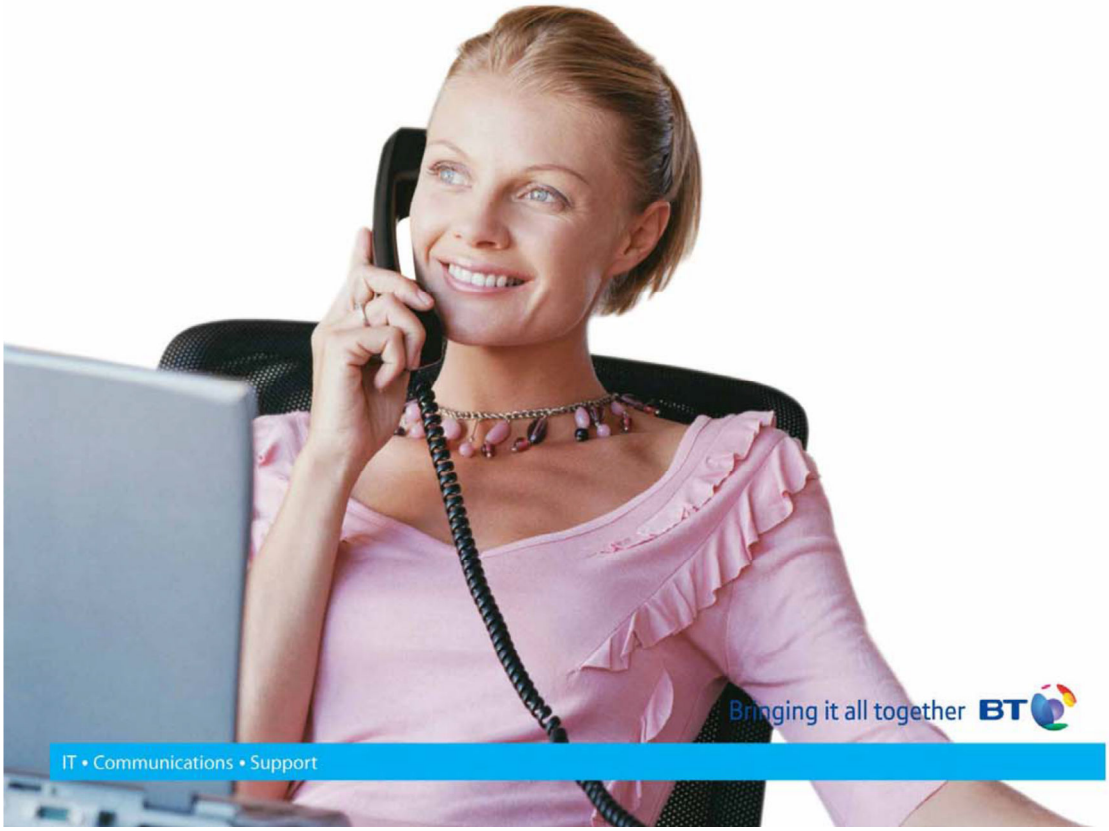
BT Business Single Cell DECT Installation & Configuration Guide



Contents

1 About This Document

This document describes the configuration, customization, management, operation, maintenance and trouble shooting of the BT Business DECT Single-Cell System (IP base, handset, and Repeater). For handset detailed user guide refer to [1].

1.1 Audience

Who should read this guide? First, this guide is intended for networking professionals responsible for designing and implementing Fijowave based enterprise networks.
Second, network administrators and IT support personnel that need to install, configure, maintain and monitor elements in a "live" BT Business DECT network will find this document helpful.

1.2 When Should I Read This Guide

Read this guide before you install the core network devices of BT Business DECT 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 also deploy a fully functional BT Business DECT System.

1.3 Important Assumptions

This document was written with the following assumptions in mind:

1) You have understanding of 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 has 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 VoIP Network	To gain knowledge about the different elements in a typical 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	VoIP Administration Interface	To learn about the Configuration Interface and define full meaning of various parameters needed to be setup in the system.
Chapter 6	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 BT Business DECT 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 BT Business DECT System functional. For such a reference to vendor specific devices, please contact the respective vendor documentation.

1.6 Abbreviations

For the purpose of this document, the following abbreviations apply:

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
RTP:	Real-time Transport Protocol
RPORT:	Response Port (Refer to RFC3581 for details)
SIP:	Session Initiation Protocol
:	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

[1]: BT Business DECT Handset Guide

2 Introduction – System Overview

In a typical telephony system, the network setup is the interconnection between Base-stations, 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 BT Business DECT single cell base station supports up to 20 registered handsets. The base supports the IP DECT CAT-IQ repeater with support for up to 5 channels simultaneous call sessions.

2.1 Hardware Setup

BT Business DECT network hardware setup can 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-station can be desk mounted or wall mounted. Radio coverage can be extended using repeaters. Repeaters are range extenders and cannot be used to increase local capacity issues.

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 6 simultaneous calls.

2.2 Components of BT Business DECT System

The BT Business DECT system is made up of (but not limited to) the following components:

- One Fijowave Base Station is connected over an IP network and using DECT as air-core interface.
- Fijowave IP DECT wireless Handset.
- BT Business DECT Configuration Interface; is a management interface for BT Business DECT Wireless Solution.

2.2.1 Fijowave Base Station

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 6 available channels.

2.2.2 BT Business DECT Administration Server/Software

This server is referred to as VoIP Configuration Interface.
The 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 Fijowave Wireless Handset

The handset is a lightweight, ergonomical and portable unit compatible with Wideband Audio (G.722), DECT, GAP standard, CAT-iq audio compliant.
The handset includes Colour 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 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)

Network capacity of relevant components can be summarised as follows:

Description	Capacity
Single Cell Setup	1
Max ## of Repeaters	3 per Base station
Max ## of Users (SIP registrations) per Base	20
Single Cell Setup: Max ## Simultaneous Calls	6
Repeater: Max ## of Calls (Narrow band)	5
Repeater: Max ## of Calls (G722)	2

3 Installation of Base Stations/Repeater

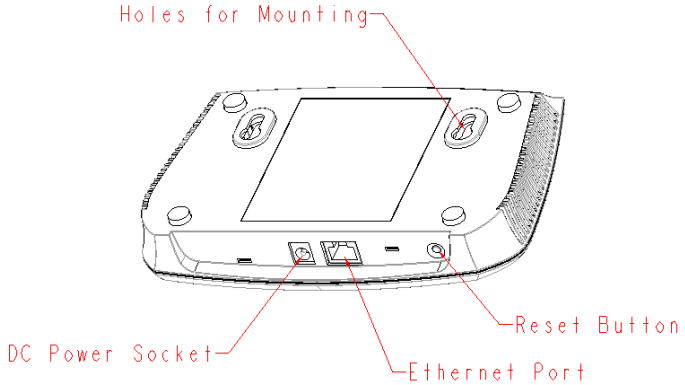
In the following we briefly describe the how to install the base station in this chapter.

3.1 Package – Contents/Damage Inspection

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:

- 1 x Cat. 5 cable (Ethernet cable)
- Base unit
- AC adaptor (Ten Pau S003GB0500060)



Back View of Base Station Unit

3.2 BT Base station Mechanics

The base station front end shows a 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 summarises the various LED states:

LED State	State
Unlit	No power in unit
Unlit/Solid red	Error condition
Blinking green	Initialisation
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 BT Engineers). Symptoms include no system/SIP debug logs are logged, etc.
Orange	Press reset button of base station.

3.3 BT 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 Configuration Interface.

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 units upside down as it significantly reduces radio coverage.

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

3.5 Find IP of Base Station

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

3.5.1 Using handset Find IP feature

On the handset press “Menu” key followed by the keys: *47* to get the handset into find bases menu. The handset will now scan for bases.

- 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. For further details refer to reference [2].

