

Konftel IP DECT 10 Manual

*Installation & Configuration
Network Deployment
Operation & Management*

Technical Reference Document
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1 About This Document

This document describes the configuration, customization, management, operation, maintenance and troubleshooting of Konftel IP DECT 10. 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 Konftel IP DECT 10.

Second, network administrators and IT support personnel that need to install, configure, maintain and monitor elements in a “live” VoIP 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 VoIP 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 functionally VoIP 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 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 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	Registration Management - Handsets	Learn how to register handset and extensions to base stations
Chapter 7	Firmware Upgrade/Downgrade Management	Provides the procedure of how to upgrade firmware to base stations and/or handsets and/or repeaters
Chapter 8	System Functionality Overview	To gain detail knowledge about the system features.

9 Appendix A	Basic Network Servers Configuration	To learn about operating the handset and base stations including detail description of handset MMI.
10 Appendix B	VLAN Setup Management	Examines how to setup VLAN in the network
11 Appendix C	Local central directory file handling	Detailed description of central directory file format and upload.

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 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 VoIP 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 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
VLAN:	Virtual Local Access Network
TOS:	Type of Service (policy based routing)
URL:	Uniform Resource Locator
UA:	User Agent

1.7 Document History

Revision	Author	Issue Date	Comments
2.4	KMR	9-Sep-2015	First version

1.8 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 Konftel 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 Phones, “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 IP DECT 10 is a single cell VoIP solution with support of up to 20 registered handsets (eg. Konftel 300Wx).

2.1 Hardware Setup

IP DECT 10 network hardware setup can be deployed as follows:

The base-stations are mounted on walls or lamp poles so that each base-station is separated from each other by up to 10m indoor.

The base-station antenna mechanism is based on space diversity feature which improves coverage. The base-stations use complete DECT MAC protocol layer and IP media stream audio encoding feature to provide up to 6 simultaneous calls.

2.2 Components of IP DECT 10

IP DECT 10 is made up of (but not limited to) the following components:

- One IP DECT 10 is connected over an IP network and using DECT as air-core interface.
- Konftel 300Wx
- Web configuration Interface; is a management interface for IP DECT 10 Wireless Solution.

2.2.1 IP DECT 10

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 VoIP 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.

2.3 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 Users (SIP registrations) per Base	20
Single Cell Setup: Max ## Simultaneous Calls	6

3 Installation of Base Station

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

3.1 Package – Contents/Damage Inspection

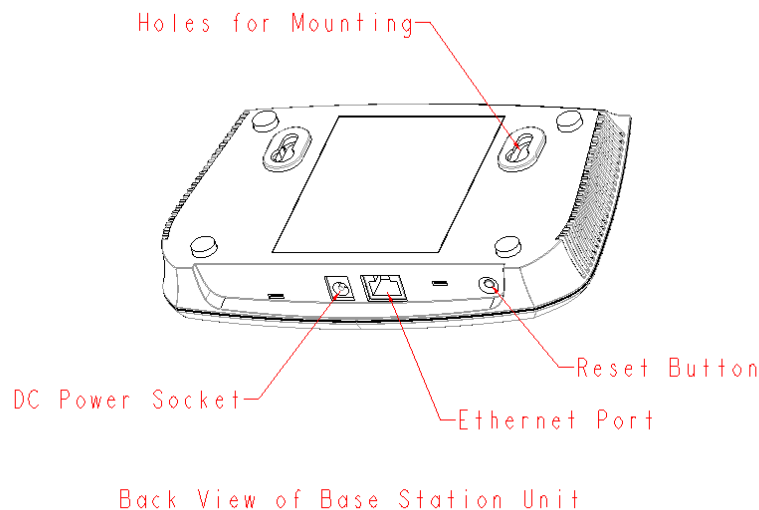
Before Package Is Opened:

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 centre 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:

- 1 x Cat. 5 cable (Ethernet cable)
- Base unit
- Power supply



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 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 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 Konftel 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 Base Station – 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 web 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:

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 Konftel 300Wx status page or using a web browser.

3.5.1 Using Konftel 300Wx to find IP address

On the handset press “Menu” then go to STATUS. Scroll to the bottom of the status view to see the IP address of the base. This can only be found when the handset (Konftel 300Wx) is connected to the base.

3.5.2 Using browser

Open any standard browser and enter the address:

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

for e.g. <http://ipdetect00087B00AA10>. 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 Configuration Interface

- STEP 1** Connect the Base station to a private network via standard Ethernet cable (CAT-5).
- STEP 2** Use section 3.5 to find the IP address and enter that in a web browser.
- 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.

4 VoIP Administration Interface

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

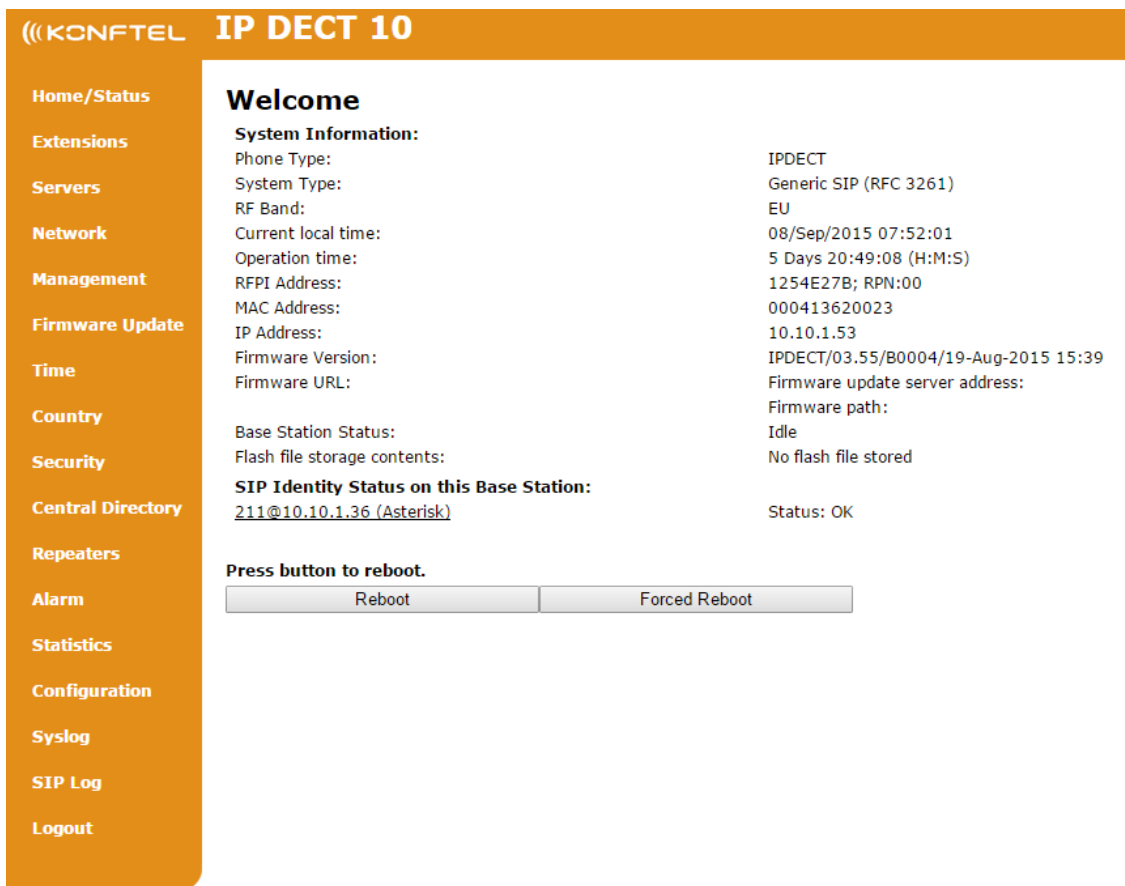
The VoIP Configuration Interface is an in-built HTTP Web Server service residing in each base station. This interface is user friendly interface and easy to handle even to a first time user.

Note: Enabling secure web is not possible. For secure configuration use secure provisioning.

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

4.1 Web navigation

We describe the left menu in the front end of the VoIP 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	<p>Typically the user configures the Network settings from here.</p> <p>NAT provisioning: allows configuration of features for resolving of the NAT – Network Address Translation. These features enable interoperability with most types of routers.</p> <p>DHCP: allows changes in protocol for getting a dynamic IP address.</p> <p>Virtual LAN: specifies the Virtual LAN ID and the User priority.</p> <p>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.</p> <p>Subnet mask: if using DHCP, leave it empty. Only write in, when you use static IP address.</p> <p>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)</p> <p>Default gateway: if using DHCP, leave it empty. Write in the IP address of your router, when you use static IP address.</p>
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	<p>Specifying the country/territory where the network is located ensures that your phone connection functions properly.</p> <p>Note: The base language and country setting are independent of each other.</p>
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	<p>Interface to common directory load of up to 3000 entries using *csv format or configuration of LDAP directory.</p> <p>Note: LDAP and central directory cannot operate at the same time.</p>
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.
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.

4.2 Home/Status

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

Screenshot

Welcome

System Information:

Phone Type:	IPDECT
System Type:	Generic SIP (RFC 3261)
RF Band:	EU
Current local time:	08/Sep/2015 07:55:14
Operation time:	5 Days 20:52:21 (H:M:S)
RFPI Address:	1254E27B; RPN:00
MAC Address:	000413620023
IP Address:	10.10.1.53
Firmware Version:	IPDECT/03.55/B0004/19-Aug-2015 15:39
Firmware URL:	Firmware update server address: Firmware path:
Base Station Status:	Idle
Flash file storage contents:	No flash file stored

SIP Identity Status on this Base Station:

<u>211@10.10.1.36 (Asterisk)</u>	Status: OK
----------------------------------	------------

Press button to reboot.



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 on this Base Station	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.

4.3 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 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.

4.3.1 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.

