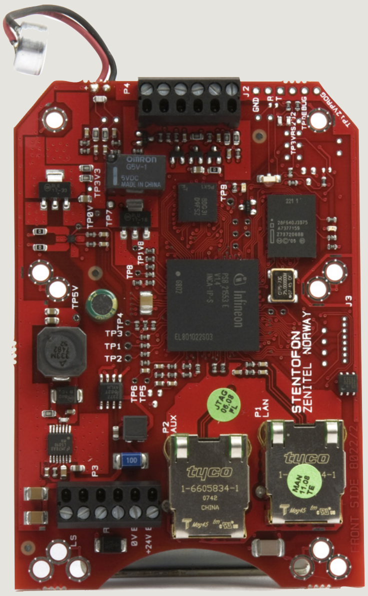
SIP Substation

INSTALLATION & CONFIGURATION GUIDE

STENTOFON



TECHNICAL MANUAL

A100K10812

About this Document

Document Scope

This document describes the installation and configuration of the STENTOFON SIP Substation and SIP Substation Kit.

Revision Information

Rev.	Date	Author	Status
1.0	23.4.2010	HKL	published
1.1	22.12.2011	HKL	Screenshots for new station SW

Contents

Figures

Tables

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.

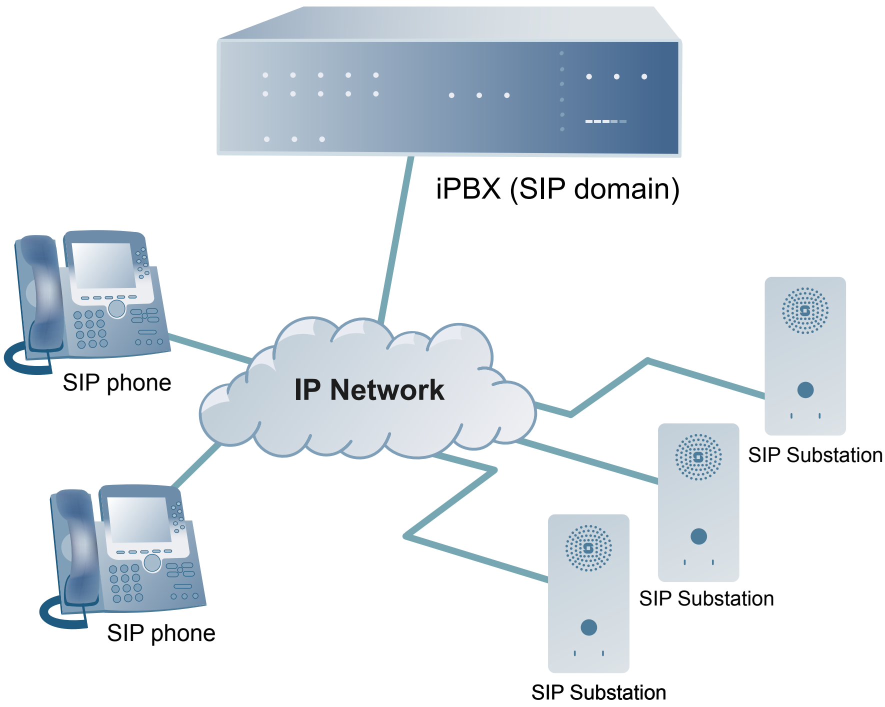


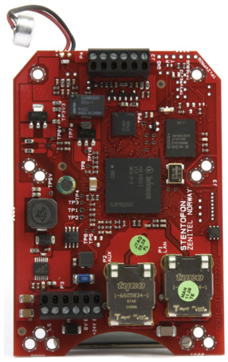
Figure 1 System Configuration

1.1 SIP Vandal Resistant Substation

- part no. 1008061100

The STENTOFON SIP Substation are custom-made IP intercom stations that integrate with any iPBX system. The SIP Vandal Proof Substation is tough, durable and resilient, and designed for use in the harshest environments. The station is typically used as a communication, information or emergency point and connects directly to the IP network, making it easy to deploy – anywhere and at any distance.

Like all STENTOFON stations, this SIP substation features superb audio quality. This is enabled through a set of advanced technologies such as active noise filtering, acoustic echo cancellation, wide band audio codec, and high power audio outputs.



1.2 SIP Substation Kit

- part no. 1008065200

The STENTOFON SIP Substation Kit features the same electronics board (PCB) that is used in its own IP substations. It is designed to build stations to the highest specification and for use in the harshest of environments. Like all STENTOFON stations, the SIP Substation Kit features superb audio quality. This is enabled through a set of advanced technologies such as active noise filtering, acoustic echo cancellation, wide band audio codec, and high power audio outputs.

1.3 Call Button Functions

The following table describes the various functions that are activated when the call button on the SIP substation is pressed.

Types of Calls:	Idle	Incoming Call	Call Attempt	Ongoing Call
Functions when call button is pressed:	Speed-dial number	Accept call	Terminate call	Terminate call

Table 1 Call Button Functions

2 Installation

2.1 Introduction

The table below is an overview of the main connectors involved when installing the STENTOFON IP Substations.

LAN	10/100 Mbps RJ45 port for LAN (uplink) connection. Supports PoE (802.3af). Draws power from either spare line or signal line.
AUX	10/100 Mbps RJ45 ports for auxiliary equipment such as PC and IP camera.
Input/Output	Pluggable screw terminal
Local Power	Pluggable screw terminal, 19-27 VDC Idle 4W, max. 8W

Table 2 Substation Connectors

2.2 Power Supply

The SIP Substation supports Power over Ethernet (PoE, IEEE 802.3 a-f) where power can be drawn from either the spare line or signal line.

If PoE is not available, the SIP Substation can be connected to a local power. A 24 VDC power supply should be used. Refer to *Appendix A: Substation Board Connections* to see where to connect local power.

2.3 Network Connection

There are two RJ45 ports on the PCB of the SIP Substation:

- The LAN port (P1) is for connecting to the network and the IPBX system.
- The AUX port (P2) is for connecting to auxiliary equipment such as a PC.

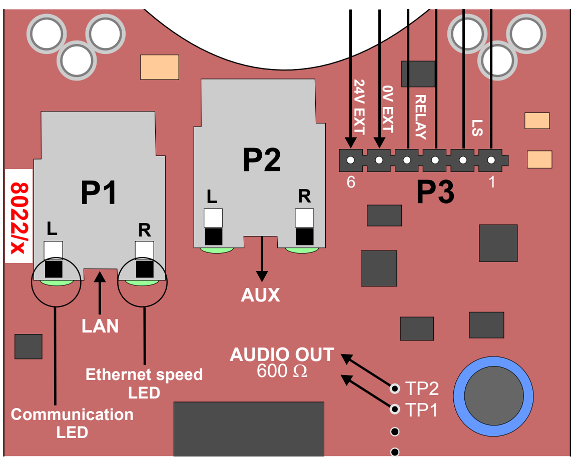


Figure 2 RJ45 Ports (P1 & P2) on PCB of SIP Substation

2.4 Input/Output Connections

The I/O connection options for the SIP Substations include:

- 3 digital inputs (P4)
- 1 relay output (P3

The relay output is typically used to open a door or gate.

The digital inputs are used to trigger a speed dial.

See section *3.4 Call Settings* on how to configure the I/O connection.

