VINGTOR STENTOFON
by ZENITEL GROUP




Configuration Manual

SIP Master Stations

About this Document

Document Scope

This document describes the configuration of the STENTOFON SIP Master Stations.

Publication Log

Rev.	Date	Author	Status
1.0	19.1.2012	HKL	published
1.1	9.6.2017	HKL	VS format, SIP server

Related Documentation

For further information, refer to the following documentation:

Doc. no.	Documentation
A100K10788	IP Master Station Installation & Configuration
A100K10935	IP Master Station Getting Started
A100K10812	SIP Substation Installation & Configuration

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1 Introduction

SIP (Session Initiation Protocol) is the de facto standard for IP telephony. The STENTOFON SIP intercom stations are specially built for easy integration with any iPBX system.

The STENTOFON SIP Stations are custom-made IP intercom stations that can integrate with any iPBX system.

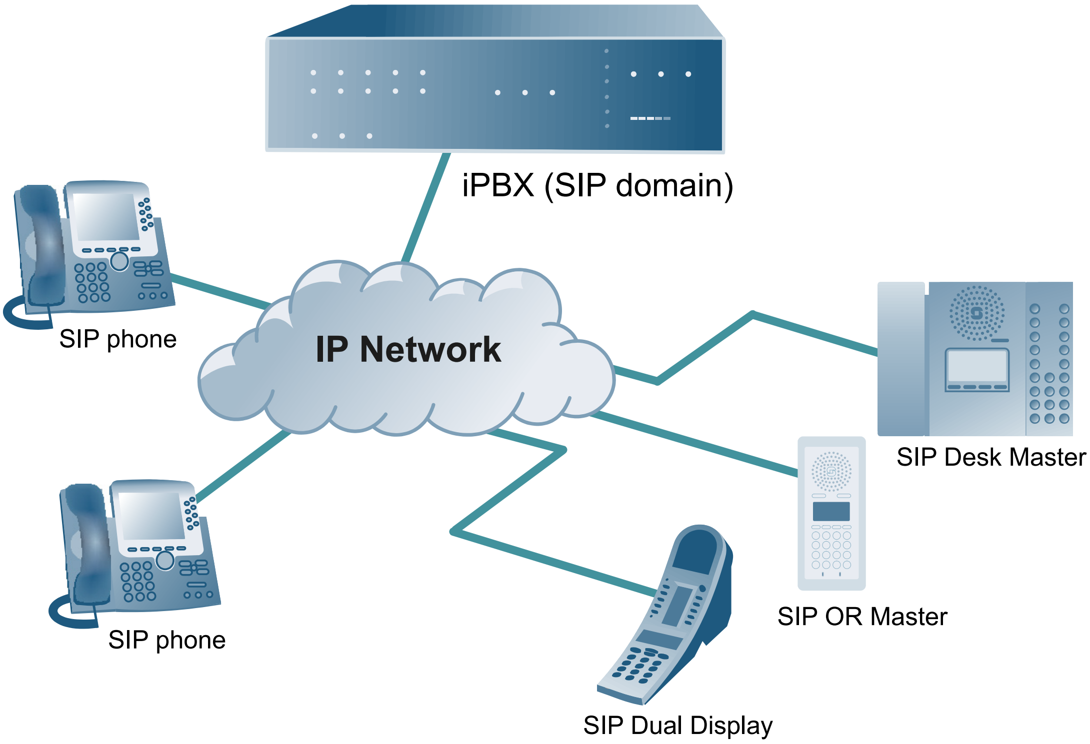


Figure 1 SIP System Configuration

1.1 IP Desktop Master Station

- item nos. 1008001000 (with handset), 1008000000 (without handset)

The IP Desktop Master is a general purpose intercom station featuring a large high contrast display with backlight showing crystal clear information. Ten direct access keys (DAK) provide single-touch access to other stations, group calls, audio monitoring, public address zones, radio channels and/or the opening of doors and gates.

The station connects directly to the IP network, making it easy to deploy anywhere at any distance. The built-in web server allows monitoring, configuration and software updates over the IP network for easy maintenance of remote stations.

1.2 IP OR Master Station

- item no. 1008015000

The IP Operating Room (OR) Master Station is an advanced intercom station intended for use in operating theatres and clean rooms. The chemical resistant and anti-bacterial front plate is completely flat and sealed to minimize bacteria accumulation. The station has an excellent audio quality. With a large backlit display and STENTOFON audio technology, the station allows users to read caller ID, listen and talk at a distance.

1.3 IP Flush Master Station

- item no. 1008031000

The IP Flush Master is intended for use in control and guard rooms. The station features a large high contrast display with backlight and up to 8 lines with 20 characters. The IP station has advanced call handling features such as call queuing. The call queue is presented to the user according to priority and time of arrival. The user can select which call to answer by scrolling through the queue. Four direct access keys (DAK) provide single-touch access to stations, group calls, audio monitoring, public address zones, radio channels, and opening of doors and gates. Each DAK key has a red and green LED to show the status.

1.4 IP Dual Display Station

- item no. 1008007000

The IP Dual Display Station is designed for desktop installation in banking/financial and office environments. It is also well suited as a control room station. The physical size makes it easy to place on desks with limited space. An optional noise cancelling gooseneck microphone module can be mounted in noisy environments. The dual backlit easy-to-read displays and navigation buttons provide single-touch access to stations, group calls, audio monitoring, public address zones, radio channels, opening of doors and gates as well as other functions. The direct access keys are easily programmed from the station and can be changed at any time.

2 SIP Configuration

There are two ways of configuring the IP Master Station for SIP:

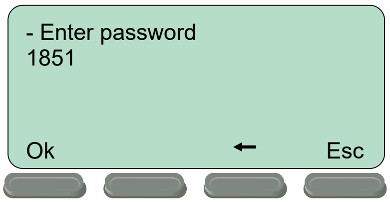
- Using the station keypad
- Using a web browser

2.1 Configuration Via Station Keypad

When the IP Master Station is not registered with the SIP server, an offline menu is displayed. The offline menu can be used to configure the station and is navigated with the 4 buttons below the display. The button on the left is used as a **Select** or **Ok** button as shown in the display, while the button on the right is used as a **Back** button. The two buttons in the middle are used to navigate up or down according to the arrows. When configuring IP settings, the **M** key is used to insert a "**.**" (dot) and the **right-middle button** is used to delete a character.

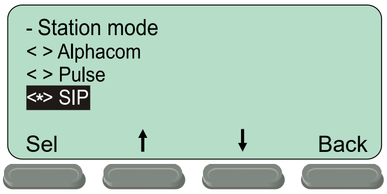
To enter the setup menu:

1. Press the **Setup** button on the left
2. Enter the password **1851**
3. Press the **Ok** button



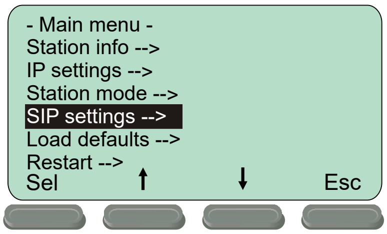
Station mode

If **SIP settings** is not listed as a main menu option, select **Station mode** to set the station to **SIP** mode.



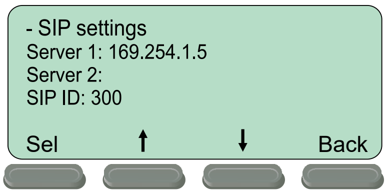
Main menu

Use the two arrow buttons in the middle to navigate to **SIP settings** and press the **Sel** button on the left. When entering data, the left-arrow is used for deleting characters.



SIP settings

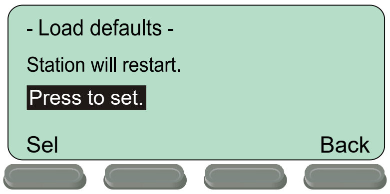
- Enter the IP address of **Server 1** (SIP server) that the station shall connect to, and the **SIP ID** (directory number) of the station.



Load defaults

This will load the factory default settings.

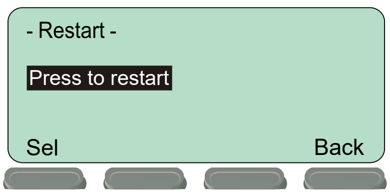
- Press the **Sel** button to load the default settings.



Restart

This will restart the station.

- Press the **Sel** button to restart the station.



Restart the station to apply new settings.

2.2 SIP Station Web Interface

The SIP Station features an embedded web server, which allows users to log in via a standard web browser.

At commissioning, the SIP Station needs to be configured to make it possible for the SIP Station to register in the iPBX system.

To do this, your PC and the IP station have to be connected together via a PoE switch using network cables:

- Connect the PC to the PoE switch
- Connect the **LAN** port on the SIP station to the PoE switch

There is one RJ45 port located at the bottom of the IP Dual Display station that is used as the LAN port.

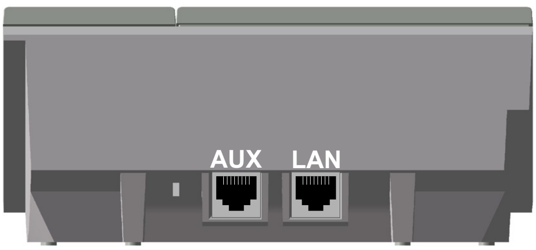


Figure 2 RJ45 Ports at Rear of IP Desktop Master Station

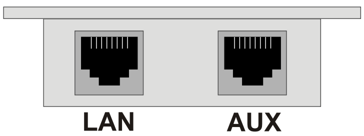


Figure 4 RJ45 Ports on IP Flush/OR Master Stations

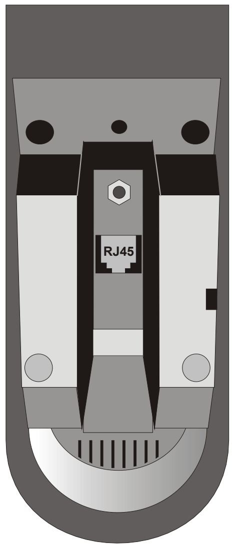


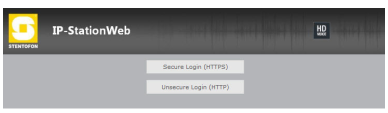
Figure 3 RJ45 Port at Bottom of IP Dual Display Station

The factory default IP address of the station is **169.254.1.100**. In order for your PC to communicate with the station it is necessary to change its **Internet Protocol Properties** to use an IP address that is in the same range as 169.254.1.100, e.g. 169.254.1.1 with subnet mask 255.255.255.0.

