Exigo Network Amplifier for Rolling Stock
ENA2060-DC1
Installation & Configuration Manual

Contents

1 Introduction

1.1 Document Scope

This document describes the mounting, installation and configuration of the ENA2060-DC1 Amplifier for Rolling Stock that can be integrated with the Train Communication Network.

Item Number	Item Name	Description
1023122061	ENA2060-DC1	Exigo Network Amplifier for Rolling Stock

1.2 Publication Log

Revision	Date	Author	Status/Comments
1.0	17.10.2017	HKL	Published
1.1	26.3.2018	HKL	UIC Priority signal input
1.2	16.7.2018	HKL	General Purpose Input

1.3 Related Documentation

Document No.	Documentation
A100K11460	Exigo Technical Manual

1.4 Product Features

- Two SIP addressable audio channels – 2x60W
- Supports wide set of IP and networking standards
- Easily integrated into existing information concepts
- 100V speaker line technology – ease of cabling with galvanic separation
- Supports direct audio routing to wide range of induction loop amplifiers
- Designed, manufactured and tested according to EN50155 and EN45545
- Speaker loop monitoring
- Local audio inputs
- Additional I/Os for various integration options
- Fanless design
- Supports MIB2 and has the system MIB in place

1.5 Standards & Certifications

The ENA2060-DC1 Amplifier conforms to the following standards and certifications:

Standard	Description
EN50155	Railway Applications - Electronic Equipment used on Rolling Stock
EN50121-3-2	Railway Applications - Electromagnetic Compatibility - Part 3-2: Rolling Stock – Apparatus
EN45545	Railway Applications - Fire protection on railway vehicles
ETSI EN300 019-1-3	Stationary use at weather-protected locations. At altitudes with air pressure 70kPa – 106kPa.
IEC/EN 61373 Category 1, Class B	Railway applications – Rolling stock equipment – Shock and vibration tests
UIC 558/568	Standardized Connection Cable for PA for Mainline Trains

2 Mounting the Amplifier

The amplifier's mechanical construction is rigid enough to be mounted using the six slots in the mounting flanges to secure it to the mounting surface in the rail carriage. For the rolling stock environment, it is considered good practice to **mount support rails to better secure the amplifier**.

- Use six **5 mm diameter bolts (M5)** that are suitable for the mounting surface
- Secure the amplifier to the mounting surface by **fastening the bolts in the slots on the mounting flanges** (see Figure 1: Mounting Measurements)
- Mount the amplifier in a position that allows the **free flow of air through its cooling fins**