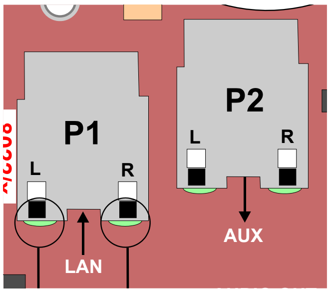
For pin settings on the connectors on the substation board, see *Appendix A: Substation Board Connections*.

3 Configuration

3.1 SIP Substation Web Interface

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

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



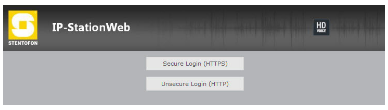
Connect both the PC and the SIP substation to a PoE switch and the LAN port (P1) on the IP intercom substation to the PC via the switch.

The factory default IP address of the substation is **169.254.1.100**. In order for your PC to communicate with the substation 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.

After the IP properties have been changed, access the substation 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 substation Login page is displayed.

To log into the substation:



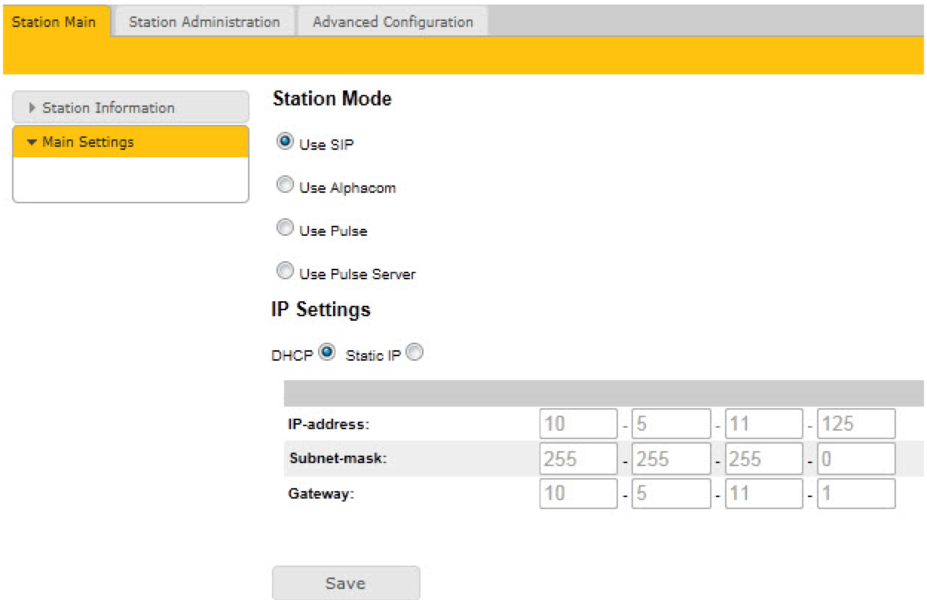
1. Click **Login**
2. Enter the default User name: **admin**
3. Enter the default password: **alphaadmin**

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

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

3.2 Station Main Settings

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



Station Mode

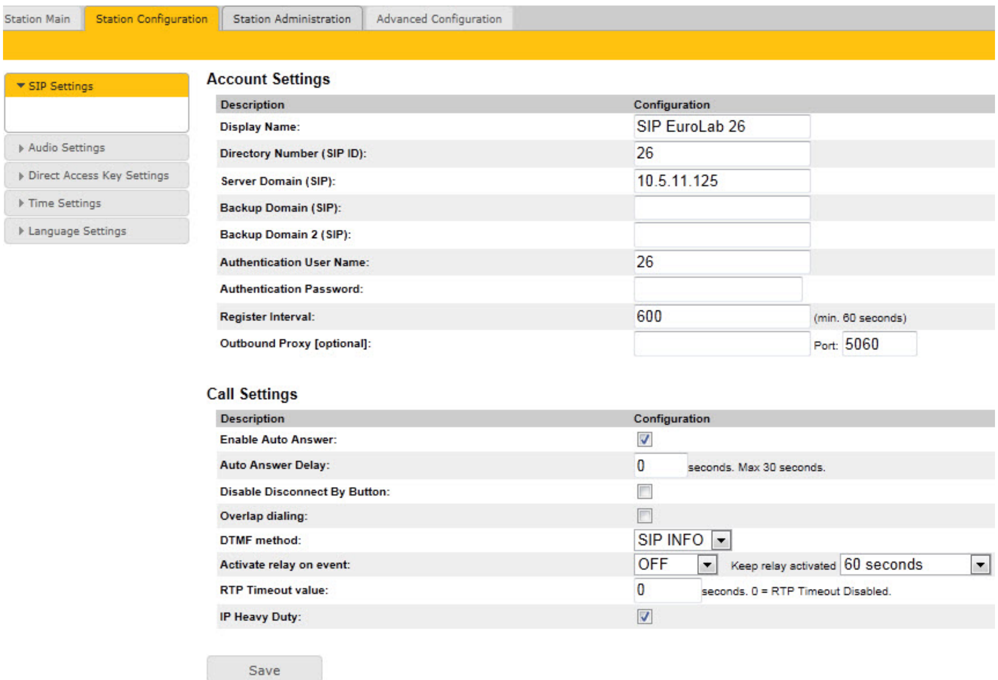
- Select the **Use SIP** radio-button

IP Settings

- **DHCP** – Use this option if the SIP Substation shall receive IP Settings from a DHCP server.
- **Static IP** – Use this option if the SIP Substation 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.

3.3 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

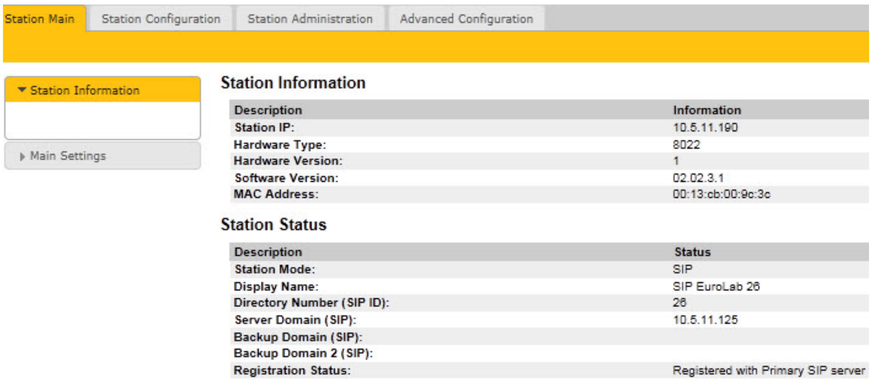
- When enabled, the station will activate the relay when receiving the specified DTMF character in the RTP stream. Select from the dropdown menu. Options are OFF, 1-9, *, In call or Ringing. The default setting is OFF.
- Select the number of seconds to keep the relay open in the range 1 to 240 from the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

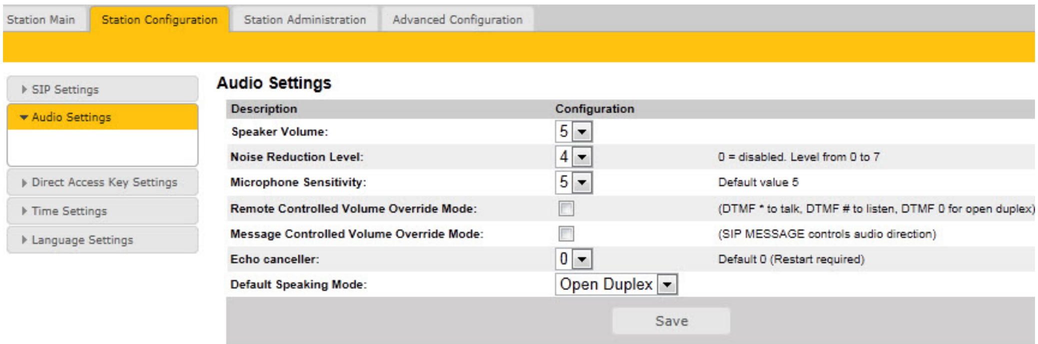
After completing the SIP configuration, click **Station Main** > **Station Information** and the main page may look like the following:



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

3.4 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

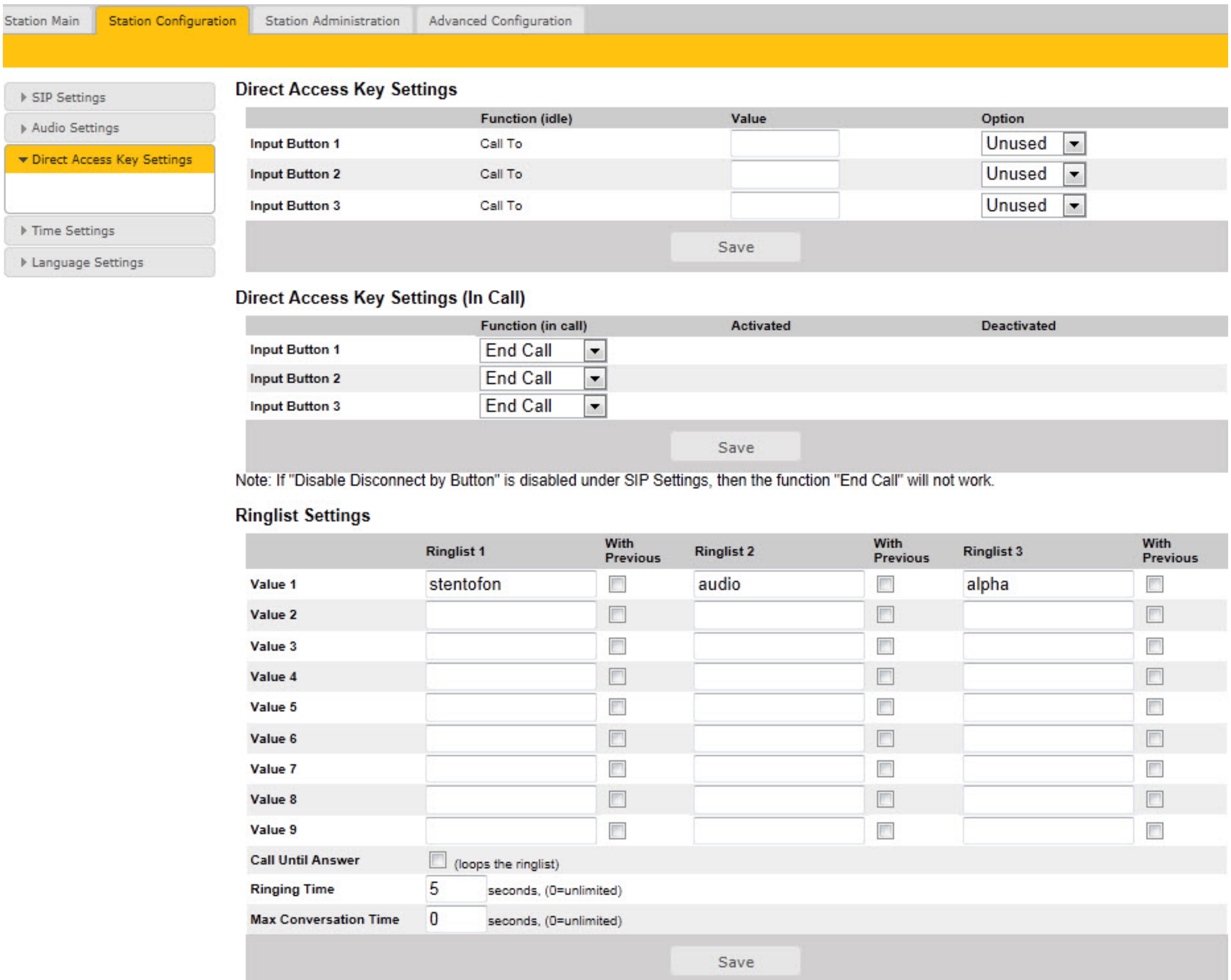
Default Speaking Mode

- Select between Open Duplex or Push-To-Talk

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

3.5 Direct Access Key Settings

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



<u>Direct Access Key Settings</u>

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.

<u>Direct Access Key Settings (In Call)</u>

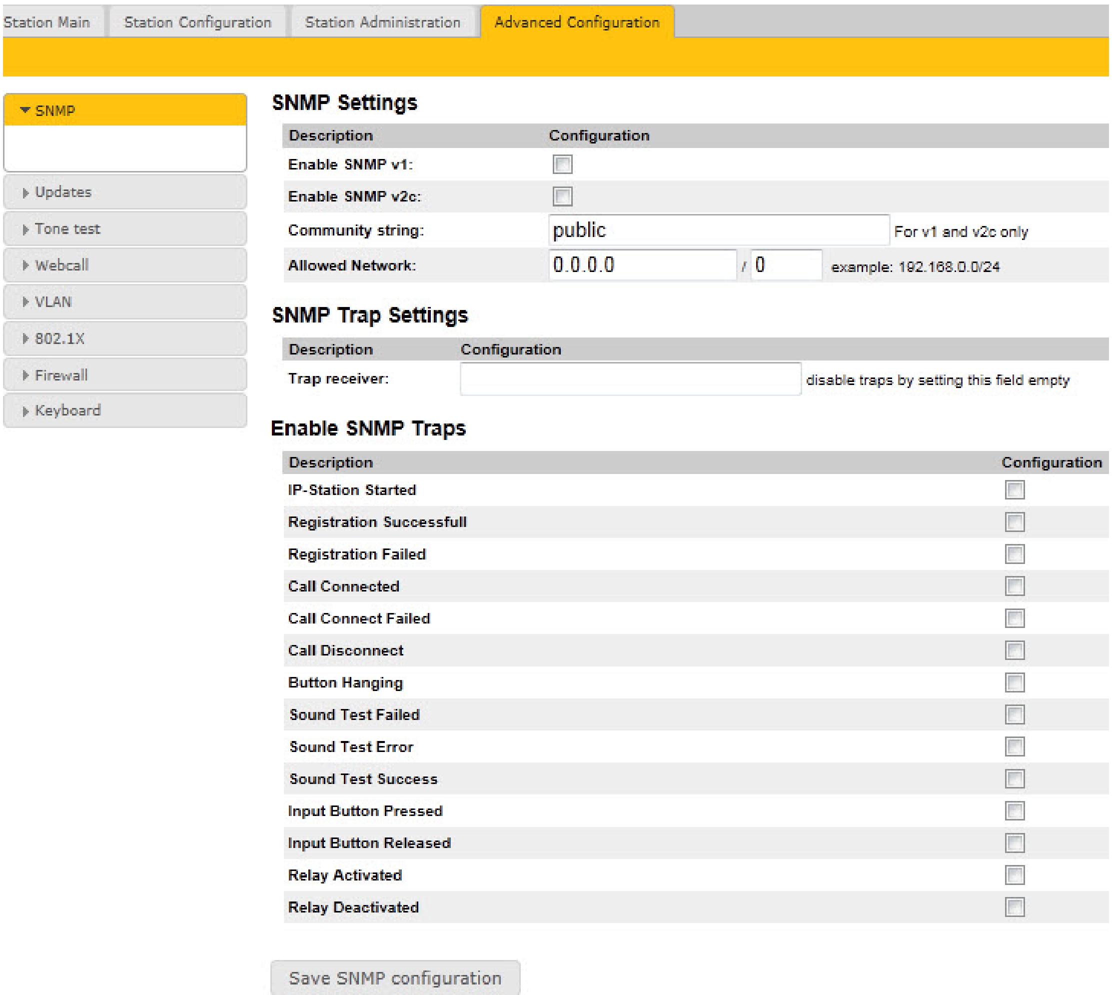
- Select input buttons 1 - 3 for direct access calls while in conversation.
- Options are: End Call, Do Nothing, Send Text, Send DTMF

