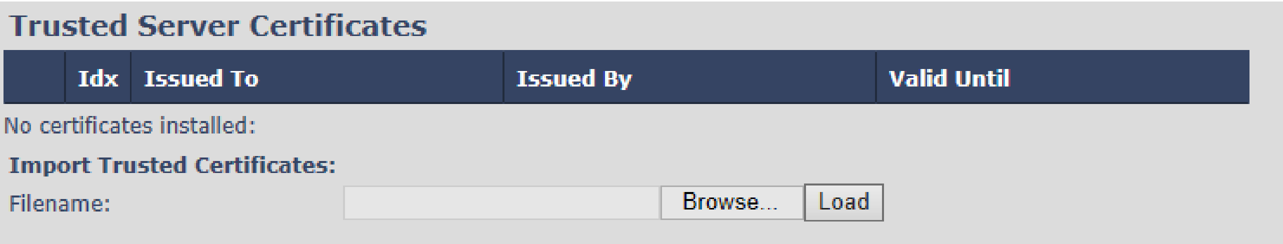
The certificate and personal key used by the base when acting as server or when the server requires client authentication in the SSL handshake procedure.

Screenshot

5.9.5 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

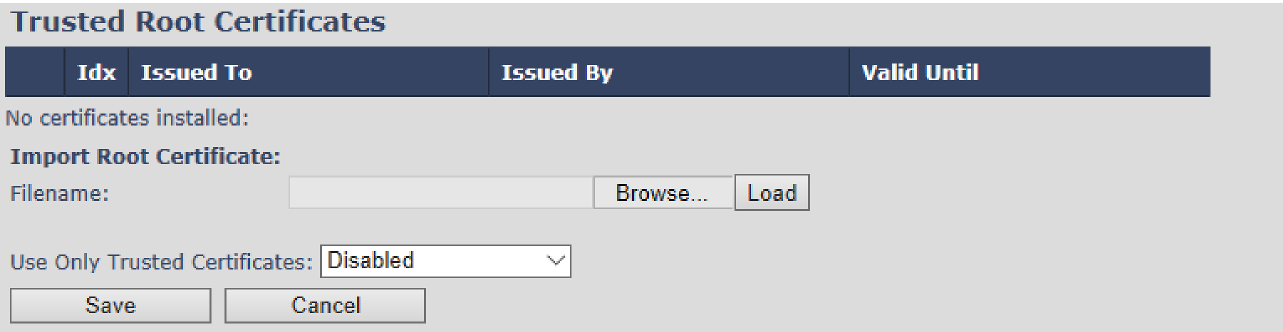
Screenshot



5.9.6 Trusted Root Certificates

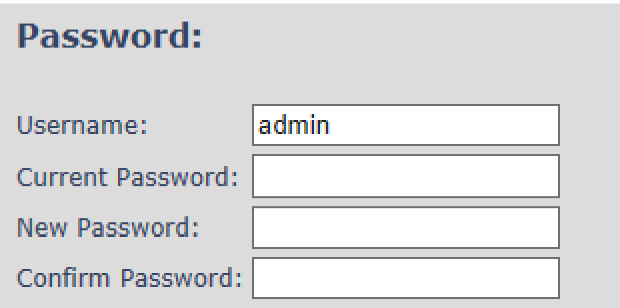
Root certificates (self-signed) trusted by the base. Used to validate received root certificates sent by the server during the SSL handshake procedure.

Screenshot



5.9.7 Password

In the below the password parameters are defined.



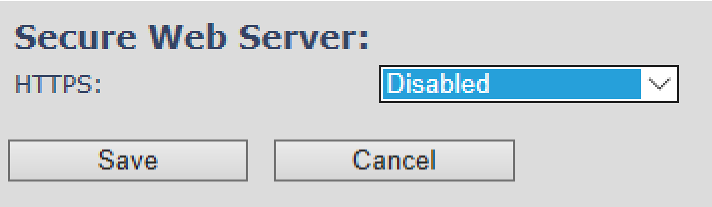
PARAMETER	Default Values	Description
USERNAME	Admin	Can be modified to any supported character and number Maximum characters: 15
CURRENT PASSWORD	Admin	Can be modified to any supported character and number
NEW PASSWORD	Empty	Change to new password Maximum characters: 15
CONFIRM PASSWORD	Empty	Confirm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

5.9.8 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.

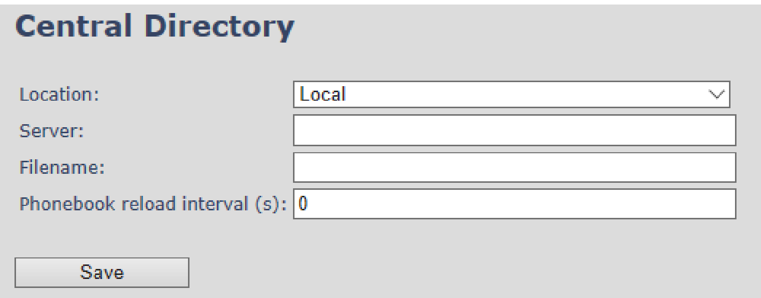
5.10 Central Directory and LDAP

The SME VOIP system support two types of central directories, a local central directory or LDAP directory. For both directories' caller id look up is made with match for 6 digits of the phone number.

5.10.1 Local Central Directory

Select local and save for local central directory.

Screenshot



PARAMETER	DEFAULT VALUES	DESCRIPTION
LOCAL	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 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.

5.10.1.1 Import Central Directory

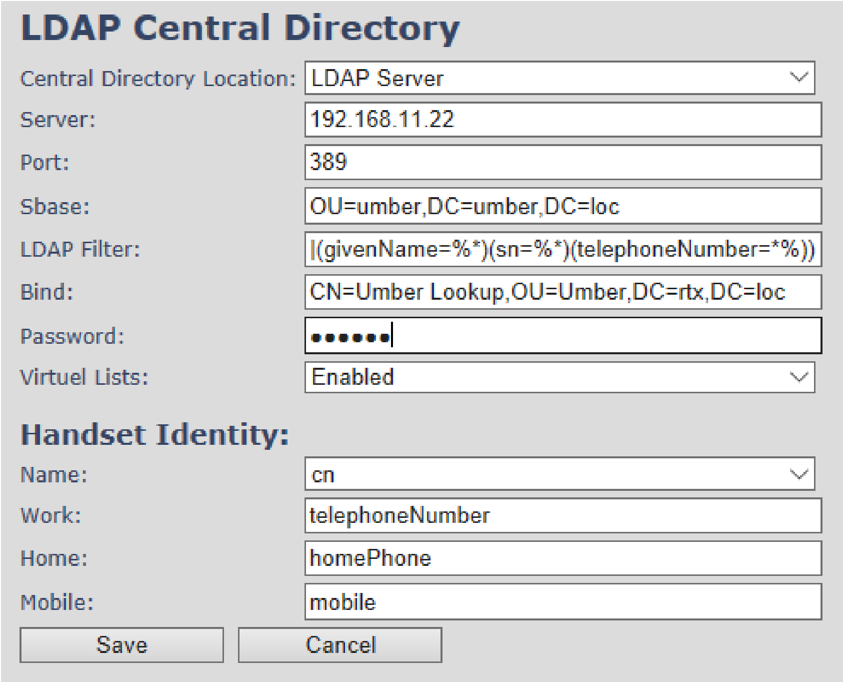
The import central directory feature is using a browse file approach. After file selection press the load button to load the file. The system supports only the original *.csv format. Please note that some excel csv formats are not the original csv format. The central directory feature can handle up to 3000 contacts (Max file size 100kb). For further details of the central directory feature refer to appendix.

5.10.2 LDAP

Select LDAP Server and save for LDAP server configuration.

Screenshot

Screenshot



PARAMETER	DEFAULT VALUES	DESCRIPTION
LDAP SERVER	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>
PORT	Empty	The server port number that is open for LDAP connections.
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
VIRTUEL LISTS	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

5.10.3 Characters supported

The below table shows which characters are supported in the communication between RTX8660 and handset.

	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
0				0	@	P	`	p	€	İ		°	À	Đ	à	ð
1			!	1	A	Q	a	q	ı	'	¡	±	Á	Ñ	á	ñ
2			"	2	B	R	b	r	‚	'	¢	Č	Â	Ò	â	ò
3			#	3	C	S	c	s	ƒ	"	£	č	Ã	Ó	ã	ó
4			$	4	D	T	d	t	„	"	¤	´	Ä	Ô	ä	ô
5			%	5	E	U	e	u	…	•	¥	µ	Å	Õ	å	õ
6			&	6	F	V	f	v	†	–	¦	¶	Æ	Ö	æ	ö
7			'	7	G	W	g	w	‡	—	§	·	Ç	×	ç	÷
8			(	8	H	X	h	x	ˆ	˜	¨	¸	È	Ø	è	ø
9			)	9	I	Y	i	y	Ř	ř	Ů	Ď	É	Ù	é	ù
A			*	:	J	Z	j	z	Š	š	ů	ď	Ê	Ú	ê	ú
B			+	;	K	[	k	{	‹	›	«	»	Ë	Û	ë	û
C			,	<	L	\	l	|	Œ	œ	Ě	Ť	Ì	Ü	ì	ü
D			-	=	M	]	m	}	Ş	ş	ě	ť	Í	Ý	í	ý
E			.	>	N	^	n	~	Ž	ž	Ň	ň	Î	Þ	î	þ
F			/	?	O	_	o	Ğ	ğ	Ÿ	¯	¿	Ï	ß	ï	ÿ

5.11 Multi-cell Parameter Definitions

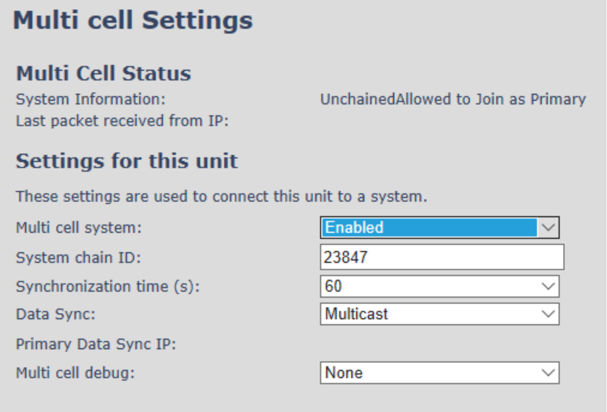
NOTE: To join 2 or more Base Station in a Multi Cell system you need to have one handset added to the system, for details and Step-by-Step guide to Multi Cell please see Appendix

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

5.11.1 Settings for Base Unit

Description of Settings for Specific Base units is as follows:

Screenshot



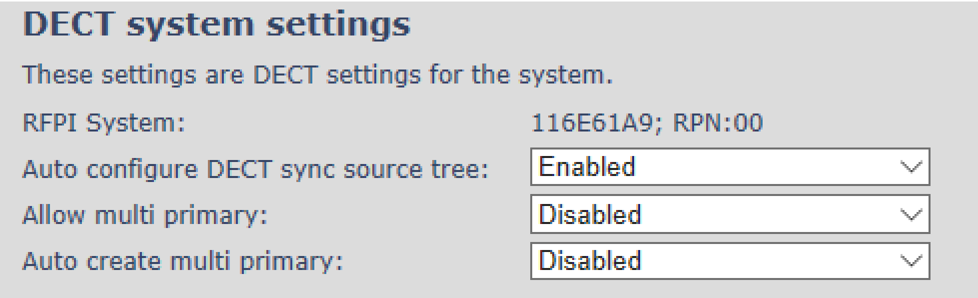
Multicell status covers status of data synchronization. The status “Keep-alive” means normal operation.

PARAMETER	DEFAULT VALUES	DESCRIPTION
MULTI CELL SYSTEM	Disabled	Enable this option to allow the Base unit to be set in multi-cell mode (can be set either as master or slave in the multi-cell chain system – refer to **MAC-units in Chain** section for details). **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. The Chain ID use up a SIP account with this value. **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. 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.
DATA SYNC:	Multicast	To select between multicast or Peer to Peer data synchronization mode. The multicast port range and IP addresses used is calculated from the chain id. 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:** 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 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

5.11.2 DECT System Settings

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

Screenshot



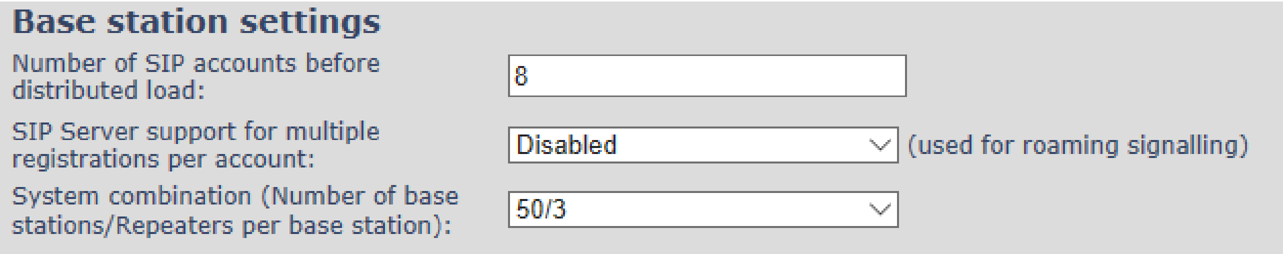
PARAMETER	DEFAULT VALUES	DESCRIPTION
DECT SYSTEM RFPI	Not able	This is a radio network identity accessed by all Base units in a specific multi-cell system. It composed of 5 octets. It is actually 5 different variables combined together. **RFPI Format:** XX XX XX XX XX (where XX are HEX values)
ALLOW 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	By enabled the system can 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 be enabled for this feature to also be enabled.
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.

NOTE: To run with 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.

5.11.3 Base System 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 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.

5.11.4 Base Station Group

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

Screenshot:

Base Station Group

	ID	RPN	Version	MAC-Address	IP-Address	IP Status	DECT sync source	DECT property	Base Station Name
☐	0	00	280	00087B0A00B3	192.168.11.159	This Unit	Select as primary	**Primary**	1
☐	1	04	280	00087B09FECA	192.168.11.116	Connected	Primary:RPN00 (-24dBm)	Locked	2
☐	2	08	280	00087B09FE45	192.168.11.113	Connected	Level 1:RPN04 (-24dBm)	Locked	3
☐	3	0C	280	00087B09FF08	192.168.11.109	Connected	Level 2:RPN08 (-24dBm)	Locked	4
☐	4	10	280	00087B09FE4A	192.168.11.166	Connected	Level 3:RPN0C (-24dBm)	Locked	5
☐	5	14	280	00087B079205	192.168.11.133	Connected	Level 4:RPN10 (-24dBm)	Locked	6

Check All / Uncheck All
With selected: Remove from chain

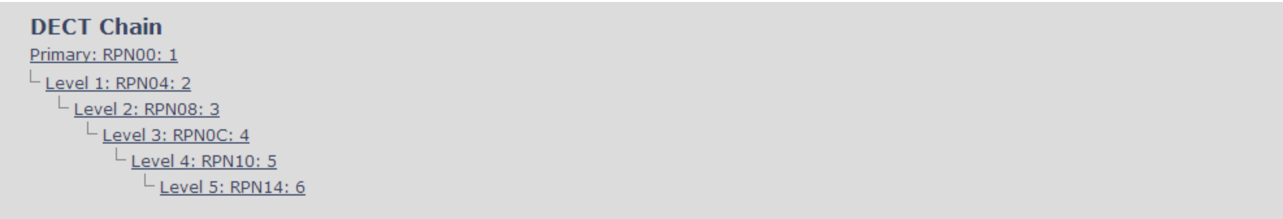
PARAMETERS	DESCRIPTION
ID	Base unit identity in the chained network. **Permitted Output:** Positive Integers
RPN	The Radio Fixed Part Number is an 8-bit DECT cell identity allocated by the installer. The allocated RPN within the SME must be geographically unique. **Permitted Output:** 0 to 255 (DEC) **OR** 0x00 to 0xFF (HEX)
VERSION	Base station current firmware version. **Permitted Output:** positive Integers with dot (e.g. 273.1)
MAC ADDRESS	Contains the hardware Ethernet MAC address of the base station. It varies from Base station to Base stations.
IP STATUS	Current Base station behavior in the SME network. Possible Outputs **Connected:** The relevant Base station(s) is online in 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 three will automatically be generated. If manual configured the administrator should choose the relevant “multi cell chain” level its wants a specific Base unit be placed. Maximum number of “multi-cell chain” levels is 12. 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 unto which all other nodes in the chain synchronizes 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 are 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 stay 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.