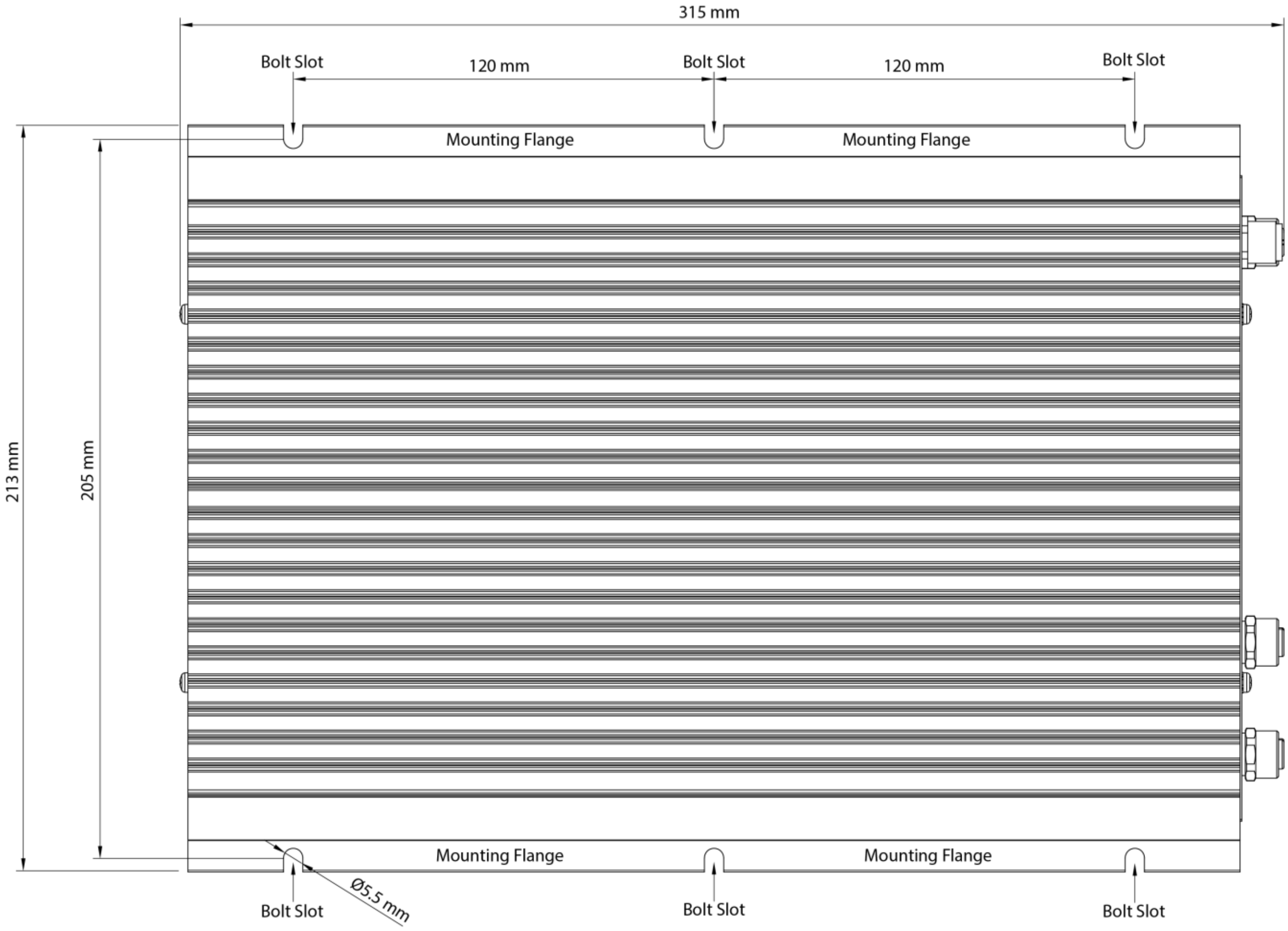


Figure 1: Mounting Measurements

3 Amplifier Connectors

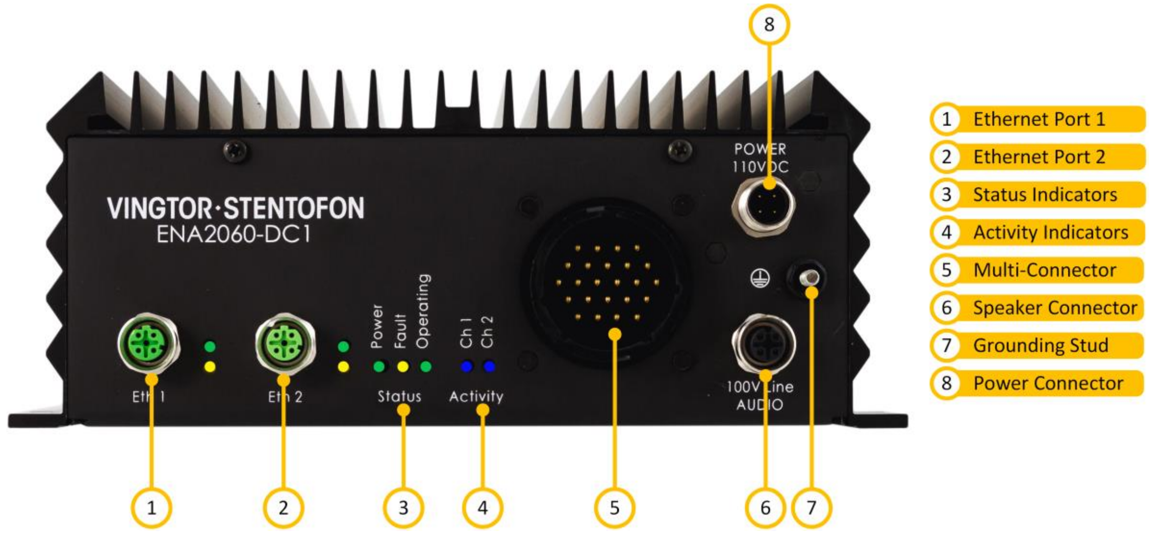


Figure 2: Connectors

3.1 Ethernet Ports 1 & 2

Ethernet Ports 1 & 2 are connected to the Train Communication Network (TCN) for exchanging data packets with the SIP server and other systems such as Network Monitoring System (NMS). Port 1 is the main Ethernet connector while Port 2 is the backup Ethernet connector. The amplifier shall be connected to the network using a **4-pin M12 D-Coded** male connector.

There are two LED indicators next to each Ethernet port:

Link: **Green** LED is lit when the amplifier is connected to the network

Activity: **Yellow** LED is lit when data is being transmitted

3.2 Status Indicators

The status indicators display the status of important parameters such as power supply and faults.

Power: **Green** LED is lit when the system has power and is hardware controlled.

Fault: **Yellow** LED is lit when the device has detected any faults in hardware or on the loudspeaker line (software controlled). The LED is lit when the processor is not running correctly (hardware controlled).

Operating: **Green** LED is in low frequency flashing pattern to indicate CPU is running and system is OK.

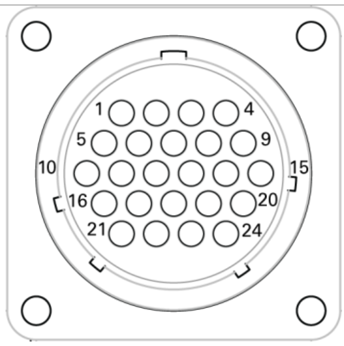
3.3 Activity Indicators

Activity: **Blue** LEDs (one for each audio channel **Ch1** and **Ch2**) that will be lit when audio is present on the audio outputs.