✍ *Pin connections for the three call buttons are located on the P4 connector. See Appendix A: Substation Board Connections for more information.*

3.6 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

ipsStarted

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

sipRegistered

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

sipRegisterFailed

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

callConnect

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

callConnectFailed

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

callDisconnect

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

3.7 Automatic Configuration using TFTP

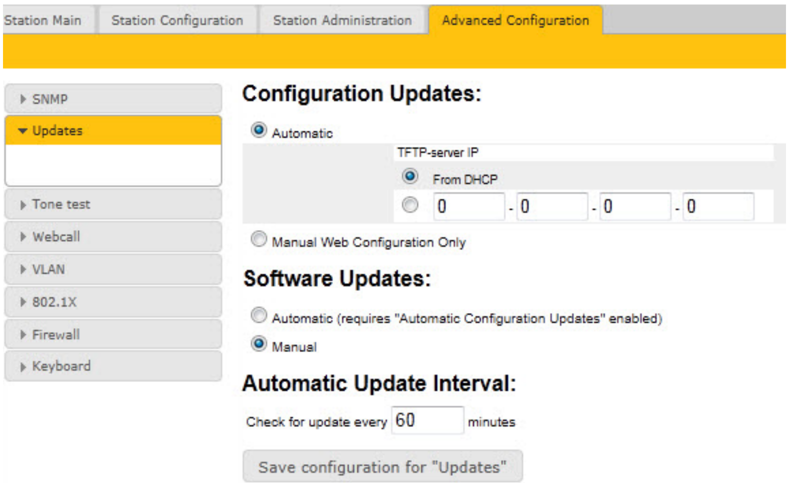
A SIP substation may be set up to automatically poll configuration settings for SIP, Call and SNMP from a TFTP server. The IP address of this TFTP server can be obtained using DHCP procedures or be manually configured.

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 *Appendix F: Configuration File Parameters*.
- Follow the procedures described in section *4.1 TFTP Server Program*.

To carry out automatic configuration from the substation 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 Substation 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 substation will now contact the TFTP server and run the configuration file to carry out the configuration procedure according to the set time interval.

3.8 Advanced Configuration Options

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

3.8.1 VLAN

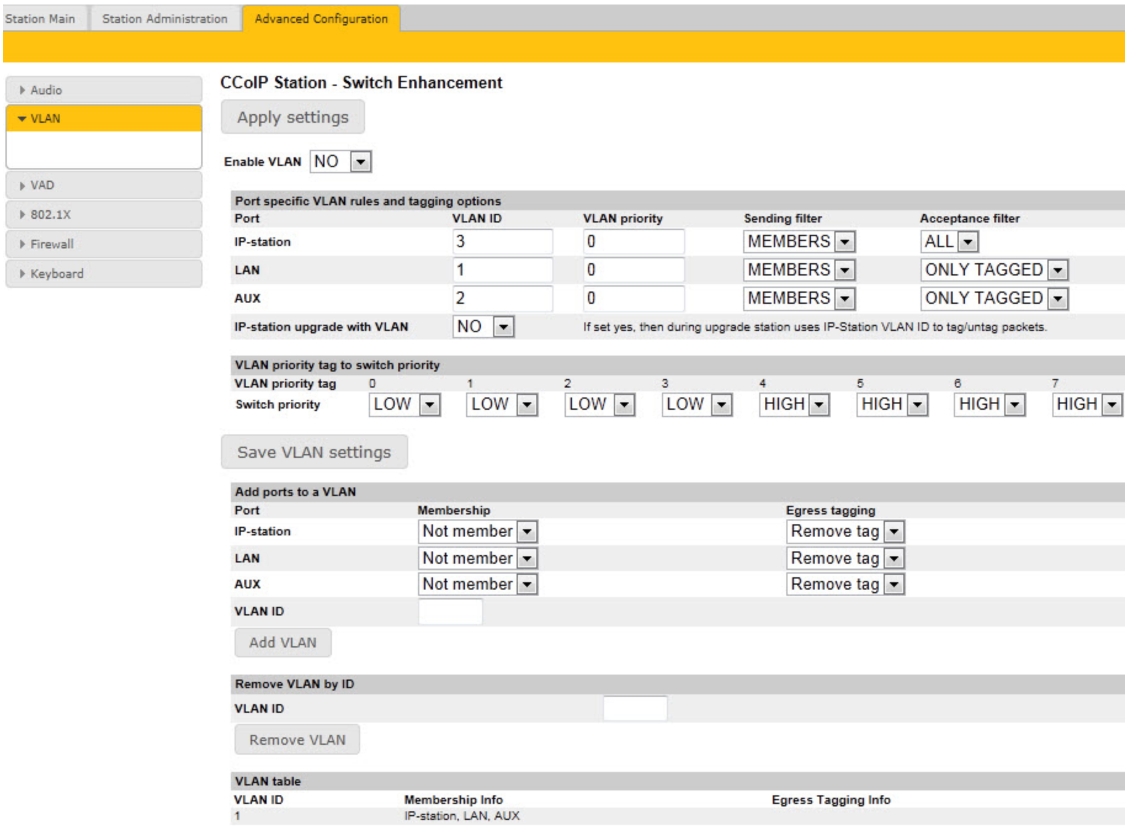
VLAN Tagging or IEEE 802.1Q is a networking standard allowing multiple bridged networks to transparently share the same physical network link without leakage of information between networks. IEEE 802.1Q — along with its shortened form *dot1q* — is commonly used to refer to the encapsulation protocol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

3.8.2 Network Access Control

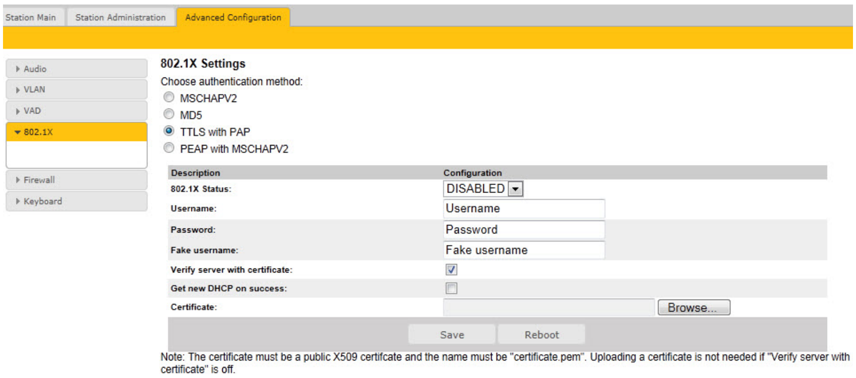
IEEE 802.1X is an IEEE Standard for Port-based Network Access Control (PNAC) By “port” we mean a single point of attachment to the LAN infrastructure. It provides an authentication mechanism to devices wishing to attach to a LAN, either establishing a point-to-point connection or preventing it if authentication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

✍ *STENTOFON SIP Substations 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.