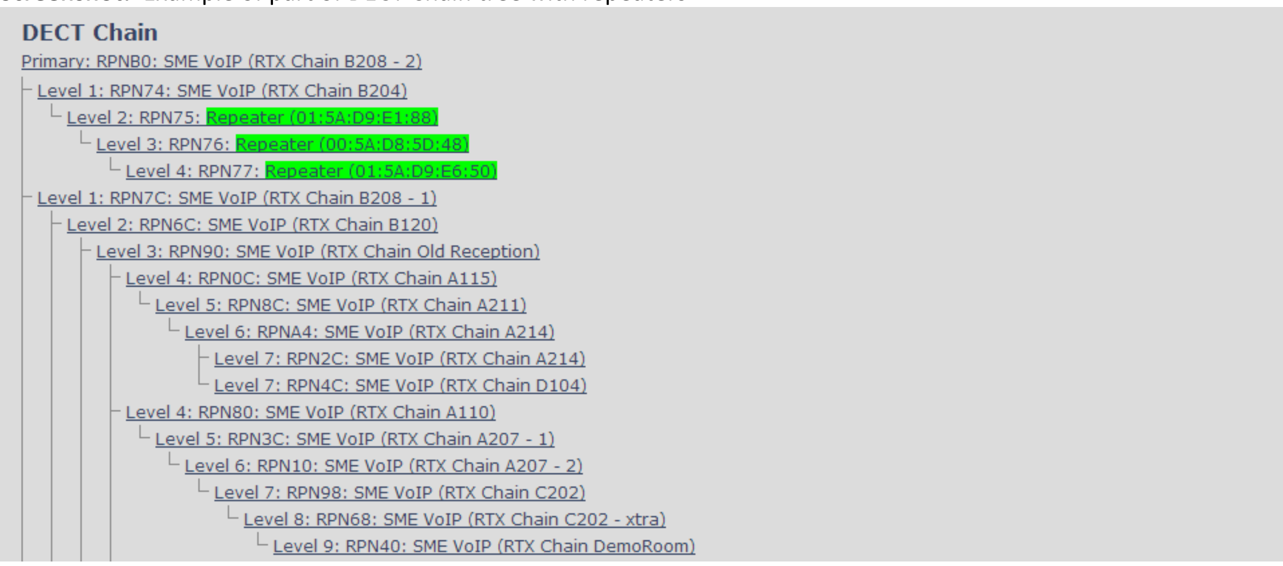
5.11.5 DECT Chain

Below the Base Group Table is the DECT Chain tree. The DECT Chain tree is a graphical presentation of the Base Group table levels and connections. Repeaters are shown with green highlight.

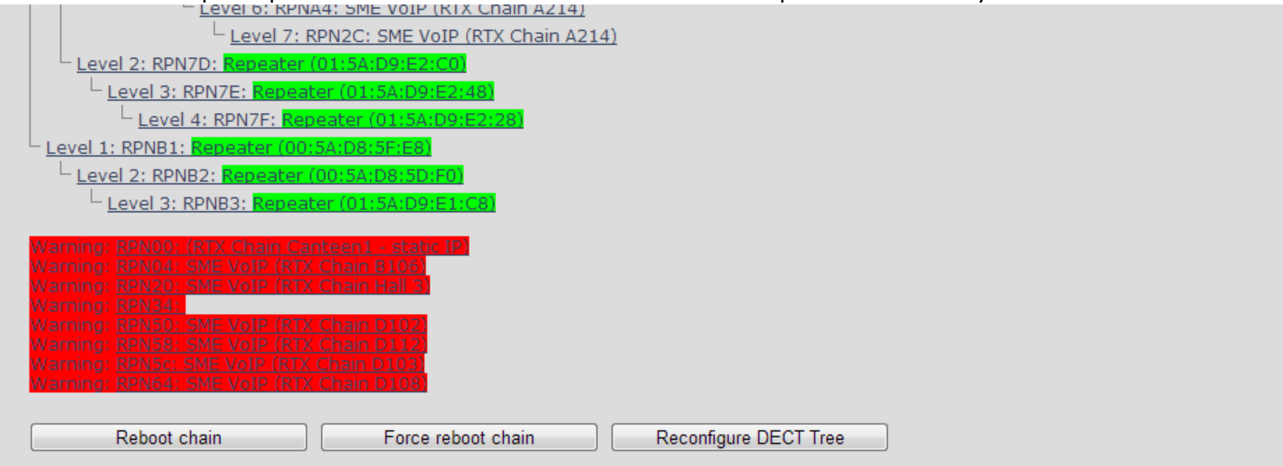
Screenshot: DECT Chain tree of above configuration



Screenshot: Example of part of DECT Chain tree with repeaters



Screenshot: Example of part of DECT Chain tree with units in Base Group but not in tree by various reasons.



When a base or repeater has not joined the tree, it will be shown with read background below the tree.

5.11.6 RTX8660 -RTX8663 Mixed mode

RTX8663 base station can be added to existing systems using RTX8660 base station. Because the RTX8663 have more powerful hardware and additional features, there will be some limitations.

A system running mixed mode, is limited to RTX8660 features.

NOTE: LAN SYNC will not work in mixed mode.

The system will display a warning message on the Home/Status page.

Screenshot:

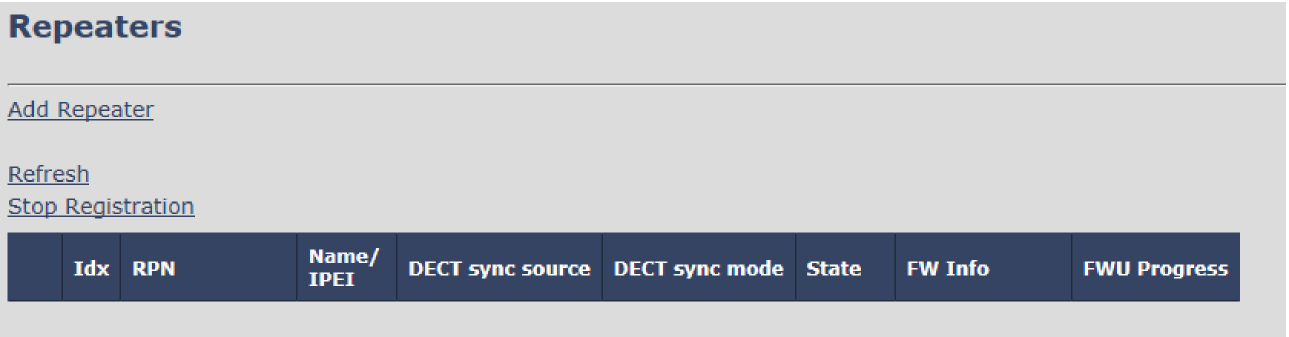
5.12 Repeaters

Within this section we describe the repeater parameter, and how to operate the repeater.

5.12.1 Add repeater

From repeaters web select "Add Repeater"

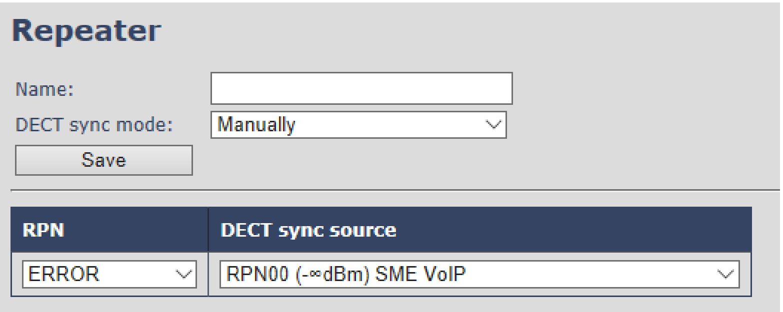
Screenshot



Then select "DECT Sync mode"

PARAMETERS	DESCRIPTION
NAME	Repeater name. If no name specified, the field will be empty
DECT SYNC MODE	**Manually:** User controlled by manually assign "Repeater RPN" and "DECT sync source RPN" **Local Automatical:** Repeater controlled by auto detects best base signal and auto assign RPN.

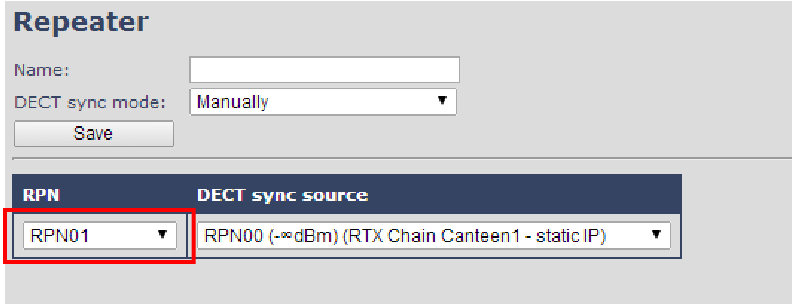
Screenshot



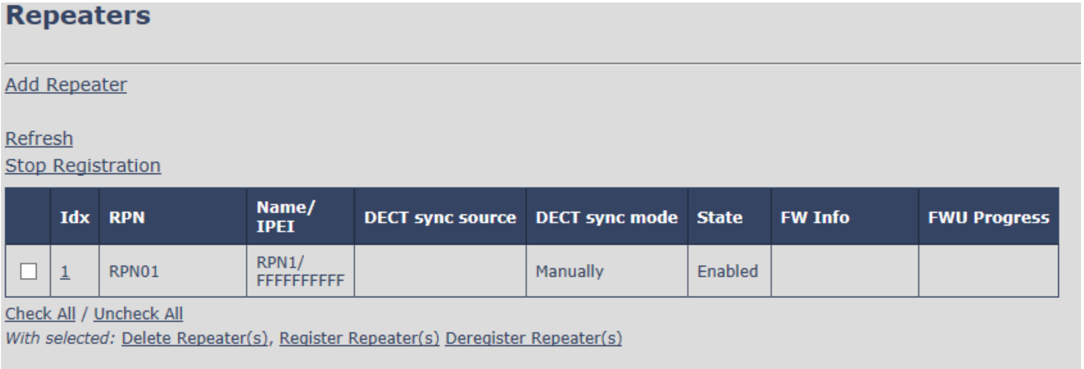
5.12.1.1 Manually

User controlled by manually assign “Repeater RPN” and “DECT sync source RPN”. The parameters are selected from the drop-down menu and Save.

Screenshot



Screenshot



PARAMETERS	DESCRIPTION
IDX	System counter
RPN	SINGLE CELL SYSTEM: The base has always RPN00, first repeater will then be RPN01, second repeater RPN02 and third RPN03 (3 repeaters maximum per base) MULTI CELL SYSTEM: Bases are increment by 2^2 in hex, means first base RPN00 second base RPN04 etc., in between RPN01, 02, 03 addressed for repeaters at Primary base and 05, 06, 07 addressed for Secondary base (3 repeaters maximum per base)
NAME/IPIE	Name and IPIE number of the repeater
DECT SYNC MODE	DECT Sync mode – Manually or Automatic
STATE	State of the repeater Enabled/Disabled
FW INFO	Firmware version
FWU PROGRESS	How many percentages of the firmware is loaded / Off if no firmware is being loaded

Screenshot

Repeaters

Add Repeater

Refresh
Stop Registration

	Idx	RPN	Name/ IPEI	DECT sync source	DECT sync mode	State	FW Info	FWU Progress
☐	1	RPN01	RPN1/ 005AD85D90	RPN00 (-26dBm)	Manually	Present@RPN00	41.1	Off

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

Good practice when adding repeaters to a Multi Cell system is to use manually registration, because then you can control what base station the repeater(s) connects to.

5.12.1.2 Local Automatical

Repeater controlled by auto detects best base signal and auto assign RPN. The RPN and DECT sync source are greyed out.

Screenshot

Repeater

Name:	RPN2
DECT sync mode:	Local Automatical

Save

The repeater RPN is dynamic assigned in base RPN range.
With local automagical mode repeater on repeater (chain) is not supported.

5.12.2 Register Repeater

Adding a repeater makes it possible to register the repeater. Registration is made by select the repeater and pressing register repeater. The base window for repeater registration will be open until the registration is stopped. By stopping the registration all registration on the system will be stopped inclusive handset registration.

	Idx	RPN	Name/ IPEI	DECT sync source	DECT sync mode	State	FW Info	FWU Progress
☐	2		RPN2/ FFFFFFFFFF		Local Automatical	Enabled		

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

5.12.3 Repeaters list

Screenshot

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)

The number of repeaters allowed on each base station is defined on the Multi cell page.
System combination: 50/3 – 127/1 -254/0.
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 have no function.

Example:
System combination 50/3:
Base stations are named RPN00 – RPN04 – RPN08. Etc. jumping 4 numbers each time (HEX numbers)
Repeaters connect to base station RPN00 will be called RPN01 – RPN02 – RPN03 (HEX numbers)
Repeaters connect to base station RPN04 will be called RPN05 – RPN06 – RPN07 (HEX numbers)
Etc.

System combination 127/1:
Base stations are named RPN00 – RPN02 – RPN04. Etc. jumping 2 numbers each time (HEX numbers)
Repeaters connect to base station RPN00 will be called RPN01 (HEX numbers)
Repeaters connect to base station RPN02 will be called RPN05 (HEX numbers)
Etc.

System combination 254/0:
Repeater registration not possible.

PARAMETERS	DESCRIPTION
IDX	Repeater 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)
NAME/IPEI	Contains the name and the unique DECT serial number of the repeater. If name is given the field will be empty.
DECT SYNC SOURCE	The “multi cell chain” connection to the specific Base/repeater unit. Maximum number of chain levels is 12. Sync. source format: “RPNyy (-zz dBm)” yy: RPN of source

	zz: RSSI level seen from the actual repeater
DECT SYNC MODE	**Manually:** User controlled by manually assign "Repeater RPN" and "DECT sync source RPN" **Local Automatical:** Repeater controlled by auto detects best base signal and auto assign RPN. **Chaining Automatical:** Base controlled by auto detects best base or repeater signal and auto assign RPN. This feature will be supported in a future version
STATE	Present@unit means connected to unit with RPN yy
FW INFO	Firmware version
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 "**Conn. term. wait**" (Repeater): All FWU is complete and is now waiting for connections to stop before repeater restart. Complete HS/repeater: FWU complete **Error:** Not able to fwu e.g. file not found, file not valid etc.