3.4 Multi-Connector

The Multi-Connector can be used for various functions as described below.

Pin Configuration Table



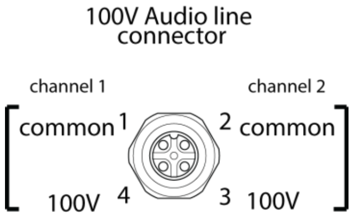
Pin	Function	Description
1	UIC Pri +	UIC Priority Signal Input
2	UIC Pri -	
3	GPO1 +	24 VDC Signal Output (physical volume controller override)
4	GPO1 -	0 VDC Signal Output
5	UIC ON/OFF +	UIC On/Off Signal Input
6	UIC ON/OFF -	
7	GPI1 +	General Purpose Input
8	GPO2 +	24 VDC Signal Output (physical volume controller override)
9	GPO2 -	0 VDC Signal Output
10	UIC IN +	UIC Audio Signal Input
11	UIC IN -	
12	GPI2 +	General Purpose Input (PTT key or audio triggered by other equipment)
13	GPI1 -	
14	CH1 0dB + (OUT)	Balanced Line Out for audio channel (inductive loop system)
15	CH1 0dB – (OUT)	
16	LINE IN +	Audio Input, Line Input (local audio source)
17	LINE IN -	
18	GPI2 -	General Purpose Input (PTT key or audio triggered by other equipment)
19	CH2 0dB + (OUT)	Balanced Line Out for audio channel (inductive loop system)
20	CH2 0dB – (OUT)	
21	MIC IN +	Audio Input for Microphone (ambient noise sensing or local announcement)
22	MIC IN -	
23	MIC IN CHASSIS	
24	SPARE/TEST	

3.5 Audio Line Connector

The Audio Line / Speaker connector is a **4-pin M12 A-Coded female** connector that has two channels with 100-volt output per channel.

Audio Outputs

- 2 x 60W, 100V Audio
- Channel Outputs
- Monitoring Signals



3.6 Grounding Stud

This is an **M4x20 stud** for fixing a tab to the chassis and grounding the amplifier.

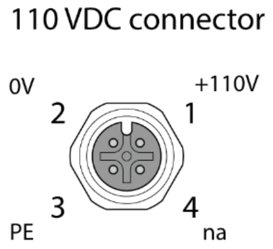
NOTE

If the amplifier isn't connected to ground, it will not be able to detect ground faults or short-circuits to ground.

3.7 Power Connector

The power connector is a **4-pin M12 A-Coded male** plug connector for 110 VDC power supply.

The **green LED** indicator for Power will be lit when the amplifier is powered up.



4 Configuration

Configuration of the amplifier can be done manually via the amplifier's web interface or automatically with DHCP provisioning via a TFTP server (see section 4.6).

After making sure that the amplifier is connected to the same LAN and logical subnet as your PC, follow the procedure described below.

When the amplifier is connected to the network, the IP address is automatically obtained in one of two ways:

- An IP address is obtained from a DHCP server if there is one.
- If there is no DHCP server, an IP address in the range 169.254.x.x will be assigned.

To be able to hear the amplifier announce its IP address:

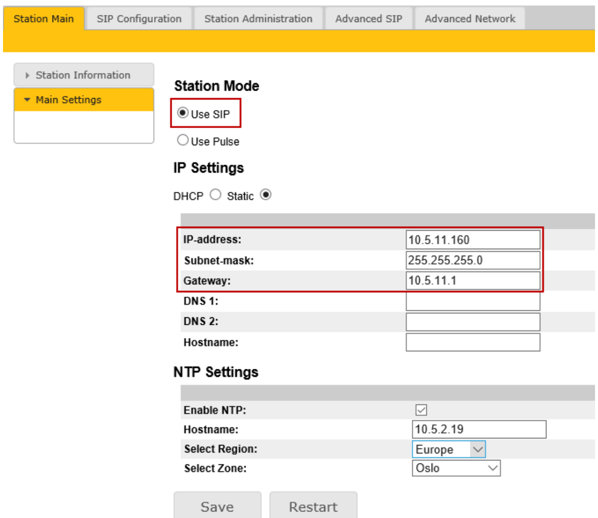
1. Either connect a 100 V speaker to Channel 1 on the 100V Audio Line (see section 3.5) or connect a 50-ohm speaker (or headset) to channel output 1 (pins 14 and 15) on the Multi-Connector (see section 3.4)
2. Ensure that the amplifier is NOT connected to any SIP server
3. After powering up the amplifier, short input 1 between pins 7 and 13 on the Multi-Connector (see section 3.4)

- The connected speaker will now announce the amplifier's IP address

4.1 Main Settings

To configure the IP address and directory number:

1. Open a web browser and enter the amplifier's IP address
2. Log in with username: **admin** and password: **alphaadmin**
3. Select **Station Main** > **Main Settings**