4 Software Upgrade

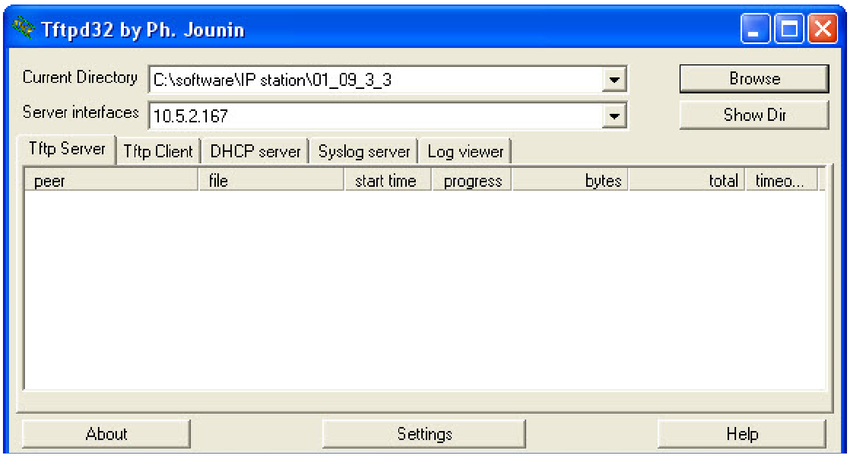
Software upgrade is carried out via the web server of the substation.

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

- Manual Upgrade
- Automatic Upgrade

4.1 TFTP Server Program

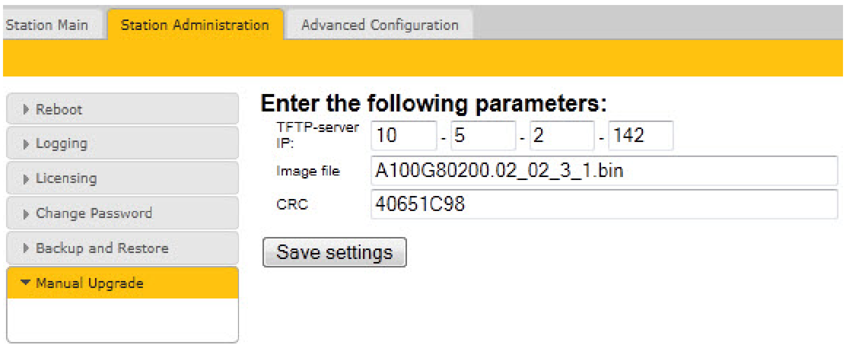
Both upgrade methods require that an TFTP Server is available and that the latest software image file has been downloaded from Zenitel's support website (AlphaWiki). During the upload process, the IP substation will connect to the TFTP Server and download the software. The TFTP Server program must already 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 substation 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.



4.2 Manual Software Upgrade

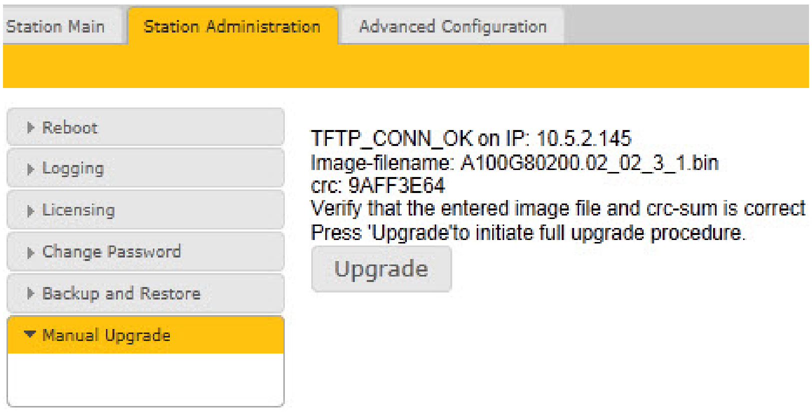
To carry out a manual software upgrade from the substation web server:

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 Substation 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 substation 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 substation. 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**.*

4.3 Automatic Software Upgrade

A SIP substation 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.

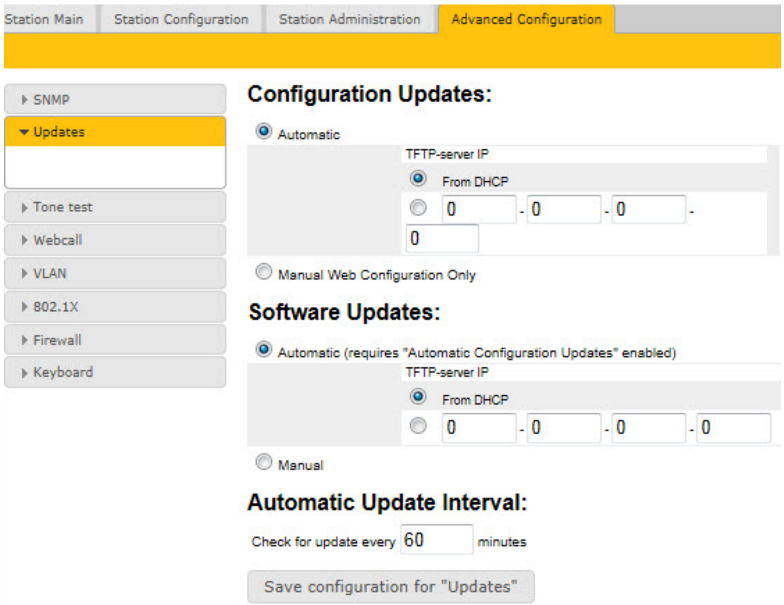
A configuration file should first be created. The relevant parameters in the configuration file are described in *Appendix F: Configuration File Parameters*.

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 substation 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 Substation 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 substation will now contact the TFTP server, download the software image and carry out the upgrade.