After the IP properties have been changed, access the station by logging into the web interface using a standard web browser:

1. Open a web browser
2. In the browser's Address bar, type **http://169.254.1.100**, and press the ENTER key
   - The station Login page is displayed.

To log into the station:



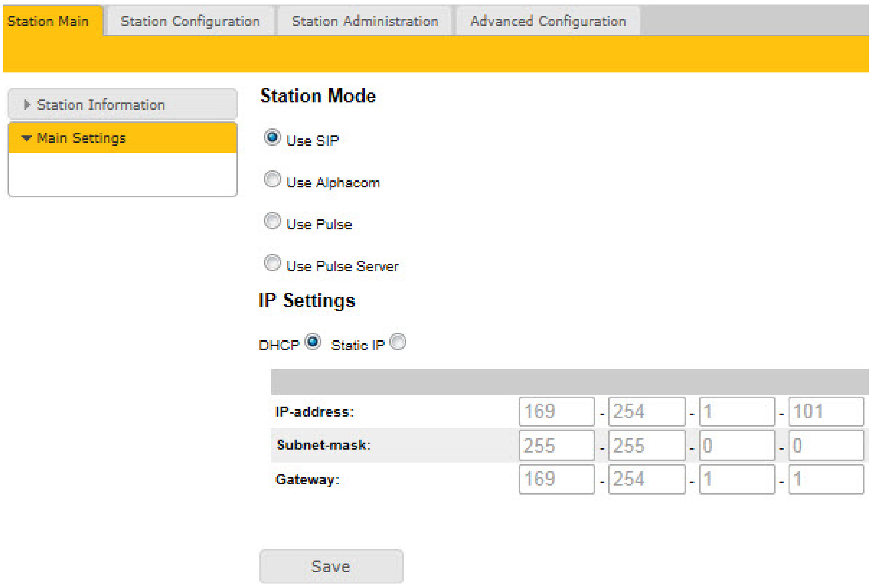
1. Click **Login**
2. Enter the default User name: **admin**
3. Enter the default password: **alphaadmin**

The main page will now be displayed, showing the Station settings including the MAC address.

Use the menu bar at the top of each page to browse through the different pages.

2.3 Station Main Settings

- Click **Station Main** > **Main Settings** to access the page for configuring station mode and IP parameters.



Station Mode

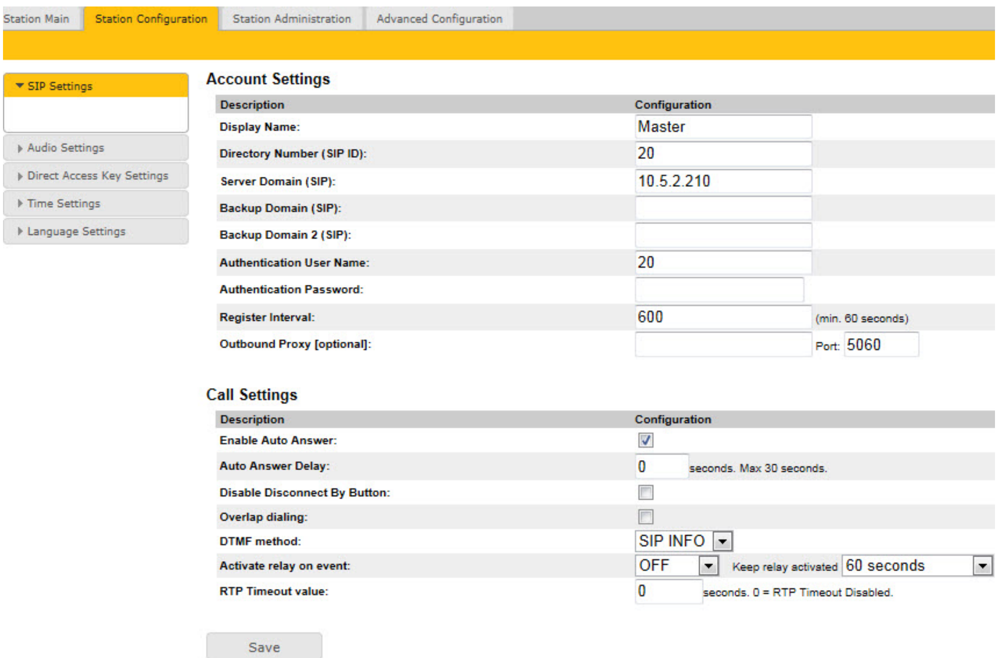
- Select the **Use SIP** radio-button

IP Settings

- **DHCP** – Use this option if the SIP Station shall receive IP Settings from a DHCP server.
- **Static IP** – Use this option if the SIP Station shall use a static IP address. Enter the **IP address**, **Subnet mask** and **Gateway** address.
- Click **Save** followed by **Apply** to apply the new configuration settings.

2.4 SIP Settings

- Click **Station Configuration** > **SIP Settings** to access the page for configuring SIP parameters.



Account Settings

Display Name

- Enter a name that will be shown on the display at the remote party.

Directory Number (SIP ID)

- This is the identification of the station in the SIP domain, i.e. the phone number for the station. This parameter is mandatory. Enter the SIP ID in integers according to the SIP account on the SIP domain server.

Server Domain (SIP)

- This parameter is mandatory and specifies the primary domain for the station and is the IP address for the SIP server (e.g. Asterisk or Cisco Call Manager). Enter the IP address in regular dot notation, e.g. 10.5.2.138.

Backup Domain (SIP)

- This is the secondary (or fallback) domain. If the station loses connection to the primary SIP domain, it will switch over to the secondary one. Enter the IP address in regular dot notation.

Backup Domain 2 (SIP)

- This is the tertiary backup domain.

Authentication User Name

- This is the authentication user name used to register the station to the SIP server. This is required only if the SIP server requires authentication and is normally the same as the SIP ID.

Authentication Password

- The authentication user password used to register the station to the SIP server. This is required only if the SIP server requires authentication

Register interval

- This parameter specifies how often the station will register, and reregister in the SIP domain. This parameter will affect the time it takes to detect that a connection to a SIP domain is lost.
- Enter the values in number of seconds from 60 to 999999. The default interval is 600 seconds.

Outbound Proxy [optional]

- Enter the IP address of the outbound proxy server in regular dot notation, e.g. 10.5.2.100

Port

- Enter the port number used for SIP on the outbound proxy server. The default port number is 5060.

Call Settings

Enable Auto Answer

- This is not required. Enables automatic answer after a set number of seconds.

- Check the checkbox to enable this function and enter the delay in seconds in the field for **Auto Answer Delay**. The default delay setting is 0 and the maximum is 30 seconds.

Disable Disconnect By Button

- This disables disconnect with the speed dial during and when setting up a conversation. Check the checkbox to enable this function.

Overlap dialing

- This will lead to the phone starting to dial each time a digit is entered and the SIP proxy replying with ‘Number incomplete’ until such time as the number has been entered and the call can be initiated successfully without the enter key having to be pressed.

DTMF method

- Choose between SIP INFO or RFC 2833 to select DTMF signalling method.

Activate relay on event

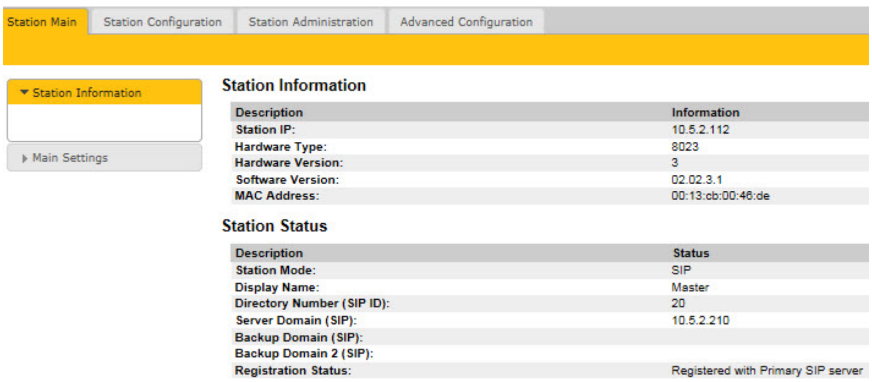
- When enabled, the station will activate the relay when receiving the specified DTMF character in the RTP stream. Select from the dropdown menu. Options are OFF, 1-9, *, In call or Ringing. The default setting is OFF.
- Select the number of seconds to keep the relay open in the range 1 to 240 from the dropdown menu. The default setting is 60 seconds.
- Options are: 1 - 240 seconds, during call, during ringing, until DTMF # or 0.

RTP Timeout value

- This cancels a call if the station does not receive RTP packets from the remote party. Enter values in the range 0-9999 seconds. The default setting is 0 which means RTP timeout is disabled.

After entering all the desired values, click **Save** and then click **Reboot** to enable the SIP settings.