4.3.2 Add extension

Screenshot

IP DECT 10

Edit extension

Extension:
 Authentication User Name:
 Authentication Password:
 Display Name:
 Mailbox Name:
 Mailbox Number:
 Server:

Call waiting feature:

BroadWorks Feature Event Package:

Forwarding Unconditional Number:

Forwarding No Answer Number: s

Forwarding on Busy Number:

Select Handset(s)

Idx	IPEI
<input type="checkbox"/>	Add Handset N/A
<input checked="" type="checkbox"/>	1 0134239EC0
<input type="checkbox"/>	2 0134209268

Parameter	Default Value(s)	Description
Extension	Empty	Handset phone number 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 Latin 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 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.
Forwarding Unconditional	Empty	Number to which incoming calls must be re-routed to irrespective of the current state of the handset.

Number	Disabled	Forwarding Unconditional must be enabled to function. Note: Feature must be enabled in the SIP server before it can function in the network Note: Feature will be automatically disabled in case the handset or extension is part of a group
Forwarding No Answer Number	Empty	Number to which incoming calls must be re-routed to when there is no response from the SIP end node. 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
	Disabled	Note: Feature must be enabled in the SIP server before it can function in the network
	90	Specify delay from call to forward in seconds. Note: Feature will be automatically disabled in case the handset or extension is part of a group
Forwarding On Busy Number	Empty	Number to which incoming calls must be re-routed to when SIP node is busy. Forwarding On Busy Number must be enabled to function.
	Disabled	Note: Feature must be enabled in the SIP server before it can function in the network Note: Feature will be automatically disabled in case the handset or extension is part of a group

When an extension is added (or edited) it can be selected (right side check box) 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

4.3.3 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.

Screenshot

Extensions and Handset

AC:

Local Call Groups: ▼

Extensions / Handset

	Idx	Extension	Display Name	Server	Server Alias	State	IPEI
<input type="checkbox"/>	1	<u>211</u>	300Wx 210	10.10.1.36	Asterisk	SIP Registered	<u>0134239EC0</u>
<input type="checkbox"/>	2	<u>212</u>		10.10.1.36	Asterisk		<u>0134209268</u>
<input type="checkbox"/>	<u>3</u>			10.10.1.36	Asterisk		
<input type="checkbox"/>	<u>4</u>			10.10.1.36	Asterisk		
<input type="checkbox"/>	<u>5</u>			10.10.1.36	Asterisk		
<input type="checkbox"/>	<u>6</u>			10.10.1.36	Asterisk		
<input type="checkbox"/>	<u>7</u>			10.10.1.36	Asterisk		

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 a unique DECT identification number. Group call: One extension can be associated to up to 20 IPEI's. The IPEI's will be listed in this cell.

4.3.4 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.

Screenshot

Extensions and Handset

AC:

Local Call Groups:

Extensions / Handset

Add Handset

	<u>Idx</u>	<u>IPEI</u>	<u>Handset State</u>	<u>Handset Type FW Info</u>	<u>FWU Progress</u>	<u>Extension</u>
<input type="checkbox"/>	1	0134239EC0	Present	3Party 00/00/00 00:00	Off	211
<input type="checkbox"/>	2	0134209268				212
Check All /						
Uncheck All						

With selected: [Delete Handset\(s\)](#) [Register Handset\(s\)](#) [Deregister Handset\(s\)](#)

Parameter	Description
Idx	Select / deselect for delete, register and deregister handsets
IPEI	Handset IPEI. IPEI is unique DECT identification number.
Handset state	The state of the given handset: Present: The handset is DECT located at the base Detached: The handset is detached from the system (e.g. powered off) Removed: The handset has been out of sight for a specified amount of time (~one hour).
Handset Type FW info	Handset type and firmware version of handset

FWU Progress	<p>Possible FWU progress states:</p> <p>Off: Means sw version is specified to 0 = fwu is off</p> <p>Initializing: Means FWU is starting and progress is 0%.</p> <p>X% : FWU ongoing</p> <p>Verifying X%: FWU writing is done and now verifying before swap</p> <p>"Waiting for charger" (HS) / "Conn. term. wait" (Repeater): All FWU is complete and is now waiting for handset/repeater restart.</p> <p>Complete HS/repeater: FWU complete</p> <p>Error: Not able to fwu e.g. file not found, file not valid etc</p>
Extension	<p>Given extension is displayed.</p> <p>Group call: The cell will show all the extensions associated with this handset and IPEI.</p>

4.3.4.1 Handset and extension list top/sub-menus

The handset extension list menu is used to control pairing 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

[Check All /](#)
[Uncheck All](#)

With selected: [Delete Handset\(s\)](#) [Register Handset\(s\)](#) [Deregister Handset\(s\)](#)

In the below table each command is described.

Actions	Description
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).

4.3.5 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.

4.4 Servers

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

Screenshot

Servers

Asterisk:

10.10.1.36

[Add Server](#)

[Remove Server](#)

Asterisk:

Server Alias:	<input type="text" value="Asterisk"/>
NAT Adaption:	<input type="button" value="Enabled"/>
Registrar:	<input type="text" value="10.10.1.36"/>
Outbound Proxy:	<input type="text"/>
Conference Server:	<input type="text"/>
Call Log Server:	<input type="text"/>
Reregistration time (s):	<input type="text" value="600"/>
SIP Session Timers:	<input type="button" value="Disabled"/>
Session Timer Value (s):	<input type="text" value="1800"/>
SIP Transport:	<input type="button" value="UDP"/>
Signal TCP Source Port:	<input type="button" value="Enabled"/>
Use One TCP Connection per SIP Extension:	<input type="button" value="Disabled"/>
Keep Alive:	<input type="button" value="Enabled"/>
Show Extension on Handset Idle Screen:	<input type="button" value="Enabled"/>
Hold Behaviour:	<input type="button" value="RFC 3264"/>
Local Ring Back tone:	<input type="button" value="Enabled"/>
Attended Transfer Behaviour:	<input type="button" value="Hold 2nd Call"/>
Directed Call Pickup:	<input type="button" value="Disabled"/>
Directed Call Pickup Code:	<input type="text"/>
Group Call Pickup:	<input type="button" value="Disabled"/>
Group Call Pickup Code:	<input type="text"/>
Use Own Codec Priority:	<input type="button" value="Disabled"/>
DTMF Signalling:	<input type="button" value="RFC 2833"/>
DTMF Payload Type:	<input type="text" value="101"/>
Remote Caller ID Source Priority:	<input type="button" value="PAI - FROM"/>
Codec Priority:	<input type="text" value="G722"/> <input type="text" value="G711U"/> <input type="text" value="G711A"/> <input type="text" value="G726"/> <input type="button" value="Up"/> <input type="button" value="Down"/>
RTP Packet Size:	<input type="button" value="20 ms"/>
Secure RTP:	<input type="button" value="Disabled"/>
Secure RTP Auth:	<input type="button" value="Disabled"/>
SRTP Crypto Suites:	<input type="text" value="AES_CM_128_HMAC_SHA1_32"/> <input type="text" value="AES_CM_128_HMAC_SHA1_80"/> <input type="button" value="Up"/> <input type="button" value="Down"/>
<input type="button" value="Reset Codecs"/> <input type="button" value="Remove"/>	
<input type="button" value="Reset Crypto Suites"/> <input type="button" value="Remove"/>	
<input type="button" value="Save"/> <input type="button" value="Cancel"/>	

Parameter	Default value	Description
Server Alias	Empty	Parameter for server alias
NAT Adaption	Disabled	To ensure all SIP messages goes directly to the NAT gateway in the SIP aware router.
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".
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 of 8660 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 24nalyse24n 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 1.0
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 (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).
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 25analyse25n 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 sendonly, recvonly or inactive dependant of the context RFC 2543: The "old" way of 25analyse25ng 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 2 nd 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 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 <u>same</u> 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 <u>different</u> 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.722 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. With G.729 add on DSP module for the base is required.
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.

4.5 Network

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

4.5.1 IP Settings

Screenshot

IP settings

DHCP/Static IP:	<input type="text" value="DHCP"/>
IP Address:	<input type="text" value="10.10.1.53"/>
Subnet Mask:	<input type="text" value="255.255.255.0"/>
Default Gateway:	<input type="text" value="10.10.1.1"/>
DNS (Primary):	<input type="text" value="10.10.1.10"/>
DNS (Secondary):	<input type="text" value="10.10.1.18"/>

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: 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.
------------------------	----	----------------------------------

4.5.2 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

VLAN Settings

ID:

User Priority:

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.

4.5.3 DHCP Options

Screenshot

DHCP Options

Plug-n-Play:

Parameter	Default Values	Description
Plug-n-Play	Enabled	Enabled: DHCP option 43 to automatically provide PBX IP address to base.