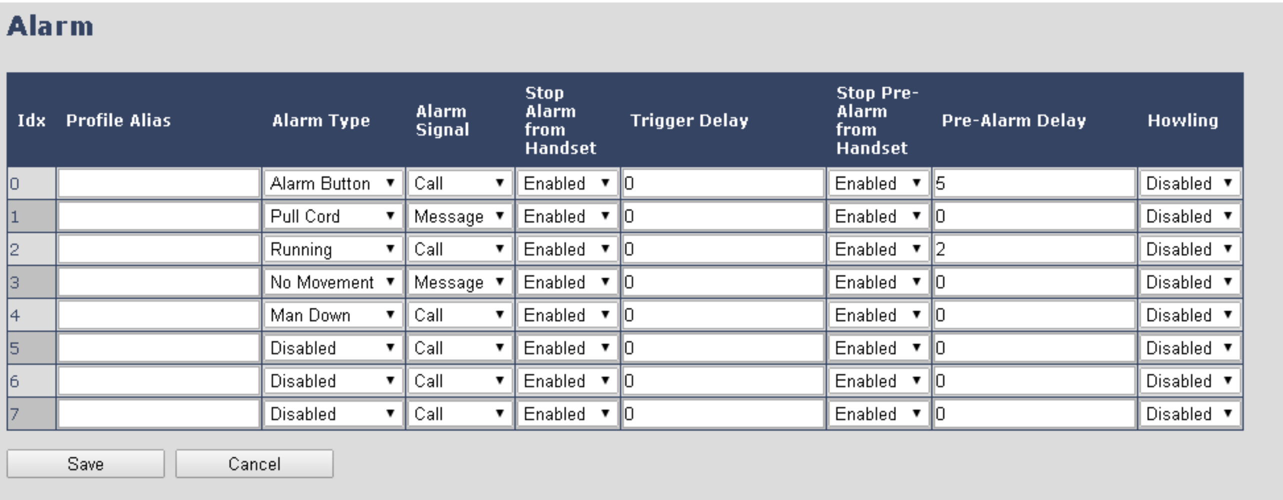
For detailed description on how to operate repeaters please see Repeater HOW-TO guide. Link is found in Appendix.

5.13 Alarm

In the Alarm Settings menu, it is controlled how an alarm appears on the handset. For example, if the handset detects "Man Down", then it is defined in this menu what alarm signal this type of alarm will send out and if a pre-alarm shall be signaled etc.

The Alarm is activated by a long press on the Alarm key (3 sec).

Screenshot



Alarm

Idx	Profile Alias	Alarm Type	Alarm Signal	Stop Alarm from Handset	Trigger Delay	Stop Pre-Alarm from Handset	Pre-Alarm Delay	Howling
0		Alarm Button	Call	Enabled	0	Enabled	5	Disabled
1		Pull Cord	Message	Enabled	0	Enabled	0	Disabled
2		Running	Call	Enabled	0	Enabled	2	Disabled
3		No Movement	Message	Enabled	0	Enabled	0	Disabled
4		Man Down	Call	Enabled	0	Enabled	0	Disabled
5		Disabled	Call	Enabled	0	Enabled	0	Disabled
6		Disabled	Call	Enabled	0	Enabled	0	Disabled
7		Disabled	Call	Enabled	0	Enabled	0	Disabled

Save Cancel

All configuration of the handset Alarm Settings is done from the base station. The concept is that on the "Alarm" page on the web server, eight different alarm profiles can be configured. Afterwards for each handset, it can be selected which of the configured alarm profiles, the given handset shall subscribe to. When this is done the selected alarm, profiles are sent to the handset.

See section 5.3.3: Edit handset.

PARAMETERS	DESCRIPTION

IDX	Indicates the index number of a specific alarm.
PROFILE ALIAS	An alias or user-friendly name to help identify the different profiles when selecting which profiles to enable for the individual handsets.
ALARM TYPE	The type of alarm is dependent of what kind of event that has triggered the alarm on the handset. The type of alarms supported is handset related. **RTX8632/RTX8633:** Alarm button **RTX8830:** Alarm button Man Down No Movement Running Pull Cord Emergency Button Disabled
ALARM SIGNAL	The way the alarm is signaled as it received on the handset. **Message:** A text message to an alarm server. **Call:** An outgoing call to the specified emergency number.
STOP ALARM FROM HANDSET	**Enable/Disable** the possibility to stop/cancel the alarm from the handset.
TRIGGER DELAY	The period from when the alarm has fired until the handset shows a pre-alarm warning. If 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 e.g. 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 the pre-alarm warning is shown until the actual alarm is signaled. The maximum value is 255.
HOWLING	**Enable/Disable** if howling shall be started in the handset, when the alarm is signaled. If disabled, only the configured signal is sent (call or message).

NOTE: This alarm feature is only available on some types of handsets (e.g. RTX8632, RTX8633 and RTX8830)
After configuration, the handset must be rebooted.

5.13.1 Use of Emergency Alarms

As described above, it can be configured if it shall 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 is considered stopped when the call is terminated. If it is not allowed to stop the alarm from the handset, it will not be possible to terminate the call from 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. If Howling is enabled, the handset will also start the Howling tone. The alarm will not stop until a call is made, and since auto answer mode is enabled, the emergency center can make the call, and the person with the handset does not have to do anything to answer the call, it will answer automatically. Again, the alarm is considered stopped, when the call is terminated with the same restrictions as for the Call alarm signal.

All type of alarms has the same priority. This means that once an alarm is active, it cannot be overruled by another alarm until the alarm 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, and therefore alarms with no pre-alarm, are given higher priority than alarms in pre-alarm state.

The Emergency Button could be an example of an alarm which would be configured without pre-alarm. Thus, when the Emergency Button is pressed you want to be sure the alarm is sent. However, if another alarm was already in pre-alarm state, it could potentially be cancelled, and if the Emergency Button alarm was ignored in this case, no alarm would be sent. This is the reason alarms with no pre-alarm, are given higher priority than alarms in pre-alarm state.

For detailed description on how to alarm please see Alarms HOW-TO guide. Link is found in Appendix.

5.14 Statistics

The statistic feature is divided into four administrative web pages, which can be access from any base.

1. System
2. Calls
3. Repeater
4. DECT data
5. Call quality

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

5.14.1 System data

The system data web is accessed by http://ip/SystemStatistics.html and data is organized in a table as shown in below example.

Screenshot

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
192.168.11.106 SME VoIP	0-00:54:22/ 0-18:09:25	0-00:53:43	0	0-00:00:00	15647	0	3	0	0
0.0.0.0	0-00:00:00/ 0-00:00:00	0-00:00:00	0	0-00:00:00	0	0	0	0	0
Sum	0-00:54:22/ 0-18:09:25	0-00:53:43	0	0-00:00:00	15647	0	3	0	0

The table is organized with headline row, data pr. base rows and with last row containing the sum of all base parameters.

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base station name from management settings
OPERATION/DURATION D-H:M:S	Operation is operation time for the base since last reboot. Duration is the operation time for the base since last reset of statistics, or firmware upgrade.
BUSY	Busy Count is the number of times the base has been busy.
BUSY DURATION D-H:M:S	Busy duration is the total time a base has been busy for speech (8 or more calls active).
SIP FAILED	Failed SIP registrations count the number of times a SIP registration has failed
HANDSET REMOVED	Handset removed count is the number of times a handset has been marked as removed
SEARCHING	Base searching is the number of times a base has been searching for its sync source
FREE RUNNING	Base free running is the number of times a base has been free running
DECT SOURCE CHANGED	Number of time a base has changed sync source

5.14.2 Free Running explained

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

The state Free running, tells the application that the base has not gotten any synchronization data from its synchronization source base station in the last 10 seconds.

The reason for this can be several:

1. The two bases are using the same DECT slots and can therefore not see each other.
2. Many simultaneous voice or data calls.
3. Suddenly change of environment (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 trigged, several recovery mechanisms are activated:

1. Move DECT slot to avoid using same DECT slot as its synchronization source base state.
2. Use information from all other base station, how they are seeing this base station in the DECT air.
   This is marked by changing to state Assisted lock

The state Assisted lock can be stabile for a long time and normally 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 station is not seen and no data is available for the assisted lock mechanism, the base station will change to a new state after 2 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.

5.14.3 Call data

The call data web is accessed by http://ip/CallStatistics.html and data are organized in a table as shown in below example.

Screenshot

Statistics

Export Clear

System / **Calls** / Repeater / DECT / Call quality

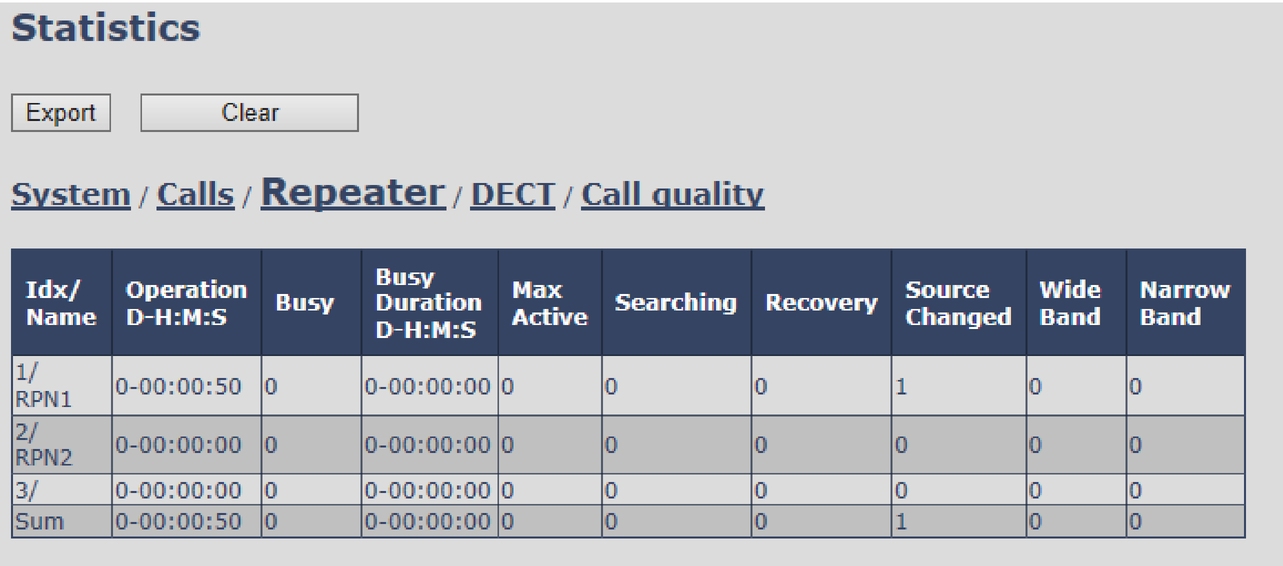
Base Station Name	Operation/ Duration D-H:M:S	Count	Dropped	No Response	Duration D-H:M:S	Active	Max Active	Codec G711U: G711A: G722: G726:	Handover Attempt Success	Handover Attempt aborted	Audio Not Detected
192.168.11.106 SME VoIP	0-00:57:36/ 0-18:12:39	0	0	0	0-00:00:00	0	0	0:0:0:0	0	0	0
0.0.0.0	0-00:00:00/ 0-00:00:00	0	0	0	0-00:00:00	0	0	0:0:0:0	0	0	0
Sum	0-00:57:36/ 0-18:12:39	0	0	0	0-00:00:00	0	0	0:0:0:0	0	0	0

The table is organized with headline row, data pr. base rows and with last row containing the sum of all base parameters.

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base station name from management settings
OPERATION TIME/DURATION	Total operation time for the base since last reboot or reset Duration is the time from data was cleared or system has been firmware upgraded.
COUNT	Counts number of calls on a base.
DROPPED	Dropped calls are the number of active calls that was dropped. E.g. if a user has an active call and walks out of range, the calls will be counted as a dropped call. An entry is stored in the syslog when a call is dropped.
NO RESPONSE	No response calls are 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 no response.
DURATION	Call duration is total time that calls are active on the base.
ACTIVE	Active call shows how many active calls that are active on the base station (Not active DECT calls, but active calls). On one base there can be up to 10 active calls in single mode and 8 in Multi Cell mode.
MAX ACTIVE	Maximum active calls are the maximum number of calls that has been active at the same time.
CODECS	Logging and count of used codec types on each call.
HANDOVER ATTEMPT SUCCESS	Counts the number of successful handovers.
HANDOVER ATTEMPT FAILED	Counts the number of failed handovers.
AUDIO NOT DETECTED	Counts the number of times where audio connection was not established.

5.14.4 Repeater data

Screenshot



Idx/ Name	Operation D-H:M:S	Busy	Busy Duration D-H:M:S	Max Active	Searching	Recovery	Source Changed	Wide Band	Narrow Band
1/ RPN1	0-00:00:50	0	0-00:00:00	0	0	0	1	0	0
2/ RPN2	0-00:00:00	0	0-00:00:00	0	0	0	0	0	0
3/	0-00:00:00	0	0-00:00:00	0	0	0	0	0	0
Sum	0-00:00:50	0	0-00:00:00	0	0	0	1	0	0

The table is organized with headline row, data pr. base rows and with last row containing the sum of all base parameters.

PARAMETERS	DESCRIPTION
IDX/NAME	Base IP address and base station name from management settings
OPERATION D-H:M:S	Total operation time for the repeater since last reboot or reset Duration is the time from data was cleared or system has been firmware upgraded.
BUSY	Busy Count is the number of times the repeater has been busy.
BUSY DURATION D-H:M:S	Busy duration is the total time a repeater has been busy for speech (5 or more calls active).
MAX ACTIVE	Maximum active calls are the maximum number of calls that has been active at the same time.
SEARCHING	Repeater searching is the number of times a repeater has been searching for it's sync source
RECOVERY	In case the sync source is not present anymore the repeater will go into lock on another base or repeater and show recovery mode
DECT SOURCE CHANGED	Number of time a repeater has changed sync source
WIDE BAND	Number of wideband calls on repeaters
NARROW BAND	Number of narrow band calls on repeaters

5.14.5 DECT data

The DECT data web is accessed by http://ip/DectStatistics.html and data is organized in a table as shown in below example.

Screenshot

Statistics

Export Clear

System / Calls / Repeater / DECT / Call quality

	Slot0	Slot1	Slot2	Slot3	Slot4	Slot5	Slot6	Slot7	Slot8	Slot9	Slot10	Slot11
Frequency0	1666	1542	1595	1590	1647	1645	1646	1635	1616	1541	1543	1622
Frequency1	1159	1184	1205	1170	1191	1164	1184	1151	1164	1074	1205	1199
Frequency2	1234	1193	1213	1199	1260	1227	1209	1163	1211	1224	1226	1294
Frequency3	1059	1175	1153	1128	1091	1171	1069	1119	1176	1141	1123	1101
Frequency4	933	1044	984	1029	1022	996	1012	1004	996	957	937	1031
Frequency5	1060	1090	986	962	1030	983	975	980	947	985	982	987
Frequency6	1073	981	1017	1013	1066	959	1013	974	978	992	1019	962
Frequency7	1049	1060	981	995	1065	1082	1005	1033	1017	949	1085	1068
Frequency8	968	972	941	968	988	964	941	963	975	912	934	961
Frequency9	1349	1398	1292	1336	1279	1313	1338	1293	1348	1248	1331	1320

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

5.14.6 Call quality

The Call quality web is accessed by http://ip/CallQuality.html and the data is organized in a table as shown in below example.