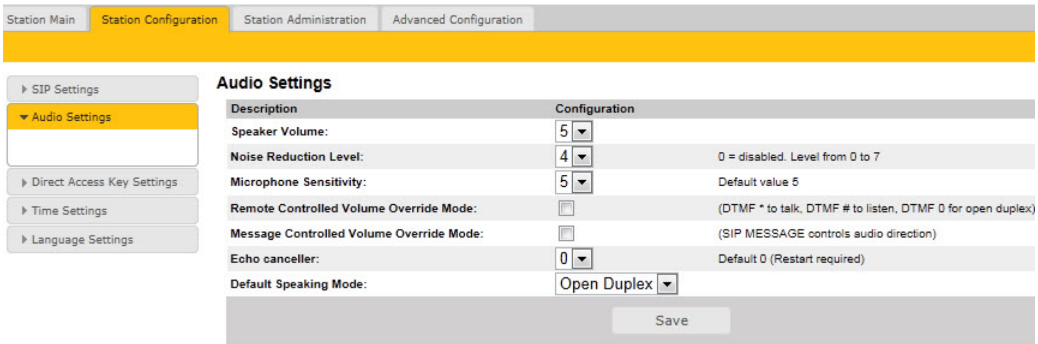
After completing the SIP configuration, click **Station Main** > **Station Information** and the main page may look like the following:



ⓘ *The IP Properties on your PC has to be changed to the same IP domain as that of the SIP station.*

2.5 Audio Settings

Click **Station Configuration** > **Audio Settings**



Speaker Volume

- Select the volume level in the range 0 to 7 from the dropdown menu. The default setting is 5.

Noise Reduction Level

- Level 0 means that the function is disabled
- Level 1 gives a maximum noise reduction of 0.2 dB
- Level 2 gives a maximum noise reduction of 6.2 dB
- Level 3 gives a maximum noise reduction of 12.2 dB
- Level 4 gives a maximum noise reduction of 18.3 dB
- Level 5 gives a maximum noise reduction of 24.3 dB
- Level 6 gives a maximum noise reduction of 30.3 dB
- Level 7 gives a maximum noise reduction of 36.3 dB

Microphone Sensitivity

- Select the sensitivity level in the range 0 to 7 from the dropdown menu. The default setting is 5.

Remote Controlled Volume Override Mode

- This acts as simplex mode. This feature is activated after the first DTMF * or # is received from the remote station. Send DTMF * to talk and # to listen. Check the checkbox to enable this function.

Message Controlled Volume Override Mode

Check the box to enable the following messages:

- SIP MESSAGE ”Audio_receive_only”: Turns the microphone off and loudspeaker on
- SIP MESSAGE ”Audio_send_only”: Turns microphone on and loudspeaker off
- SIP MESSAGE ”Audio_send_receive”: Turns both microphone and loudspeaker on

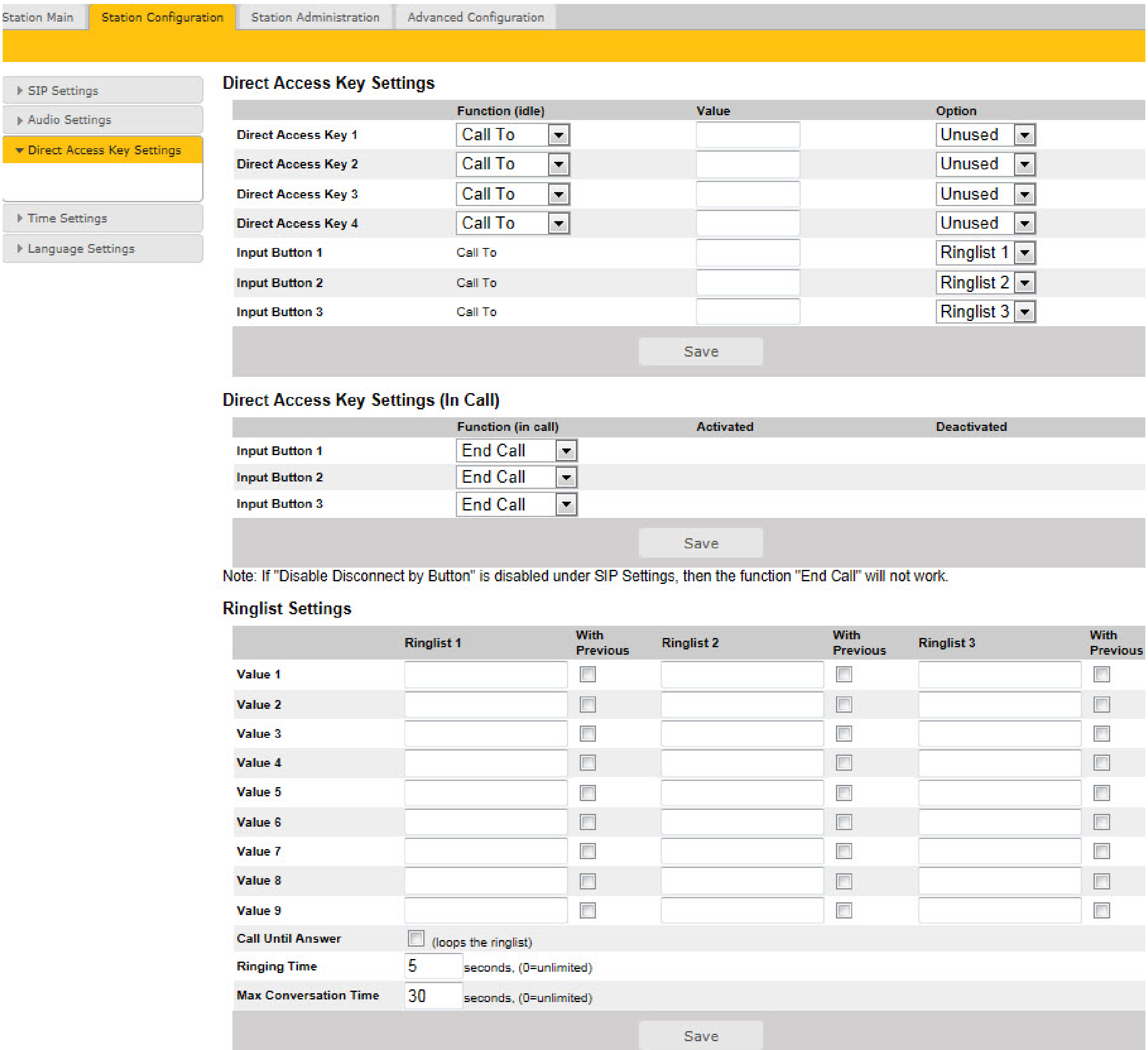
Default Speaking Mode

- Select between Open Duplex or Push-To-Talk

After entering all the desired values, click **Save** to enable the audio settings.

2.6 Direct Access Key Settings

- Click **Station Configuration** > **Direct Access Key Settings** to access the page for configuring DAKs.



<u>Direct Access Key Settings</u>

Direct Access Key 1 - Direct Access Key 4

Enter the numbers to call in the **Value** field.

ⓘ *The Desktop and Dual Display master stations have 10 Direct Access Keys and do not have the 3 Input Buttons described below.*

Input Button 1

This is the SIP ID for the extension to be called when call button no. 1 is pressed, i.e. the SIP ID number of the receiving party.

Input Button 2

This is the SIP ID for the extension to be called when call button no. 2 is pressed, i.e. the SIP ID number of the receiving party.

Input Button 3

This is the SIP ID for the extension to be called when call button no. 3 is pressed, i.e. the SIP ID number of the receiving party.

Direct Access Key Settings (In Call)

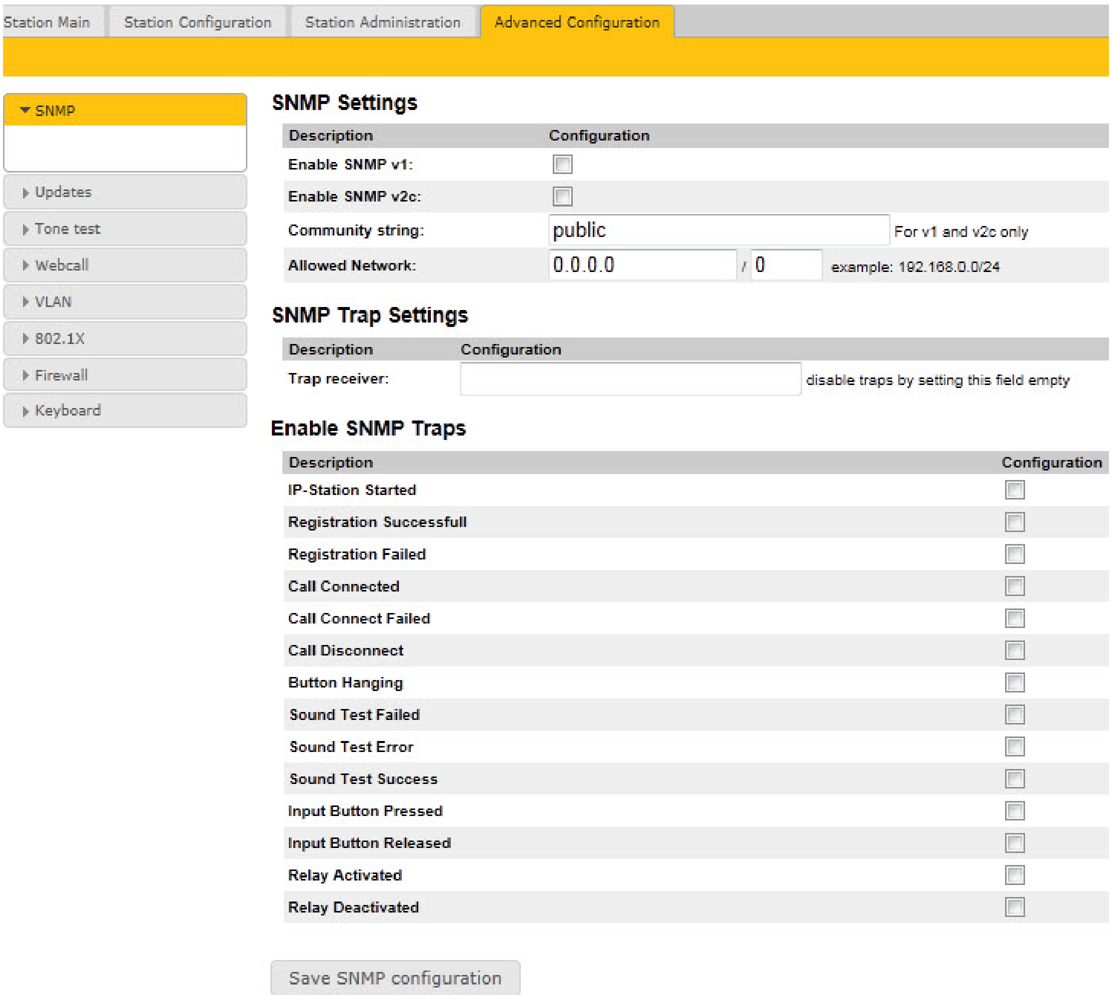
- Select input buttons 1 - 3 for direct access calls while in conversation.
- Options are: End Call, Do Nothing, Send Text, Send DTMF

ⓘ *Pin connections for the three input buttons are located on the P4 connector. See Section 4: Station Board Connections for more information.*

2.7 SNMP Settings

SNMP (Simple Network Management Protocol) is a protocol for centralizing the management of devices in IP networks.