4.5.4 NAT Settings

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

Screenshot

NAT Settings

Enable STUN:	<input type="text" value="Disabled"/>
STUN Server:	<input type="text"/>
STUN Bindtime Determine:	<input type="text" value="Enabled"/>
STUN Bindtime Guard:	<input type="text" value="80"/>
Enable RPORT:	<input type="text" value="Disabled"/>
Keep alive time:	<input type="text" value="90"/>

Parameter	Default Values	Description
Enable STUN	Disabled	Enable to use STUN
STUN Server	NA	Permitted value(s): AAA.BBB.CCC.DDD (Currently only Ipv4 are supported) or url
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

4.5.5 SIP/RTP Settings

These are some definitions of SIP/RTP settings:

Screenshot

SIP/RTP Settings

Use Different SIP Ports:	<input type="text" value="Disabled"/>
RTP Collision Detection:	<input type="text" value="Enabled"/>
Always reboot on check-sync:	<input type="text" value="Disabled"/>
Local SIP port:	<input type="text" value="5060"/>
SIP ToS/QoS:	<input type="text" value="0x68"/>
RTP port:	<input type="text" value="50004"/>
RTP port range:	<input type="text" value="40"/>
RTP ToS/QoS:	<input type="text" value="0xB8"/>

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	
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. <ul style="list-style-type: none"> 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

4.6 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

Management Settings

Base Station Name:

Settings

Management Transfer Protocol:

HTTP Management upload script:

User Name:

HTTP Management password:

Enable Automatic Prefix:

Set Maximum Digits of Internal Numbers:

Set Prefix for Outgoing Calls:

Configuration

Configuration Server Address:

Configuration File Download:

Base Specific File:

DHCP Controlled Config Server:

DHCP Custom Option:

DHCP Custom Option Type:

Text Messaging

Text Messaging:

Text Messaging & Alarm Server:

Text Messaging Port:

Text Messaging Keep Alive (m):

Text Messaging Response (s):

Text Messaging TTL:

Syslog/SIP Log

Upload of SIP Log:

SIP Log Server Address:

Syslog Level:

Syslog Server IP Address:

Syslog Server Port:

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	Empty	Password that should be entered in order to have access to the

Management password		configuration server. Permitted value(s): 8-bit string length
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
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.
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 Typr	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.
Text Messaging	Disabled	Disable/enable messaging with Mobicall server The third option is to "Enable Without Server". With this setting handset can send messages to other handsets, which support messaging. Note: Contact Mobicall to get the proper version and setup for Mobicall server
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
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 33nalyse level also contains the messages from normal operation. Debug: Used by IP DECT 10 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.

For further details refer to doc reference [3].

4.7 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

Firmware Update Settings

Firmware update server address:

Firmware path:

Image path:

Type	Required version	Required branch	Startup image	Background image
8630	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text"/>	<input type="text"/>

Update Base Stations

Required version	Required branch
<input type="text"/>	<input type="text"/>

Parameter	Default Value(s)	Description
Firmware update server address	Empty	IP address or DNS of firmware update files source Valid Input(s): AAA.BBB.CCC.DDD or <URL> Example: 10.10.104.41
Firmware path	Empty	Location of firmware on server (or firmware update server path where firmware update files are located). Example: /East_Fwu Note: Must begin with (/) slash character
Required Version Type	Empty	Version of firmware to be upgraded (or downgraded) on handset type or repeater. Valid Input(s): 8-bit string length. E.g. 280 Note: Value version 0 will disable firmware upgrade for handsets and/or repeater Note: Two handset types will be serial firmware upgraded. First type 8630 then type 8430.
Required Version Base	Empty	Version of firmware to be upgraded (or downgraded) on Base station. Base units are referred to as gateways over here. Valid Input(s): 8-bit string length. E.g. 280

4.8 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

Time Settings

	<input type="button" value="Time PC"/>
Time Server:	<input type="text" value="10.10.1.10"/>
Allow broadcast NTP:	<input checked="" type="checkbox"/>
Refresh time (h):	<input type="text" value="24"/>
Set timezone by country/region:	<input type="checkbox"/>
Timezone:	<input type="text" value="+1:00"/> ▼
Set DST by country/region:	<input checked="" type="checkbox"/>
Daylight Saving Time (DST):	<input type="text" value="Automatic"/> ▼
DST Fixed By Day:	<input type="text" value="Use Month and Day of Week"/> ▼
DST Start Month:	<input type="text" value="March"/> ▼
DST Start Date:	<input type="text" value="0"/>
DST Start Time:	<input type="text" value="2"/>
DST Start Day of Week:	<input type="text" value="Sunday"/> ▼
DST Start Day of Week Last in Month	<input type="text" value="Second First In Month"/> ▼
DST Stop Month:	<input type="text" value="November"/> ▼
DST Stop Date:	<input type="text" value="0"/>
DST Stop Time:	<input type="text" value="2"/>
DST Stop Day of Week:	<input type="text" value="Sunday"/> ▼
DST Stop Day of Week Last in Month	<input type="text" value="First In Month"/> ▼
<input type="button" value="Save and Reboot"/> <input type="button" value="Save"/> <input type="button" value="Cancel"/>	

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
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 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.

4.9 Country

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

Screenshot

Country

Select country:

State / Region:

Select Language:

Set timezone by country/region:

Set DST by country/region:

Notes:

Parameter	Default Values	Description
Select Country	Germany	Supported countries: Australia, Belgium, Brasil, Denmark, Germany, Spain, France, Ireland, Italia, Luxembourg, Nederland, New Zealand, Norway, Portugal, Swiss, Finland, Sweden, Tyrkey, United Kingdom, US/Canade, Austria
State / Region	NA	Only shown by country selection US/Canada, Auustralia, Brasil
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, Brasil

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

4.10 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 is server dependent and in order to configure user must use the web option Servers (refer to servers web).

4.10.1 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

Security

Certificates:

	Idx	Issued To	Issued To	Valid Until
<input type="checkbox"/>	0			
<input type="checkbox"/>	1			
<input type="checkbox"/>	2			
<input type="checkbox"/>	3			

[Check All](#) / [Uncheck All](#)

With selected: [Delete Certificate\(s\)](#)

Import Trusted Certificates:

Filename:

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

Use Only Trusted Certificates:

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

4.10.2 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

SIP Client Certificates:

	Idx	Issued To	Issued To	Valid Until
<input type="checkbox"/>	0			
<input type="checkbox"/>	1			

[Check All /Uncheck All](#)

With selected: [Delete Certificate\(s\)](#)

Import SIP Client Certificate and Key Pair:

Filename:

4.10.3 Password

In the below the password parameters are defined.

Screenshot

Password:

Username:

Current Password:

New Password:

Confirm Password:

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 accidentally wrong changes of passwords

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

Password valid numbers: 0-9
 Password valid letters: a-z and A-Z

4.11 Central Directory and LDAP

The 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.

4.11.1 Local Central Directory

Select local and save for local central directory.

Screenshot

Central Directory

Location: 

Server:

Filename:

Phonebook reload interval (s):

Import Central Directory:

Filename:

Parameter	Default Values	Description
Local	Local	Drop down menu to select between local central directory and LDAP based central directory
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.

4.11.1.1 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 support 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. For further details of the central directory feature refer to appendix.

4.11.2 LDAP

Select LDAP Server and save for LDAP server configuration.

Screenshot

LDAP Central Directory

Central Directory Location:

Server:

Port:

Sbase:

LDAP Filter:

Bind:

Password:

Virtual Lists:

Handset Identity:

Name:

Work:

Home:

Mobile:

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 depends 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
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

4.12 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

All four 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.

4.12.1 System data

The system data web is access by <http://ip/SystemStatistics.html> and data is organised in a table as shown in below example.

Screenshot

Statistics



System / Calls / Repeater / DECT

Base Station Name	Operation/Duration D-H:M:S	Busy	Busy Duration D-H:M:S	SIP Failed	Handset Removed	Searching	Free Running	DECT Source Changed
Sum	0-00:31:12/ 5-21:27:59	0		127	1	0	0	0

The table is organised 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	Total operation time for the base
Busy Count	Busy Count is the number of times the base has been busy.
Busy Duration	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 it's 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

4.12.2 Call data

The call data web is access by <http://ip/CallStatistics.html> and data are organised in a table as shown in below example.

Screenshot

System / Calls / Repeater / DECT

Base Station Name	Operation/ Duration D-H:M:S	Count	Dropped	No Response	Duration D-H:M:S	Active	Max Active	Codec G711U: G711A: G729: G722: G726:	Handover Success	Handover Failed	Audio Not Detected
Sum	0-00:31:45/ 5-21:28:32	106	1	0	0-00:26:32	0	3	10:41:0:36:0	0	0	46

The table is organised 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 is the number of calls that have no response, e.g. if a 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 (Not active DECT calls, but active calls). On one base there can be up to 30 active calls.
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 Success	Counts the number of successful handovers.
Handover Failed	Counts the number of failed handovers.

4.12.3 DECT data

The DECT data web is access by <http://ip/DectStatistics.html> and data is organised in a table as shown in below example.

Screenshot

System / Calls / Repeater / DECT

	Slot0	Slot1	Slot2	Slot3	Slot4	Slot5	Slot6	Slot7	Slot8	Slot9	Slot10	Slot11
Frequency0	11	0	6	0	16	0	11	0	7	0	7	0
Frequency1	11	0	16	0	16	0	13	0	16	0	12	0
Frequency2	13	0	15	0	15	0	15	0	10	0	10	0
Frequency3	15	0	16	0	12	0	17	0	12	0	9	0
Frequency4	9	0	22	0	11	0	16	0	10	0	15	0
Frequency5	21	0	19	0	16	0	14	0	22	0	17	0
Frequency6	8	0	5	0	10	0	3	0	4	0	4	0
Frequency7	13	0	16	0	19	0	10	0	14	0	8	0
Frequency8	22	0	16	0	18	0	12	0	11	0	16	0
Frequency9	19	0	15	0	14	0	22	0	24	0	24	0

Please note 3 frequencies are manually removed in the example system.

4.13 Settings – Configuration File Setup

This page provides non editable information showing the native format of entire 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:

- 1) Using the VoIP Configuration interface to make changes. Each page of the HTTP web interface is a template for which the user can customise settings in the configuration file.
- 2) Retrieving the relevant configuration file from the TFTP and modify and enter new changes. This should be done with an expert network administrator.
- 3) Navigate to the settings page of the VoIP 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.

For details refer to [3].

An example of contents of settings is as follows:

```
~RELEASE=UMBER_FP_V0054
%GMT_TIME_ZONE%: 16
%COUNTRY_VARIANT_ID%: 18
%FWU_POLLING_ENABLE%: 0
%FWU_POLLING_MODE%: 0
%FWU_POLLING_PERIOD%: 86400
%FWU_POLLING_TIME_HH%: 3
%FWU_POLLING_TIME_MM%: 0
%DST_ENABLE%: 2
%DST_FIXED_DAY_ENABLE%: 0
%DST_START_MONTH%: 3
%DST_START_DATE%: 1
. . . .
. . . .
```

4.14 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.