3.5.2 Using browser IPDECT

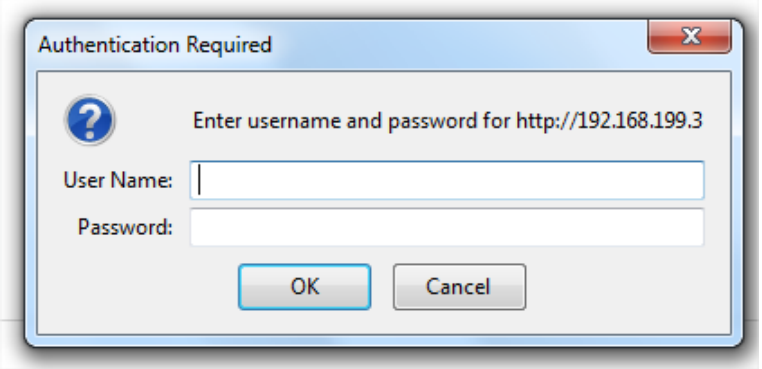
Open any standard browser and enter the address:
http://ipdect<MAC-Address-Base-Station>
for e.g. http://ipdect38b74d000afb. This will retrieve the HTTP Web Server page from the base station with hardware address 38b74d000afb.
This feature requires an available DNS server.

3.6 Login to Base Configuration Interface

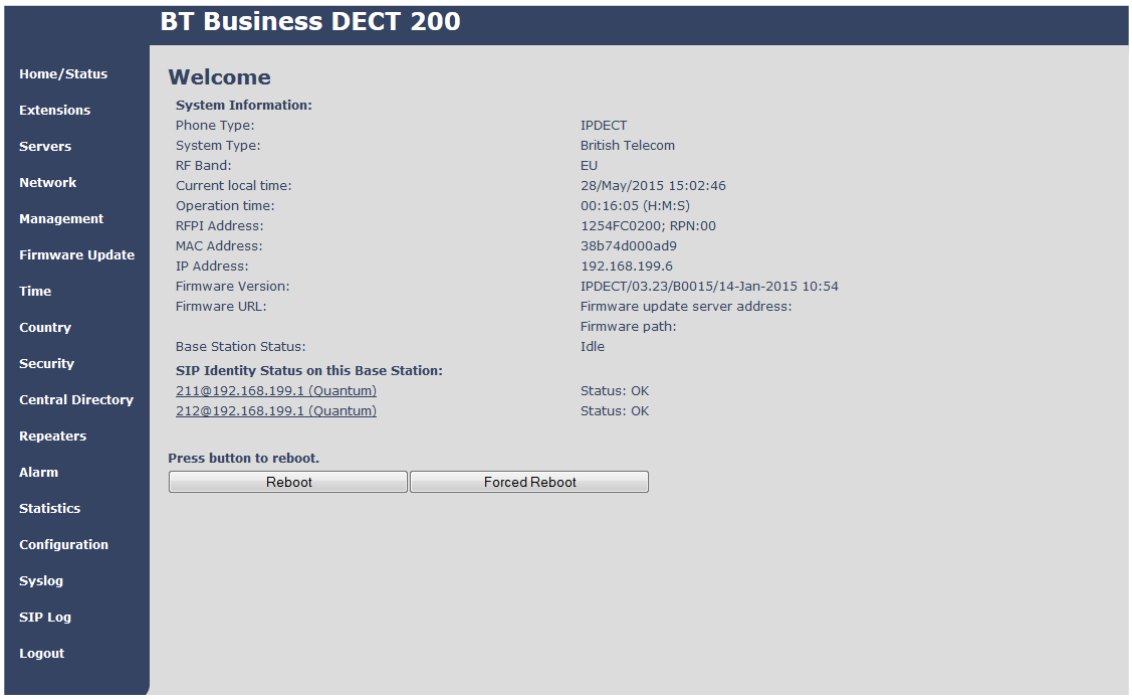
STEP 1 Connect the Base station to a private network via standard Ethernet cable (CAT-5).

STEP 2 Use the IP find menu in the handset (Menu *47*) to determine the IP-address of the base station by matching the MAC address on the back of the base station with the MAC address list in the handset.

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



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


BT Business DECT 200
Home/Status
Extensions
Servers
Network
Management
Firmware Update
Time
Country
Security
Central Directory
Repeaters
Alarm
Statistics
Configuration
Syslog
SIP Log
Logout
Welcome
System Information:
Phone Type: IPDECT
System Type: British Telecom
RF Band: EU
Current local time: 28/May/2015 15:02:46
Operation time: 00:16:05 (H:M:S)
RFPI Address: 1254FC0200; RPN:00
MAC Address: 38b74d000ad9
IP Address: 192.168.199.6
Firmware Version: IPDECT/03.23/B0015/14-Jan-2015 10:54
Firmware URL: Firmware update server address:
Firmware path:
Base Station Status: Idle
SIP Identity Status on this Base Station:
211@192.168.199.1 (Quantum) Status: OK
212@192.168.199.1 (Quantum) Status: OK
Press button to reboot.
Reboot
Forced Reboot

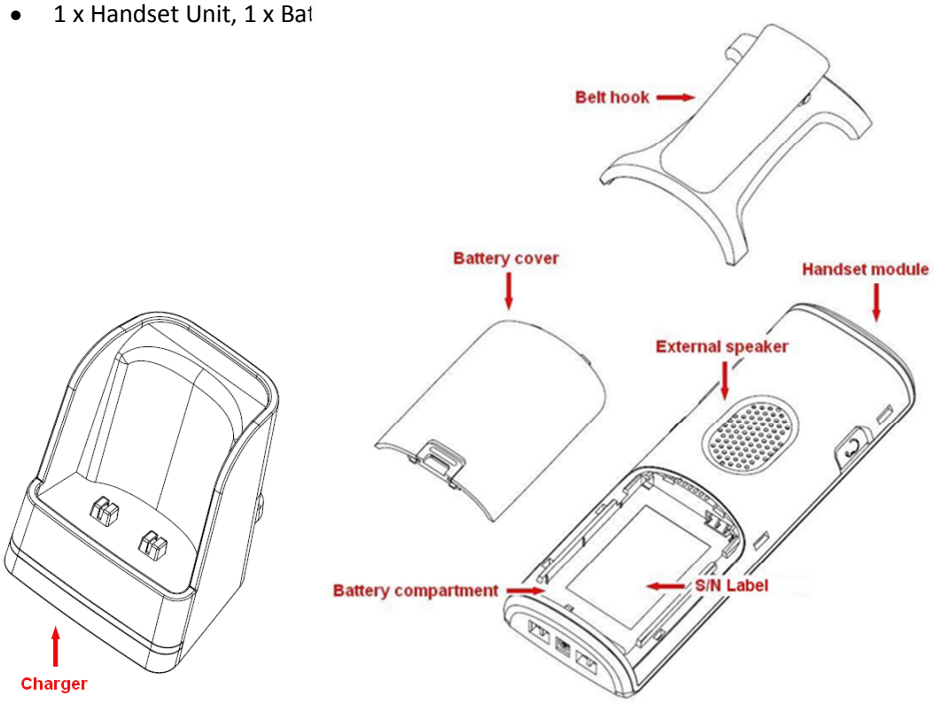
4 Preparing the Handset

We briefly describe how to prepare the handset for use, install, insert and charge new batteries.

Package – Contents

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

- 1 x Handset hook
- 1 x A/C Adaptor (Ten Pau S008CM0550060)
- 1 x Rechargeable Battery (Lithium-Ion battery 3.7V, 1100mAh)
- 1 x Charger
- 1 x Handset Unit, 1 x Bat



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

Installing the Battery

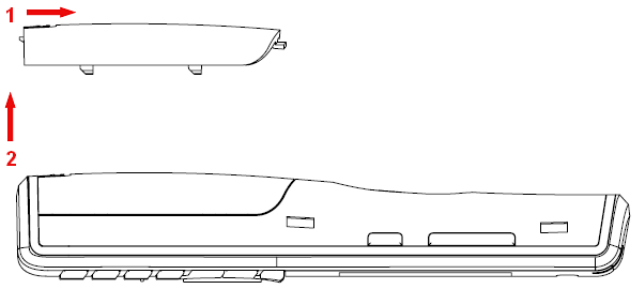
1. Never dispose battery in fires, otherwise it will explode.
2. Never replace the batteries in potentially explosive environments, e.g. close to inflammable liquids/ gases.
3. ONLY use approved batteries and chargers from the vendor or operator.
4. Do not disassemble, customise or short circuit the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

Using the Charger

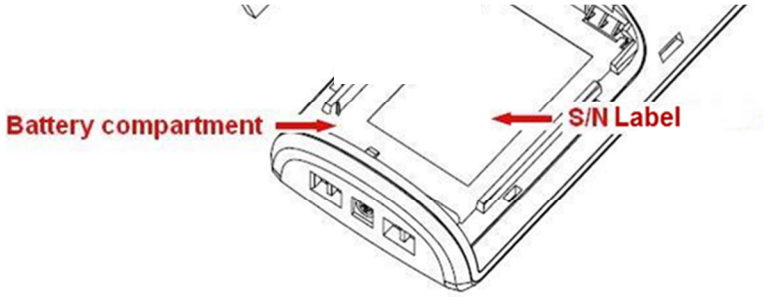
Each handset is charged through the use of a handset charger. The charger is a compact desktop unit designed to charge and automatically maintain the correct battery charge levels and voltage.
The charger Handset is powered by AC supply from 110-240VAC that supplies 5.5VDC at 600mA.
When charging the 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 0°C and 25°C/32°F and 77°F. Do not place the handset in direct sunlight. The battery has a built-in heat sensor which will 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 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.