- Click **Advanced Configuration** > **SNMP** to access the page for configuring SNMP parameters.



SNMP Settings

Enable SNMP v1

- This enables reading of the MIB using SNMP version 1.

Enable SNMP v2c

- This enables reading of the MIB using SNMP version 2c.

Community string

- Enter a text string used as a password for authentication.

Allowed Network

- This is used, together with the network mask, to determine the allowed network for reading the MIB on the station.
- The IP address is entered in regular dot notation, e.g. 10.5.2.100. For example with an allowed network 10.5.2.0 and a network mask of 24, any station with an IP address in the range 10.5.2.0 to 10.5.2.255 can access the MIB.

SNMP Trap Settings

Trap receiver

- Enter the IP address of the server receiving SNMP traps. This is disabled if the field is left empty.

Enable SNMP Traps

IP-Station Started

- If enabled, the station will send an SNMP trap when the station application is started.

Registration Successfull

- If enabled, the station will send an SNMP trap when successfully registered in the SIP domain.

Registration Failed

- If enabled, the station will send an SNMP trap if registration in the SIP domain failed.

Call Connected

- If enabled, the station will send an SNMP trap when a call is connected.

Call Connect Failed

- If enabled, the station will send an SNMP trap if a call to the station fails to connect for any reason (busy etc.).

Call Disconnect

- If enabled, the station will send an SNMP trap when a call is disconnected.

Button Hanging

- If enabled, the station will send an SNMP trap when a button is hanging.

Sound Test Failed

- If enabled, the station will send an SNMP trap when a sound test has failed.

Sound Test Error

- If enabled, the station will send an SNMP trap when there is a sound test error.

Sound Test Success

- If enabled, the station will send an SNMP trap when a sound test is successful.

Input Button Pressed

- If enabled, the station will send an SNMP trap when an input button is pressed.

Input Button Released

- If enabled, the station will send an SNMP trap when an input button is released.

Relay Activated

- If enabled, the station will send an SNMP trap when a relay is activated.

Relay Deactivated

- If enabled, the station will send an SNMP trap when a relay is deactivated.

2.8 Automatic Configuration using TFTP

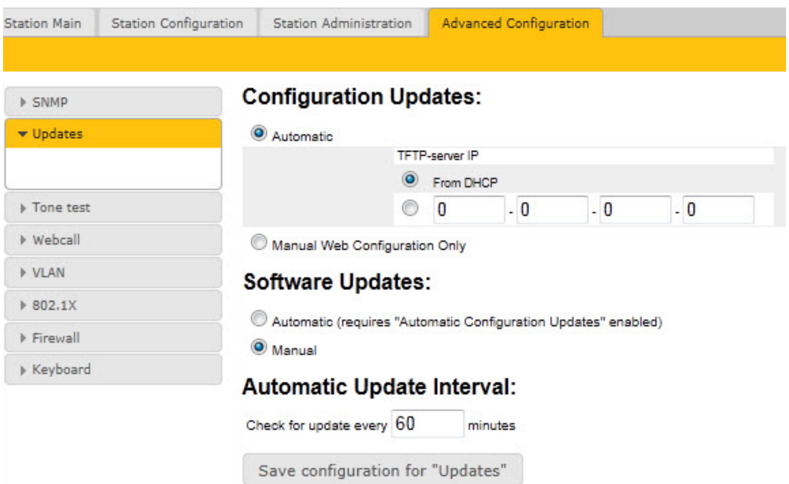
A SIP station may be set up to automatically poll configuration settings for SIP, Call and SNMP from a TFTP server. The IP address of this TFTP server can be obtained using DHCP procedures or be manually configured.

Before you start the automatic configuration procedure:

- A configuration file should first be created. The relevant parameters for SIP, Call and SNMP in the configuration file are described in *Section 7: Configuration File Parameters*.
- Follow the procedures described in *Section 3.1: TFTP Server Program*.

To carry out automatic configuration from the station web server:

1. Start the TFTP server program and set the server path by browsing to the directory where the configuration file is located.
2. Log on to the SIP Station web server.
3. Select **Advanced Configuration** > **Updates**



4. Under **Configuration Updates** select the radio button for **Automatic**
5. Either select the radio button for **From DHCP** or enter the IP address of the **TFTP server** (your PC IP address)
6. Under **Automatic Update Interval** enter the interval in minutes for checking updates.
   - The value must be between 1 and 999 and the default setting is 60.
7. Click **Save configuration for "Updates"**

The station will now contact the TFTP server and run the configuration file to carry out the configuration procedure according to the set time interval.

2.9 Advanced Configuration Options

ⓘ *The configuration settings described in this section are not mandatory.*

2.9.1 VLAN

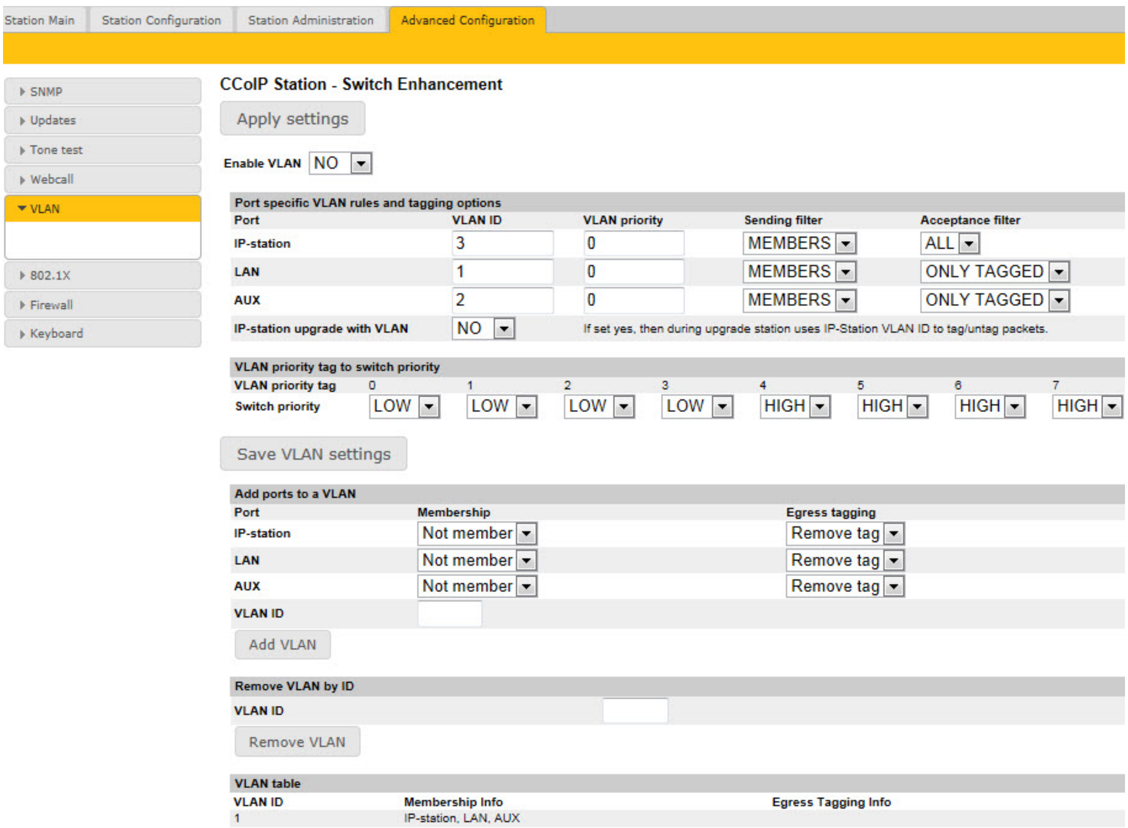
VLAN Tagging or IEEE 802.1Q is a networking standard allowing multiple bridged networks to transparently share the same physical network link without leakage of information between networks. IEEE 802.1Q — along with its shortened form *dot1q* — is commonly used to refer to the encapsulation protocol used to implement this mechanism over Ethernet networks.

ⓘ *STENTOFON IP Stations support 802.1Q as from firmware version 01.09.3.0.*

User interface

VLAN is configured in the IP station web interface.

- Select **Advanced Configuration** > **VLAN** from the menu



Clicking the **Apply settings** button will apply the chosen settings. With the exception of a restart, the saved settings will not come into effect until **Apply settings** is clicked.

Enable VLAN

This option determines whether the switch uses 802.1Q or not. If this is enabled, the switch is VLAN aware. Select **YES** or **NO** from the dropdown menu.

Port specific VLAN rules and tagging options

Here, it is possible to specify which VLAN ID and priority the ports should assign untagged packets to. Tagged packets are not changed.