4.15 SIP Logs

This page shows SIP server related messages that are logged during the operation of the 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.

5 Firmware Upgrade Procedure

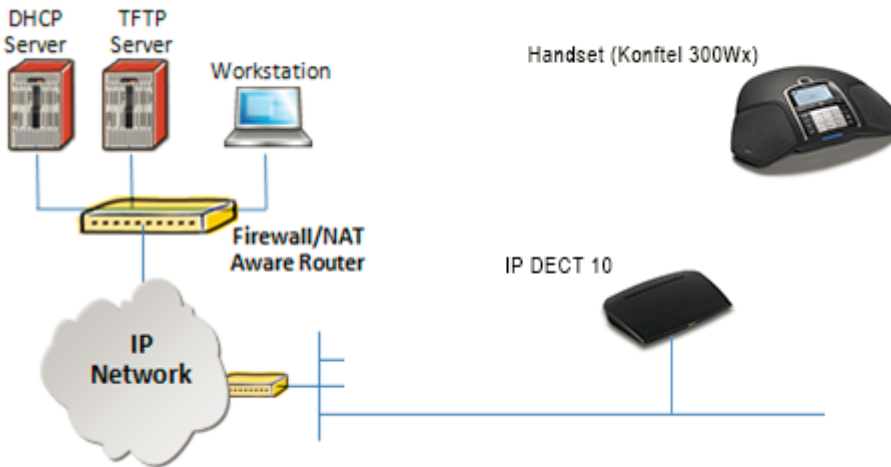
This step-by-step chapter describes how to upgrade or downgrade base station(s) and/or handset(s) / repeater (s) to the relevant firmware provided by Konftel.

5.1 Network Dimensioning

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

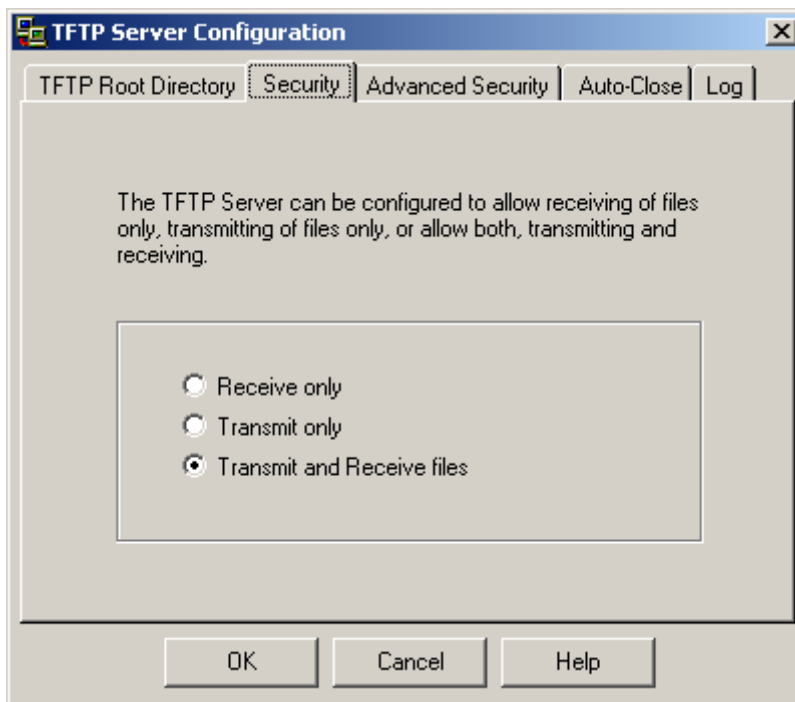
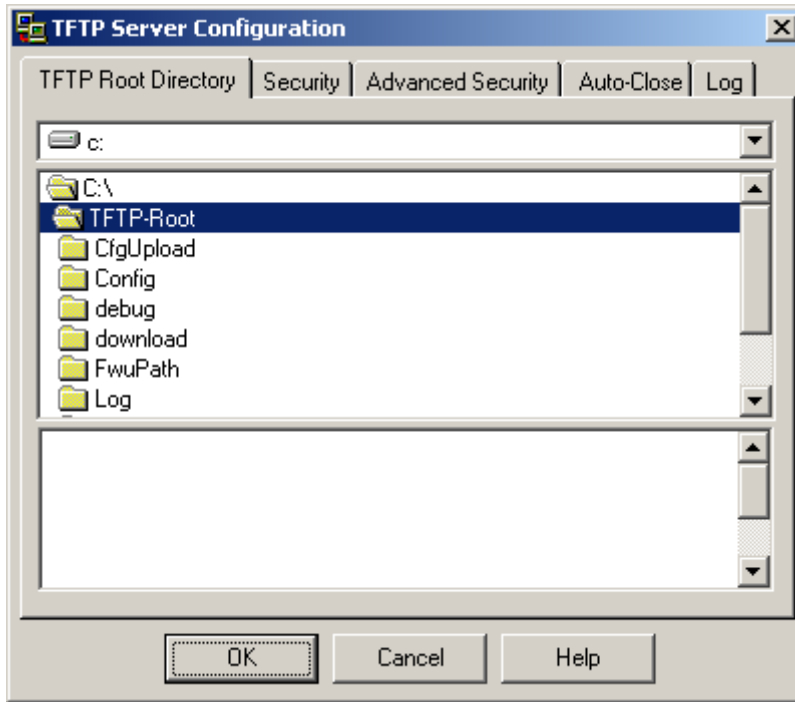
The minimum hardware and software components that are required to be able update via TFTP include the following (but not limited to):

- 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



5.2 TFTP Configuration

This section illustrate 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.



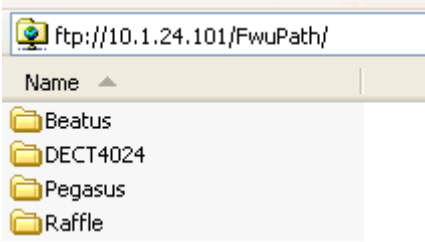
NOTE: If TFTP server timeout settings are too short firmware upgrade might not complete. Recommended time out setting is more than 3 seconds.

5.3 Create Firmware Directories

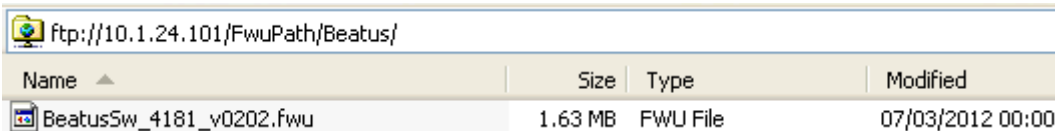
The admin from the service provider’s side must create the relevant firmware directory in the server where both old and new firmware(s) can be placed in it. (See the STEP above)

5.3.1 Base:

On the TFTP server root, create directory "9430".



Copy Base station firmware to the named directory.



IMPORTANT:

The 9430 directory name cannot be changed.

5.4 Firmware Update Settings

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

Firmware Update Settings

Firmware update server address:

Firmware path:

Image path:

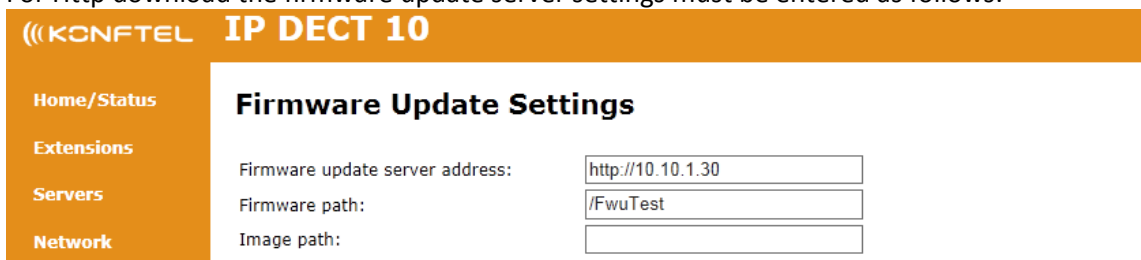
Type	Required version	Required branch	Startup image
8630	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text"/>

Update Base Stations

Required version	Required branch
<input type="text"/>	<input type="text"/>

Type IP address and firmware path followed by save.

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



5.5 Base Station Firmware Upgrade

On the **Firmware Update Settings** page > scroll down to the **Update Gateways** section > Enter the relevant firmware version of the base station to upgrade or to downgrade. Enter 202 for base version V0202.

Update Base Stations

Required version	Required branch
<input type="text"/>	<input type="text"/>
<input type="button" value="Save/Start Update"/>	

Efter entering required version choose **Start update** button > select **OK** button from the dialog window to start the update/downgrade procedure.

The base station will automatically reboot and retrieve the firmware specified from the server and update itself accordingly.

The base firmware update behaviour 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 immediately the firmware update procedure starts.

5.5.1 Base firmware confirmation

Base station firmware version status can be seen on the web Welcome page.