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

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

A Substation Board Connections

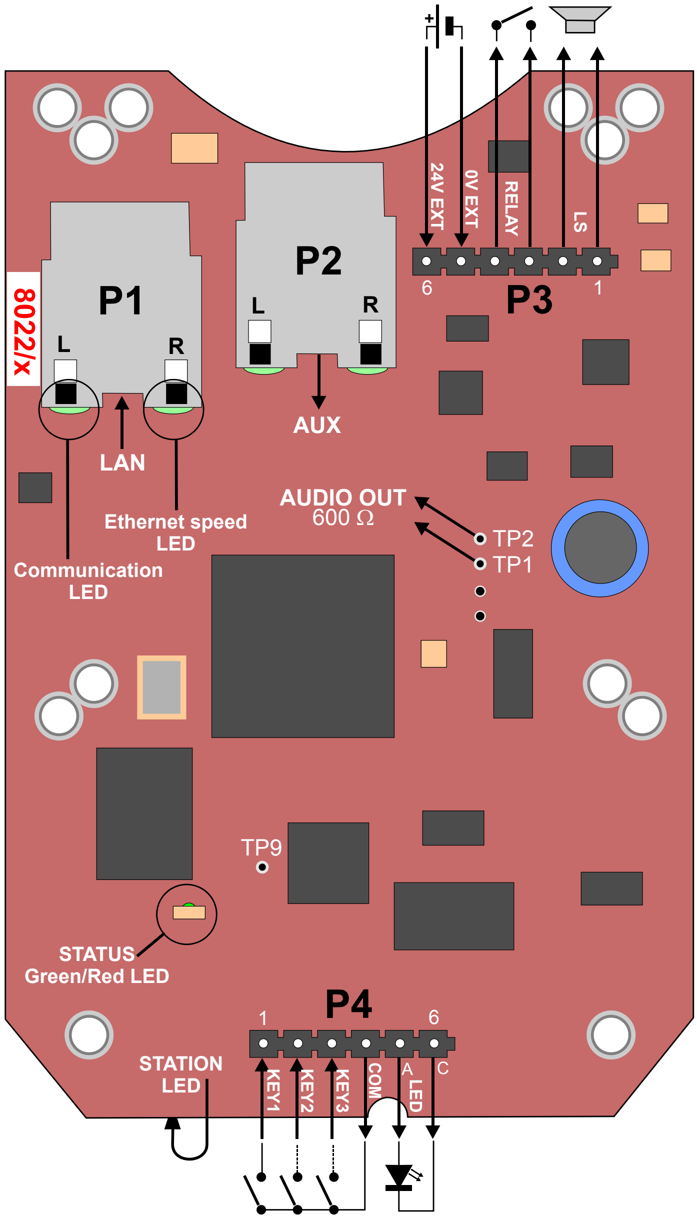


Figure 3 Substation Board Connections

There are four connectors on the SIP Substation board: P1, P2, P3, and P4.

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).

P2: RJ45 connector for auxiliary equipment like IP camera, PC or a second IP station.
This port does not have an individual IP address nor carry power for auxiliary equipment.

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

Pin 1/2: Connected to station loudspeaker. May also be used for 8-20 ohm / 2W external loudspeaker in parallel.

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

Pin 5/6: Connected to 24 VDC for station power when power is not distributed via LAN. (see *Appendix E: Substation Specifications* for details) Pin 6 is positive.

P4: 6-pin plug-on screw terminal for internal connections.

Pin 1/4: Call button no. 1

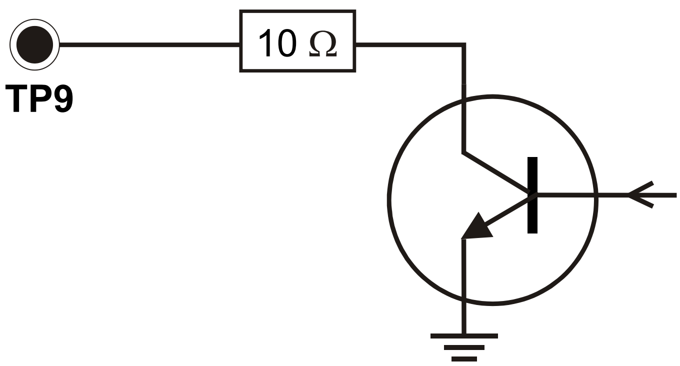
Pin 2/4: Call button no. 2 or logical input

Pin 3/4: Call button no. 3 or logical input

Pin 5/6: Station LED for call and message information

TP1/TP2: 0 dB, 600 ohm balanced audio output for connection to a power amplifier.

TP9: Logical output for a spare relay driver, max. 30 mA.



✍ *Note that the amplifier input MUST BE AN AC COUPLED INPUT (transformer, capacitors, etc.).*

B Substation Indication LEDs

B.1 Station LED (on board and front plate)

Steady light:	There is an ongoing call.
Blinking:	There is a call attempt or incoming call.
No light:	There are no calls.

B.2 LAN LEDs (on LAN and AUX RJ45 ports)

Left LED

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

Right LED

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

C Dimensions & Mounting Instructions

	Dimensions (HxWxD)	Weight
SIP Vandal Resistant Substation	180 x 92 x 46 mm	0.8 kg
SIP Substation Kit	110 x 72 x 20 mm	0.1 kg

Table 3 Substation Dimensions

C.1 SIP Substation Dimensions

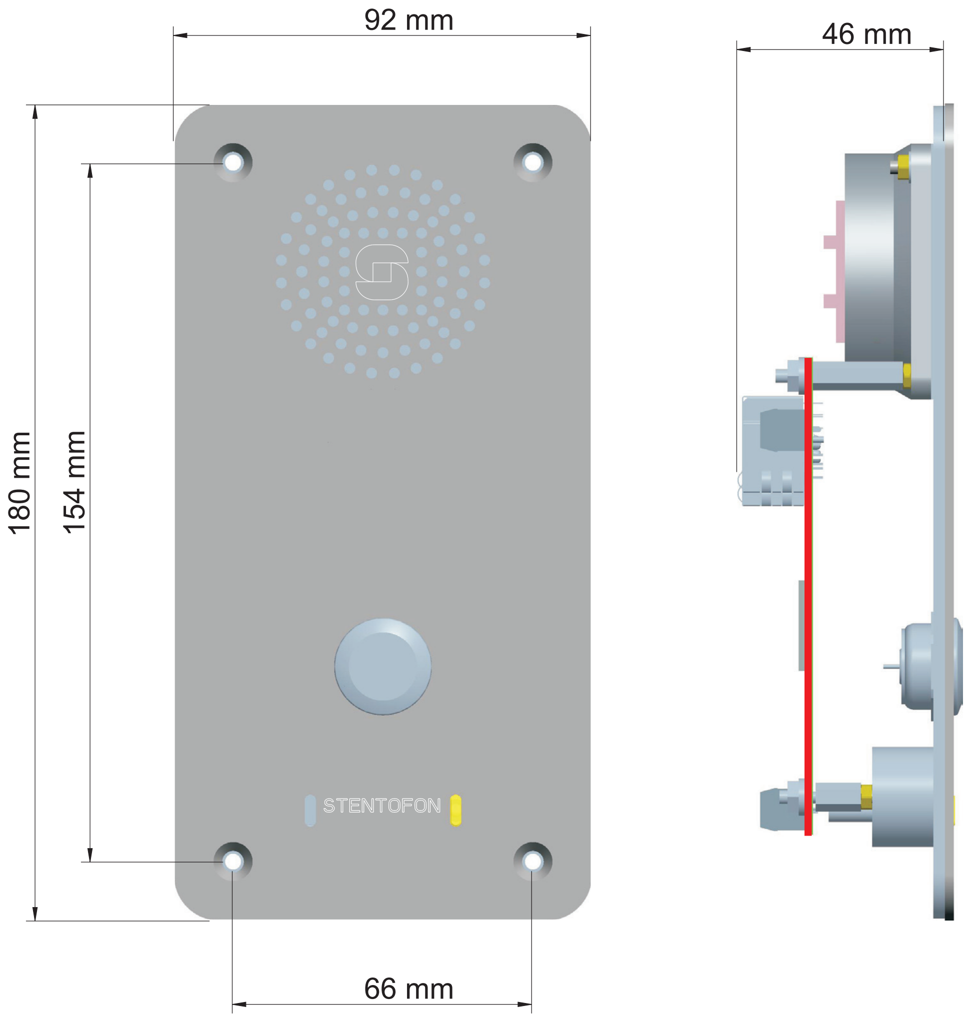


Figure 4 SIP Substation Dimensions

C.2 SIP Substation Flush Mounting

The SIP Vandal Resistant Substation (part no. 1008061100) can be mounted in a flush mount backbox (part no. 1008098100) or a surface mount backbox (part no. 1008098000).