- **VLAN ID** has a value range from 0 to 4094. It specifies which VLAN ID tag to add to a packet.
- **VLAN priority** has a value range from 0 to 7. It specifies which VLAN priority tag to add to a packet.
- **Sending filter** specifies whether a given port will only send to VLANs which it is a member of or all VLANs. For example, if the chosen option is **MEMBERS** then a packet with VLAN ID 1 at the LAN port will only reach another port which is a member of VLAN ID 1. Select **MEMBERS** or **ALL** from the dropdown menu.
- **Acceptance filter** specifies whether a port will accept only tagged packets or all packets. The option **ONLY TAGGED** should only be used against VLAN aware devices which tag packets. Select **ONLY TAGGED** or **ALL** from the dropdown menu.

VLAN priority tag to switch priority

Here, it is possible to specify how the switch should queue the packets with **VLAN priority tag**.

- **Switch priority**: Select **HIGH** or **LOW** from the dropdown menu. By default, packets with VLAN priority tags from 4 to 7 are set to the **HIGH** priority queue.

Add ports to a VLAN

Here, it is possible to determine whether the ports should be members of the specified VLAN. There is also a setting for specifying whether the ports should strip or keep the VLAN tag when sending egress packets.

- **Membership** determines whether the port is a member of the specified VLAN or not. Select **Not member** or **Member** from the dropdown menu.
- **Egress tagging** determines whether the port should remove VLAN tags or keep them for the specified VLAN. Select **Remove tag** or **Keep tag** from the dropdown menu.

Clicking the **Add VLAN** button will add the current chosen settings to the **VLAN table** below. If a VLAN in the **VLAN table** already exists with the chosen **VLAN ID**, then the settings will be updated.

Remove VLAN by ID

Here, it is possible to determine which VLAN is to be removed from the **VLAN table** by specifying the **VLAN ID**, then clicking the **Remove VLAN** button.

VLAN table

The VLANs that the ports are members of are listed under the **Membership Info** column. The table also lists the ports that keep the VLAN tag when sending egress packets; this is shown under the **Egress Tagging Info** column. The **VLAN table** can accommodate a maximum of 63 VLANs.

ⓘ *The DHCP address is received before the switch is VLAN aware (during startup). Either trunk all VLANs or set the DHCP server which should reach the IP station on a native VLAN.*

2.9.2 Network Access Control

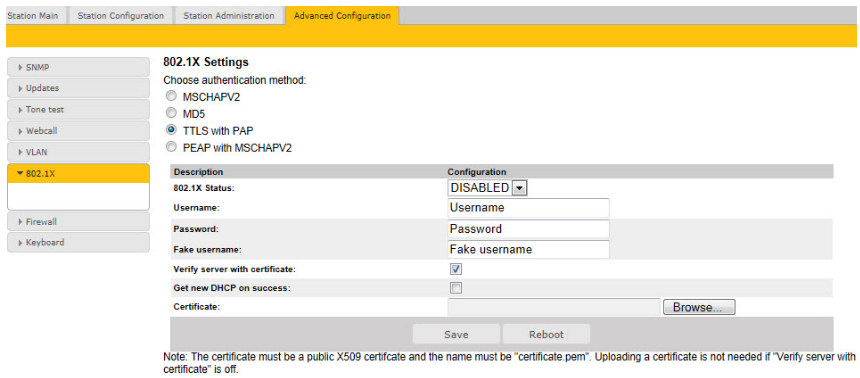
IEEE 802.1X is an IEEE Standard for Port-based Network Access Control (PNAC) By “port” we mean a single point of attachment to the LAN infrastructure. It provides an authentication mechanism to devices wishing to attach to a LAN, either establishing a point-to-point connection or preventing it if authentication fails.

ⓘ *STENTOFON SIP Stations support 802.1X as from firmware version 01.09.3.0.*

User interface

802.1X Network Access Control is configured from the IP station web interface.

- Select **Advanced Configuration** > **802.1X** from the menu.



The radio button list lets the user choose the authentication method to configure and use.

The different authentication methods are:

- MSCHAPV2
- MD5
- TTLS with PAP
- PEAP with MSCHAPV2

MSCHAPV2 and **MD5** will encrypt the password.

TTLS with PAP and **PEAP with MSCHAPV2** will encrypt both the **Username** and **Password**.

The parameters to configure depend on the authentication method:

802.1X status: Enable or disable 802.1X.

Username: The user name that identifies a station.

Password: The password associated with the user name.

Fake username: The fake user name sent outside of encrypted tunnel with **TTLS with PAP** and **PEAP with MSCHAPV2**. The user name is encrypted.

If **TTLS with PAP** or **PEAP with MSCHAPV2** is chosen, a certificate must be uploaded to the station by clicking **Browse.** The certificate must either be in Privacy Enhanced Mail (PEM) or Distinguished Encoding Rules (DER) format, and it must be named *certificate.pem*.

- Click **Save** to save the current settings
- Click **Reboot**
  - The new 802.1X settings will only come into effect after the reboot.

3 Software Upgrade

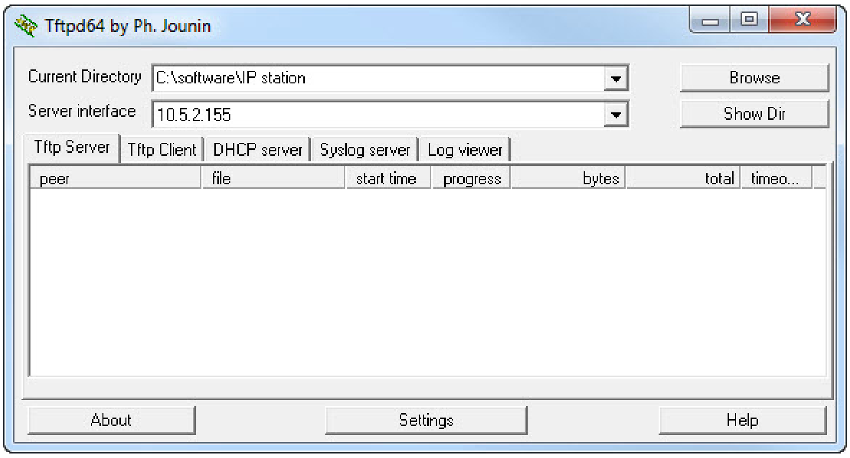
Software upgrade is carried out via the web server of the station.

There are two ways of upgrading the software on the SIP station:

- Manual Upgrade
- Automatic Upgrade

3.1 TFTP Server Program

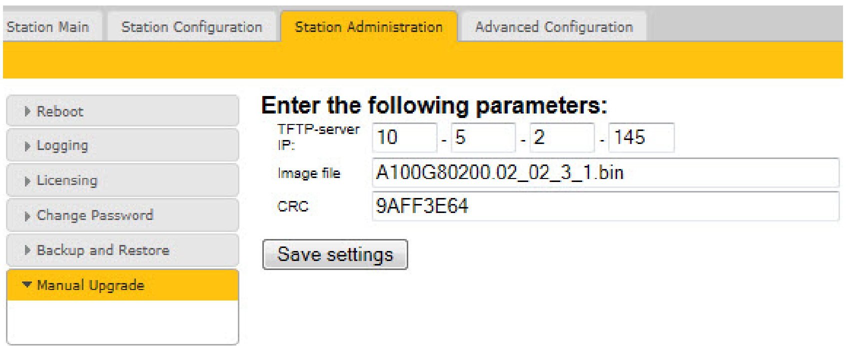
Both upgrade methods require that an TFTP Server is available and that the latest software image file has been downloaded from Zenitel's support website (AlphaWiki). During the upload process, the IP station will connect to the TFTP Server and download the software. The TFTP Server program must already be installed on your PC/server with a defined IP address. A free TFTP Server program can be downloaded from http://tftpd32.jounin.net. Before starting the IP station upgrade procedure, the TFTP Server program must be running and the directory where the software image file is located must be selected by using the **Browse** button in the program interface.



3.2 Manual Software Upgrade

To carry out a manual software upgrade from the station web server:

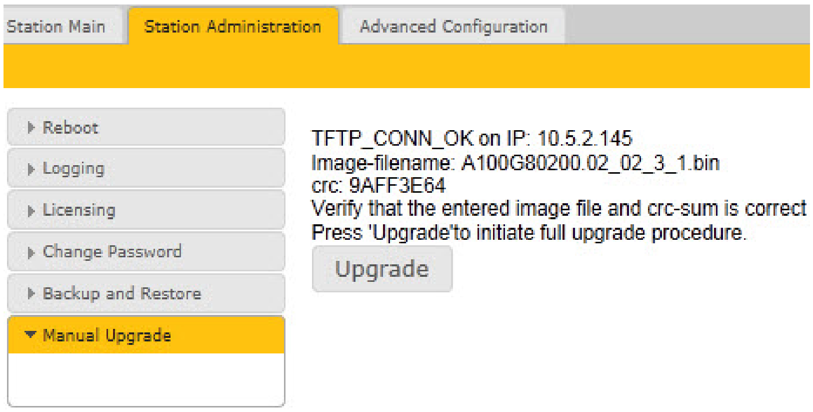
1. Start the TFTP server program and set the server path by browsing to the directory where the software file is located.
2. Log on to the SIP Station web server.
3. Select **Station Administration** > **Manual Upgrade**



4. Enter the IP address of the **TFTP server** (your PC IP address)

5. Enter the name of the software **Image file** (include *bin* file extension)
6. Enter the **CRC** checksum (found in the text file from the downloaded software package)
7. Click **Save settings** to store the data

The station will now try to contact the TFTP server. If the connection cannot be established or the file *tftp_test.txt* is missing from the directory, the message *TFTP_CONN_ERROR* is displayed. If the response is *TFTP_CONN_OK* the settings are saved, and the **Upgrade** button will appear.



- Click the **Upgrade** button to start the software upgrade procedure for the SIP station.
  - The upgrade procedure takes approximately 3 minutes.

The upgrade process can be monitored by clicking the **Log viewer** tab in the TFTP server program.