Welcome

System Information:

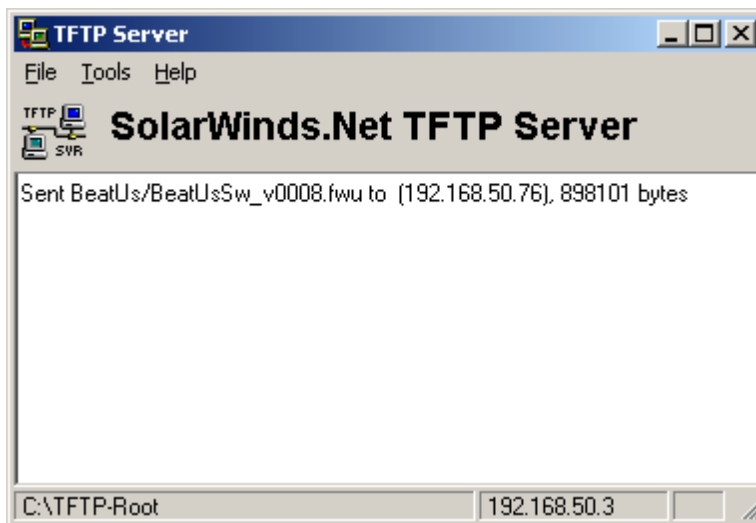
Phone Type:	IPDECT
System Type:	Generic SIP (RFC 3261)
RF Band:	EU
Current local time:	08/Sep/2015 08:58:53
Operation time:	00:59:12 (H:M:S)
RFPI Address:	1254E27B; RPN:00
MAC Address:	000413620023
IP Address:	10.10.1.53
Firmware Version:	IPDECT/03.55/B0004/19-Aug-2015 15:39
Firmware URL:	Firmware update server address: http://10.10.1.30
	Firmware path: FwuTest
Base Station Status:	Idle

5.5.2 Verification of Firmware Upgrade

Syslog information when Management Syslog level is set to “Debug”

```
[ FWU Downloading File tftp://10.1.24.101/FwuPath/Beatus/BeatusSw_4181_v0202.fwu]
[ Base FWU started]
[ Base FWU ended with exit code 2101 (NE_FILE_TRANSFER_EOF): End of file]
```

The log window of the TFTP server:



6 Functionality Overview

So far we have setup our VoIP system. Next, in this chapter we list what features and functionalities are available in the system. The 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 VOIP network functionalities are:

- **Outgoing/incoming voice call management:** The 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 in order to provide differentiated services and tariff management. Within a DECT traffic area, established calls can seamlessly be handover between Base-station and repeaters using connection handover procedures.
- **Security:** The IP DECT 10 system also supports robust security functionalities for Base-station. Most security¹ functionality is intrinsically woven into the VOIP network structure so that network connections can be encrypted and terminal authentication can be performed.

6.1 Base Station Interfaces

Interfaces	
Power	Input: 100-240 VAC 50-60Hz (90 – 265 VAC) Output Nom: 5VDC 1000mA Type: Switch mode single or multi-plug solution Plugs: UK, EU, US and AUS
LAN Interface	Standard : 10BASE-T(IEEE 802.3 100Mbps) Connector: RJ45 8/8
Keys	
	1: Reset key, Page and Default
LED indicator	
	One Status LED (multicolour, red, green, orange)
RF	
Frequency Bands	1880 – 1900 MHz (EMEA) 1910 – 1930 MHz (Latam) 1920 – 1930 MHz (USA) Factory setting which can't be modified after production
Output Power	250 mW or 140mW depending on country version
Antenna	Two antennas for diversity
Software upgrade	
Downloadable	Remote firmware update using HTTP, HTTPS or TFTP
Temperatures	
Operation	0°C to 40°C

¹ With active security 4 channels is supported

6.2 Software Features

CODEC's	
G.711 PCM A-law & U-law	Yes
G.722	Yes
G.726	Yes
G.729	A/AB (including VAD), max 4 coders G729 licence not included
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
RFC3326	The Reason Header Field for the Session Initiation Protocol
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	Secure SIP
In-band DTMF	No
SRTP	Yes, packet authentication will limit the number of calls to 4
SIP registrations	max 20
RTP streams	max 10
SIP transport	UDP, TCP or TLS
Web server	
	Embedded web server, accessed using HTTP
Other features	
IP quality	Warning – Network outage, VoIP service outage
Jitter buffer	Yes, adaptive
Automatic DST	Yes
Tone Scheme	Country Depend Tone Scheme
Provisioning	Yes
Re-direct server	Yes
SIP configuration	Yes, from web page or configuration file
Call groups	Yes
IP features	
IPv4	Yes
IPv6	Hardware ready, software not included

TCP/IP/UDP	Yes
DHCP Support	Yes
DHCP option	66, 120
Static IP	Yes
DNS srv	Yes
VLAN	Yes, 802.1p/q
Quality of service	Type of Service (ToS) including DiffServ Tagging, and QoS per IEEE 802.1p/q
TLS	Yes, 1.0
Certificates	Yes, X.509 (certificate not included)
TFTP	Yes, for firmware and configuration file download
HTTP server	Yes
HTTP client	Yes, for firmware and configuration file download
HTTPS	Yes, for firmware and configuration file download
SNTP	Yes, For internet clock synchronization
DECT	
DECT handover	Yes, inter-cell handover for repeater support
CAT-IQ v1.0	HD audio or NB audio support
Repeater support	Yes
Intercom	No
DECT encryption	Yes
DECT Authentication	Yes
Group TPUI support	Yes, for call groups
GAP compliant	No
CAT-IQ compliant	No
Handset registrations	20

6.3 Call Features

Call supported	5 simultaneous call supported
Simultaneous calls/base	5 Wideband calls (g.722). 5 narrowband calls (PCMA, PCMU, G.726) or 4 when using G729
Simultaneous calls/handset	2
Call features	Codec Negotiation
	Codec Switching
	Missed call notification
	Voice message waiting notification
	Date and Time synchronization
	Parallel calls
	Call Hold
	Call Retrieve
	Call transfer unannounced
	Call transfer announced
	Conference (3PTY)
	Conference, Network
	Call Waiting Indication
	Calling line identity
	Outgoing call
	Call Toggle/Swap
	Incoming call
	Line identification
	Multiple Lines

	Multiple calls
	Call identification
	Calling Name Identification Presentation (CNIP)
	Calling Line Identification Presentation (CLIP)
	Call Completed Elsewhere
	Distinctive Ringing
Central Phone Book:	
- LDAP	Yes
- XML	Yes, remote or file load from web interface
- CSV	Yes, file load through web interface
DND:	Yes
Call Forward:	Configurable from base or handset (Not with Call Group active)
- CFU	Yes
- CFNA	Yes
- CFB	Yes
Call groups:	Yes, 1-20 handsets/SIP account

Appendix

7 Appendix A: Basic Network Server(s) Configuration

In this chapter we describe how to setup the various server elements in the system.

7.1 Server setup

In the network, the server environment is installed as a centralized system.

The main server types hosted on the network include SIP, DNS/DHCP and HTTP/TFTP Servers. These servers can be hosted both in one or multiple windows and/or Linux Server environment.

Management servers are normally installed to monitor and manage the network in detail. Each Base-station status can be checked. Each Repeater and each Subscriber Terminal can be monitored over the air from a centralized location.

Further, new software can be uploaded to all system elements from the centralized location (typically a TFTP server) on an individual basis. This includes Subscriber Handsets where the latest software is downloaded over the air.

7.2 Requirements

Regardless of whether or not you will be installing a centrally provisioned system, you must perform basic TCP/IP network setup, such as IP address and subnet mask configuration, to get your organization's phones up and running.

7.3 DNS Server Installation/Setup

Name server is a name server service installed in a server for mapping or resolution of humanly memorable domain names and hostnames into the corresponding numeric Internet Protocol (IP) addresses.

The customer should refer to the platform vendor either windows or Linux vendor for detail step-by-step guide on how to install and configure Domain Name System for internet access. In this section, we briefly describe hints on how to setup DNS behind NAT or Firewall.

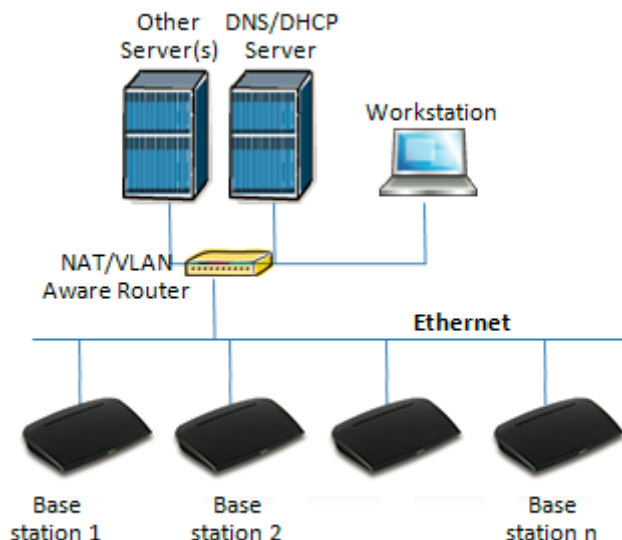
7.3.1.1 Hints on how to Configure DNS behind a Firewall/NAT

Proxy and Network Address Translation (NAT) devices can restrict access to ports. Set the DNS to use UDP port 53 and TCP port 53. For windows Servers, set the RCP option on the DNS Service Management console and configure the RCP to use port 135.

These settings should be enough to resolve some of potential issues that may occur when you configure DNS and firewalls/NAT.

7.4 DHCP Server Setup

A DHCP Server allows diskless clients to connect to a network and automatically obtain an IP address. This server is capable of supplying each network client with an IP address, subnet mask, default gateway, an IP address for a WINS server, and an IP address for a DNS server. This is very often used in enterprise networks to reduce configuration efforts. All IP addresses of all computers/routers/bases are stored in a database that resides on a server machine.



The network administrator should contact the relevant vendors for detail information or step-by-step procedure on how to install and setup DHCP process or service on windows/Linux servers. In this section, we will provide some hints of how to resolve potential problems to be encountered you setup DHCP Servers.

7.4.1 Hint: Getting DHCP Server to Work

Windows Server:

1) Clients are unable to obtain an IP address

If a DHCP client does not have a configured IP address; it generally means that the client has not been able to contact a DHCP server. This is either because of a network problem or because the DHCP server is unavailable. If the DHCP server has started and other clients have been able to obtain a valid address, verify that the client has a valid network connection and that all related client hardware devices (including cables and network adapters) are working properly.

2) The DHCP server is unavailable

When a DHCP server does not provide leased addresses to clients, it is often because the DHCP service has failed to start. If this is the case, the server may not have been authorized to operate on the network. If you were previously able to start the DHCP service, but it has since stopped, use Event Viewer to check the system log for any entries that may explain the cause.