Screenshot


Statistics

Export | Clear

System / Calls / Repeater / DECT / **Call quality**

Base Station Name	Type	Call count	Local/remote side	Jitter [ms]	Round trip latency [ms]	Packet loss [%]	R-value	MOS-value
192.168.11.106 SME VoIP	Call	0	Local	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00
			Remote	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00
	Relay conn	0	Local	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00
			Remote	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00
0.0.0.0	Call	0	Local	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00
			Remote	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00
	Relay conn	0	Local	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00
			Remote	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.000 Max: 0.000 Avg: 0.000	Min: 0.0 Max: 0.0 Avg: 0.0	Min: 0.00 Max: 0.00 Avg: 0.00	Min: 0.00 Max: 0.00 Avg: 0.00

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base station name from management settings
TYPE	Call: Relay conn:
CALL COUNT	Count the number of calls
LOCAL/REMOTE SIDE	Local: Remote:
JITTER[MS]	Measures how the RTP packets are received, the lower the Jitter is the better
ROUND TRIP LATENCY [MS]	Measures the time it takes for RTP packets to reach it destination.
PACKET LOSS [%]	Percentages of packets lost.
R-VALUE	A way to measure call quality, from 0-120

USER SATISFACTION LEVEL	MOS	R-Factor
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

PARAMETERS	DESCRIPTION
MOS-VALUE	**MOS** measures subjective call quality for a call. MOS scores range from 1 for unacceptable to 5 for excellent. VOIP calls often are in the 3.5 to 4.2 range See table above.

5.15 Diagnostics

This page provides information about the Ethernet connection to each base station and Extension.

Screenshot

Diagnostics

Base stations / Extensions

Base Station Name	Active Dect Ext (Mm/Ciss/CcOut/CcIn)	Active Dect Rep (Mm/Ciss/CcOut/CcIn)	Active RTP (Lcl/Rx BC)	Active Relay RTP (Lcl/Remote)	Latency [ms] (Avg.Min/Average/Avg.Max)
192.168.11.106 SME VoIP	0/0/0/0	0/0/0/0	0/0	0/0	NA
192.168.11.169 SME VoIP	0/0/0/0	0/0/0/0	0/0	0/0	2/2/3
Sum	0/0/0/0	0/0/0/0	0/0	0/0	2/2/3

PARAMETERS	DESCRIPTION
BASE STATION NAME	Base IP address and base station name from management settings
ACTIVE DECT EXT (MM/CISS/CCOUT/CCIN)	Number of active DECT MAC connections to extensions in the different base stations. Types of connection is (mm/Ciss/CcOut/CcIn)
ACTIVE DECT REP (MM/CISS/CCOUT/CCIN)	Number of active DECT MAC connections to repeaters in the different base stations. Types of connection is (mm/Ciss/CcOut/CcIn)
ACTIVE RTP (LCL/RX BC)	Number of active RTP Streams used. Types of stream (Local RTP stream/Broadcast Receive RTP stream)
ACTIVE RELAY RTP (LCL/REMOTE)	Number of active RTP Relay Streams used. Types of stream (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)

Screenshot

Diagnostics

Base stations / **Extensions**

Idx	No of HS restarts	Last HS restart (dd/mm/yyyy hh:mm:ss)
1	0	01/01/1970 00:00:00

PARAMETERS	DESCRIPTION
IDX	Extension Index number
NO OF HS RESTARTS	Number of times that the Handset have restarted
LAST HS RESTART (DD/MM/YYYY HH:MM:SS)	Date and time of the last time the Handset have restarted

5.16 Settings – Configuration File Setup

This page provides non-editable information showing the native format of entire SME VoIP Configuration parameter settings. The **settings** format is exactly what is used in the configuration file. The configuration file is found in the TFTP server.
The filename for the configuration server is **<MAC_Address>.cfg**. The configuration file is saved in the folder **/Config** in the TFTP sever.

There are three ways to edit the configuration file or make changes to the **settings** page:

- Using the SME VoIP Configuration interface to make changes. Each page of the web interface is a template for which the user can customize settings in the configuration file.
- Retrieving the relevant configuration file from the TFTP and modify and enter new changes. This should be done with an expert network administrator.
- Navigate to the settings page of the VoIP SME Configuration interface > copy the contents of settings > save them to any standard text editor e.g. notepad > modify the relevant contents, make sure you keep the formatting intact > Save the file as **<Enter_MAC_Address_of_RFP>.cfg** > upload it into the relevant TFTP server.

An example of contents of settings is as follows:

```
~RELEASE=BEATUS_FP_V0400_B0001
~System Mode=51/51
%GMT_TIME_ZONE%:0x06
%COUNTRY_VARIANT_ID%:0x12
%COUNTRY_REGION_ID%:0x00
%TIMEZONE_BY_COUNTRY_REGION%:0x01
%DST_BY_COUNTRY_REGION%:0x01
%DST_ENABLE%:0x02
%DST_FIXED_DAY_ENABLE%:0x00
%DST_START_MONTH%:0x03
%DST_START_DATE%:0x00
………
```

For detailed description on how to use provisioning please see Provisioning HOW-TO guide. Link is found in Appendix.

5.17 Sys log

This page shows live feed of system level messages of the current base station. The messages the administrator see here depends on what is configured at the Management settings. The Debug logs can show only **Boot Log** or **Everything** that is all system logs including boot logs.
The Debug log is saved in the file format **<Time_Stamp>b.log** in a relevant location in the TFTP server as specified in the upload script.

A sample of debug logs is as follows:

```
0101000013 [N](01):DHCP Enabled
0101000013 [N](01):IP Address: 192.168.10.101
0101000013 [N](01):Gateway Address: 192.168.10.254
0101000013 [N](01):Subnet Mask: 255.255.255.0
0101000013 [N](01):TFTP boot server not set by DHCP. Using Static.
0101000013 [N](01):DHCP Discover completed
0101000013 [N](01):Time Server: 192.168.10.11
0101000013 [N](01):Boot server: 10.10.104.63 path: Config/ Type: TFTP
0101000013 [N](01):RemCfg: Download request of Config/00087b077cd9.cfg from 10.10.104.63 using TFTP
0101000014 [N](01):accept called from task 7
0101000014 [N](01):TrelAccept success [4]. Listening on port 10010
0101000019 [N](01):RemCfg: Download request of Config/00087b077cd9.cfg from 10.10.104.63 using TFTP
0101000019 [W](01):Load of Config/00087b077cd9.cfg from 10.10.104.63 failed
```

To dump the log simply copy and page the full contents.

5.18 SIP Logs

This page shows SIP server related messages that are logged during the operation of the SME system. The full native format of SIP logs is saved in the TFTP server as **<MAC_Address><Time_Stamp>SIP.log**
These logs are saved in 2 blocks of 17Kbytes. When a specific SIP log is fully dumped to one block, the next SIP logs are dumped to the other blocks.

An example of SIP logs is shown below:

```
.....
Sent to udp:192.168.10.10:5080 at 12/11/2010 11:56:42  (791 bytes)
REGISTER sip:192.168.10.10:5080 SIP/2.0
Via: SIP/2.0/UDP 192.168.10.101:5063;branch=z9hG4bKrlga4nkuhimpnj4.qx
Max-Forwards: 70
From: <sip:Ext003@192.168.10.10:5080>;tag=3o5l314
To: <sip:Ext003@192.168.10.10:5080>
Call-ID: p9st.zzrfff66.ah8
CSeq: 6562 REGISTER
Contact: <sip:Ext003@192.168.10.101:5063>
Allow: INVITE, CANCEL, BYE, ACK, REGISTER, OPTIONS, REFER, SUBSCRIBE, NOTIFY, MESSAGE, INFO, PRACK
Expires: 120
User-Agent: Generic-DPV-001-A-XX(Generic_SIPEXT2MLUA_v1)
Content-Type: application/X-Generic_SIPEXT2MLv1
Content-Length: 251
.....
```

To dump the log simply copy and page the full contents.

6 Appendix – How-To setup a Multi Cell System

This chapter we describe how to setup a multi cell system, add and synchronize one or multiple base stations to the network.

NOTE: It is possible to have RTX8660 and RTX8660 in the same chain.

6.1 Adding Base stations

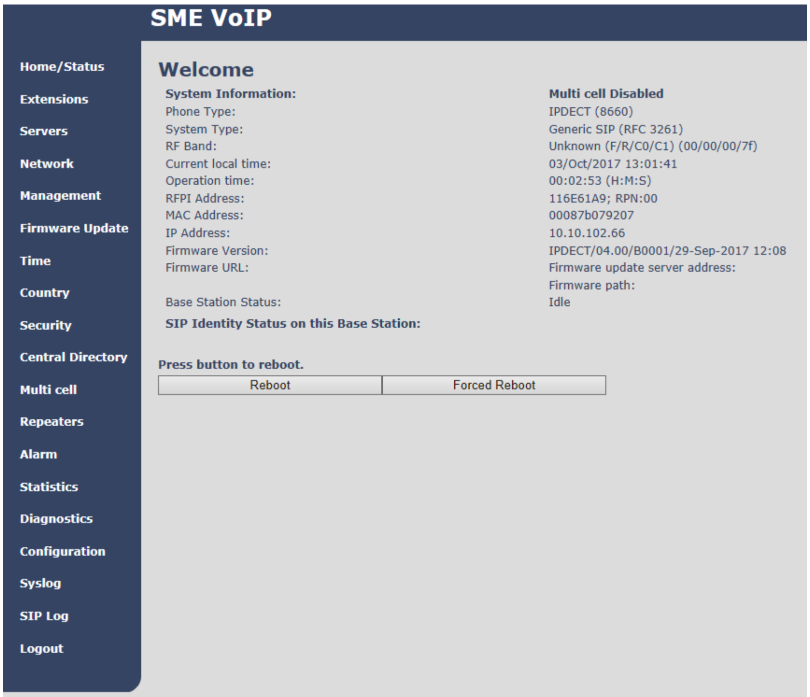
Here are the recommended steps to add Base stations to network:

STEP 1:
Connect the Base station to a private network via standard Ethernet cable.

STEP 2:
Use one of the two methods to determine the base station IP address.
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.
Use the IPdect feature.

STEP 3:
Open browser on the computer and type in the IP address of the base. Press “Enter” to access the base Login to base station.
Once you have authenticated, the browser will display front end of the SME Configuration Interface. The front end will show relevant information of the base station.

Screenshot

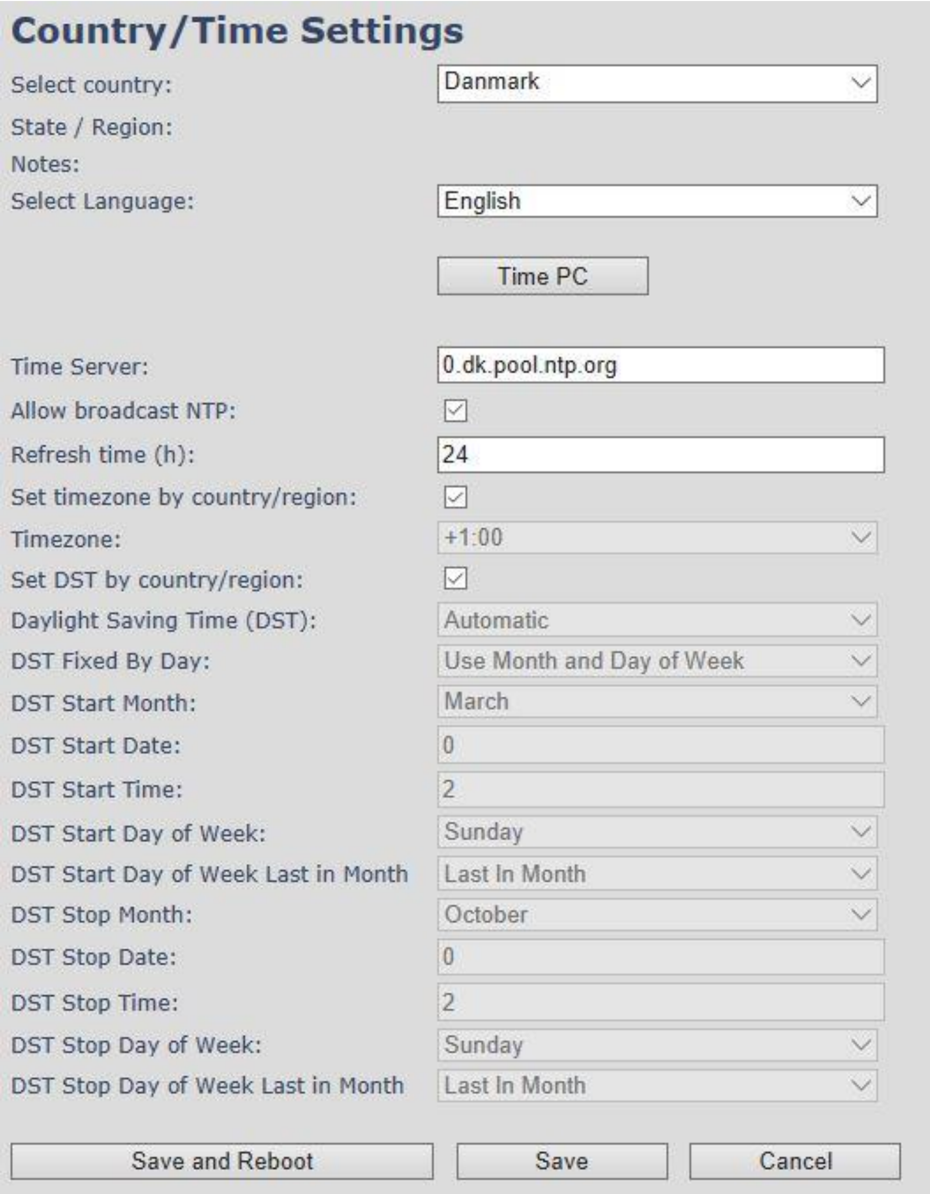


6.1.1 Country and Time Server Setup

STEP 4:
Navigate to the Country page and configure it country and time settings.
Use the PC time feature or enter the relevant parameters on this page and press the **Save and Reboot** button. Make sure there is contact to the "Time server" otherwise the Multi-cell feature will not work.

You can verify whether the Time server is reachable if rebooting the base station by verifying the correct Time Server IP address is still in place.