ⓘ *Windows Explorer may be set to hide known file extensions so the file may appear without the **.bin** extension. The name of the software image file has to be entered with the extension **.bin**.*

3.3 Automatic Software Upgrade

A SIP station may be set up to automatically poll software upgrade configuration from a TFTP server. The IP address of this TFTP server can be obtained using DHCP procedures or be manually configured.

A configuration file should first be created. The relevant parameters in the configuration file are described in *Section 7: Configuration File Parameters*.

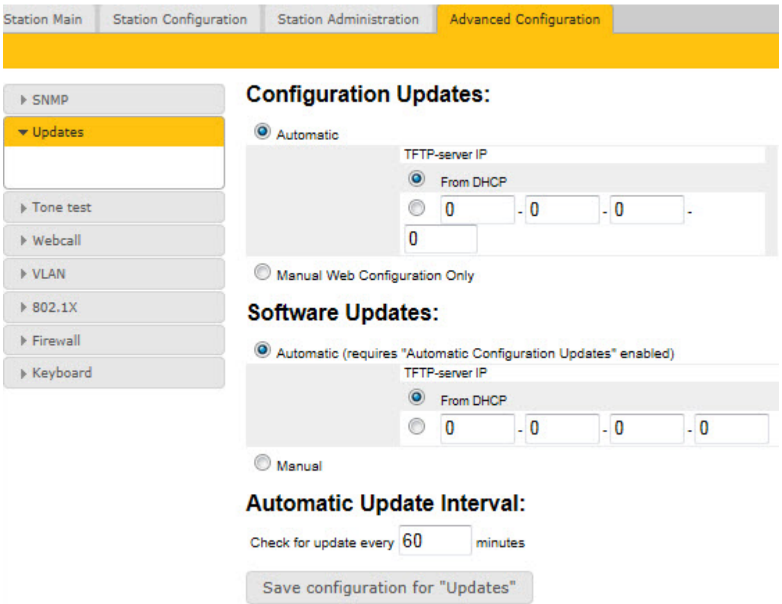
An example of the parameters for software upgrade in the configuration file is as follows:

```
auto_update_interval=10
auto_update_image_type=A100G80200.01_10_1_2.bin
auto_update_image_crc=C1466499
```

To carry out automatic software upgrade from the station web server:

1. Start the TFTP server program and set the server path by browsing to the directory where the software file is located.
2. Log on to the SIP Station web server.

3. Select **Advanced Configuration** > **Updates**



4. Under **Configuration Updates** select the radio button for **Automatic**
   - Automatic Configuration Updates has to be enabled
5. Either select the radio button for **From DHCP** or enter the IP address of the **TFTP server** (your PC IP address)
6. Under **Software Updates** select the radio button for **Automatic**
7. Either select the radio button for **From DHCP** or enter the IP address of the **TFTP server** (your PC IP address)
8. Under **Automatic Update Interval** enter the interval in minutes for checking updates.
   - The value must be between 1 and 999 and the default setting is 60.
9. Click **Save configuration for "Updates"**

The station will now contact the TFTP server, download the software image and carry out the upgrade.

ⓘ *During an upgrade, the station switch will not be VLAN aware. Make sure the IP station can reach the TFTP server from the native VLAN.*

ⓘ *During an upgrade of the station, 802.1X will not be running. Thus if 802.1X reauthentication is enabled and is performed during the upgrade, the station may lose contact with the TFTP server (depending on the configuration when 802.1X authentication fails). If the station loses contact with the TFTP server, it will not be upgraded.*

4 Station Board Connections

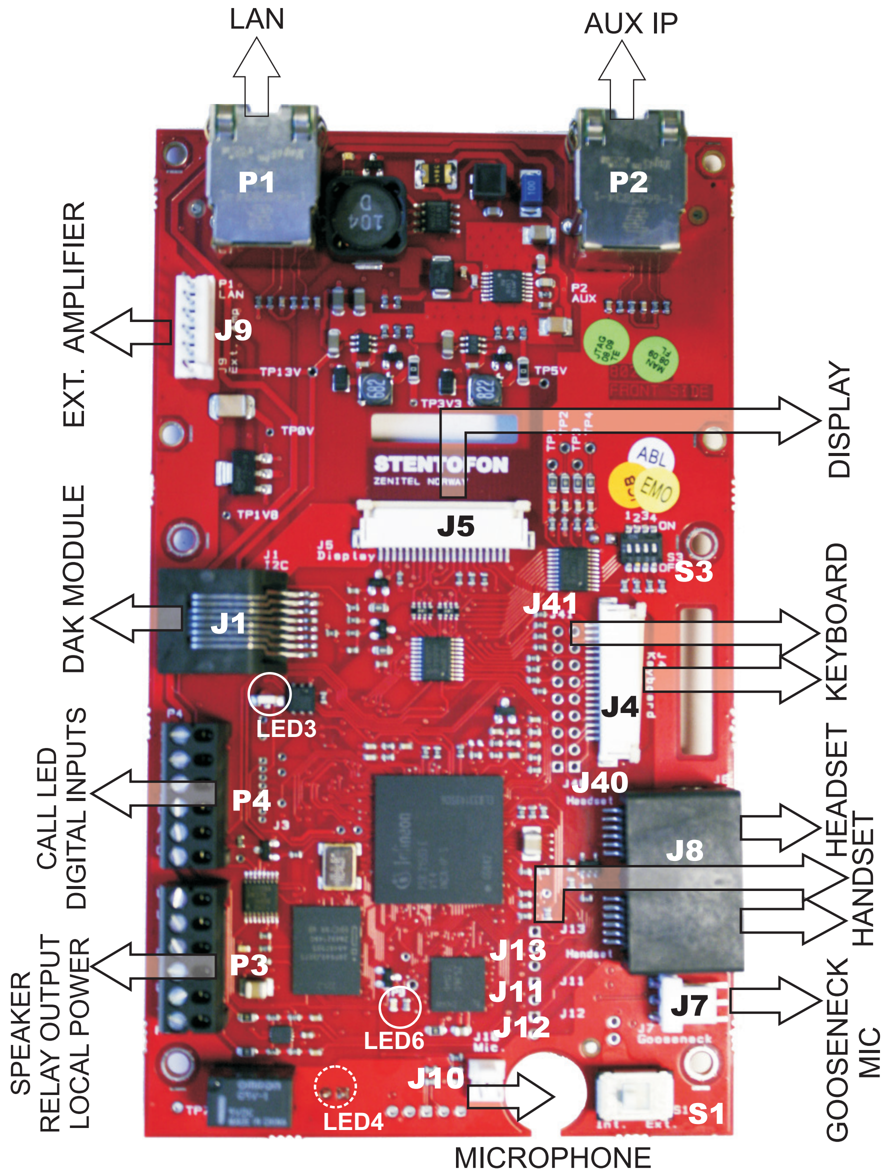
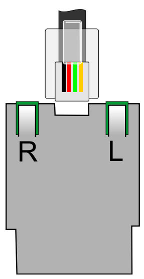


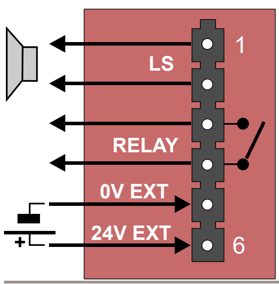
Figure 5 Station Board Connections



P1 RJ45 LAN connector for 10/100 Mbit Ethernet connection.
The station can be powered from this connection if the line supports Power over Ethernet (PoE).

The connector has two LEDs in front where the right (R) LED indicates Ethernet speed and the left (L) LED verifies Ethernet link and traffic. (L and R as seen from the connector side).

RJ45 connector for auxiliary equipment like IP camera, PC or a second IP station.
This port does not have an individual IP address. It does not carry power for AUX equipment.

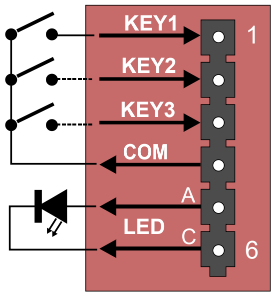


P3 6-pin plug-on screw terminal for external connections.

Pin 1/2 Connect to 1.5 W loudspeaker.
Typical impedance: 8 Ω
Recommended impedance: 6-25 Ω.

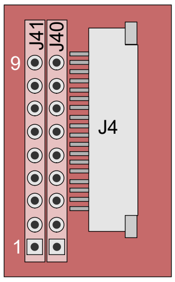
Pin 3/4 Internal NO relay contact for door lock control etc.

Pin 5/6 Connect 24 VDC for station power if PoE is not used. Pin 6 is positive.



P4 6-pin plug-on screw terminal for external connections.

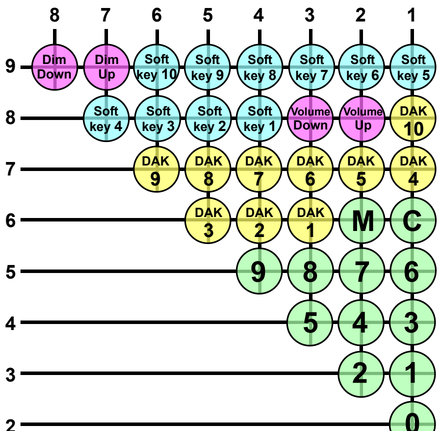
Pin 1/4	Input 1
Pin 2/4	Input 2
Pin 3/4	Input 3
Pin 5/6	Station LED for call message info.



J4 18-pin ZIF-connector for keyboard.
The keyboard may have up to 10 dialling keys, M and C keys, 10 DAK keys, 10 soft keys, volume up/down and light dim up/down keys connected in a matrix according to the drawing.

J40 Optional keyboard connection.

J41 Optional DAK LED connection.
Connection to J40/J41 can either be to a pin header or soldered directly to holes in the PCB.