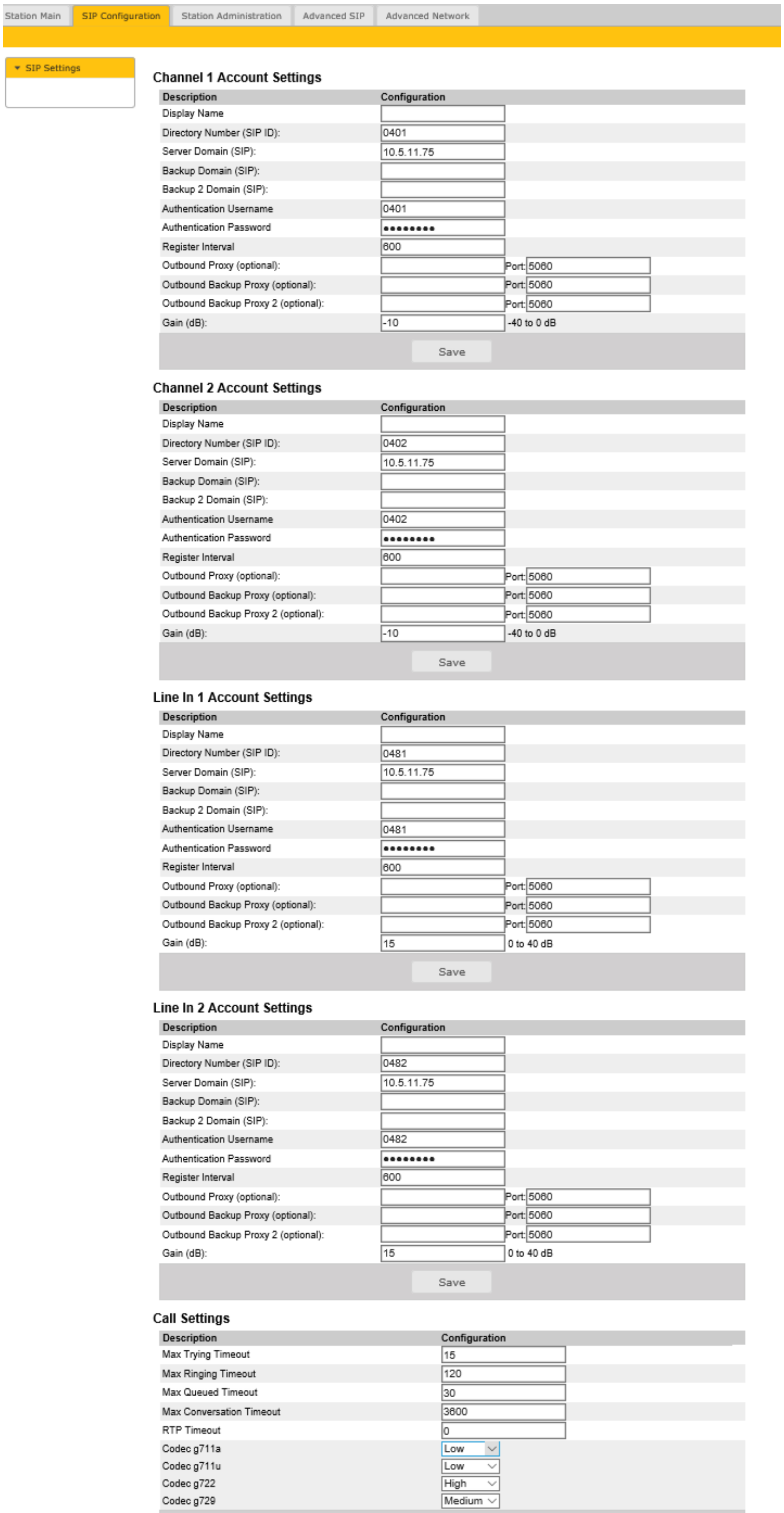


Set the following values:

- Station Mode: **Use SIP**
- IP Settings: **Static IP** or **DHCP**
  - IP-address: **10.5.11.160** (example IP address of Amplifier)
  - Subnet-mask: **255.255.255.0**
  - Gateway: **10.5.11.1**
- NTP Settings:
  - Enable NTP: **Check the box to enable NTP**
  - Hostname: **10.5.2.19** (example IP address of Hostname)
  - Select Region: **Europe** (example Region)
  - Select Zone: **Oslo** (example Zone)
- Click **Save** followed by **Restart**

4.2 SIP Settings

- Select **SIP Configuration** > **SIP Settings**



SIP Settings have to be done for Channel 1, Channel 2, Line In 1, and Line In 2.

Channel 1 & Channel 2 / Line In 1 & Line In 2 Account Settings

Display Name

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

Directory Number (SIP ID)

This is the identification of the amplifier in the SIP domain, i.e. the ID number for the amplifier. This parameter is mandatory. Enter the SIP ID in integers according to the SIP account on the SIP domain server. Channel 1 and Channel 2, Line In 1 and Line In 2 have different SIP ID numbers.