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

5 BT Business DECT Administration Interface

The VoIP Administration Interface is the main interface through which the system is managed and debugged.

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

Note: Enabling secure web will decrease web server speed perceived by the user. The MS internet explorer caches more data and as such this browser is recommended in secure web mode.

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 BT Business DECT Administration Interface.

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 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.
Time	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.
Country	Specifying the country/territory where the network is located ensures that your phone connection functions properly. Note: The base language and country setting are independent of each other.
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.
Repeaters	Administration and configuration of repeaters of the system
Emergency	Administration and configuration of the emergency 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.
Configuration	This shows detail and complete 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.

5.2 Home/Status

We describe the parameters found in the Welcome front end home/status of the BT Business DECT 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
Current local time	This base local time
Operation time	Time from last boot of base
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
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" 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 call, directory access, firmware update active
Forced Reboot	Reboot immediately even active calls are ongoing.

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. This section also describes the group call feature.

The system can handle maximum 20 extensions matching 20 handsets which can be divided between servers. When 20 handsets are registered it is not possible to add more extensions.

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 Group call

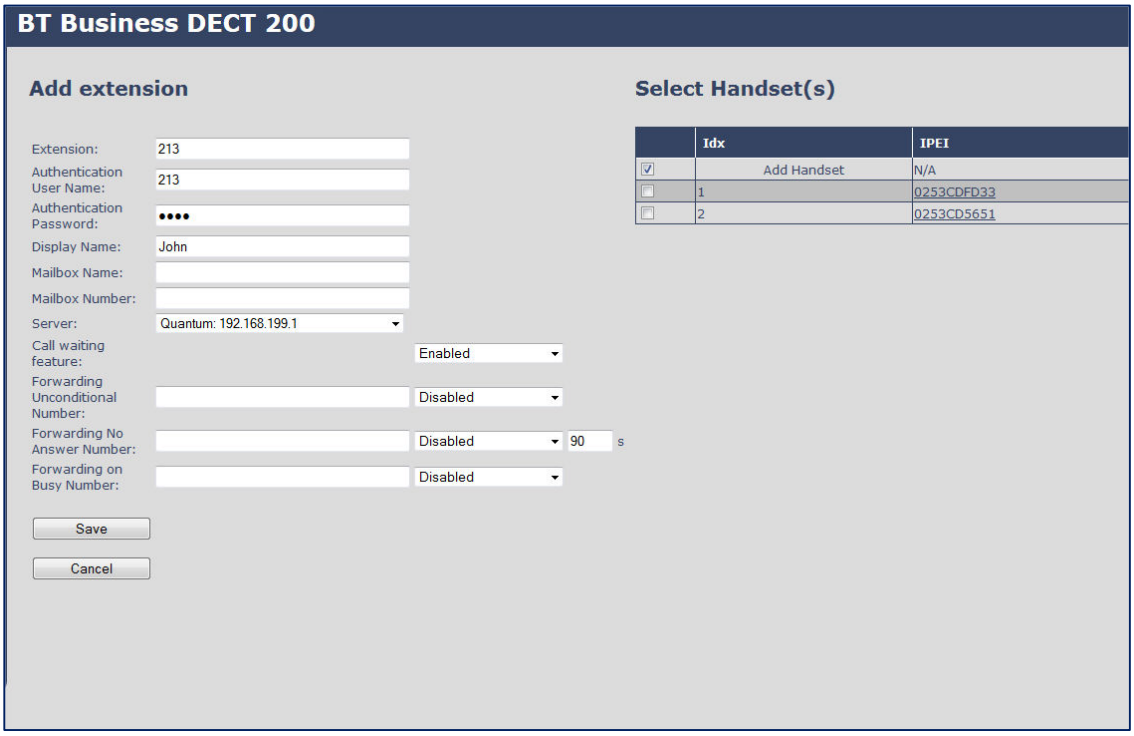
Call Group is a SIP extension where multiple handsets are associated. All handsets that subscribes to a given extension (and hence Call Group) can receive incoming calls and initiate outgoing calls on the given extension. It is possible for any handset to perform any call action which is possible without the Call Group feature. That is, call actions as Hold, Transfer etc. are possible if the PBX supports them.

When an incoming call arrives to a given Call Group, all Call Group subscribed handsets will alert. Thus, if a Call Group contains 20 handsets, all 20 handset will alert.
An alerting handset cannot receive another incoming call, and therefore if a handset subscribes for multiple Call Groups, and a call arrives for a 2nd Call Group while the handset is alerting, the handset will not receive this call. If DND is enabled for a given handset, it will not receive the incoming call.

For outgoing calls, it can be selected in the handset which line (i.e. Call Group) to use for the call. The maximum number of lines is 20. For any outgoing actions, the settings for the selected line (SIP extension) will be used.

5.3.2 Add extension

Screenshot



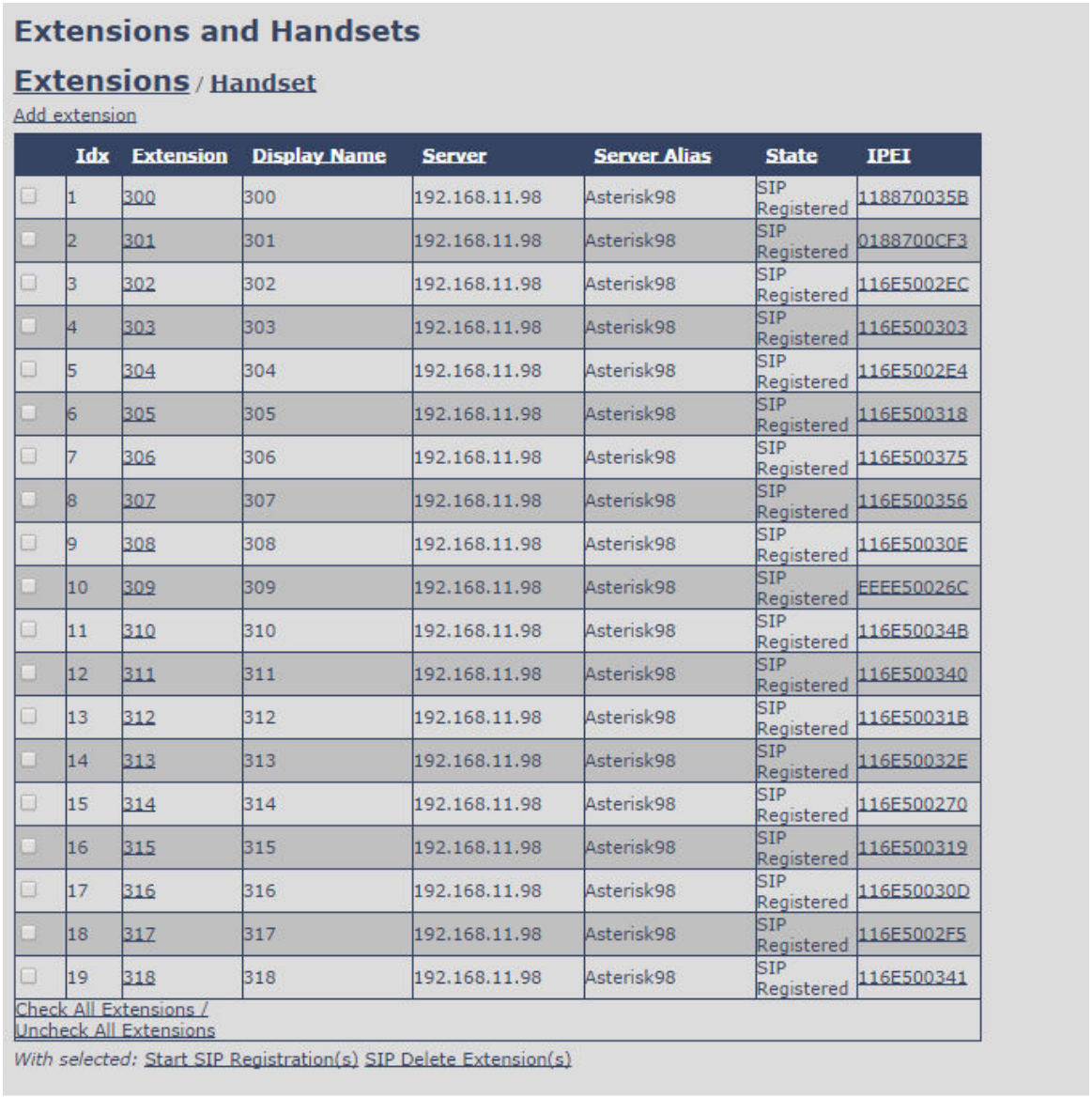
Parameter	Default Value(s)	Description
Extension	Empty	Handset phone number or SIP username depending on the setup. **Possible value(s):** 8-bit string length **Example: 1024, etc.** **Note:** The Extension must also be configured in SIP server in order for this feature to function.
Authentication User 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
Mailbox Name	Empty	Name of centralised system used to store phone voice messages that can be retrieved by recipient at a later time. **Valid Input(s):** 8-bit string characters for the Name
Mailbox Number	Empty	Dialled 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 VoIP Service provider.
Call waiting	Enabled	Used to enable/disable Call Waiting feature. When disabled a