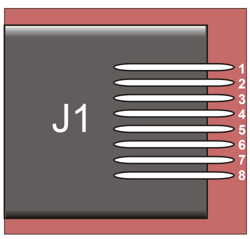
J4 and J40 pin matrix

J4 & J40 pin1-9 Keyboard matrix

J4/10 - J41/1	DAK1 Green LED
J4/11 - J41/2	DAK1 Red LED
J4/12 - J41/3	DAK2 Green LED
J4/13 - J41/4	DAK2 Red LED
J4/14 - J41/5	DAK3 Green LED
J4/15 - J41/6	DAK3 Red LED
J4/16 - J41/7	DAK4 Green LED
J4/17 - J41/8	DAK4 Red LED
J4/18 - J41/9	Common +3.3 V

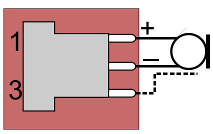
The DAK keys and the DAK LEDs are programmed using IND commands in the Event Handler. DAKs 1 to 4 and Soft keys 1 to 4 are used in current master stations.
Privacy on/off is accomplished by pressing the **C** key for 3 seconds. A Ⓟ is shown in the display during Privacy mode.

J5 20-pin ZIF connector for LCD display.
Separate display panels are available as a kit of 5 units (order no. 100 8099 000).



J1 RJ45 Connector for I2C interface.
Used for connection of DAK modules.

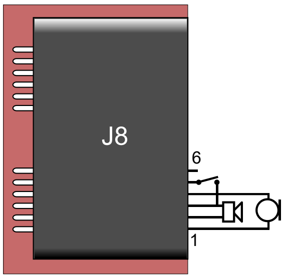
Pin 1	GND
Pin 2	A0 (GND)
Pin 3	A1 (GND)
Pin 4	NC (GND)
Pin 5	IRQ
Pin 6	SCL (Clock)
Pin 7	SDA (Data)
Pin 8	+13 V



J7 3-pin header for connecting gooseneck microphone.

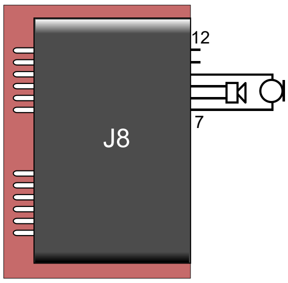
Pin 1	MIC+
Pin 2	MIC-
Pin 3	GND

J8 Dual RJ11 for handset and headset.
Handset can optionally be connected via J11, J12 and J13.



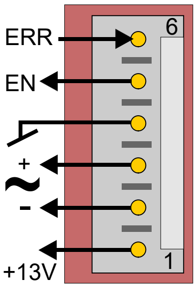
Handset:

Pin 1	Mic+
Pin 2	Spk+
Pin 3	Spk- (0 V)
Pin 4	Mic-
Pin 5	Hook-switch (must be connected to pin 3 with a switch)
Pin 6	PTT (May not be supported by SW)



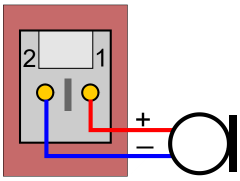
Headset:

Pin 7	Mic+
Pin 8	Spk+
Pin 9	Spk-
Pin 10	Mic-
Pin 11	PTT Ground
Pin 12	PTT (May not be supported by SW)



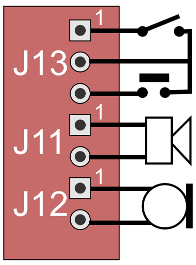
J9 6-pin header for connecting external amplifier.
12V/5W can be supplied to the external amplifier.
Signal to amplifier is balanced 0dBm 600 ohm.

Pin 1	+13 V	max. 500 mA to amplifier
Pin 2	LSN	negative line
Pin 3	LSP	positive line
Pin 4	GND	
Pin 5	AMP_EN	enable signal to amplifier
Pin 6	AMP_ERR	error signal from amplifier



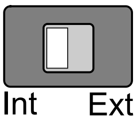
J10 Internal microphone

Pin 1	MIC+
Pin 2	MIC-

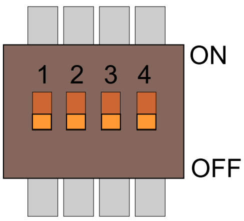


J11, J12, J13 Pin header for handset connection.
Same as J8 pin 1 - 6.

J11	Pin 1	Spk+
J11	Pin 2	Spk-
J12	Pin 1	Mic+
J12	Pin 2	Mic-
J13	Pin 1	OFFHOOK
J13	Pin 2	GND
J13	Pin 3	PTT



S1 Slide-switch to select internal microphone (Int) or gooseneck microphone (Ext).



S3 4-way DIP switch for future use.
Leave all in default OFF position.

5 Station Indication LEDs

5.1 Station LED

LED4 - Red 5mm LED on board rear side, also seen through front plate.

Flashing at 1 second intervals

- Station has no connection to the SIP server.

Possible reasons:

- No connection to Ethernet
- Wrong SIP server IP address configured
- Invalid IP address
- No gateway or wrong gateway to the SIP server

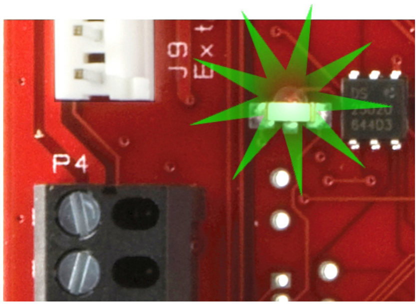
No flashing

Possible reasons:

- Station connected and registered to the SIP server
- Station not powered up (if no other LEDs are active)

5.2 Status LED

LED3 – Bicolor SMD LED on board component side.



Flashing 2 red + 1 green

- Station has no connection to the SIP server.

Flashing 1 red + 2 green

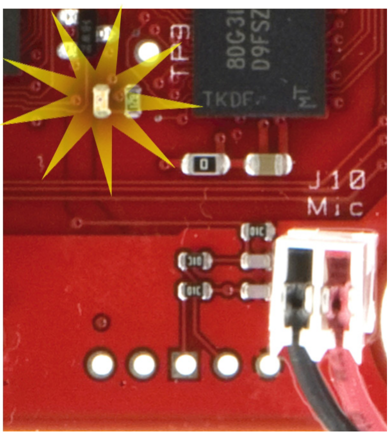
- Station connected but NOT registered in the SIP server.

Flashing 3 green

- Station connected and registered in the SIP server.

5.3 Power LED

LED6 – Yellow SMD LED on board component side.



Light: 13 V board power OK.

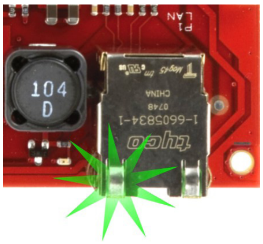
No light: 13 V board power faulty if no other LEDs are lit.

Possible reasons:

- 24 VDC or PoE not connected
- Faulty board

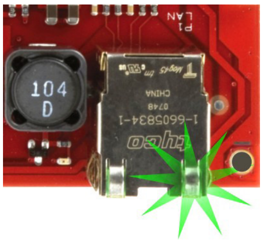
5.4 LAN LEDs

These LEDs are on the LAN (P1) and AUX (P2) RJ45 ports.



Left LED

Steady light:	Ethernet connection OK
Flashing:	Ethernet traffic
No light:	No Ethernet connection



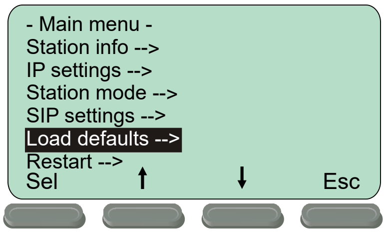
Right LED

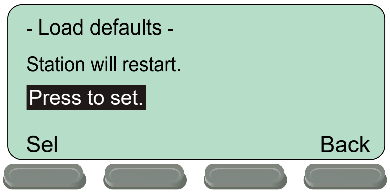
Light:	100 Mbit Ethernet connection
No light:	10 Mbit Ethernet connection

6 Restoring Factory Defaults

An IP Master Station may have to be reset to its original factory default settings if, for instance, the password to the web server is forgotten.

- Connect 24 VDC to P3 5/6 to power up the station.

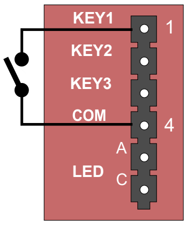




Stations with display

1. Make sure the Master Station is disconnected from the SIP server.
2. Press the **Setup** button beneath the display
3. Enter the password **1851** and press the **Ok** button.
4. Navigate to **Load defaults** and press the **Sel** button.
5. Press the **Sel** button again to load factory defaults.

The station will now restart with factory settings.



Stations without display

1. Connect a switch (KEY1) between pin 1 and 4 on the P4 connector.
2. Keep KEY1 pressed while powering up the station.
3. Release KEY1 after precisely two flashes of the station LED, then press KEY1 briefly again.

The station will then come up with factory settings.

7 Configuration File Parameters

7.1 Remote Provisioning using TFTP

An IP station may be set up to automatically poll configuration from a TFTP server. The IP address of this TFTP server can be obtained using DHCP procedures or be manually configured.

The IP station will first try to download the global configuration file:

```
ipst_config.cfg
```

Then the IP station will download a device specific configuration file:

```
ipst_config_01_02_03_04_05_06.cfg
```

where `01_02_03_04_05_06` is the MAC address of the IP station.

If the same parameter is found in both files, the value from the device specific file takes precedence.

7.2 General Parameters

```
auto_update_interval
```

Required: No. If this parameter is not set in the file, the function will be disabled.

Description: This parameter enables the station to automatically look for software updates on the TFTP server.

Values: Number of minutes to wait between each server request. Value must be between 1 and 999.

```
auto_update_image_type
```

Required: If auto_update_interval is set.

Description: The name of the software image file to be uploaded.

Values: Text giving the name of the software image file. The full name of the file, including extension, is required. This parameter must be set if the auto update function is enabled.

```
auto_update_image_crc
```

Required: If auto_update_interval is set.

Description: The CRC checksum calculated for the software image file specified by the auto_update_image_type parameter. This is used to check the integrity of the software file before updating the station.