Server Domain (SIP)

This parameter is mandatory and specifies the primary domain for the amplifier 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. 10.5.11.75.

Backup Domain (SIP)

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

Backup 2 Domain (SIP)

Optional - This is the tertiary SIP Server Domain used as backup in case both the primary and secondary domains fail.

Authentication Username

This is the authentication user name used to register the amplifier 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

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

Register Interval

This parameter specifies how often the amplifier will register, and re-register 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.

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

Outbound Backup Proxy 1&2 (optional)

Enter the IP address and **Port** number of the backup outbound proxy server.

Gain (dB)

Enter the audio gain value in decibels in the range **-40 to 0 dB for Channel 1/2** and **0 to 40 dB for Line In 1/2**

- Click **Save** for each Account Setting that has been completed

Call Settings

Max Ringing Time

How long a PA call can be ringing before hanging up.

Max Conversation Time

How long a PA call can be in conversation before hanging up.

Max Queued Time

How long a PA call can be queued before hanging up.

RTP Timeout

This cancels a PA call if the amplifier 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.

Codec g729 / Codec g722 / Codec g711a / Codec g711u

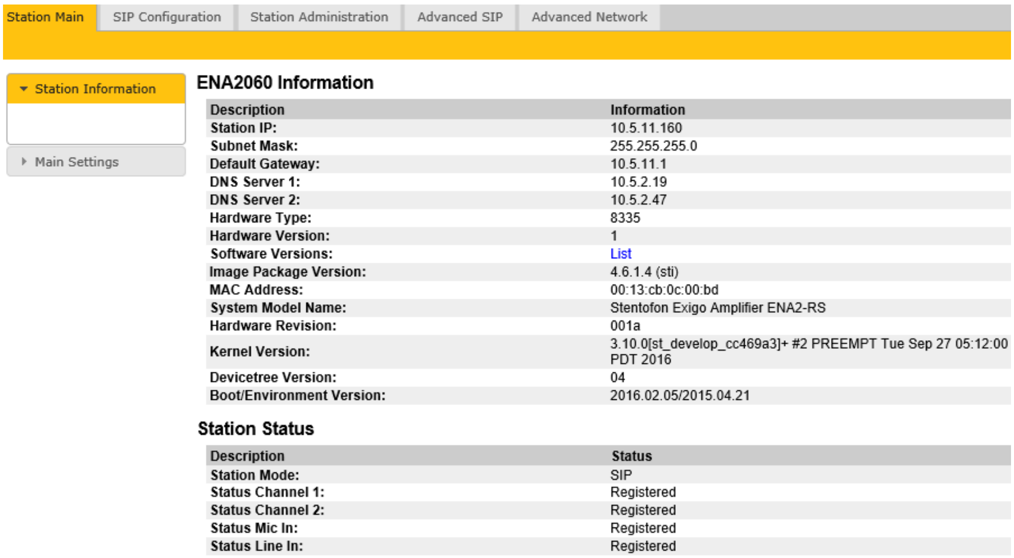
Options are: **Unused**, **Low**, **Medium**, **High**.

- Click **Save**

4.3 Station Information

To confirm that configuration has been done correctly, open the Station Information page:

- Select **Station Main** > **Station Information**



Description	Information
Station IP:	10.5.11.160
Subnet Mask:	255.255.255.0
Default Gateway:	10.5.11.1
DNS Server 1:	10.5.2.19
DNS Server 2:	10.5.2.47
Hardware Type:	8335
Hardware Version:	1
Software Versions:	List
Image Package Version:	4.6.1.4 (sti)
MAC Address:	00:13:cb:0c:00:bd
System Model Name:	Stentofon Exigo Amplifier ENA2-RS
Hardware Revision:	001a
Kernel Version:	3.10.0[st_develop_cc469a3]+ #2 PREEMPT Tue Sep 27 05:12:00 PDT 2016
Devicetree Version:	04
Boot/Environment Version:	2016.02.05/2015.04.21