Screenshot



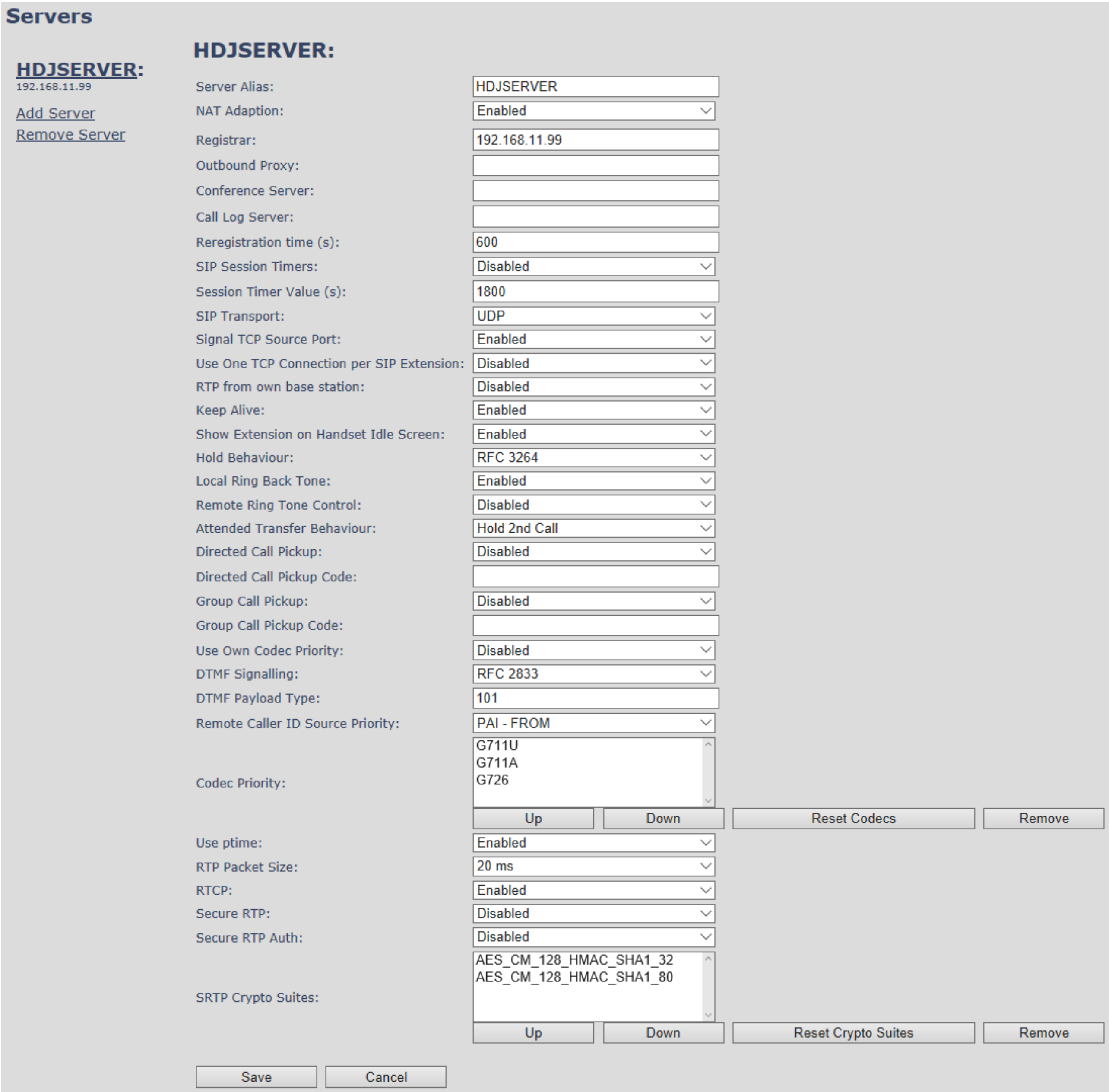
6.1.2 SIP Server (or PBX Server) Setup

STEP 5:

Create the relevant SIP server (or PBX Server) information in the system. Each service provider/customer should refer SIP server vendor on how to setup SIP servers.

a. Click the link "**Server"** at the left-hand column of home page, you can add your SIP server for base station use.
b. Next, from the Server page, click on the **Add Server** URL and enter the relevant SIP server information (an example is shown below).
c. Choose "**Disabled**" on NAT adaption parameter if NAT function of the SIP aware router is not enabled. Enter the relevant parameters based on the description in the table below. Click **Save**.

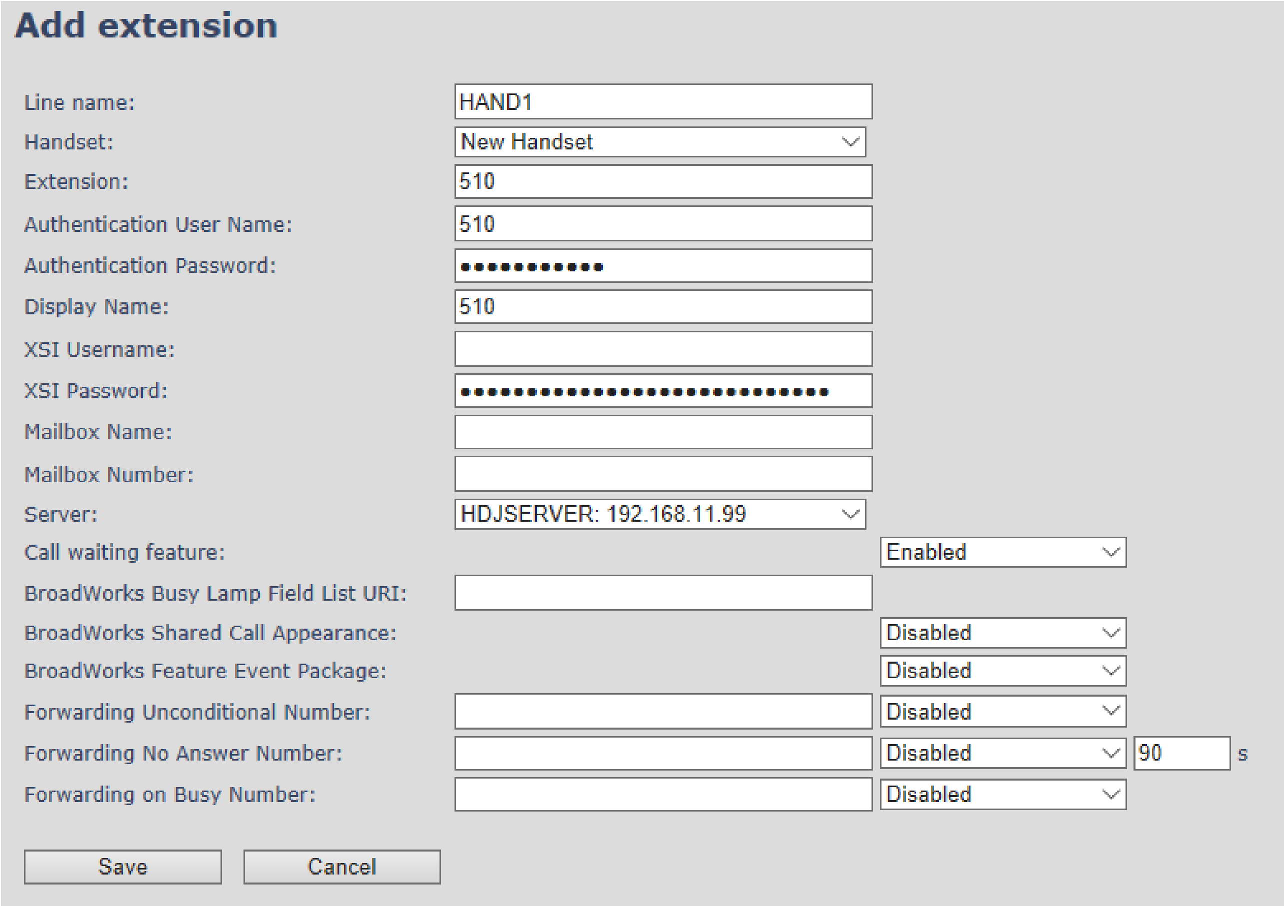
Screenshot



6.1.3 Add an extension

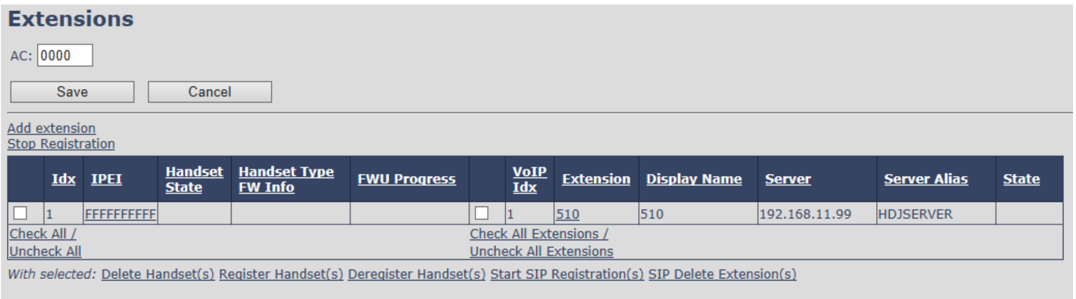
STEP 6:
And an extension before you move to the Multi Cell page. Go to Extensions – Add Extension.
Fill out the extension data, and save

Screenshot



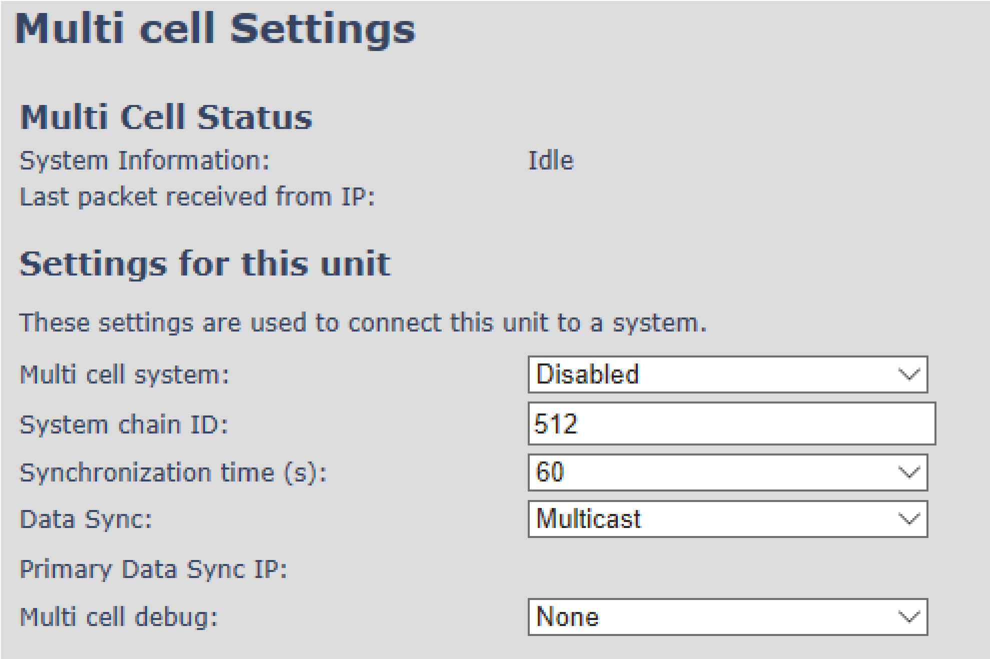
You will now see the extension on the extension page, you do not need to fully register the extension

Screenshot



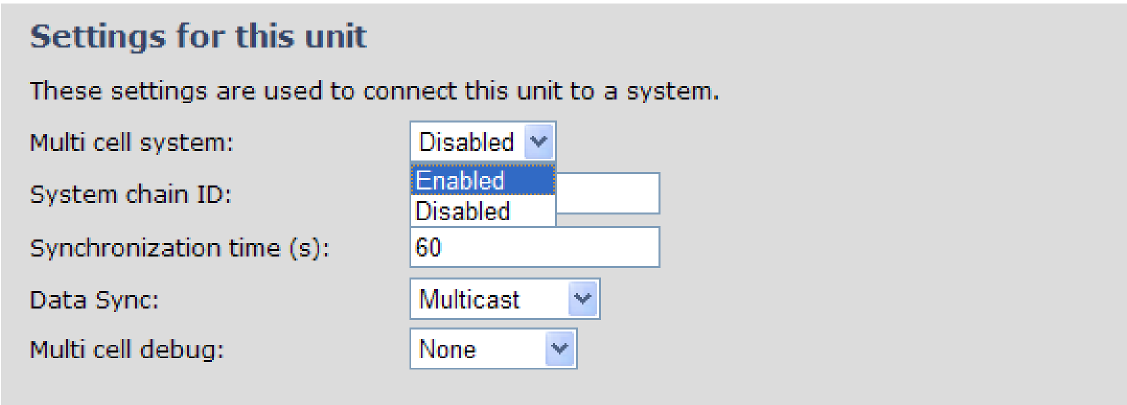
STEP 7:
Click on Multi Cell URL link in the SME VoIP Configuration to view the current Multi cell settings status of the current base station. Brand new base stations have **Multi cell system** feature disabled by default

Screenshot



STEP 8:
Next, the system administrator needs to create and Enable Multi Settings profile for the current base station. On the **Multi Cell settings** Page, choose **Enable** option from the drop-down menu of the **Multi cell system** parameter. Enable the **Multi cell debug** option if the system administrator wants some Multi-cell related logs to be catalogued by the system.

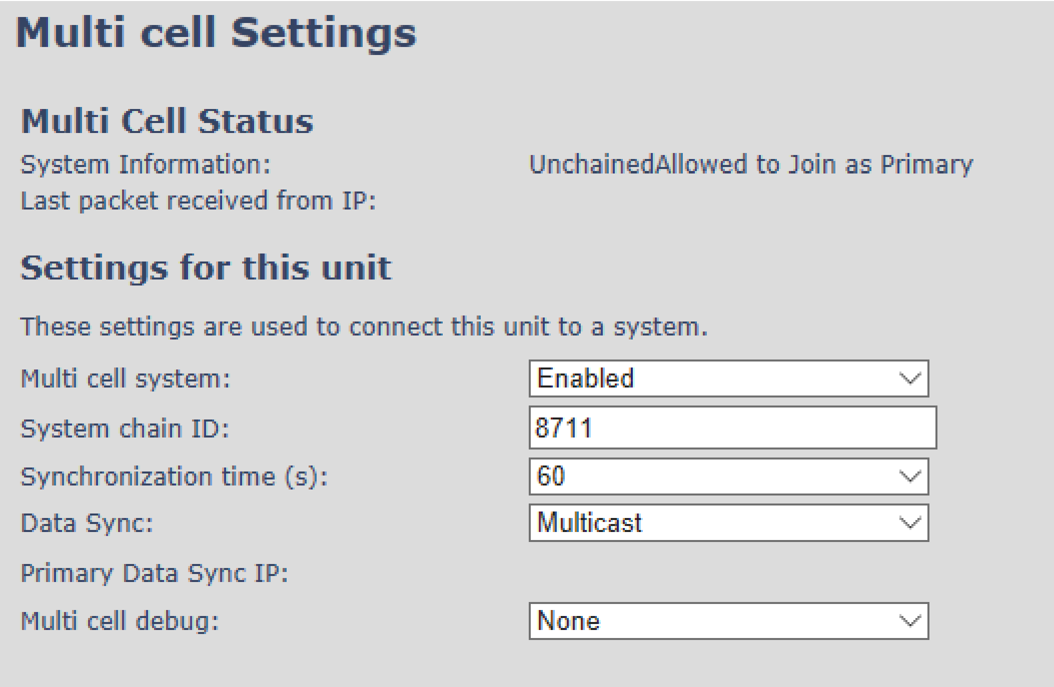
Screenshot



STEP 9:
On the same **Multi Cell Settings** page > Enter the relevant values for **System chain ID** and **Synchronization time (s)** respectively. The **System chain ID** is a geographically unique DECT cell identity allocated to bridge several base stations together in a chain. An example is **55555**. The **Synchronization time (s)** parameter is defined as window/period of time in seconds a specific base station synchronizes to the master base station unit (by default 60).