feature:		second incoming call will be rejected. If enabled a second call will be presented as call waiting.
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

When an extension is added (or edited) it can be selected which handsets shall subscribe to the given extension, and hence be a part of this call group, see above figure. It is also possible to choose to add a new handset entry at this point, and if this is done, DECT registration for the new entry can be enabled afterwards on the handsets subpage

5.3.3 Extensions 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.



Extensions and Handsets

Extensions / Handset

Add extension

	Idx	Extension	Display Name	Server	Server Alias	State	IPEI
☐	1	300	300	192.168.11.98	Asterisk98	SIP Registered	118870035B
☐	2	301	301	192.168.11.98	Asterisk98	SIP Registered	0188700CF3
☐	3	302	302	192.168.11.98	Asterisk98	SIP Registered	116E5002EC
☐	4	303	303	192.168.11.98	Asterisk98	SIP Registered	116E500303
☐	5	304	304	192.168.11.98	Asterisk98	SIP Registered	116E5002E4
☐	6	305	305	192.168.11.98	Asterisk98	SIP Registered	116E500318
☐	7	306	306	192.168.11.98	Asterisk98	SIP Registered	116E500375
☐	8	307	307	192.168.11.98	Asterisk98	SIP Registered	116E500356
☐	9	308	308	192.168.11.98	Asterisk98	SIP Registered	116E50030E
☐	10	309	309	192.168.11.98	Asterisk98	SIP Registered	EEEE50026C
☐	11	310	310	192.168.11.98	Asterisk98	SIP Registered	116E50034B
☐	12	311	311	192.168.11.98	Asterisk98	SIP Registered	116E500340
☐	13	312	312	192.168.11.98	Asterisk98	SIP Registered	116E50031B
☐	14	313	313	192.168.11.98	Asterisk98	SIP Registered	116E50032E
☐	15	314	314	192.168.11.98	Asterisk98	SIP Registered	116E500270
☐	16	315	315	192.168.11.98	Asterisk98	SIP Registered	116E500319
☐	17	316	316	192.168.11.98	Asterisk98	SIP Registered	116E50030D
☐	18	317	317	192.168.11.98	Asterisk98	SIP Registered	116E5002F5
☐	19	318	318	192.168.11.98	Asterisk98	SIP Registered	116E500341

Check All Extensions / Uncheck All Extensions

With selected: Start SIP Registration(s) SIP Delete Extension(s)

Parameter	Description
Idx	Select / deselect for delete, register and deregister handsets
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.
IPEI	Handset IPEI. IPEI is unique DECT identification number.

5.3.3.1 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.

Screenshots

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).

5.3.4 Edit Extension

To edit extension use the mouse to click the link of the extension.
Edit extension will open the same configuration possibilities as add extension. Refer to the above add extension section.

5.3.5 Handset list

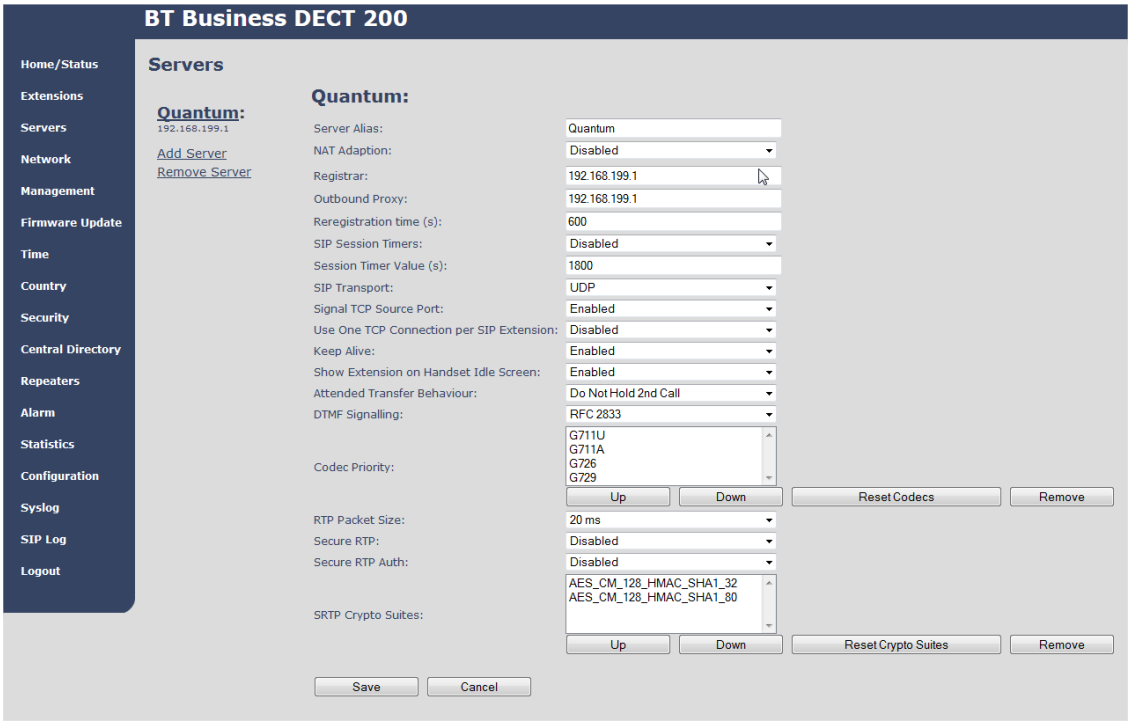
The added handsets will be shown in the handset lists.
The list can be sorted by any of the top headlines, by mouse click on the headline link.

Parameter	Description
Idx	Select / deselect for delete, register and deregister handsets
IPEI	Handset IPEI. IPEI is unique DECT identification number.
Handset state	
FW info	
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.
Extension	

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.
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”.
Re-registration time	600	The “expires” value in SIP REGISTER requests. This value indicates how long the current SIP registration is valid, and hence 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 1.0
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 will not be used. The “Signal TCP Source Port” parameter specifies if the used source port shall be signalled 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).
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 behaviour by handset hold feature. RFC 3264: Hold is 24nalyse24n 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, recv only or inactive dependant of the context RFC 2543: The ”old” way of 24nalyse24ng HOLD. The connection information part of the SDP is set to 0.0.0.0, and the direction attribute is sendonly, recvonly or inactive dependant of the context
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 PBXs do not expect that the 2nd call is put on hold, and therefore attended transfer fails on these PBXs. The "Attended Transfer Behaviour" feature defines whether or not 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
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 signalling 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).
Codec Priority	G.711U G.711A G.726	Defines the codec priority that base stations uses 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 in order 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.
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
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 is 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.