Next, restart the DHCP service, click **Start**, click **Run**, type **cmd**, and then press ENTER. Type **net start dhcpserver**, and then press ENTER.

Linux Platform:

Troubleshooting DHCP, check the following:

- 1) Incorrect settings in the `/etc/dhcpd.conf` file such as not defining the networks for which the DHCP server is responsible;
- 2) NAT/Firewall rules that block the DHCP **bootp** protocol on UDP ports 67 and 68;
- 3) Routers failing to forward the **bootp** packets to the DHCP server when the clients reside on a separate network. Always check your `/var/logs/messages` file for dhcpd errors.
- 4) Finally restart the **dhcpd** service daemon

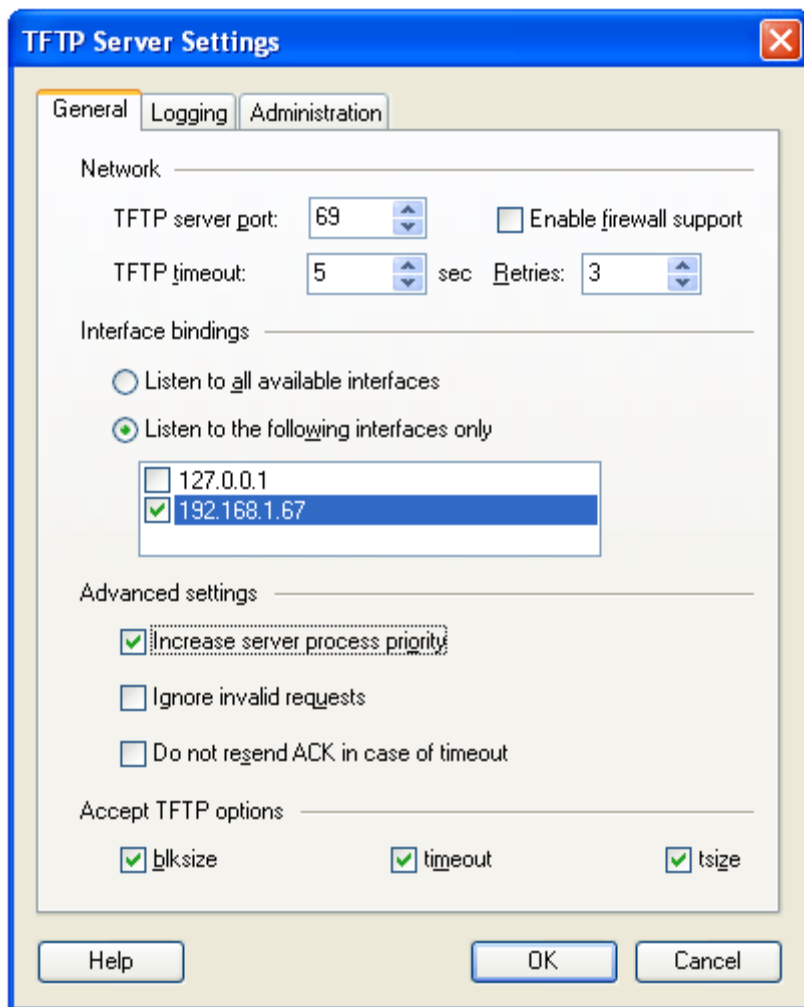
7.5 TFTP Server Setup

There are several TFTP servers in the market place; in this section we describe how to setup a commonly used TFTP Server.

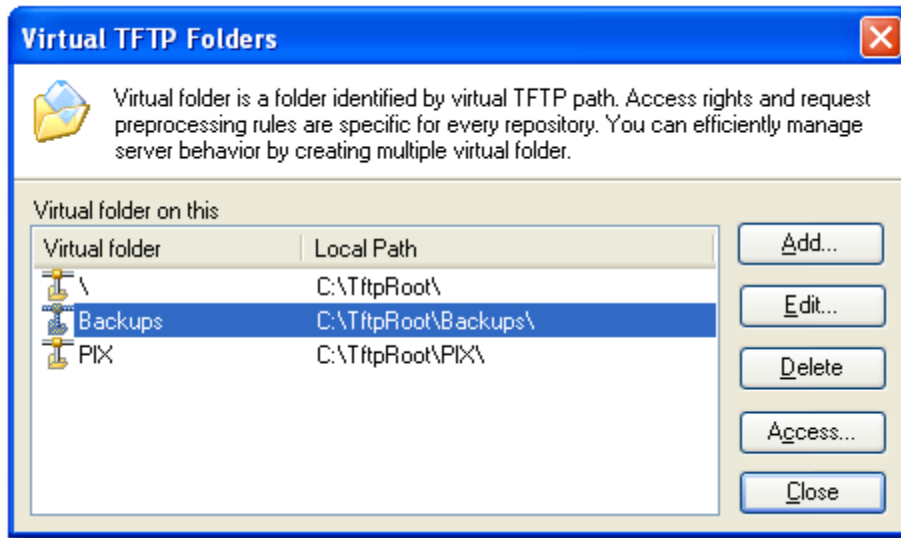
7.5.1 TFTP Server Settings

The administrator must configure basic parameters of the TFTP application:

- Specify UDP 69 port – for TFTP incoming requests and TCP 12000 – for remote management of the server. For file transmission the server opens UDP ports with random numbers. In case the option **Enable NAT or firewall support** is activated on the server, the server uses the same port for files transmission and listening to the TFTP incoming requests (UDP 69 port on default).
- Specify the interface bindings, TFTP root directory, port which the TFTP Server will listen, timeout and number of retries, and TFTP options supported by the server.



- Configure the relevant TFTP virtual folder in the server. The TFTP virtual folder is the file folder, visible for TFTP clients under a certain name. You can set security settings separately for every virtual TFTP folder. Next, set rights to access TFTP folders according to the relevant clients.



7.6 SIP Server Setup

SIP server is one of the main components of a network, dealing with the setup of all SIP calls in the network. A SIP server is also referred to as a SIP Proxy or a Registrar.

Although the SIP server is the most important part of the SIP based phone system, some servers only handles call setup and call tear down. It does not actually transmit or receive any audio. This is done by the media server in RTP.

The IP DECT 10 is fully interoperable with the most of SIP Server applications. There are many off-the-shelf vendor and open source SIP servers. In this section, we will briefly explain settings required to take full advantage of FreePBX SIP Server feature set. The settings are similar for other SIP servers.

7.6.1 FreePBX SIP Server

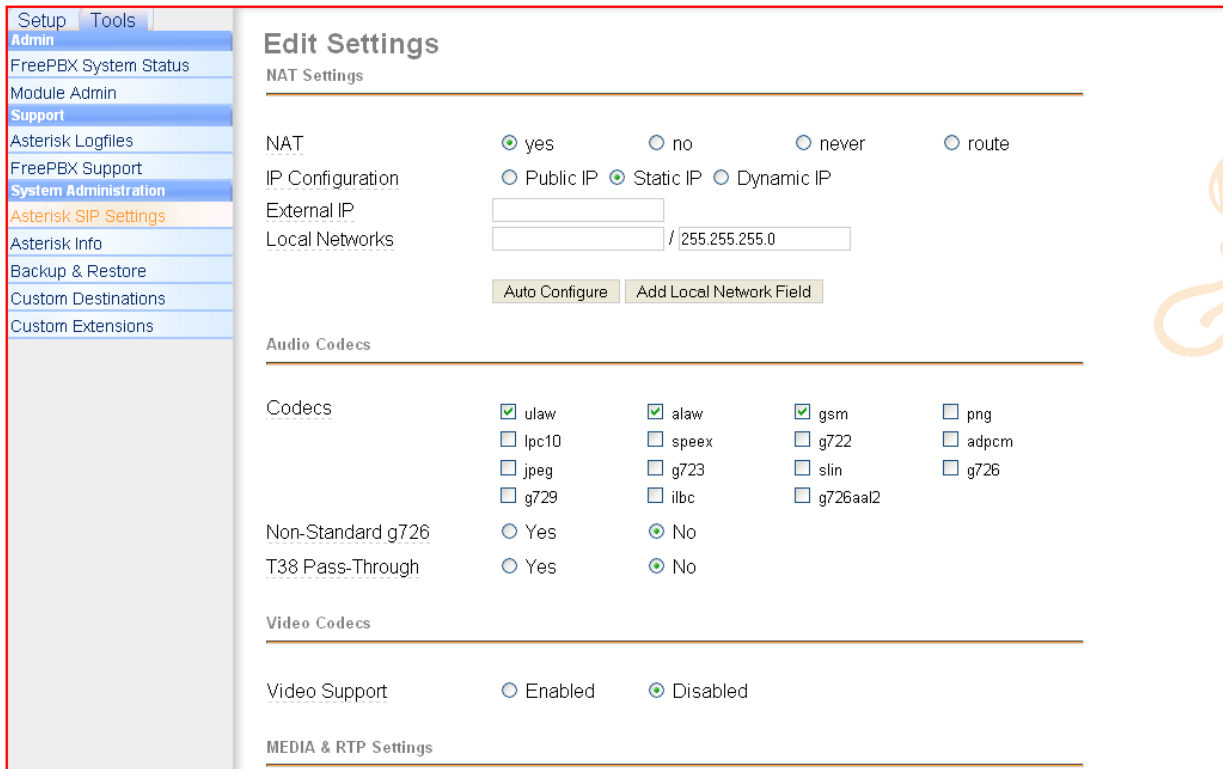
FreePBX is an easy to use GUI (graphical user interface) that controls and manages Asterisk, which the most popular open source telephony engine software.

The administrator should refer to the relevant detail step-by-step procedure of how to install FreePBX SIP server. This section briefly describes SIP Server setup parameters.