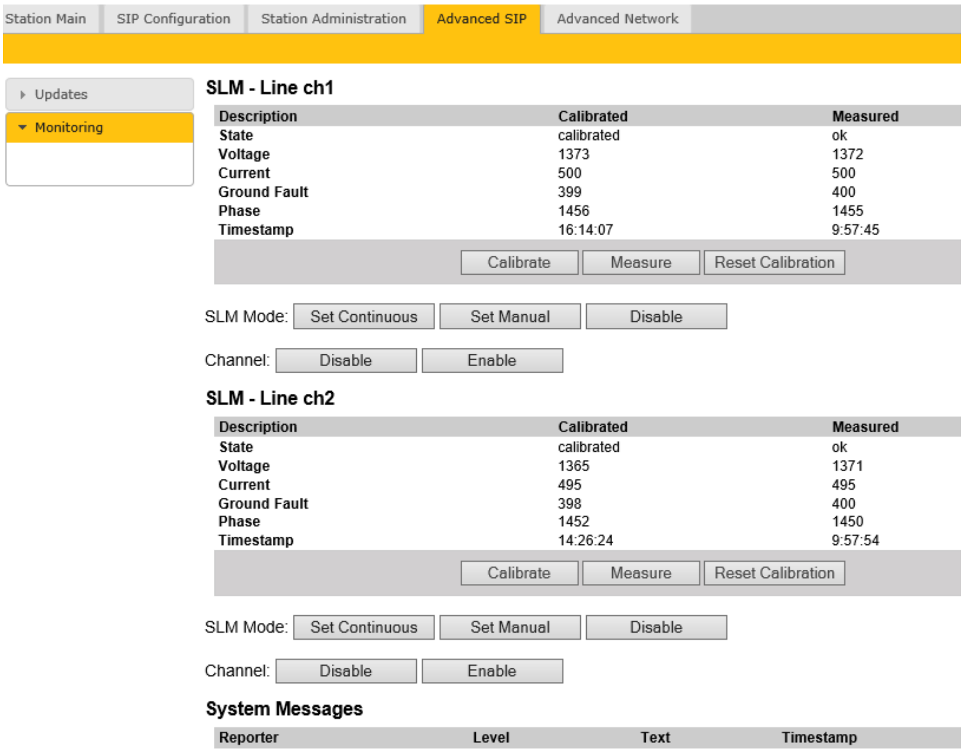
Description	Status
Station Mode:	SIP
Status Channel 1:	Registered
Status Channel 2:	Registered
Status Mic In:	Registered
Status Line In:	Registered

4.4 Speaker Line Monitoring

The ENA2060-DC1 amplifier has the option to monitor each speaker line to detect line faults such as **Open**, **Shorted** or **Ground Fault**.

To configure Speaker Line Monitoring:

- Select **Advanced SIP** > **Monitoring**



This **Monitoring** page shows the current status for each speaker line and whether there are any System Messages reported.

After the speaker line is set up with all the speakers and the line is properly connected to the amplifier, the speaker line that is to be monitored must be calibrated. When calibration is carried out, the amplifier makes an initial measurement of the speaker line to check whether the line load is within acceptable limits. The calibration values are used as a reference and are compared with later speaker line measurements for any significant changes that will cause line fault reporting. After the calibration, the speaker line load must not be changed, or else false fault messages may be reported.

SLM – Line

The **SLM – Line** status section shows the speaker line calibration status in the **Calibrated** column and the latest measured line values in the **Measured** column. The various statuses are:

State: This shows the line's calibrated state, i.e. **calibrated**, **uncalibrated**, **ok**, **open**, **shorted** or **ground fault** under the **Calibrated** column and the latest measured line state, i.e. **unknown**, **ok**, **open**, **shorted** or **ground fault** under the **Measured** column.

Voltage / **Current** / **Ground Fault** / **Phase**: These show the line's measured calibration values under the **Calibrated** column and the latest measured values under the **Measured** column.

Timestamp: This shows the time when either the **Calibrate** or **Reset Calibration** button were clicked and the time when the last SLM measurement or any operation that affects the line measurement values were executed.

The 3 buttons in this section have the following functions.

Calibrate: Click to start the speaker line calibration sequence. The Channel must be ON (**Enable** must be clicked) and a proper line load must be attached.

Measure: Click to start a manual SLM measurement. The line state must be calibrated as **ok** and **Set Manual** must be clicked.

Reset Calibration: Click to clear the calibrated state (state=**uncalibrated**) and remove any line fault related **System Messages**.

SLM Mode:

Set Continuous: Click to monitor the speaker line once every xx minutes and report any fault detected.

Set Manual: Click to have an operator/technician manually execute an SLM operation by clicking the **Measure** button. The manual operation may take up to 15 seconds.

For either SLM mode used, the status and timestamp for the latest measurement will be shown in the **Measured** column and any faults will be visible in the **System Messages** section.

Any faults reported will be removed if subsequent SLM measurements result in an **ok** state or if the **Reset Calibration** button is clicked.

Channel:

Disable: Click to switch OFF the amplifier output for that channel.

Enable: Click to switch ON the amplifier output for that channel.

To calibrate a speaker line:

1. Click the **Enable** button to switch ON amplifier output
2. Click either **Set Continuous** or **Set Manual** button
3. Click the **Calibrate** button

Wait for the calibration state to be updated (approx. 5-10 seconds).

State will report either **ok**, **open**, **shorted** or **ground fault**.

If the state is **ok** the amplifier's Speaker Line Monitoring (SLM) feature is ready to operate.

If the state is either **open**, **shorted** or **ground fault**, the line must be corrected and the calibration procedure must be repeated.

The four benchmarks for the evaluation of line state is Voltage, Current, Phase and Ground Fault.

The values measured on Voltage, Current and Phase will together provide an evaluation of the line state regarding whether it is Shorted, Open or OK.