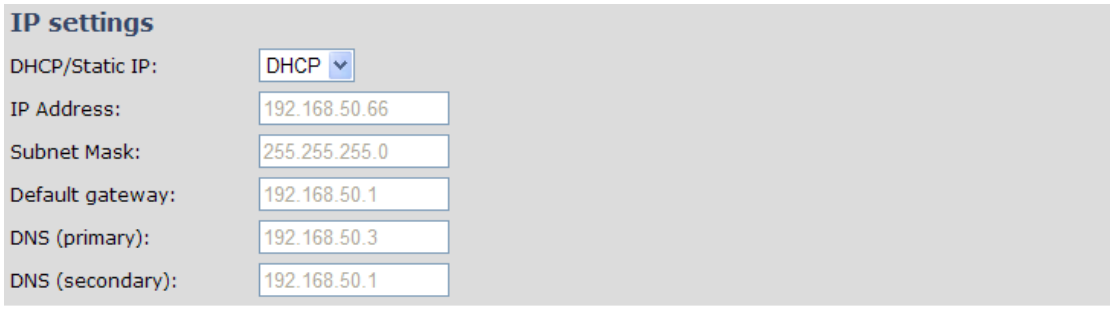
Note: Within servers or even with multi servers, extensions must always be unique. This means same extension number on server 1 cannot be re-used on server 2.

5.5 Network

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

5.5.1 IP Settings

Screenshot



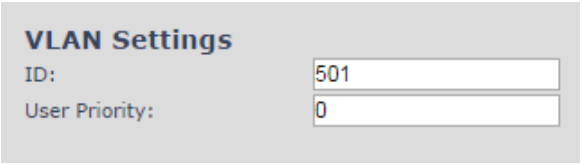
Parameter	Default Values	Description
DHCP/Static IP	DHCP	If DHCP is enabled, the device automatically obtains TCP/IP parameters. **Possible value(s):** Static, DHCP **DHCP:** IP addresses are allocated automatically from a pool of leased addresses. **Static IP:** IP addresses are manually assigned by the network administrator. 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 other 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.

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)

For further help on VLAN configuration refer to Appendix.

5.5.3 DHCP Options

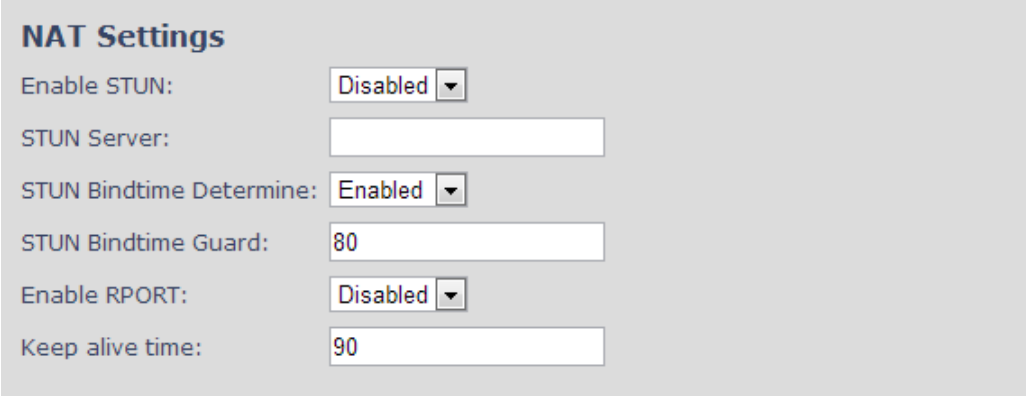
Screenshot

Parameter	Default Values	Description
Plug-n-Play	Disabled	Enabled: DHCP option 43 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

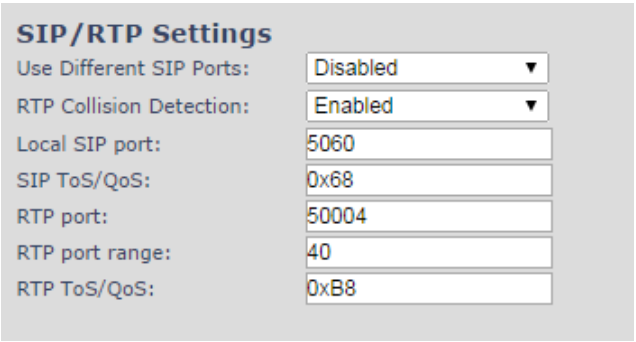


Parameter	Default Values	Description
Enable STUN	Disabled	Enable to use STUN
STUN Server	NA	**Permitted value(s): AAA.BBB.CCC.DDD** (Currently only Ipv4 are supported) or **url** (e.g.: firmware.Fijowave.net).
STUN Bindtime Determine	Enabled	
STUN Bindtime Guard	80	**Permitted values:** Positive integer 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

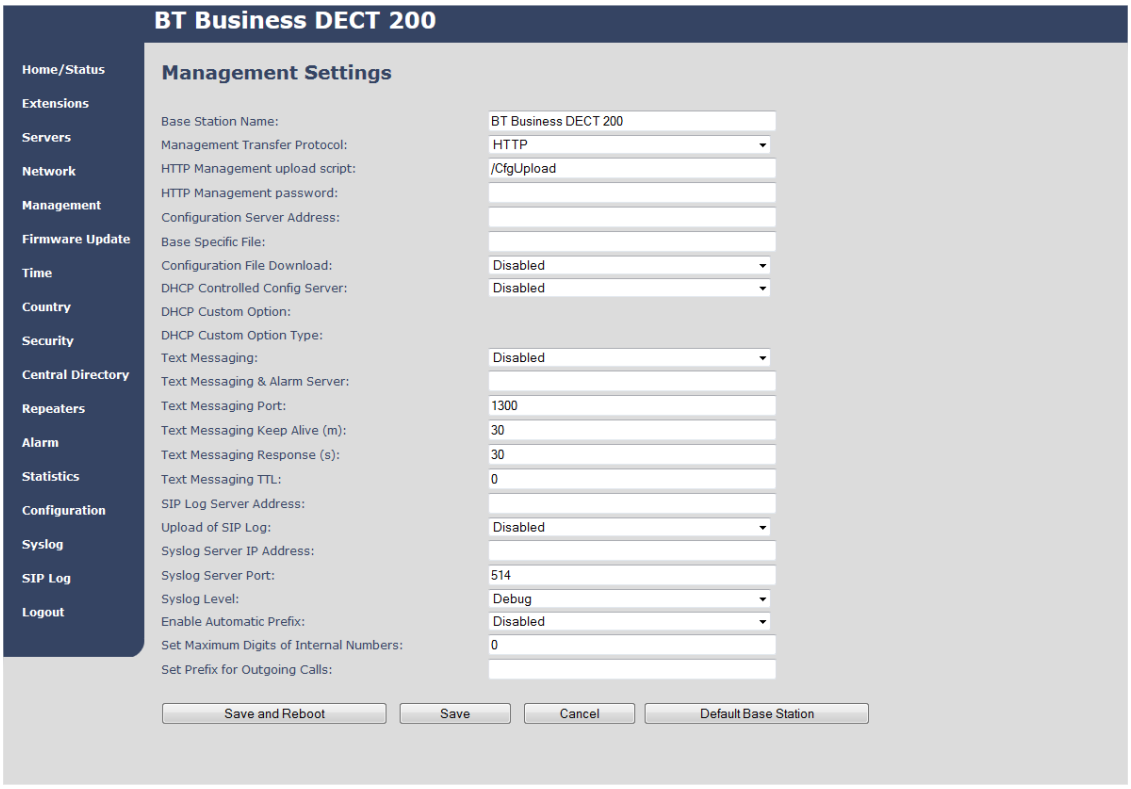


Parameter	Default Values	Description
Use Different SIP Ports	Disabled	If disabled, the Local SIP port parameter specifies the source port used for SIP signalling in the system. If enabled, the Local SIP Port parameter specifies the source port used for first user agent (UA) instance. Succeeding UA's will get succeeding ports.
RTP Collision Detection	Enabled	Enable: If two sources with same SSRC, the following one is discarded. Disabled: No check – device will accept all sources.
Local SIP port	5060	The source port used for SIP signalling **Permitted values:** Port number default 5060.
SIP ToS/QoS	0x68	Priority of call control signalling 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