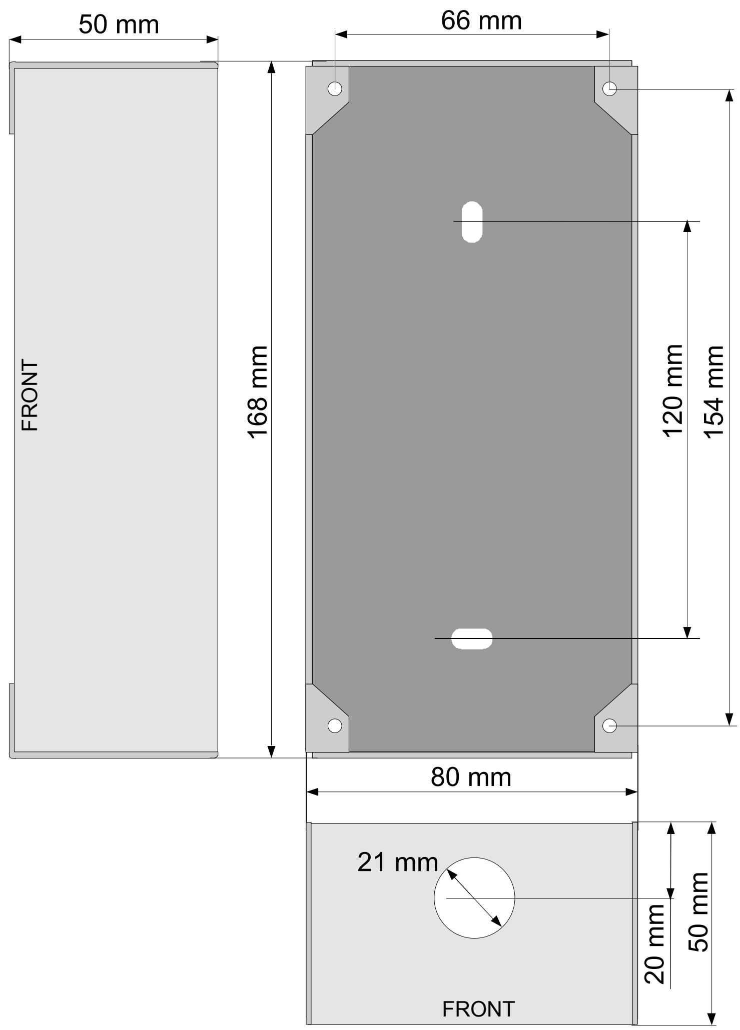


Figure 5 SIP Substation - Flush Mount Backbox

C.3 SIP Substation Surface Mounting

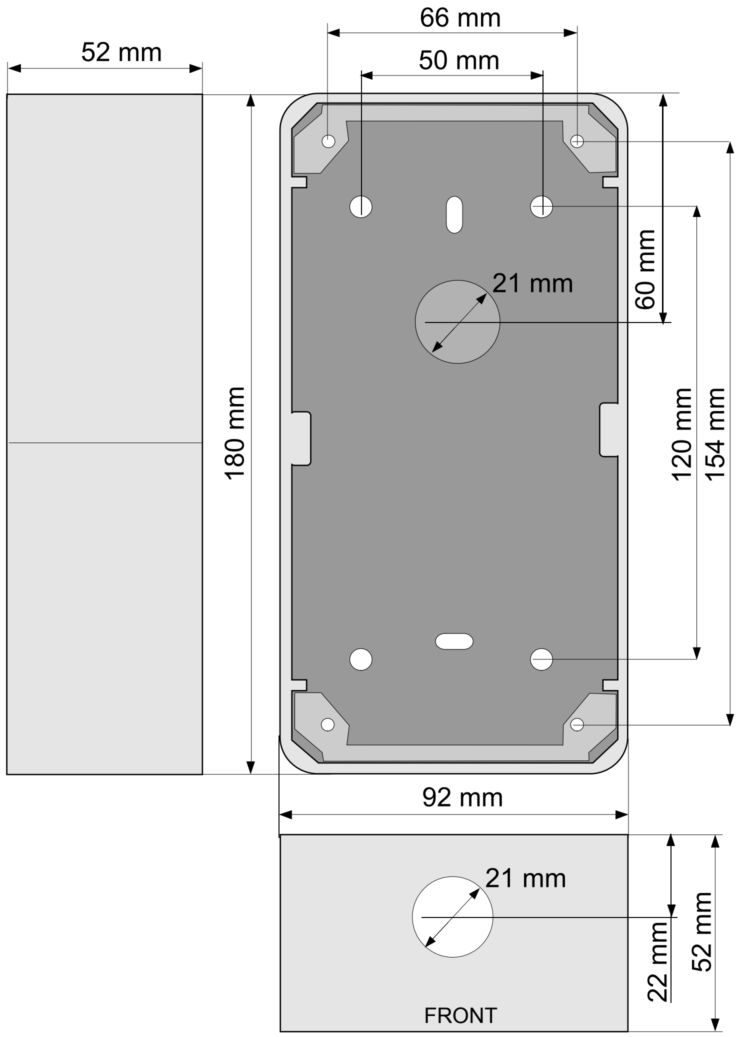


Figure 6 SIP Substation - Surface Mount Backbox

C.4 Substation Kit Dimensions

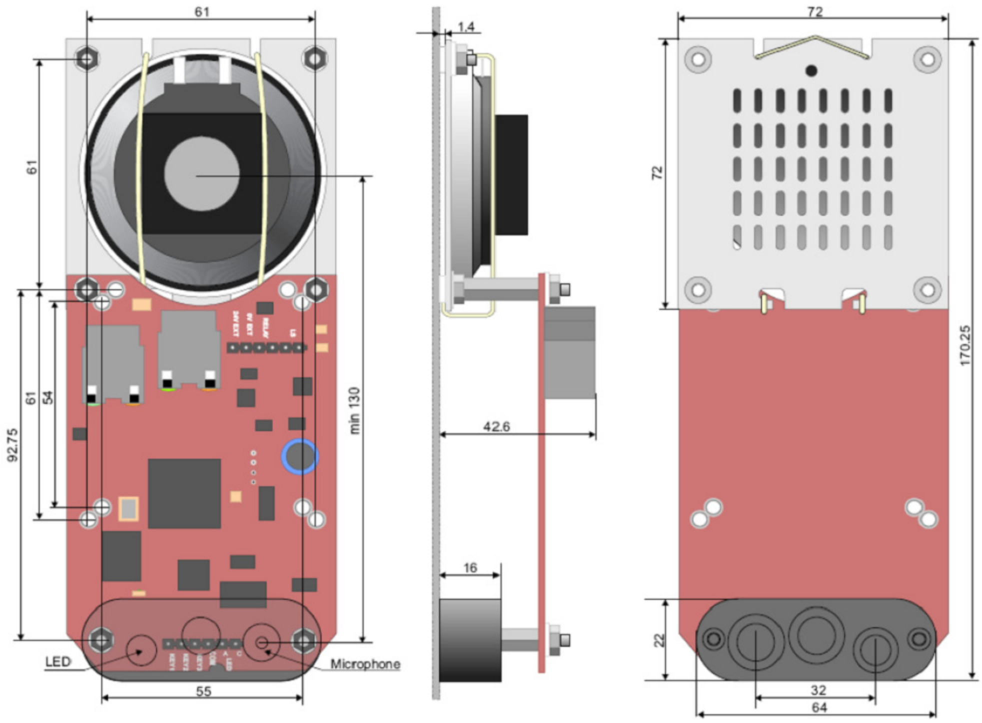


Figure 7 Substation Kit Mounting Dimensions (mm)

C.5 Mounting & Assembly Kit for Substation

- part no. 1008091000

The mounting and assembly kit includes gaskets, a 2-inch loudspeaker, loudspeaker housing, and microphone with mounting block for call LED.