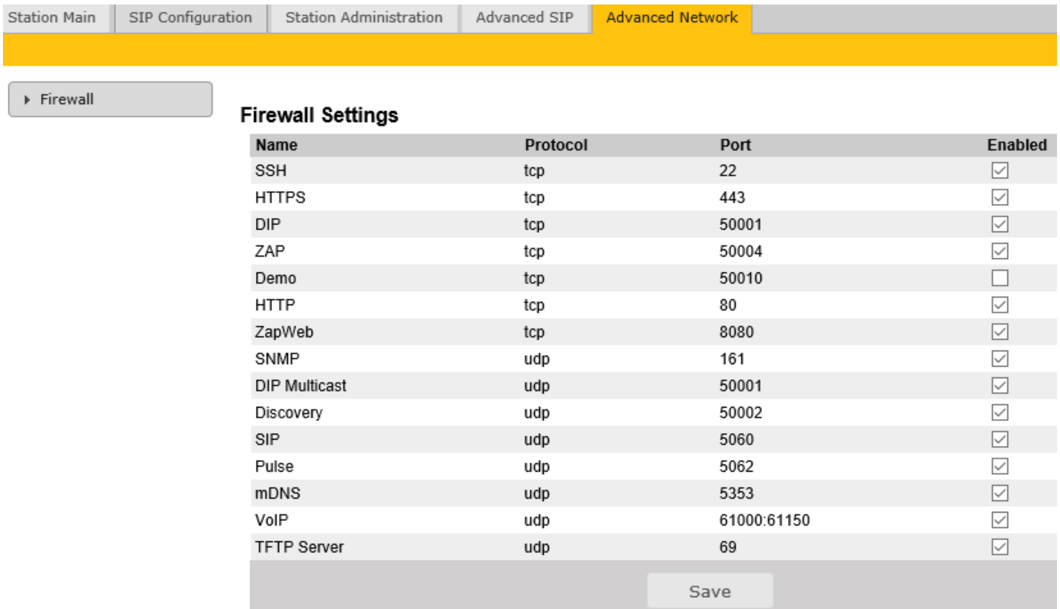
Typical SLM values for a 100 m line with 15W speaker load, calibrated values and various Shorted, Open and Ground Fault situations are as follows:

	Calibration OK	Shorted @ENA	Shorted @100M	Open @ENA	Open @100M	Ground Fault
Voltage	985	403	403	780	1015	400..1500
Current	1045	1370	1365	435	1065	400..1500
Phase	1075	765	925	1545	975	400..1500
Ground Fault	405	405	405	405	405	1580

4.5 Firewall Settings

To be able to upload configuration data, the amplifier network port must be opened for certain protocols.

- Select **Advanced Network** > **Firewall**



Make sure that the port for **SIP**, **SSH**, **DIP**, **HTTP**, **ZAP**, **SNMP**, **TFTP server**, **VoIP** is enabled by checking the relevant boxes as shown.

4.6 Automatic Configuration using TFTP

The EN2060-DC1 Amplifier may be set up to automatically poll configuration settings for SIP and SNMP from a TFTP server. The IP address of this TFTP server can be obtained using DHCP procedures or be manually configured.

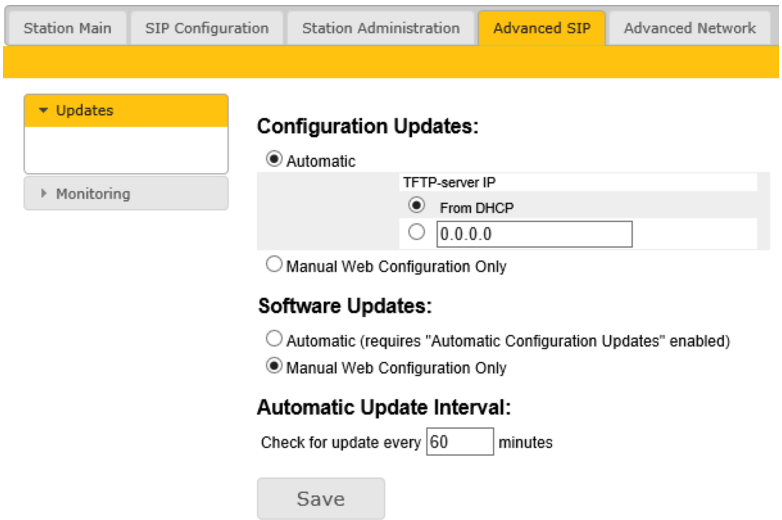
Before you start the automatic configuration procedure:

- Create a **configuration file**

An example configuration file can be found in section *6 Example Configuration File*.

To implement automatic configuration from the amplifier web interface:

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 amplifier web interface
3. Select **Advanced SIP** > **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**

The amplifier will then contact the TFTP server and automatically run the configuration file to carry out the configuration procedure according to the set time interval.