5.6 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



Parameter	Default value	Description
Base Station Name:	VoIP	It indicates the title that appears at the top window of the browser and is used in the multicell page.
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 password	Empty	Password that should be entered in order to have access to the configuration server. **Permitted value(s):** 8-bit string length
Configuration	Empty	Server/device that provides configuration file to base station.

server address		**Type:** DNS or IP address **Permitted value(s): AAA.BBB.CCC.DDD or <URL>**
Base Specific File	Empty	Base configuration file
Configuration File Download	Disabled	Base Specific file: Used when configuring a single cell base Multicell Specific File: Used when configuring a multicell based system Base and Multicell Specific File: Used on out of factory bases to specify VLAN and Multicell ID and settings.
Text Messaging	Disabled	
Text Messaging & Alarm server	Empty	
Text Messaging Port	Empty	
Text Messaging Keep Alive (m)	Empty	
Text Messaging Response (s)	Empty	
Text Messaging TTL		
SIP Log Server Address	Empty	**Permitted value(s): AAA.BBB.CCC.DDD or <URL>** Requires a predefined folder named: \SIP
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 Server IP-Address	NA	**Permitted value(s): AAA.BBB.CCC.DDD or <URL>**
Syslog Server Port	NA	Port number of syslog server.
Syslog Level	Off	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 32nalyse level also contains the messages from normal operation. Debug: Used by Fijowave for debug. Should not be enabled during normal operation.
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 dialled number. In case of detection the base will not complete the dialled 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 of Internal Numbers	0	Used to detect internal numbers. In case of internal numbers no prefix number will be added to the dialled number.
Set Prefix for Outgoing Calls	Empty	Prefix number for the enabled automatic prefix feature. **Permitted value(s): 1 to 9999**

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.

5.7 Time Server

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

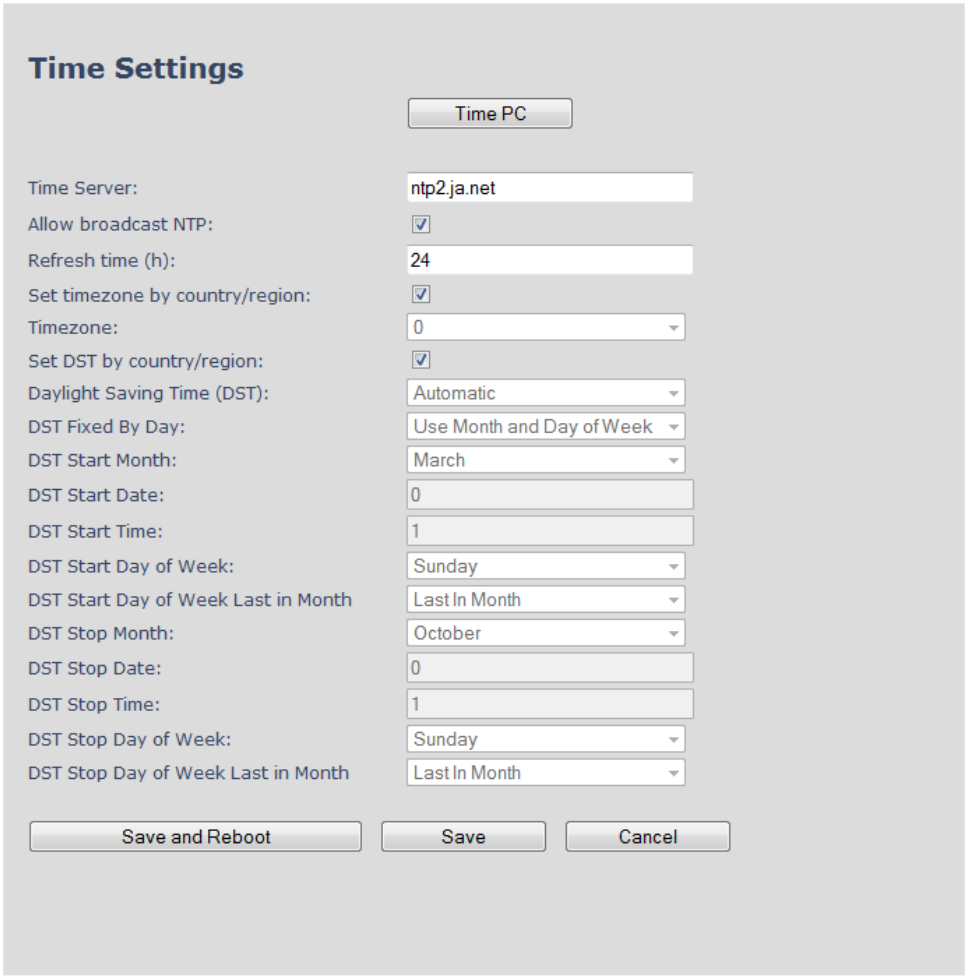
As well the time server is used in the debug logs and for SIP traces information pages, and used to determine when to check for new configuration and firmware files.

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.

Screenshot

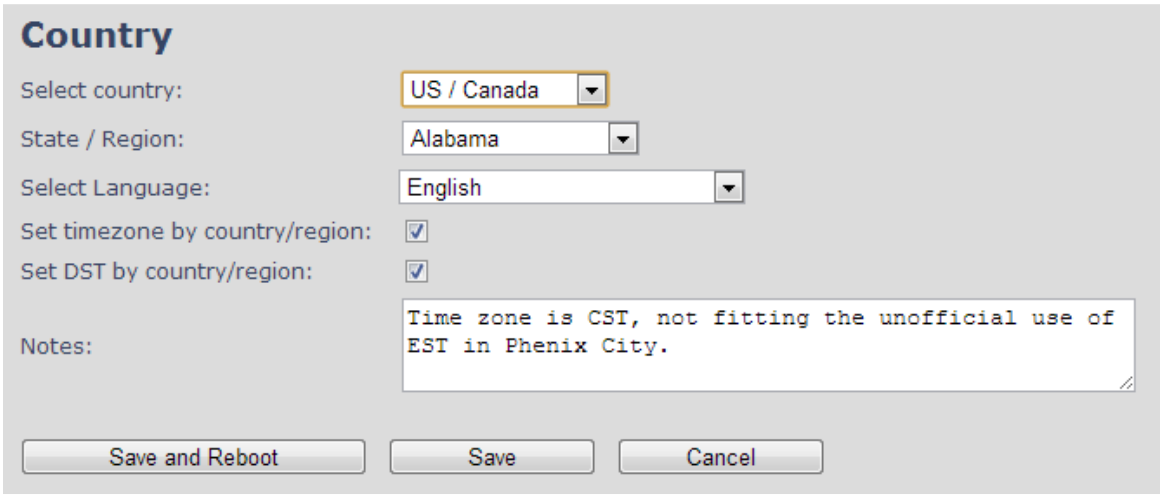


Parameter	Default Values	Description
Time Server	Empty	DNS name or IP address of NTP server. Enter the IP/DNS address of the server that distributes reference clock information to its clients including Base stations, Handsets, etc. **Valid Input(s):** AAA.BBB.CCC.DDD or URL (e.g. time.server.com) Currently only Ipv4 address (32-bit) nomenclature is supported.
Allow broadcast NTP	Checked	
Refresh time (h)	Empty	The window time in seconds within which time server refreshes. **Valid Inputs:** positive integer
Set timezone 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
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 Date	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	25	Numerical day of month DST comes to effect when DST is fixed to a specific date **Valid Inputs:** positive integer
DST Start Time	3	DST start time in the day **Valid Inputs:** positive integer
DST Start Day of Week	Monday	Day within the week DST begins
DST Start Day of Week, Last in Month	Last in Month	Specify the week that DST will actually start.
DST Stop Month	October	The month that DST actually stops.
DST Stop Date	1	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.