NOTE: Do NOT use a chain ID similar to an extension.

Screenshot



Click on **Save** button to keep modified changes of multi cell settings into the base station.

Screenshot

NOTE: That after you save the System information changes status to "Unchained Allowed to Join as Primary"

NOTE: The Multi Cell data synchronization ONLY works when the relevant **Time Server** is set in the system before Server/Subscriber profile is added or created. Refer to **STEP 5**.

IMPORTANT: Base stations must be rebooted after the time server has been set.

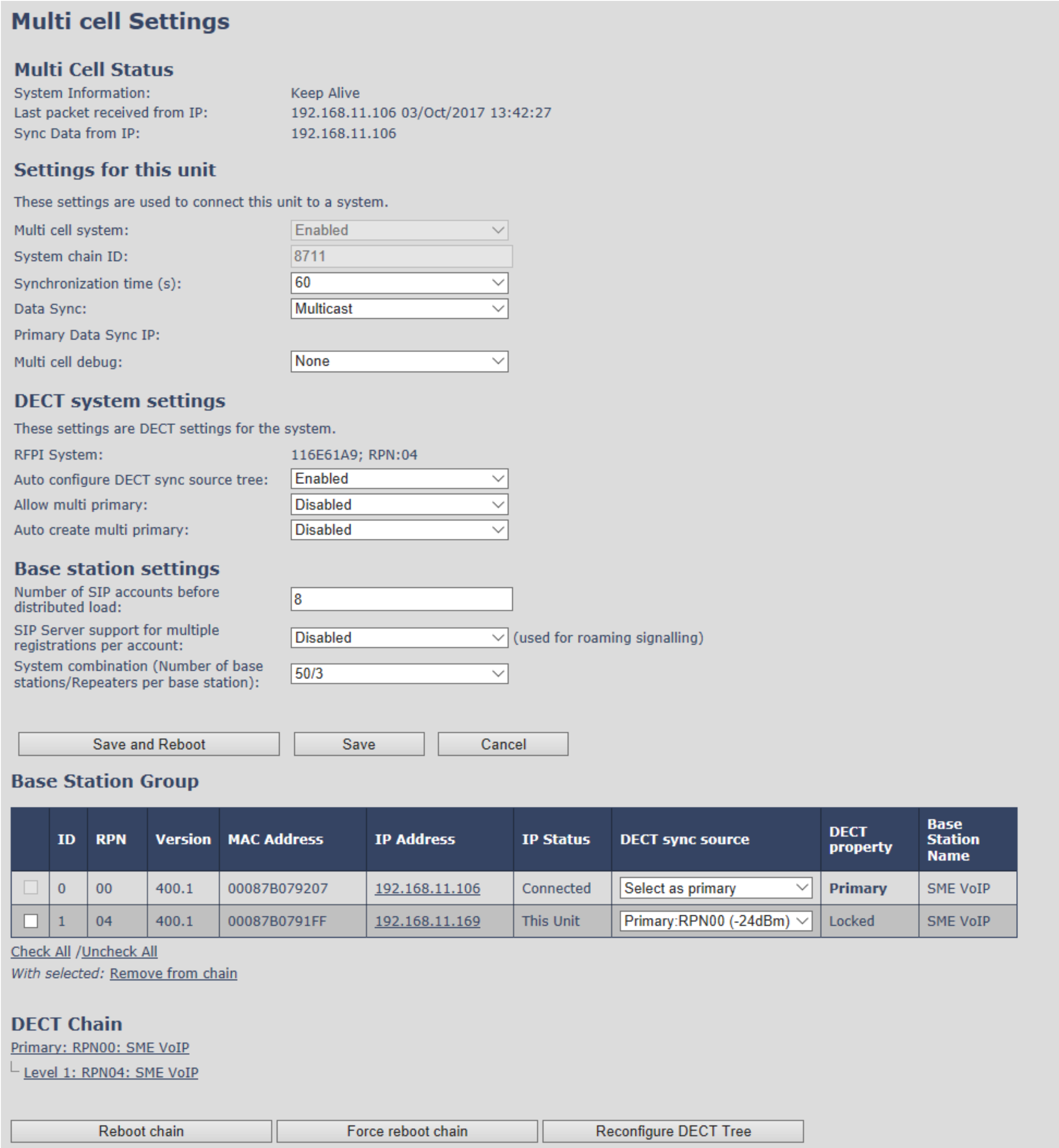
STEP 10:
Logon to the base station that you want to connect to the Multi Cell system.

STEP 11:
Navigate to the multi Cell page and "Enable" Multi Cell system and enter the Chain ID that you used on the first base Station.

STEP 12:
Press Save and Reboot

IMPORTANT: It takes up to 5 minutes (synchronization time) to add a new base station to a Multi Cell System.

Screenshot



STEP 13:

To add more Base Stations, repeat **STEP 9-12**.

7 Appendix - Adding Extensions

This section describes how to register the wireless handset to a Multi Cell system.

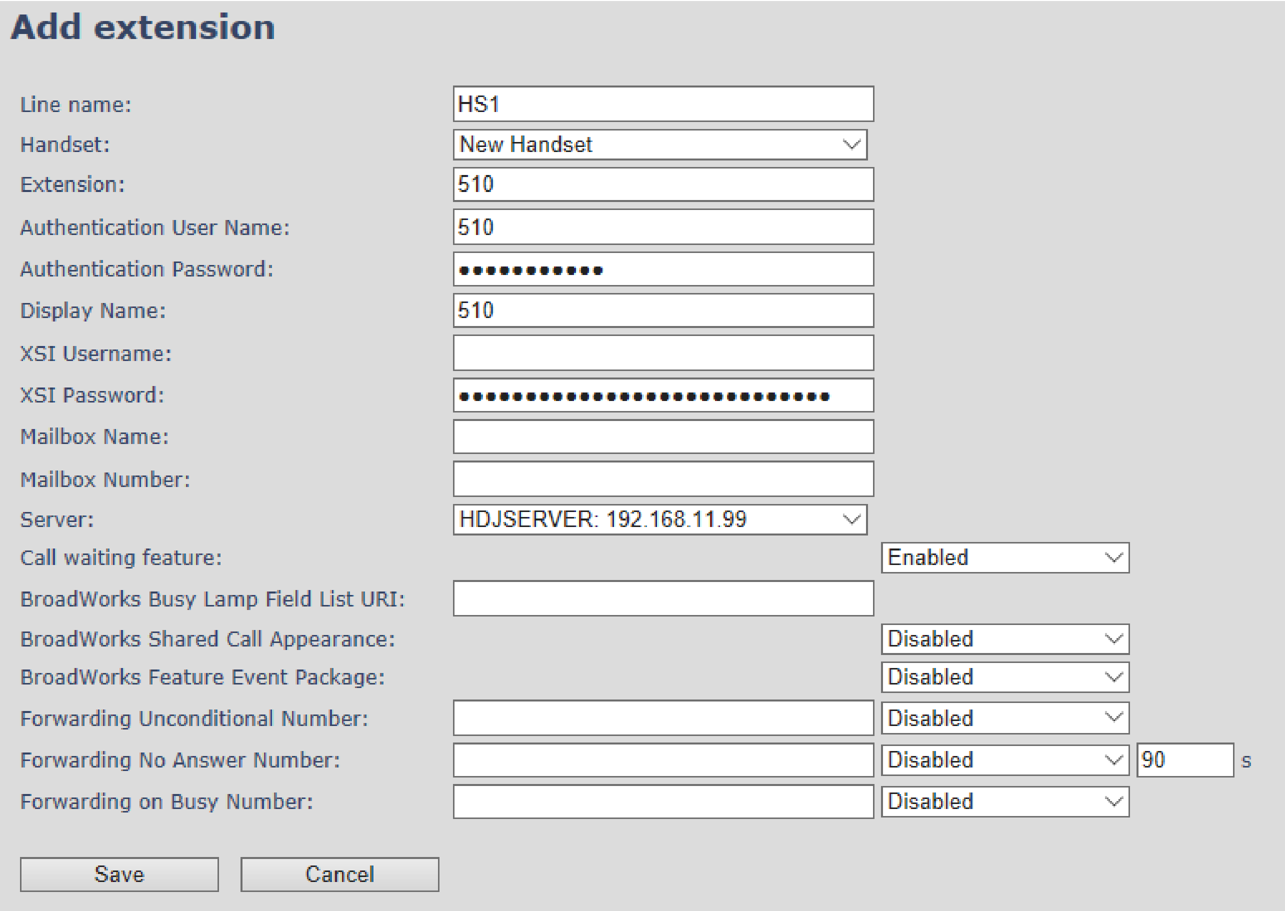
NOTE: Minimum one server must be registered to the base (system), otherwise a handset cannot be registered to the system. Please see chapter 6.1.2.

STEP 1:
Login to a base station.

STEP 2:
'Select "Extensions" URL and click "Add extension" link

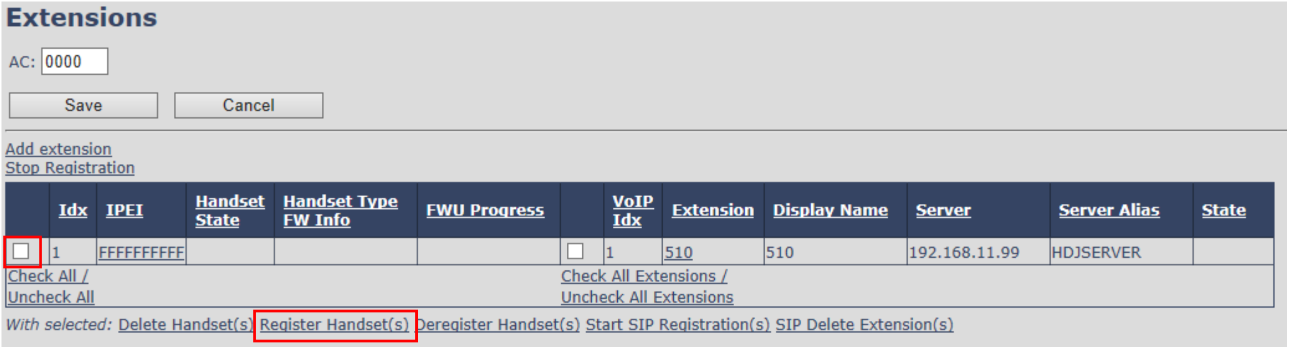
STEP 3:
Fill out the form and click "Save". In the example below, we add the extension "510" and this SIP account got the same number as "Authentication User Name", "Password" and "Display Name".

Screenshot



STEP 4:
In the handset and extensions list set a Check mark on the handset Idx, which you want to register and click "Register handset (s)". The base is now open (ready state) for handset registrations for 5 minutes.

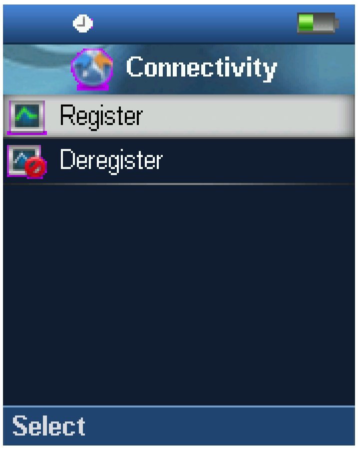
Screenshot



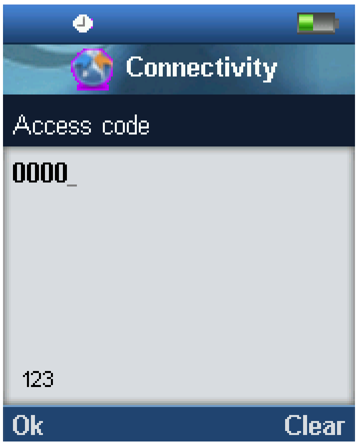
STEP 5:
Start the registration procedure on the handset by following step “a” to “d” below.

a) Select main menu “Connectivity”

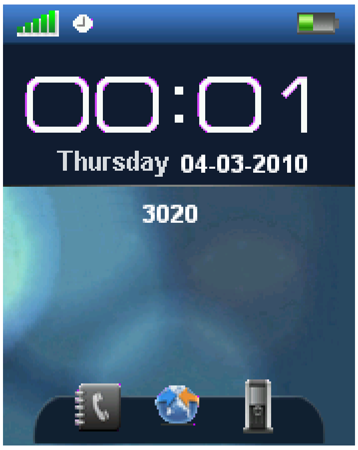
b) Select menu ”Register”



c) Type in the “AC code” and press “OK” to start the registration. The default AC code is “0000”.



d) After a while the handset is registered, and the idle display is shown



STEP 6:
confirm the registration from the unique handset IPEI is displayed in column "IPEI" when the handset is successfully registered.

NOTE: The web page must be manually updated by pressing "F5" to see that the handset is registered; otherwise the handset IPEI (International Portable Equipment Identity) isn't displayed on the web page.

Screenshot

STEP 7:
Confirm the SIP registration by SIP State in right column.

NOTE: The web page must be manually updated by pressing "F5" to see that the handset is SIP registered; otherwise the handset SIP state isn't displayed on the web page.

Repeat **STEP 2-7** for each handset you want to register.

8 Appendix - Firmware Upgrade Procedure

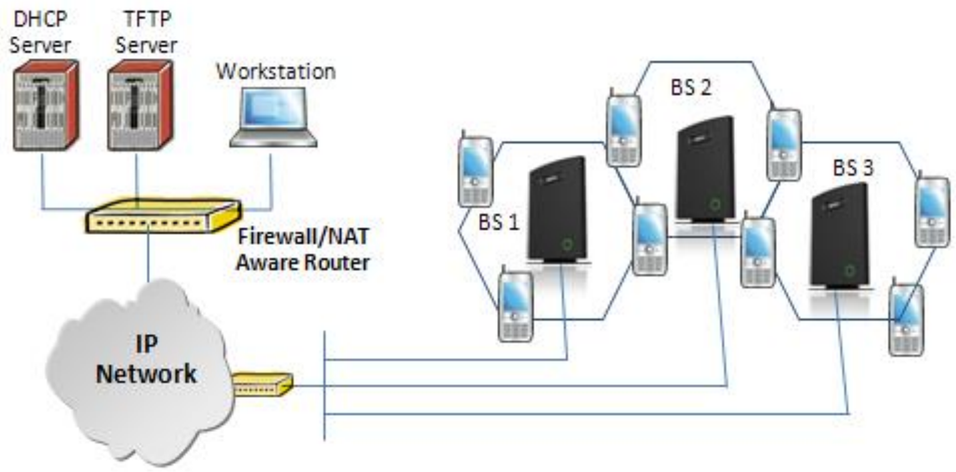
This step-by-step chapter describes how to upgrade or downgrade base station(s) and/or handset(s) / repeater (s) to the relevant firmware provided by RTX.