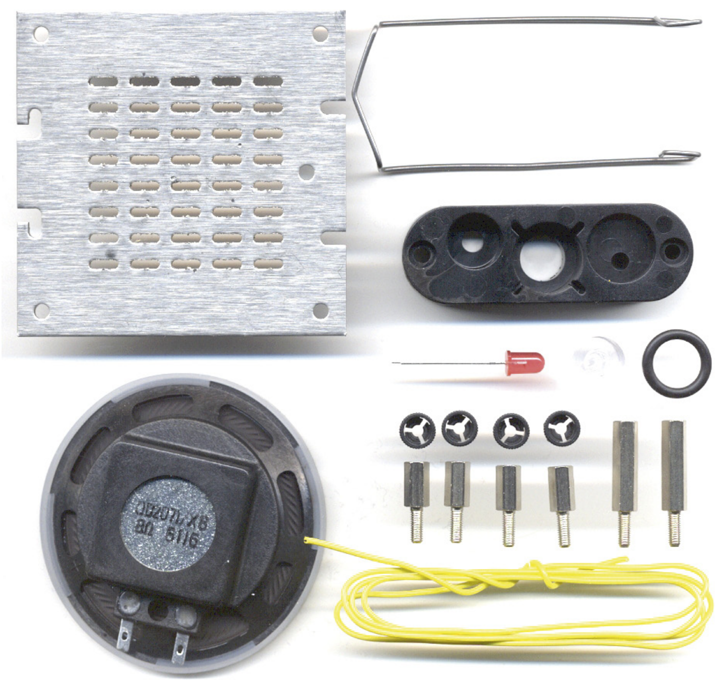


Figure 8 Substation Assembly Kit

D Restoring Factory Defaults

An IP Substation may have to be reset to its original factory default settings if, for instance, the password to the web server is forgotten. The defaults can be either be set to Static IP or Activated DHCP (when there are many new IP Substations in an installation).

D.1 Reset to Factory Default Settings with Static IP

1. While pressing the call button, power up the substation.
2. When the substation LED starts blinking, release the call button **after exactly 2 blinks**.
3. Let the LED blink for **exactly 2 more times**, then press the button again.
4. Keep the button pressed until the LED blinks fast 2 times indicating a successful reset with static IP.
5. Release the button and the substation will restart with the factory default settings.

Factory default values

Station IP address: **169.254.1.100**

Username: **admin**

Password: **alphaadmin**

D.2 Reset to Factory Default Settings with Activated DHCP

1. While pressing the call button, power up the substation.
2. When the substation LED starts blinking, release the call button **after exactly 2 blinks**.
3. Let the LED blink for **exactly 4 more times**, then press the button again.
4. Keep the button pressed until the LED blinks fast 4 times indicating a successful reset with activated DHCP.
5. Release the button and the substation will restart with the factory default settings.

Factory default values

Station IP address: (determined by DHCP server)

Username: **admin**

Password: **alphaadmin**

E Substation Specifications

E.1 SIP Vandal Resistant Substation

Dimensions (HxWxD)	180 x 92 x 46 mm
Weight	0.8 kg
Protection	Vandal resistant design, 2 mm stainless steel front, tamper proof fastening screws, buttons and loudspeaker grills
Protection class	With mounting backbox 1008098000 (surface mount): IP-44 With mounting backbox 1008098100 (flush mount): IP-55
Mounting	Flush mount in 50 mm deep backbox
Temperature range	-20˚C – +50˚C
Power	Power over Ethernet: IEEE 802.3 a-f, Class 0 Local power: 19 – 27 VDC, Idle 4W, max. 8W
Connectors	2 x RJ45 (Ethernet) 10/100 Mbps Pluggable screw terminals (audio and I/O)
Remote control	3 digital inputs, 1 relay output
SIP support	RFC 3261, SIP Info (DTMF), RFC 2833 (DTMF)
IP protocols	IP v4 - TCP - UDP - HTTPS – TFTP - RTP - RTCP -DHCP - SNMP DiffServ - TOS – STENTOFON CCoIP® - SIP
LAN protocols	Power over Ethernet (IEEE 802.3 a-f), VLAN (IEEE 802.1pq), Network Access Control (IEEE 802.1x), STP (IEEE 802.1d), RSTP (IEEE 802.1d-2004)
Audio technology	Wideband 200 Hz - 7 kHz (G.722) Telephony 3.4 kHz (G.711) Active noise filtering Acoustic echo cancellation Open duplex Adaptive jitter filter 1.5 Watt audio output 8 ohm loudspeaker impedance External audio out (0 dB, 600 ohm)
Management and operation	HTTPS (Web configuration) DHCP and static IP TFTP (firmware and configuration download) SNMP v1, v2 and v3 (monitoring) Status LED
Advanced features	Dual port managed data switch supporting VLAN Standby SIP server for redundancy

E.2 SIP Substation Kit

Dimensions (HxWxD)	110 x 72 x 20 mm
Weight	0.1 kg
Temperature range	-20˚C – +50˚C
Power	Power over Ethernet: IEEE 802.3 a-f, Class 0 Local power: 19 – 27 VDC, Idle 4W, max. 8W
Connectors	2 x RJ45 (Ethernet) 10/100 Mbps Pluggable screw terminals (audio and I/O)
Remote control	3 digital inputs, 1 relay output
SIP support	RFC 3261, SIP Info (DTMF), RFC 2833 (DTMF)
IP protocols	IP v4 - TCP - UDP - HTTPS – TFTP - RTP - RTCP -DHCP - SNMP - DiffServ - TOS – STENTOFON CCoIP® - SIP
LAN protocols	Power over Ethernet (IEEE 802.3 a-f), VLAN (IEEE 802.1pq), Network Access Control (IEEE 802.1x), STP (IEEE 802.1d), RSTP (IEEE 802.1d-2004)
Audio technology	Wideband 200 Hz - 7 kHz (G.722) Telephony 3.4 kHz (G.711) Active noise filtering Acoustic echo cancellation Open duplex Adaptive jitter filter 1.5 Watt audio output 8 ohm loudspeaker impedance External audio output (0 dB, 600 ohm)
Management and operation	HTTPS (Web configuration) DHCP and static IP TFTP (firmware and configuration download) SNMP v1, v2 and v3 (monitoring) Status LED
Advanced features	Dual port managed data switch supporting VLAN Standby SIP server for redundancy

F Configuration File Parameters

F.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.

F.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.

F.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. $60 \leq$ register_interval ≤ 999999

F.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.

F.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.

F.6 Example Configuration Files

F.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
```

F.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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