Values: Hexadecimal value.

7.3 SIP Parameters

`nick_name`

Required: No. Defaults to sip_id.

Description: The nickname for the station can be used to assign a logical name to the station. For example, a station belonging to James may be assigned the nickname “James” or “James’ station”.

Values: Text string. Max length is 64 characters.

`sip_id`

Required: Yes

Description: This is the identification of the station in the SIP domain, i.e. the phone number of the station.

Values: Integer value. Max length is 64 characters.

`sip_domain`

Required: Yes

Description: SIP domain is a server that uses SIP (Session Initiation Protocol) to manage real-time communication among SIP clients. The sip_domain parameter specifies the primary domain for the station, as opposed to sip_domain2 which specifies the secondary (or fallback) domain. The IP address for the SIP domain server (e.g. Asterisk or Cisco Call Manager) should be defined in this section.

Values: IP address given in regular dot notation, e.g. 10.5.2.100

`sip_domain2`

Required: No

Description: This is the secondary (or fallback) domain. If the station loses connection to the primary SIP domain, it will switch over to the secondary domain.

Values: IP address given in regular dot notation, e.g. 10.5.2.100

`auth_user`

Required: Only if the SIP server requires authentication.

Description: The authentication user name used to register the station to the SIP server.

Values: Text string.

`auth_pwd`

Required: Only if the SIP server requires authentication.

Description: The authentication user password used to register the station to the SIP server.

Values: Text string.

`sip_outbound_proxy`

Required: Optional

Description: Configures an outbound proxy server that receives all initiating request (INVITE and SUBSCRIBE) messages.

Values: IP address given in regular dot notation, e.g. 10.5.2.100

```
sip_outbound_proxy_port
```

Required: If proxy server is defined. Default is 5060.

Description: The UDP port on the SIP proxy server.

Values: Integer.

```
register_interval
```

Required: No. Defaults to 600 seconds.

Description: This parameter specifies how often the station will register, and reregister, in the SIP domain. This parameter will affect the time it takes to discover that a connection to a SIP domain is lost.

Values: Number of seconds. 60 ≤ register_interval ≤ 999999

7.4 Call Parameters

```
speeddial_1
```

Required: Yes

Description: This is the SIP ID for the extension to be called when the first call button is pressed, i.e. the telephone number of the receiving party.

Values: Integer value

```
speeddial_1_ip
```

Required: No

Description: If desired, an IP address can be configured as a backup for speeddial_1. If the station has no connection to any of the configured SIP domains, it can call directly to this IP address.

Values: IP address given in regular dot notation, e.g. 10.5.2.100

```
speeddial_2
```

Required: No

Description: This is the SIP ID for the extension to be called when the second call button is pressed, i.e. the telephone number of the receiving party.

Values: Integer value

```
speeddial_2_ip
```

Required: No

Description: If desired, an IP address can be configured as a backup for speeddial_2. If the station has no connection to any of the configured SIP domains, it can call directly to this IP address.

Values: IP address given in regular dot notation, e.g. 10.5.2.100

```
speeddial_3
```

Required: No

Description: This is the SIP ID for the extension to be called when the third call button is pressed, i.e. the telephone number of the receiving party.

Values: Integer value

```
speeddial_3_ip
```

Required: No

Description: If desired, an IP address can be configured as a backup for speeddial_3. If the station has no connection to any of the configured SIP domains, it can call directly to this IP address.

Values: IP address given in regular dot notation, e.g. 10.5.2.100

```
speaker_volume
```

Required: No. Defaults to 4.

Description: This parameter sets the volume of the station's speaker.

Values: Integer. 0 ≤ speaker_volume ≤ 7

```
mic_sensitivity
```

Required: No. Defaults to 5.

Description: This parameter adjusts the microphone sensitivity.

Values: Integer. 0 ≤ mic_sensitivity ≤ 7

```
rtp_timeout
```

Required: No. Defaults to 0.

Description: Cancels a call if the station does not receive RTP.

Values: Integer value: 0-9999 seconds. 0 = RTP timeout disabled.

```
remote_controlled_volume_override_mode
```

Required: No.

Description: Acts as a simplex mode after first DTMF * or # is received from remote station. Send DTMF * to talk and # to listen.

Values: Integer. 0 = disabled, 1 = enabled.

```
auto_answer_mode
```

Required: No.

Description: Enables auto-answer after a set number of seconds.

Values: Integer. 0 = disabled, 1 = enabled.

```
auto_answer_delay
```

Required: No. Defaults to 0.

Description: The number of seconds to delay the auto-answer.

Values: Integer. 0 ≤ delay ≤ 30

```
disable_disconnect_by_button
```

Required: No.

Description: Disable disconnect with the speed dial during and when setting up conversation.

Values: Integer. 0 = disabled, 1 = enabled.

```
activate_relay_event
```

Required: No. Function will be disabled if parameter not present.

Description: When enabled, the station will activate the relay when receiving the specified DTMF digit in the RTP stream. The DTMF digit must be sent according to RFC 2833.

Values: Integer. 0 ≤ activate_relay_event ≤ 9

`activate_relay_duration`

Required: No. Defaults to 60.

Description: This parameter sets the duration for the relay activation in seconds.

Values: 0 ≤ activate_relay_duration ≤ 240. 0 means that the relay stays open.

7.5 SNMP Parameters

`trap_receiver`

Required: No.

Description: The IP address of the server receiving SNMP traps.

Values: IP address given in regular dot notation, e.g. 10.5.2.100

`network`

Required: No.

Description: Used, together with the network mask, to determine the allowed network for reading the MIB on the IP station.

Values: IP address given in regular dot notation, e.g. 10.5.2.100. For example, with an allowed network of 10.5.2.0 and a network mask of 24, anyone with IP address 10.5.2.0 to 10.5.2.255 can access the MIB.

`network_mask`

Required: No.

Description: The mask used to determine the allowed network for reading the MIB.

Values: Integer. 0 ≤ network_mask ≤ 32. For example, with an allowed network of 10.5.2.0 and a network mask of 24, anyone with IP address 10.5.2.0 to 10.5.2.255 can access the MIB.

`community`

Required: No.

Description: A text string used as a password for authentication.

Values: String.

`enable_v1`

Required: No.

Description: Enables reading of MIB using SNMP version 1.

Values: Integer. 1 = enabled, 0 = disabled.

`enable_v2c`

Required: No.

Description: Enables reading of MIB using SNMP version 2c.

Values: Integer. 1 = enabled, 0 = disabled.

`enable_ipsStarted`

Required: No. Defaults to 1.

Description: If enabled, the station will send an SNMP trap when the station application is started.

Values: 0 = disabled, 1 = enabled.

`enable_sipRegistered`

Required: No. Defaults to 1.

Description: If enabled, the station will send an SNMP trap when successfully registered in the SIP domain.

Values: 0 = disabled, 1 = enabled.

`enable_sipRegisterFailed`

Required: No. Defaults to 1.

Description: If enabled, the station will send an SNMP trap if registration to the SIP domain failed.

Values: 0 = disabled, 1 = enabled.

`enable_callConnect`

Required: No. Defaults to 1

Description: If enabled, the station will send an SNMP trap when a call is connected.

Values: 0 = disabled, 1 = enabled.

`enable_callConnectFailed`

Required: No. Defaults to 1.

Description: If enabled, the station will send an SNMP trap if an incoming call to the station fails to connect for any reason (busy etc.).

Values: 0 = disabled, 1 = enabled.

`enable_callDisconnect`

Required: No. Defaults to 1.

Description: If enabled, the station will send an SNMP trap when a call is disconnected.

Values: 0 = disabled, 1 = enabled.

7.6 Example Configuration Files

7.6.1 Device Specific Configuration File

```
[general]
 auto_update_interval=10
 auto_update_image_type=A100G80200.01_10_1_2.bin
 auto_update_image_crc=C1466499

[sip]
 nick_name=Testname
 sip_id=1003
 sip_domain=10.5.2.209
 sip_domain2=10.5.2.138
 auth_user=1003
 auth_pwd=1003pass
 sip_outbound_proxy=10.5.2.138
 sip_outbound_proxy_port=5060
 register_interval=600

[call]
 speeddial_1=1000
 speeddial_1_ip=10.5.2.200
 speeddial_2=1004
 speeddial_2_ip=10.5.2.201
 speeddial_3=1005
 speeddial_3_ip=10.5.2.202
 speaker_volume=4
 mic_sensitivity=5
 rtp_timeout=60
 remote_controlled_volume_override_mode=1
 auto_answer_mode=1
 auto_answer_delay=10
 disable_disconnect_by_button=1
 activate_relay_event=*
 activate_relay_duration=10

[snmp]
 trap_receiver=10.5.2.219
 network=10.5.2.0
 network_mask=24
 community=public
 enable_v1=1
 enable_v2c=1
 enable_ipsStarted=1
 enable_sipRegistered=1
 enable_sipRegisterFailed=1
 enable_callConnect=1
 enable_callConnectFailed=1
 enable_callDisconnect=1
```

7.6.2 Global Configuration File

The global configuration file has the same parameters as the device specific file except that the four parameters below will be ignored. Hence, it is recommended that the following parameters not be used in the global configuration file.

```
nick_name
sip_id
auth_user
auth_pwd
```

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The WEEE Directive does not legislate that Zenitel, as a 'producer', shall collect 'end of life' WEEE.

This 'end of life' WEEE should be recycled appropriately by the owner who should use proper treatment and recycling measures. It should not be disposed to landfill.

Many electrical items that we throw away can be repaired or recycled. Recycling items helps to save our natural finite resources and also reduces the environmental and health risks associated with sending electrical goods to landfill.

Under the WEEE Regulations, all new electrical goods should now be marked with the crossed-out wheeled bin symbol shown.

Goods are marked with this symbol to show that they were produced after 13th August 2005, and should be disposed of separately from normal household waste so that they can be recycled.

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