8.1 Network Dimensioning

In principle, several hardware and software components should be available or be satisfied before base station/handset update can be possible.

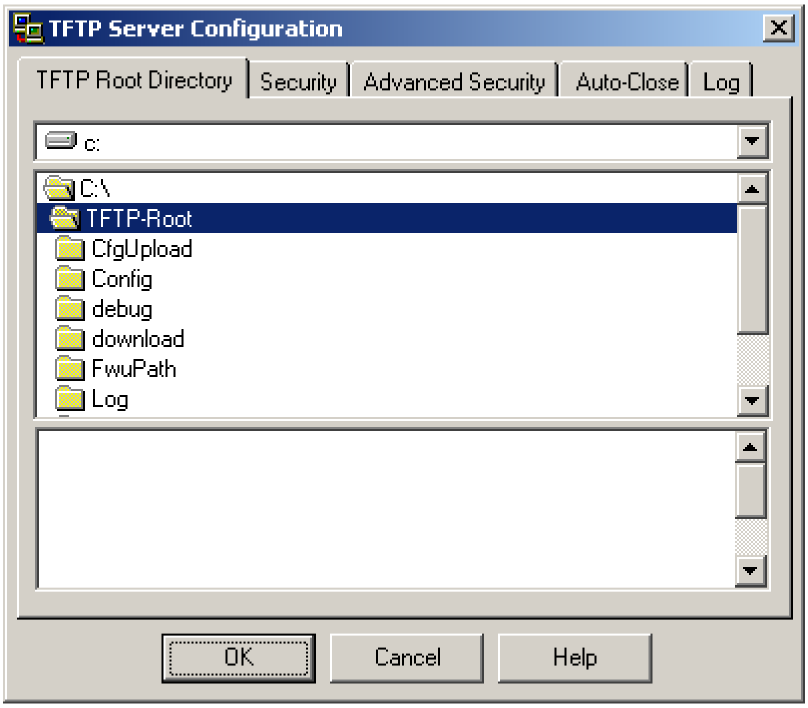
The minimum hardware and software components that are required to be able update via TFTP include the following (but not limited to):

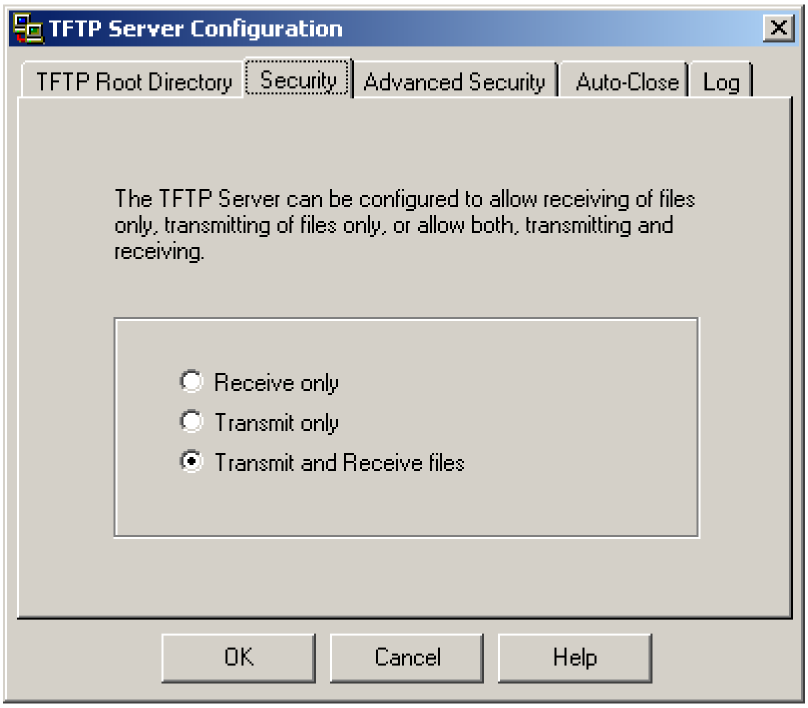
- Handsets
- Base stations
- TFTP Server (Several Windows and Linux applications are available)
- DHCP Server (Several Windows and Linux applications are available)
- Workstation (e.g. Normal terminal or PC)
- Any standard browser (e.g. Firefox)
- Public/Private Network



8.2 TFTP Configuration

This section illustrates TFTP Server configuration using “SolarWinds” vendor TFTP Server. Create the following relevant folders as shown in the snap shots and choose defaults settings for the remaining options and save.





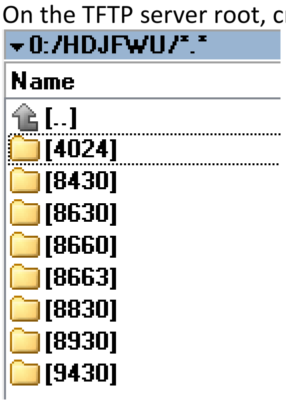
NOTE: If TFTP server timeout settings are too short firmware upgrade might not complete. Recommended time out setting is more than 3 seconds.

8.3 Create Firmware Directories

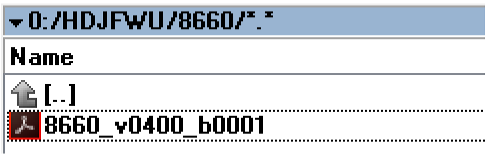
The admin from the service provider's side must create the relevant firmware directory in the server where both old and new firmware(s) can be placed in it. (See the STEP above)

8.3.1 Base:

On the TFTP server root, create directory's as in screenshot.



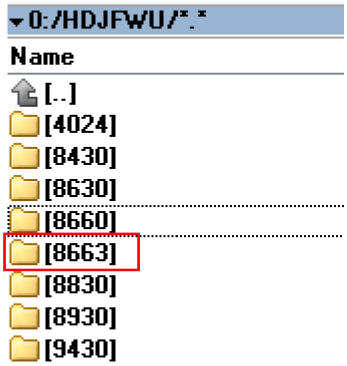
Copy Base station firmware to the named directory.



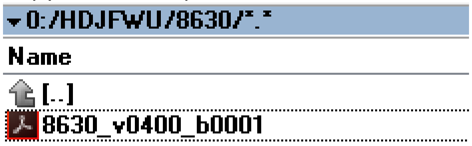
IMPORTANT: The **8663** directory name cannot be changed.

8.3.2 Handsets/Repeaters:

On the TFTP server root, create directory “8430” or “8630” or “8830” or “8930” or “4024” depending on type.



Copy handset/repeater firmware to the named directory of each model.

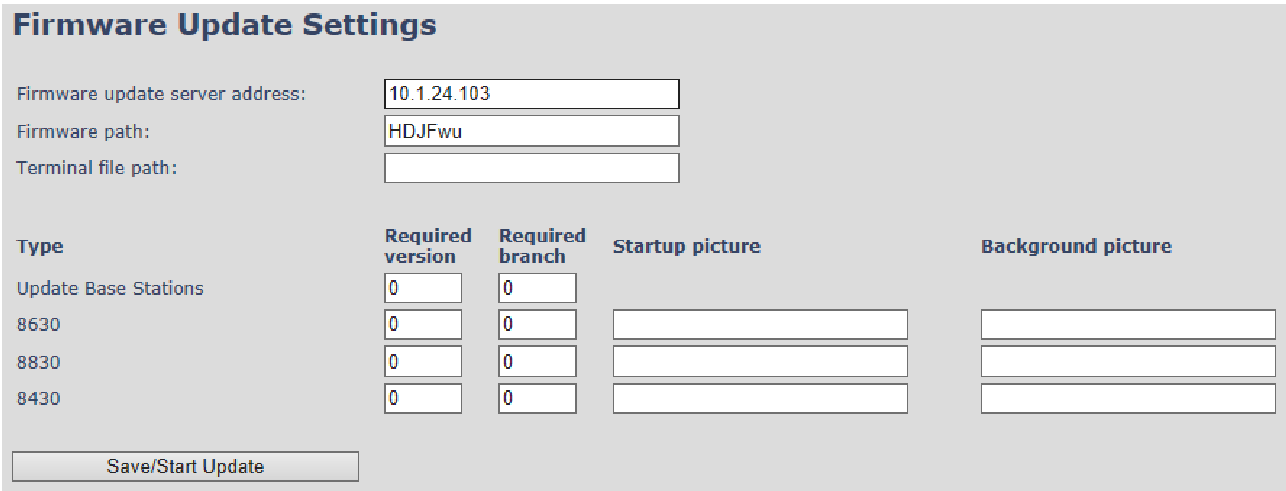


IMPORTANT: The **8430, 8630,8830 and 8930** directory names cannot be changed.

8.4 Handset Firmware Update Settings

Scroll down and Click on Firmware Update URL link in the SME VoIP Configuration Interface to view the Firmware Update Settings page.

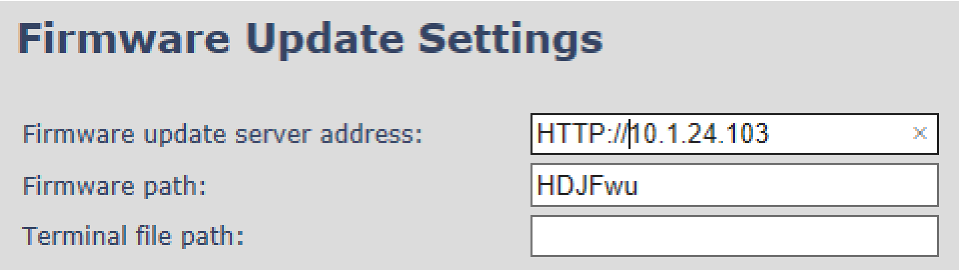
Screenshot



Type IP address and firmware path followed by save.

For Http download the firmware update server settings must be entered as follows:

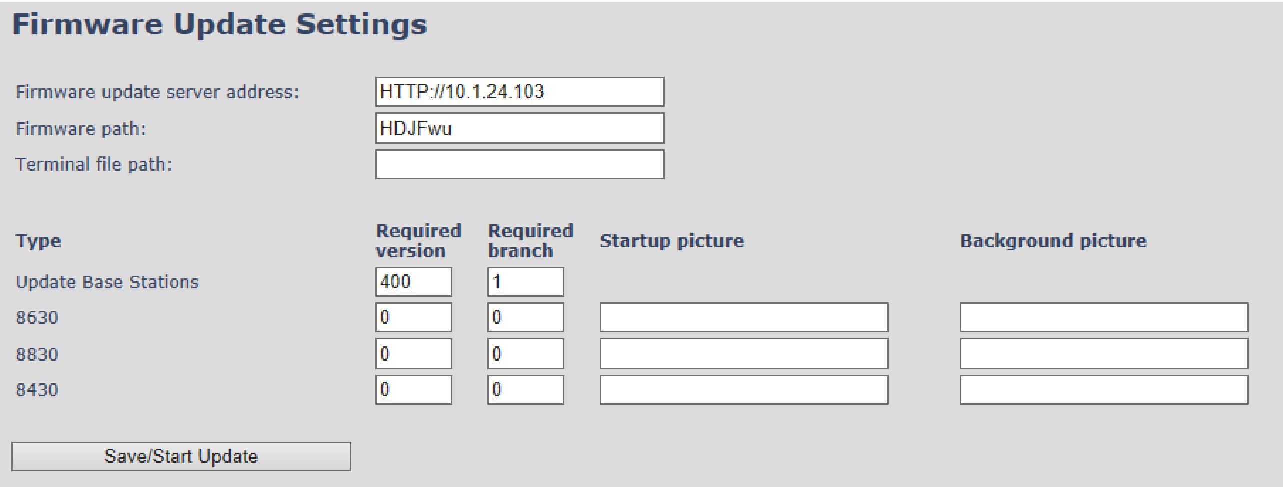
Screenshot



8.5 Handset(s) and Repeater Firmware Upgrade

On the **Firmware Update Settings** page enter the relevant handset/repeater firmware for each type and Branch name (e.g. 440 for v440 for Required Version) and (e.g.01 for Branch 01 for Required Branch) to upgrade or downgrade > press **Save** button to initialize the process of updating all handsets.

Screenshot



NOTE: To disable handset/repeater firmware process type version 0 in the required version field, followed by the save button. It is recommended to use version 0 after all units are upgraded.

NOTE: For handset TFTP/HTTP download only one handset type can be downloaded at the same time. In case two handset models are defined for fwu at the same time fwu will fail.

8.5.1 Monitor handset firmware upgrade

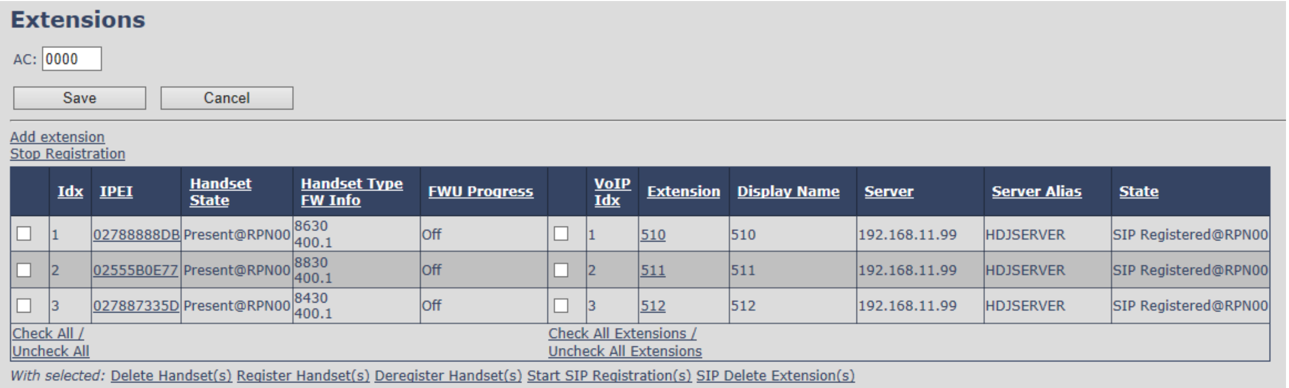
Handset firmware upgrade status is monitored on the handset extensions page, FWU Process Colum.

If the status says "Off" it means that the Required Version and Branch is set to "0" as it should be unless you're in process of updating/downgrading the firmware.

The firmware 5pgrade/Downgrade process have 6 states

- Initializing
- In progress (% from 0-100)
- Verifying (% 0-100)
- Waiting for charger (The handset must be placed in charge and NOT removed until it reboots)
- Complete
- Off

Screenshot



Extensions

AC: 0000

Save Cancel

Add extension
Stop Registration

	Idx	IPEI	Handset State	Handset Type FW Info	FWU Progress		VoIP Idx	Extension	Display Name	Server	Server Alias	State
☐	1	02788888DB	Present@RPN00	8630 400.1	Off	☐	1	510	510	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	2	02555B0E77	Present@RPN00	8830 400.1	Off	☐	2	511	511	192.168.11.99	HDJSERVER	SIP Registered@RPN00
☐	3	027887335D	Present@RPN00	8430 400.1	Off	☐	3	512	512	192.168.11.99	HDJSERVER	SIP Registered@RPN00

Check All / Uncheck All   Check All Extensions / Uncheck All Extensions

With selected: Delete Handset(s) Register Handset(s) Deregister Handset(s) Start SIP Registration(s) SIP Delete Extension(s)

Handset firmware update time from start to complete takes 20- 40 minutes.