5.8 Country

The country setting controls the in-band tones used by the system. To select web interface language go to the management page.

Screenshot



Parameter	Default Values	Description
Select Country	Germany	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, Tyrkie, Deutsch, Portuguese, Hrvatski, Srpski, Slovenian, Nederlands, Francaise, Espanol, Russian, Polski.
Set timezone by country/region	checked	When checked timezone 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 timezone and DST the parameters in web page Time will be discarded.

The following types of in-band tones are supported:

- Dial tone
- Busy tone
- Ring Back tone
- Call Waiting tone
- Re-order tone

5.9 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.

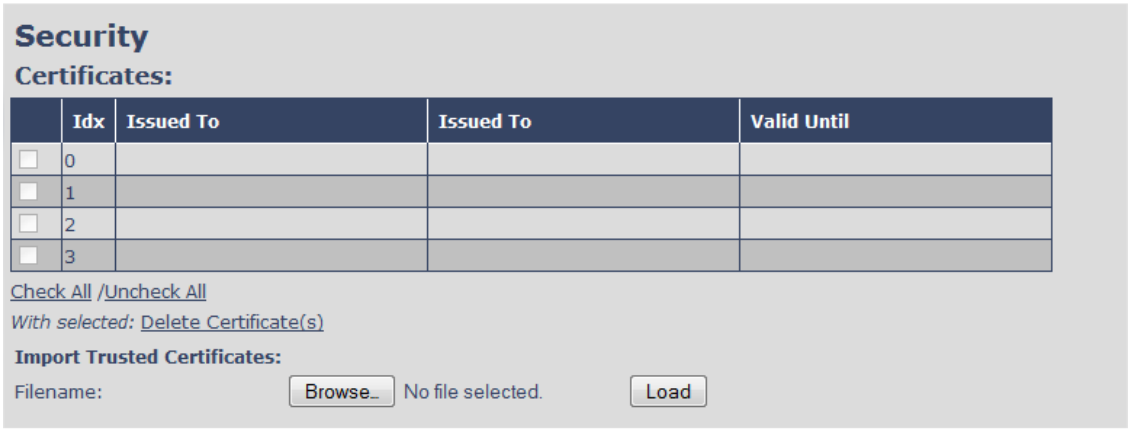
To setup secure fwu and configuration file download select HTTPs for the Management Transfer Protocol (reference 0)

SIP and RTP security is server dependent and in order to configure user must use the web option Servers (reference 5.4).

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



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	Organisation, Company – which is part of the certificate file
Valid Until	Empty	Date Time Year – which is part of the certificate file

Screenshot

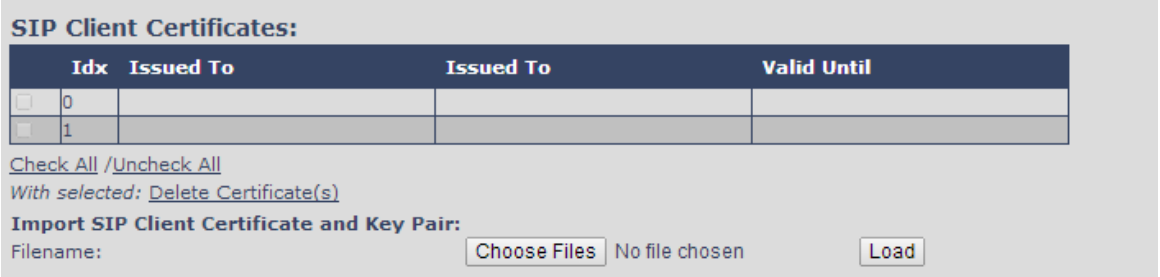
By enabling Use Only Trusted Certificates, the certificates the base will receive from the server must be valid and loaded into the system. If no valid matching certificate is found during the TLS connection establishment, the connection will fail. When Use Only Trusted Certificates is disabled, all certificates received from the server will be accepted.

5.9.2 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

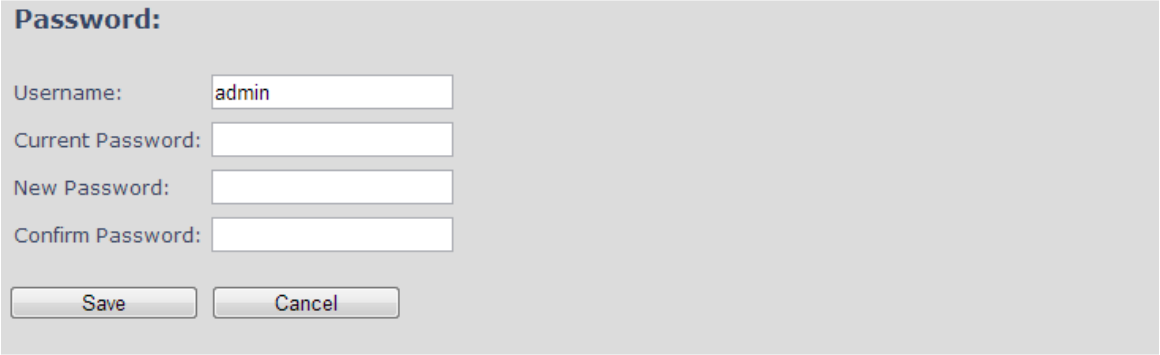
Screenshot



5.9.3 Password

In the below the password parameters are defined.

Screenshot



Parameter	Default Values	Description
Username	Admin	Can be modified to any supported character and number
Current Password	Admin	Can be modified to any supported character and number
New Password	Empty	Change to new password
Confirm Password	Empty	Confirm password to reduce accidental changes of passwords

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

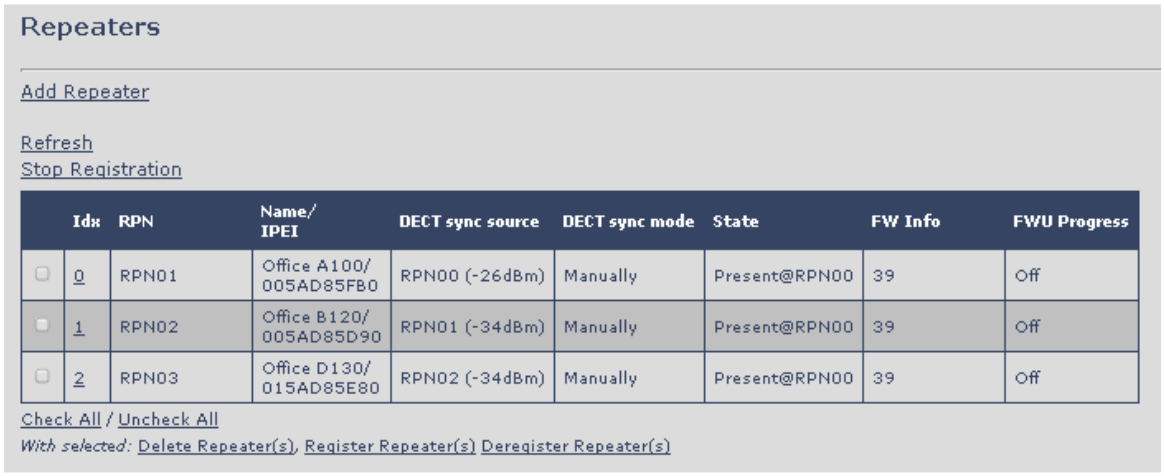
5.10 Repeaters

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

5.10.1 Add repeater

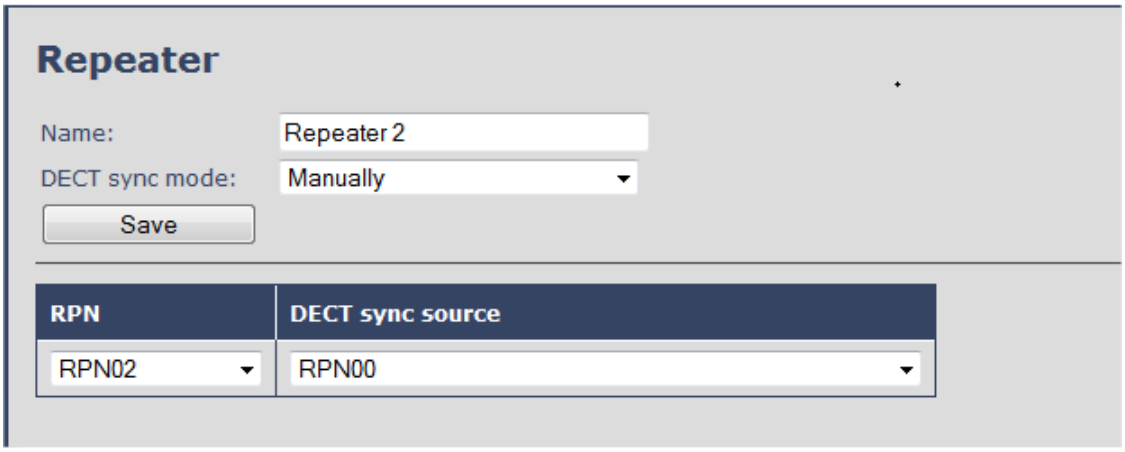
From repeaters web select “Add Repeater”