1) SIP Server Setup

Settings	Description
NAT	<p>This option determines the settings for users connecting to an asterisk server.</p> <p>Possible values: Yes, No, Never, Route</p> <p>NAT=route Asterisk will send the audio to the port and IP where it's receiving the audio from. Instead of relying on the addresses in the SIP and SDP messages. This will only work if the phone behind NAT send and receive audio on the same port and if they send and receive the signalling on the same port. (The signalling port does not have to be the same as the RTP audio port).</p> <p>NAT=No Asterisk will add an RPORT to the via header of the SIP messages</p> <p>NAT=never This will cause asterisk not to add an RPORT in the VIA line of the sip invite header</p>

Other NAT Settings	Choose the relevant option or enter the settings in IP configuration, External IP, Local Network.
Codecs	Some SIP Servers supports dynamic codec support. Codecs are algorithm used to compress or decompress speech or audio signals. The user should select the relevant Codecs and other speech compression techniques whose traffic will be routed to the network.
Video Codecs	The user should enable this option if network supports video telephony.
Media & RTP Settings	This option should be enabled to provide for deliver media streams (e.g., audio and video) or out-of-band events signalling (DTMF in separate payload type).



Edit Settings

NAT Settings

NAT yes no never route

IP Configuration Public IP Static IP Dynamic IP

External IP

Local Networks / 255.255.255.0

Audio Codecs

Codecs ulaw alaw gsm png
 lpc10 speex g722 adpcm
 jpeg g723 slin g726
 g729 ilbc g726aal2

Non-Standard g726 Yes No

T38 Pass-Through Yes No

Video Codecs

Video Support Enabled Disabled

MEDIA & RTP Settings

2) Extensions

This feature allows administrators create handset profiles in the network. In other words, Extensions describes the Dial plan for the PBX SIP system. Enter the relevant parameters

The screenshot shows the FreePBX 2.7.0.2 web interface. The top navigation bar includes 'Admin', 'Reports', 'Panel', 'Recordings', and 'Help'. The user is logged in as 'admin'. The sidebar menu on the left lists various system administration options, with 'Internal Options & Configuration' selected. The main content area is titled 'Add SIP Extension' and contains several sections: 'Add Extension' with input fields for 'User Extension', 'Display Name', 'CID Num Alias', and 'SIP Alias'; 'Extension Options' with dropdown menus for 'Ring Time' (Default), 'Call Waiting' (Enable), 'Call Screening' (Disable), and 'Pinless Dialing' (Disable); and 'Assigned DID/CID' with input fields for 'DID Description', 'Add Inbound DID', and 'Add Inbound CID'. On the right side, there is a list of existing extensions, including '117 <117>', '118 <118>', '119 <119>', '201 <201>', '400 <400>', '1234 <1234>', 'MYA <2275>', 'MYA1 <2276>', '3000 <3000>', '3001 <3001>', '3002 <3002>', '3003 <3003>', '3004 <3004>', '3005 <3005>', '3006 <3006>', '3007 <3007>', '3008 <3008>', and '3009 <3009>'. A large orange watermark of a hand is visible in the center of the page.

8 Appendix B: Using Base with VLAN Network

In this chapter we describe how to setup a typical VLAN in the network.

8.1 Introduction

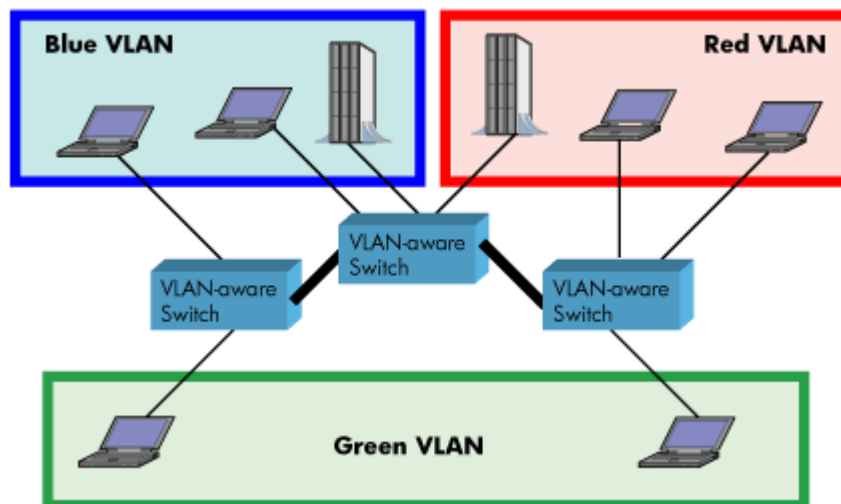
In this chapter, we describe how to setup VLAN to typical network. There are three main stages involved in this procedure:

- 1) Configure a VLAN Aware Switch to a specific (un)tagged VLAN ID, so the system can process untagged frames forwarded to it.
- 2) Setup the Time Server (NTP Server) and other relevant network servers.
- 3) Configure the HTTP server in the Base station to access the features in the PBX or system.

VLAN allows administrators to separate logical network connectivity from physical connectivity analogous to traditional LAN which is limited by its physical connectivity. Normally, users in a LAN belong to a single broadcast domain and communicate with each other at the Data Link Layer or “Layer 2”. LANs are segmented into smaller units for each IP subnets and here communication between subnets is possible at the Network Layer or “Layer 3”, using IP routers.

A VLAN can be described as a single physical network that can be logically divided into discrete LANs that can operate independently of each other.

An Illustration of using VLANs to create independent broadcast domains across switches is shown below:



The figure above highlights several key differences between traditional LANs and VLANs.

- All switches are interconnected to each other. However, there are three different VLANs or broadcast domains on the network. Physical isolation is not required to define broadcast domains. If the figure was a traditional LAN without VLAN-aware switches, all stations would belong to one broadcast domain.
- All switch ports can communicate with one another at the Data Link Layer, if they become members of the same VLAN.

- The physical location of an end station does not define its LAN boundary.
 1. An end station can be physically moved from one switch port to another without losing its “view of the network”. That is, the set of stations it can communicate with at the Data Link Layer remains the same, provided that its VLAN membership is also migrated from port to port.
 2. By reconfiguring the VLAN membership of the switch port an end station is attached to, you can change the network view of the end station easily, without requiring a physical move from port to port.

8.2 Backbone/ VLAN Aware Switches

To implement a VLAN in your network, you must use VLAN-aware switches.

Before we continue, let consider two rules to remember regarding the functioning of a regular LAN switch:

1. When the switch receives a broadcast or multicast frame from a port, it floods (or broadcasts) the frame to all other ports on the switch.
2. When the switch receives a unicast frame, it forwards it only to the port to which it is addressed.

A VLAN-aware switch changes the above two rules as follows:

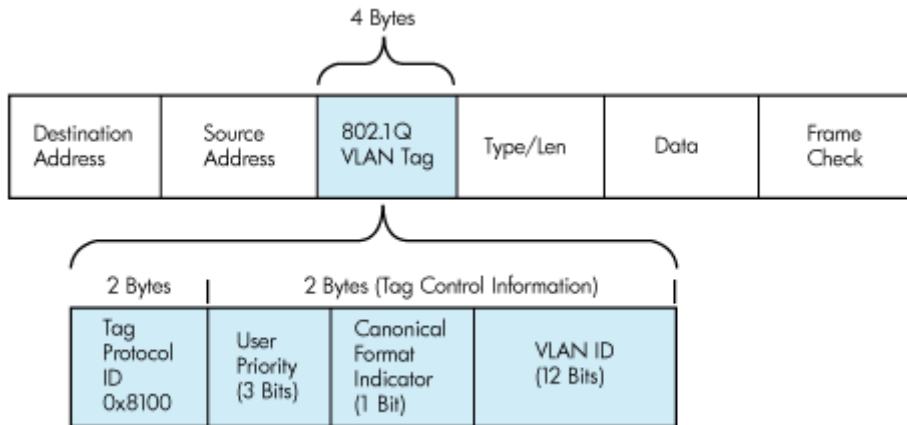
1. When the switch receives a broadcast or multicast frame from a port, it floods the frame to only those ports that belong to the same VLAN as the frame.
2. When a switch receives a unicast frame, it forwards it to the port to which it is addressed, only if the port belongs to the same VLAN as the frame.
3. A unique number called the VLAN ID identifies each VLAN.

Which VLAN Does a Frame Belong To?

The previous section notes that a frame can belong to a VLAN. The next question is—how is this association made?

- A VLAN-aware switch can make the association based on various attributes of the type of frame, destination of MAC address, IP address, TCP port, Network Layer protocol, and so on.

An illustration of IEEE 802.1Q VLAN tag in Ethernet frame is as follows:



8.3 How VLAN Switch Work: VLAN Tagging

VLAN functionality can be implemented via explicit frame tagging by switches and end stations. Network switches and end stations that know about VLANs are said to be VLAN aware. Network switches and end stations that can interpret VLAN tags are said to be VLAN tag aware. VLAN-tag-aware switches and end stations add VLAN tags to standard Ethernet frames—a process called explicit tagging. In explicit tagging, the end station or switch determines the VLAN membership of a frame and inserts a VLAN tag in the frame header (see figure above for VLAN tagging), so that downstream link partners can examine just the tag to determine the VLAN membership.