8.5.2 Monitor Repeater firmware upgrade

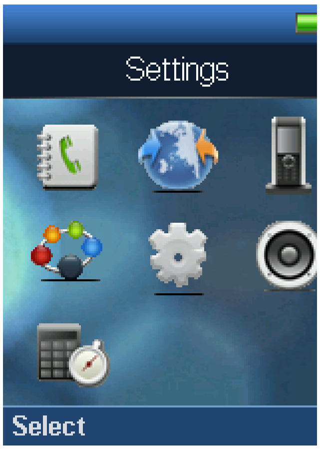
Repeater firmware upgrade status is monitored on the Repeater page, right column.

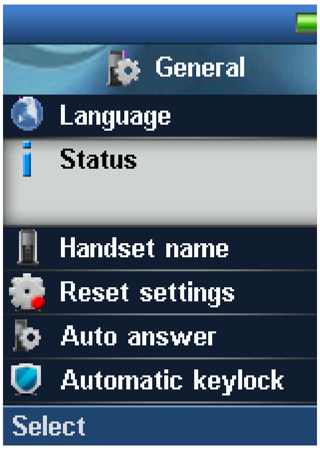
Repeater firmware upgrade time from start to complete takes minimum 20 minutes.

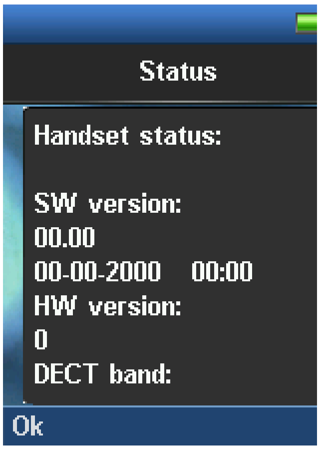
8.5.3 Verification of Firmware Upgrade

The firmware upgrade is confirmed by the FWU Progress status in the FWU Colum on the handset extension list or repeater list. The “FWU info” column contains the software version and the “FWU Progress” column contains the status. In case status is “Complete”, the unit is firmware upgraded.

Alternatively, the handset firmware can be verified from the Handset **Menu** by navigate to **Settings** > Scroll down to **Status** this will list information regarding Base station and Handset firmware versions.







8.6 Base Station(s) Firmware Upgrade

On the **Firmware Update** Page Base stations are updated in the same way as handsets and other extensions.

After entering Required Version and Required Branch choose **Start update** button > select **OK** button from the dialog window to start the update/downgrade procedure.
The relevant base station(s) will automatically reboot and retrieve the firmware specified from the server and update itself accordingly.

The base firmware update behavior is: Base will fetch the fwu file for approximately 3 minutes, then reboot and start flashing the LED - indicated by LED fast flashing for approximately 3 minutes and reboots in new version.

NOTE: All on-going voice calls are dropped from the base station(s) immediately the firmware update procedure starts.

8.6.1 Base firmware confirmation

Base station firmware version status in a multicell environment can be seen in the multicell base station group overview page, column 4 (Version).

Screenshot

Base Station Group

	ID	RPN	Version	MAC Address	IP Address	IP Status	DECT sync source	DECT property	Base Station Name
☐	0	00	400.1	00087B079207	192.168.11.106	This Unit	Select as primary	**Primary**	SME VoIP
☐	1	04	400.1	00087B0791FF	192.168.11.169	Connected	Primary:RPN00 (-26dBm)	Locked	SME VoIP

Check All /Uncheck All

With selected: Remove from chain

8.6.2 Verification of Firmware Upgrade

If the firmware upgrade/Downgrade don't start, you can check the syslog to see if the path is right.

Syslog information when Management Syslog level is set to "Debug"

[FWU Downloading File tftp://10.1.24.103/FwuPath/8663/8663_v0440_b0001.fwu]
[Base FWU started]
[Base FWU ended with exit code 2101 (NE_FILE_TRANSFER_EOF): End of file]

This is the path when the base station expects to find the firmware:
tftp://10.1.24.103/FwuPath/8663/8663_v0440_b0001.fwu

Check if the firmware file is in the correct directory.

9 Appendix – Multiline Feature

This section describes how to register the wireless handset to a system with active multiline feature.

One handset will be able to support up to 4 lines (4 different SIP accounts) ... A handset only supports 2 call appearances.
The limitation of maximum 1000 terminals in the system is maintained, and the maximum number of SIP registrations, one base station can handle, is maintained.
With 4 lines pr. terminal maximum number of terminals registered in a system are 250.
With 1-line pr. terminal maximum number of terminals registered in a system are 1000.
Still the limitation of 30 SIP accounts registered pr. base is maintained.
With 4 lines (SIP accounts) pr. terminal maximum number of terminals registered pr. base is 7.
The 4 SIP accounts pr. terminal follows the location of the terminal similar.
With multiline feature enabled 200 contacts in contact list is possible.

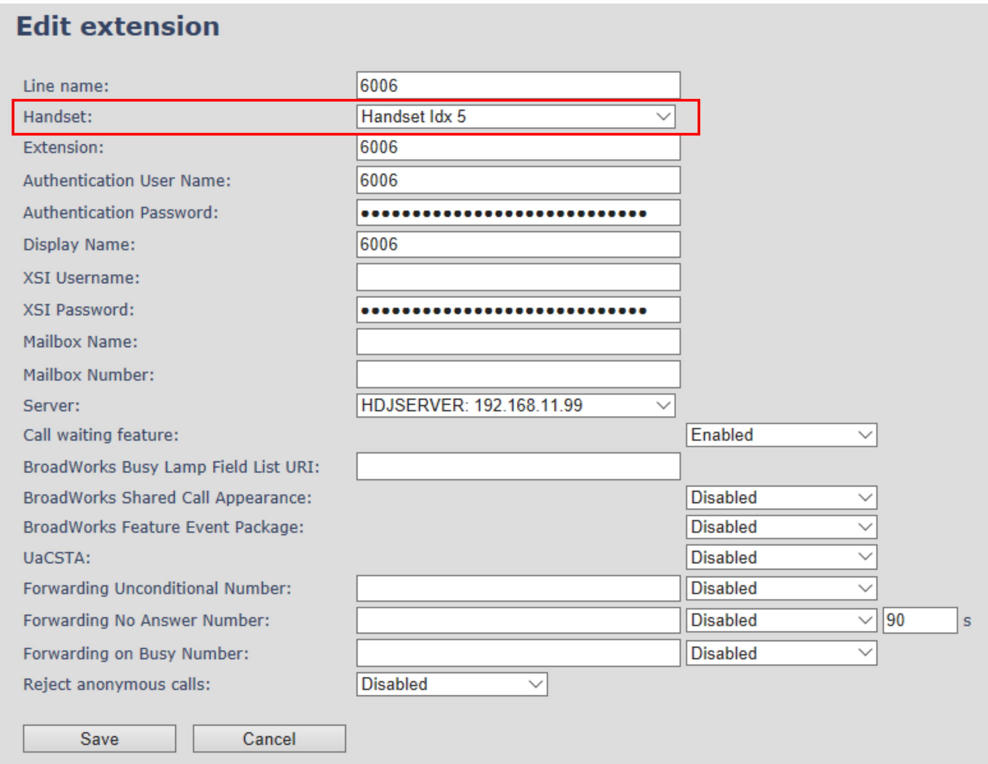
9.1 How to setup Multiline.

Step 1:

Register handset as described in point 7 (Appendix Adding Handsets).

Step 2:

Add a multiline to a handset by creating a new extension but instead for "New Handset" select the existing handset that you want to add the multiline to. (in this case Handset Idx 5)

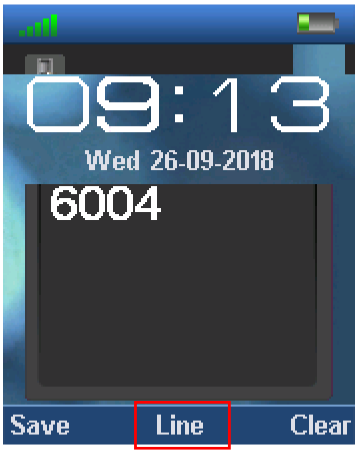


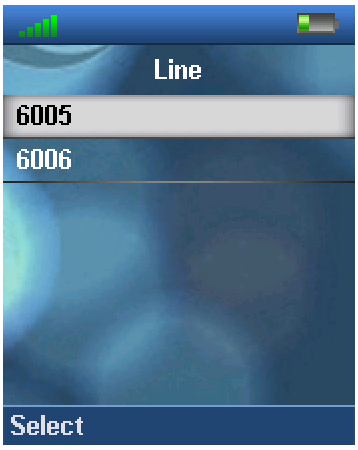
Step 3:

The extension will now show in the extension list with the same Idx and IPEI and the handset selected.
NB: the handset must be rebooted for the changes to take effect.

☐	5	02788DD16D	Present@RPN04	8630 430.1	Off	☐	5	6005	6005	192.168.11.99	HDJSERVER	SIP Registered@RPN04
☐	5	02788DD16D	Present@RPN04	8630 430.1	Off	☐	6	6006	6006	192.168.11.99	HDJSERVER	SIP Registered@RPN04

The Extension will now have two numbers 6005 and 6006.
When making call the user can chose which line to call from. Simply enter the number to call and press line.
Select the decried line and hook off to place the call from this line.





10 Appendix - Functionality Overview

So far, we have setup our SME VoIP system. Next, in this chapter we list what features and functionalities are available in the system. The SME VOIP system supports all traditional and advanced features of most telephony networks. In addition, 3rd party components handle features like voice mail, call forward, conference calls, etc. A brief description of SME VOIP network functionalities is:

- **Outgoing/incoming voice call management:** The SME VOIP system can provide multiple priority user classes. Further, up to 3 repeaters can be linked to a Base-station.
- **Internal handover**: User locations are reported to SIP Server to provide differentiated services and tariff management. Within a DECT traffic area, established calls can seamlessly be handover between Base-stations using connection handover procedures.

- **Security:** The RTX SME VOIP system also supports robust security functionalities for Base-stations. Most security[2] functionality is intrinsically woven into the SME VOIP network structure so that network connections can be encrypted, and terminal authentication can be performed.

10.1 Gateway Interface

CONNECTOR INTERFACES	
POWER	Connector: Ethernet PoE (Ethernet adaptor for normal power) IEEE 802.3: Power class 2 (3.84 – 6.49W)
LAN INTERFACE	Standard : 10BASE-T(IEEE 802.3 100Mbps) Connector: RJ45 8/8
INTERNET PROTOCOL:	• IPv4 • IPv6
KEYS	
	1 x Reset key
LED INDICATOR	
	One Status LED (multicolor, red, green, orange)
RF	
FREQUENCY BANDS	1880 – 1900 MHz (EMEA) 1910 – 1930 MHz (Latam) 1920 – 1930 MHz (USA) This is software settings and to be set when it is packed in factory.
OUTPUT POWER	<250 mW (for USA < 140mW)
ANTENNA	Two antennas for diversity
SOFTWARE UPDATE	
DOWNLOADABLE	Remote firmware update HTTPS/TFTP

[2] With active security with authentication 4 channels is supported

10.2 Detail Feature List

CODECs	
G.711 PCM A-LAW & U-LAW	Uncompressed voice Silence suppression (No)
G.722	Allows HD sound for the handset
G.726	ADPCM, 32 Kbps
G.729	A G.729.1 (ehem. G.729 EV) Note: Only with additional module, this is an extra option that requires a board connector mounted in Gateway. Per default not mounted.
SIP	
RFC2327	SDP: Session Description Protocol
RFC2396	Uniform Resource Identifiers (URI): Generic Syntax
RFC2833	In-Band DTMF/Out of band DTMF support
RFC2976	The SIP INFO method
RFC3261	SIP 2.0
RFC3262	Reliability of Provisional Responses in the Session Initiation Protocol (PRACK)
RFC3263	Locating SIP Servers (DNS SRV, redundant server support)
RFC3264	Offer/Answer Model with SDP
RFC3265	Specific Event Notification
RFC3311 RFC3325	The Session Initiation Protocol UPDATE Method P-Asserted Identity
RFC3326	The Reason Header Field for the Session Initiation Protocol (SIP)
RFC3489	STUN
RFC3515	REFER: Call Transfer
RFC3550	RTP: A Transport Protocol for Real-Time Application
RFC3581	Rport
RFC3842	Message Waiting Indication
RFC3891	Replace header support
RFC3892	The Session Initiation Protocol (SIP) Referred-By Mechanism
RFC3960	Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
RFC4475	Session Initiation Protocol (SIP) Torture Test Messages
SIPS	
SRTP	Will limit number of active calls pr. base when enabled.
WEB SERVER	
	Embedded web server HTTP
OTHER FEATURES	
QUALITY OF SERVICE	Type of Service (ToS) including DiffServ Tagging, and QoS per IEEE 802.1p/q
IP QUALITY	Warning – Network outage, VoIP service outage
	Adaptive Jitter Buffer support
AUTOMATIC DST	
TONE SCHEME	Country Depend Tone Scheme
ETHERNET FEATURES	
SPEED DUPLEX	10 & 100 duplex
VLAN	VLAN (802.1p/q)
DHCP SUPPORT	
STATIC IP	
TLS 1.1	For secure connections (AES-128)
TFTP	For configuration download.

HTTP	For configuration download.
HTTPS	For secure configuration download.
TCP/IP/UDP	
SNTP	For internet clock synchronization
QUALITY OF SERVICE	Type of Service (ToS) including DiffServ Tagging, and QoS per IEEE 802.1p/q
DHCP OPTION	66
DNS SRV	
DECT	
DECT CAP	Connectionless handover, enhanced location registration
CAT-IQ V1.0	Wideband Speech
GENERAL TELEPHONY	
HANDSET SUPPORT	10 simultaneous handsets supported (single cell) (10 call / single cell and 8 call/Multi cell) Total 1000 simultaneous call supported / system
VOIP ACCOUNTS	30 VoIP accounts per base – (maximum 254 bases per installation) Total 1000 VoIP accounts / system
	Maximum 1000 handsets per installation
SIMULTANEOUS CALLS	4 Wideband calls (g.722) or 10 single cell, 8 multi cell narrowband calls (PCMA, PCMU, G.726) or mixed wideband and narrowband.
CALL FEATURES	Codec Negotiation
	Codec Switching
	Missed call notification
	Voice message waiting notification
	Date and Time synchronization
	Parallel calls
	Common parallel call procedures
	Call transfer unannounced
	Call transfer announced
	Conference
	Call Waiting
	Calling line identity restriction
	Outgoing call
	Call Toggle
	Incoming call
	Line identification
	Multiple Lines
	Multiple calls
	Call identification
	Calling Name Identification Presentation (CNIP)
	Calling Line Identification Presentation (CLIP)
	Call Hold
	List of registered handsets
CALL LOG	50 mixed between Incoming, outgoing, missed calls
PHONE BOOK	Common Phonebook with up to 3000 entries (Import via csv format)
	Common Phonebook LDAP V2.0
	Local Phonebook (100 entries 8630 and 50 entries 8430)
DND	Do Not Disturb
CALL FORWARD	All
	No Answer
	Busy
	Individual Speed dial
	Programmable Function keys