Screenshot



Then select “DECT Sync mode”

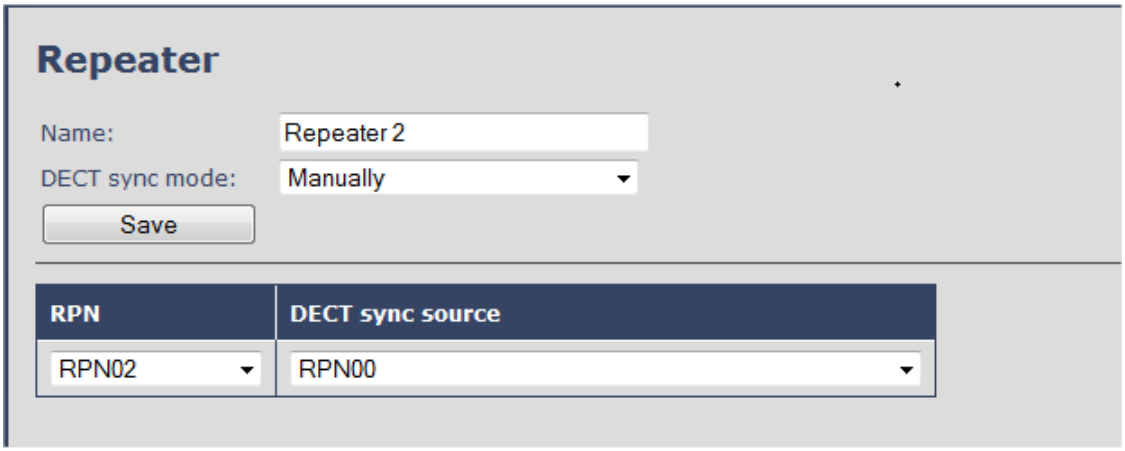
Screenshot



5.10.1.1 Manually

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

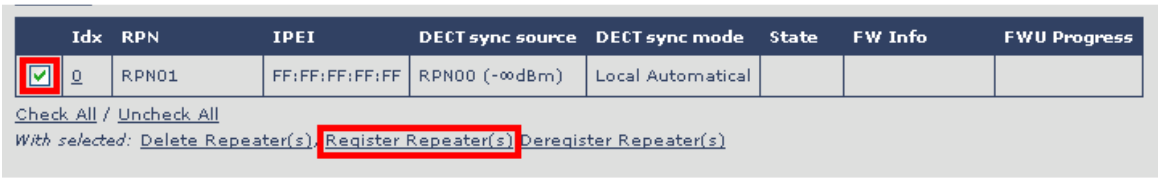
Screenshot



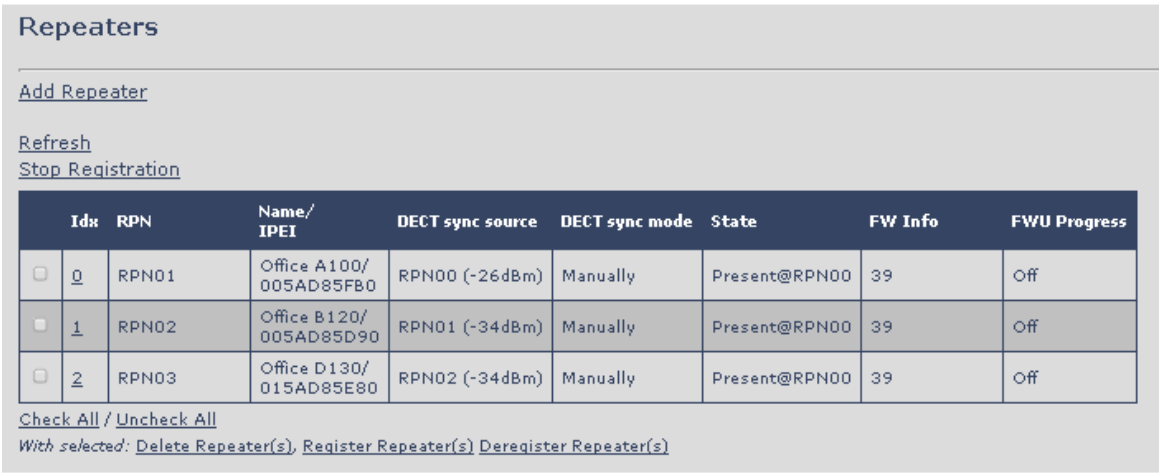
Parameters	Description
Idx	System counter
RPN	SINGLE CELL SYSTEM: The base is always RPN00, first repeater will then be RPN01, second repeater RPN02 and third RPN03 (3 repeaters maximum per base)
DECT sync source	Select the base or repeater the repeater has to be synchronized to.

5.10.2 Register Repeater

Adding a repeater makes it possible to register the repeater. Registration is made by selecting 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 including handset registration.



5.10.3 Repeaters list



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 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.
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.

6 Functionality Overview

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

- **Outgoing/incoming voice call management:** The BT Business DECT 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 in order 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 BT Business DECT system also supports robust security functionalities for Base-stations. Most security[1] functionality is intrinsically woven into the VOIP network structure so that network connections can be encrypted and terminal authentication can be performed.

6.1 System Feature List

This section gives a summary of some essential functionality within the wireless IP network.

Components	System Features
Speech Coding	10 channels ADPCM G.726 on air interface[1] 4 channels of G729 on IP interface 10 channels of G711 on IP interface Support of mixed types of Codecs in one Base Station
In-band Tones	Dial tone Busy tone Ring back tone Call waiting tone Re-order tone
Radio Access Mechanism	Bearer Handover and Connection Handover: Busy indication and support Connection re-establishment Emergency Calls[2]: Inside or outside roaming areas
SIP support	REGISTER, INVITE, and TERMINATE sessions Session Description Protocol (SDP), HTTP authentication Support 20 DECT instances (depending on VOIP configuration) locating SIP servers Support for re-INVITE

[1] With active security 4 channels is supported
[2] Emergency call is not possible if the Network connection is not working or in case of power failure.

Components	System Features
	Support for fail-over SIP proxy Message Waiting Indication Support for “302” response between UA ⇔ SIP Server
Internal Synchronization	Internal Synchronization lock, timing and transmission
Management Features	Assignment of Base-stations Logging calls and internal events, and tracking use of resources Logging system faults TFTP server for software upgrade. WEB interface for remote management of network devices Remote debugging of network devices, including log features
Temperature Range	0˚C to + 40˚C
Range (EU DECT)	50m and Outdoor: 300m

6.2 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	G.729
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	The Session Initiation Protocol UPDATE Method
RFC3325	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/HTTPS
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	
VLAN	VLAN (802.1p/q)
DHCP Support	
Static IP	
TLS 1.0	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	20 handsets supported (single cell)
VoIP Accounts	20 VoIP accounts per base
Simultaneous Calls	6 simultaneous calls per base station
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

This product is in conformity with the essential requirements of the EC directive 1999/5/EC. A copy of which is available at Fijowave, Synergy Centre, ITT Dublin, Tallaght, Dublin 24, Ireland.