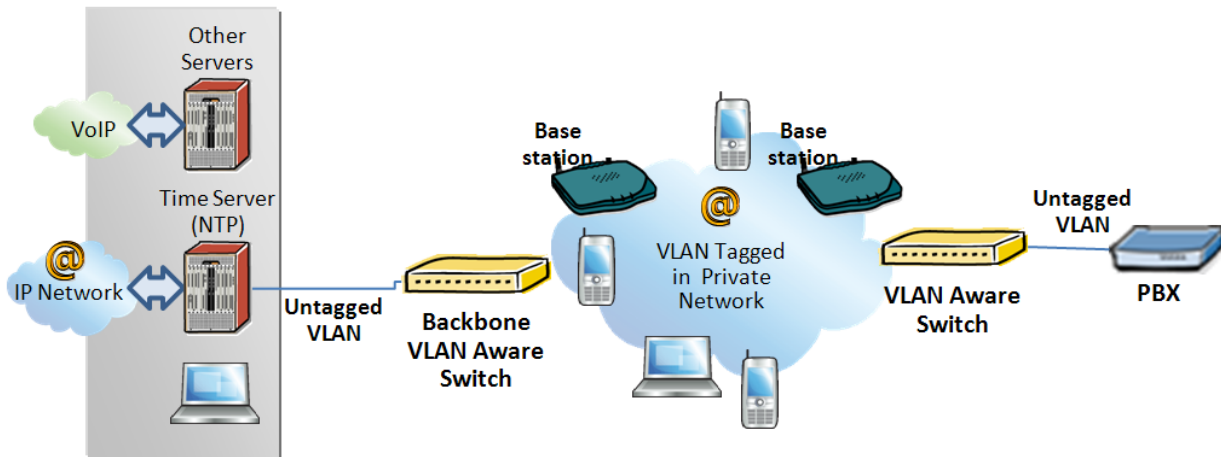
8.4 Implementation Cases

Common types of usage scenarios for VLANs on typical VLAN switches: port-based VLANs, protocol-based VLANs, and IP subnet-based VLANs. Before figuring out which usage scenario suits your needs, you must understand what each type of usage scenario implies.

- Port-based VLAN:** All frames transmitted by a NIC are tagged using only one VLAN ID. The NIC does not transmit or receive any untagged frames.

All protocols and applications use this virtual interface’s virtual PPA to transmit data traffic. Therefore all frames transmitted by that NIC port are tagged with the VLAN ID of that Virtual Interface.

- Protocol-based VLAN:** The NIC assigns a unique VLAN ID for each Layer 3 protocol (such as IPv4, IPv6, IPX, and so on). Therefore, the VLAN ID of outbound frames is different for each protocol. An inbound frame is dropped if the protocol and VLAN ID do not match.
- IP subnet-based VLAN:** The NIC assigns a unique VLAN ID for each IP subnet it belongs to. Therefore, the VLAN ID of outbound frames is different for different destination subnets. An inbound frame is dropped if the IP subnet and VLAN ID do not match.



8.5 Base station Setup

After the admin have setup the Backbone switch, next is to configure the Base station via HTTP interface.

- STEP 1** Connect the Base station to a private network via standard Ethernet cable (CAT-5).
- STEP 2** Use one of the two methods to find the base IP
- STEP 3** On the Login page, enter your authenticating credentials (the username and password is **admin** by default unless it is changed). 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.
- STEP 5** Create the relevant SIP server information in the system. Each service provider/customer should refer SIP server vendor on how to setup SIP servers.

8.6 Configure Time Server

- STEP 6** Navigate to the Time settings and configure it. Scroll on the left column and click on **Time** url link to Open the **Time Settings** Page. Enter the relevant parameters on this page and press the **Save** button.

Time Settings

Time PC

Time Server:

Allow broadcast NTP:

Refresh time (h):

Set timezone by country/region:

Timezone:

Set DST by country/region:

Daylight Saving Time (DST):

DST Fixed By Day:

DST Start Month:

DST Start Date:

DST Start Time:

DST Start Day of Week:

DST Start Day of Week Last in Month:

DST Stop Month:

DST Stop Date:

DST Stop Time:

DST Stop Day of Week:

DST Stop Day of Week Last in Month:

8.7 VLAN Setup: Base station

STEP 7 Navigate to the **Network** url > On the network page enter the relevant settings in the VLAN section > VLAN Id should be the same as those configured into the backbone.

Network Settings

IP settings

DHCP/Static IP:

IP Address:

Subnet Mask:

Default Gateway:

DNS (Primary):

DNS (Secondary):

NAT Settings

Enable STUN:

STUN Server:

STUN Bindtime Determine:

STUN Bindtime Guard:

Enable RPORT:

Keep alive time:

VLAN Settings

ID:

User Priority:

SIP/RTP Settings

Use Different SIP Ports:

RTP Collision Detection:

Always reboot on check-sync:

Local SIP port:

SIP ToS/QoS:

RTP port:

RTP port range:

RTP ToS/QoS:

DHCP Options

Plug-n-Play:

9 Appendix C: Local Central directory file handling

In this appendix the Local Central Directory file format, import and configuration is described.

9.1 Central Directory Contact List Structure

The structure of Contact List is simple. The figure below shows an example of structure of Contact List in Text format and in Xml format. **Contact name must not contain more than 23 characters and contact number must not contain more than 21 digits.**

.csv or .txt

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

.xml

```
File Edit Format View Help
<IPPhonedirectory>
<DirectoryEntry>
<Name>Mark Ross</Name>
<Telephone>100</Telephone>
<Office>+4 501234 56789</Office>
<Mobile>+4 511234 56789</Mobile>
<Fax>+4 521234 56789</Fax>
</DirectoryEntry>
</IPPhonedirectory>
```

Txt file limitations:

- Contact name must NOT be longer than 23 characters (name will be truncated)
- Contact name must NOT contain “,”
- Contact number must be limited to 21 digits (entry will be discarded, no warning)
- Contact number digits must be: +0123456789
- Contact number does not support SIP-URI
- Spaces between name section “,” and number section is not supported

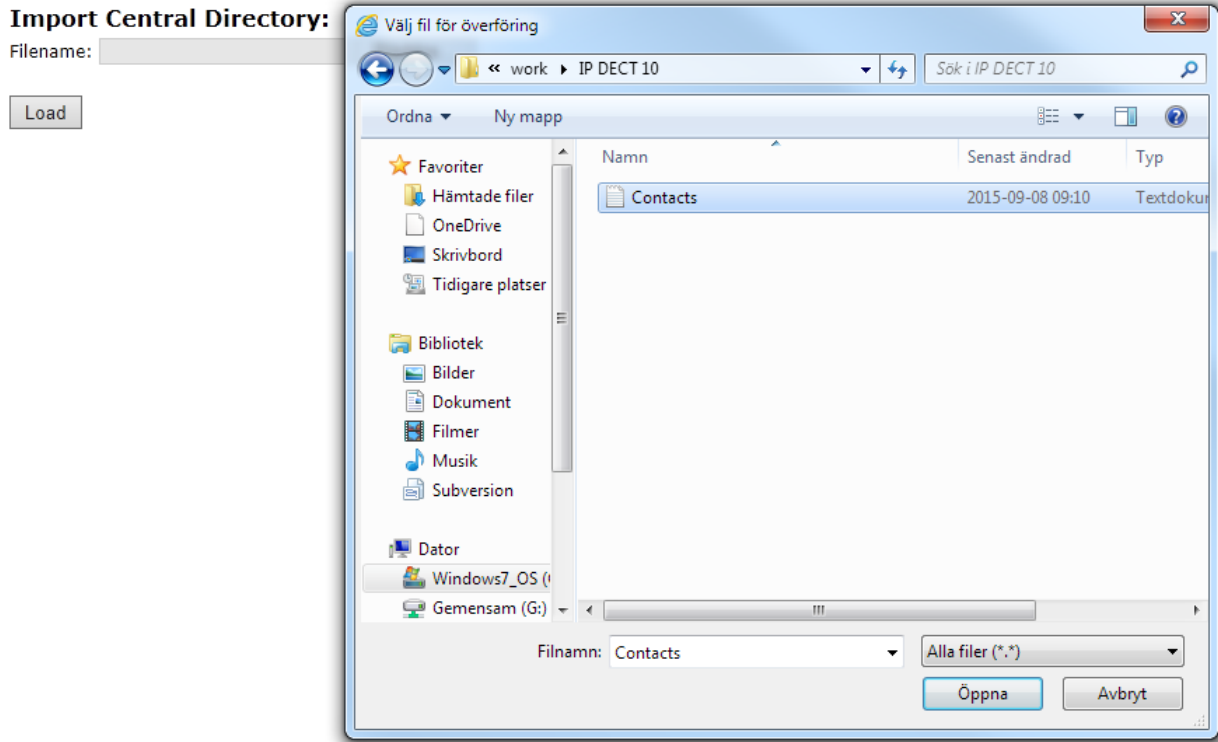
9.2 Central Directory Contact List Filename Format

The Contact list is saved as file format: .txt .csv or .xml

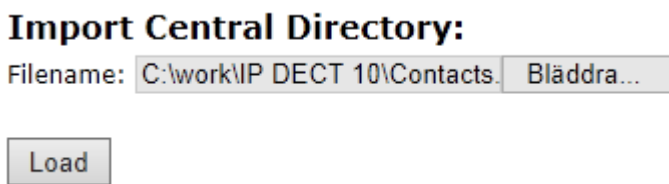
9.3 Import Contact List to Central Directory

On the **Central Directory** page, the admin should click on **Browse** button and the **Choose File to Load** dialog window will be shown.

On the **Choose File to Upload** dialog window, navigate to the directory or folder that contains the right file to be imported to the base station > Click on **Open** button.



Next, click on the **Load** button. This will import the contents of contacts in the selected file into the relevant Base station.



The figure below shows the import procedure is in process.



The parameters are successfully saved
You will be redirected after 3 seconds

9.4 Central directory using server

Alternative way to import a Contact List is to get it from a server. First click on Management url to get Management Settings page, then select the protocol of your server (TFTP/HTTP) in Management Transfer Protocol, then save the setting by clicking Save.

Settings

Management Transfer Protocol:

TFTP
HTTP
HTTPS
/CfgUpload

HTTP Management upload script:

Go back to Central Directory page and enter Server IP address (inclusive the path in the end of the address) and Filename of the contact list, then save the setting by clicking Save. (See example below).

XML Central Directory

Central Directory Location: XML Server

Server:

Save	Cancel
------	--------

Then reboot the Base station to ensure that the changes take effect.

9.5 Verification of Contact List Import to Central Directory

On the Handset, navigate to Central Directory where the correct contact list should populate to the contacts uploaded to the Base station.