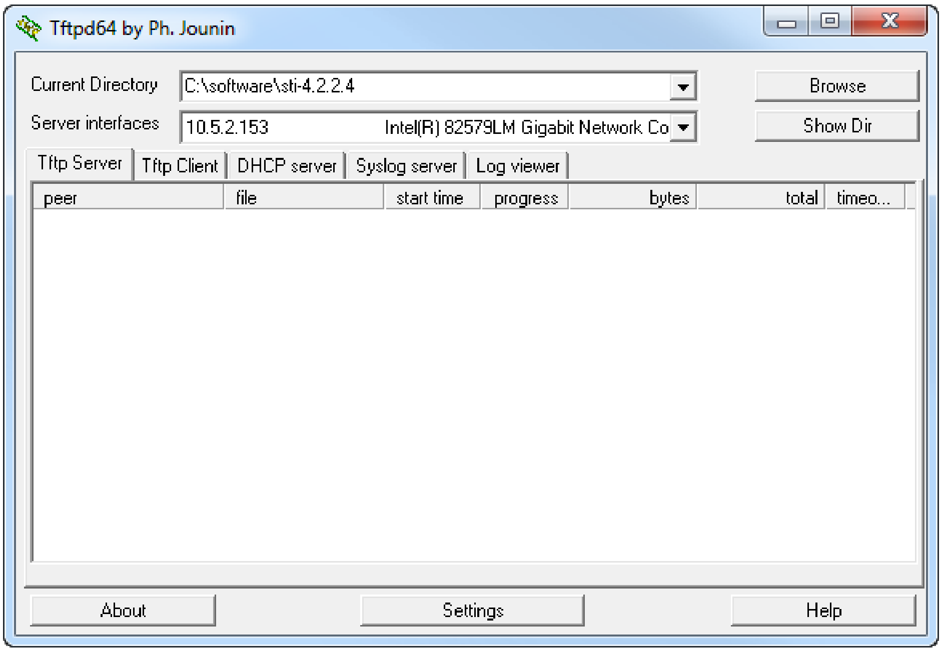
5 Software Upgrade

Software upgrade is accomplished by uploading the latest software via the web interface of the amplifier. The software upgrade process requires that an TFTP Server is available and that the latest software image files have been downloaded from Zenitel's support website (AlphaWiki). During the upgrade process, the amplifier will connect to the TFTP Server and download the software.

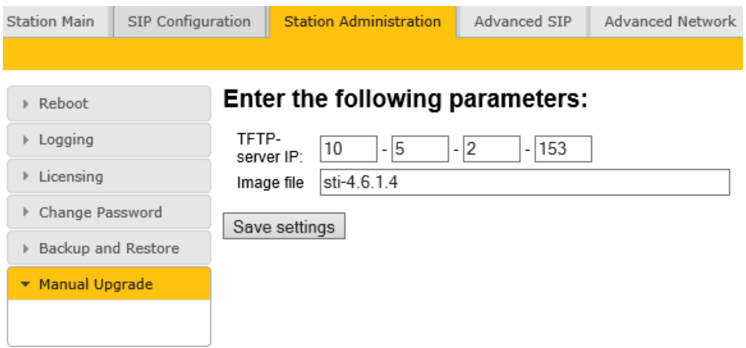
- Install the TFTP Server program on your PC.

Follow the procedure below to upgrade the amplifier software:

1. Start the TFTP server program and click **Browse** to select the folder where the software image files are located



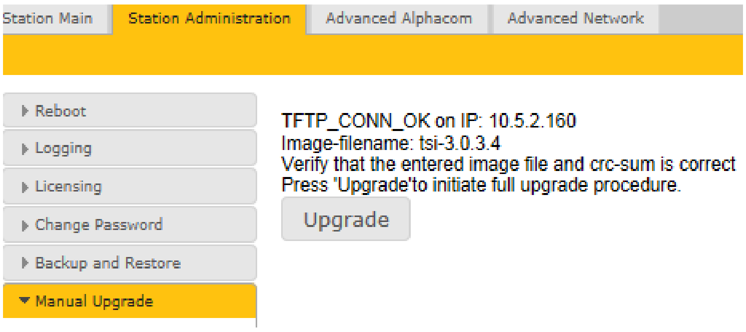
2. Log into the amplifier web interface
3. Select **Station Administration** > **Manual Upgrade**



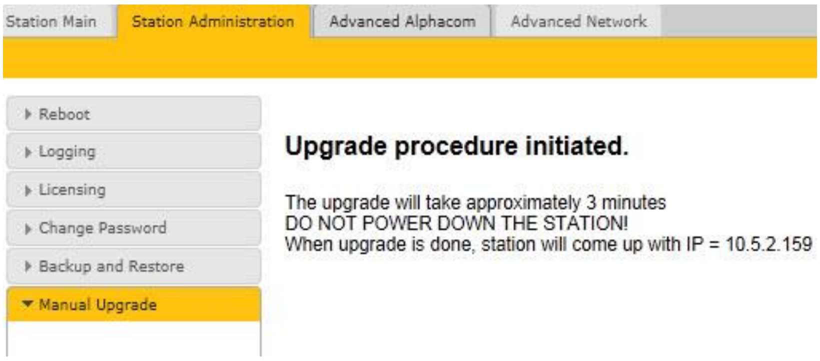
4. Enter the **IP address of the TFTP server** (your PC's IP address)
5. Enter the prefix (e.g. **sti-x.x.x.x**) to the software image files in the Image file field
6. Click **Save settings** to store the data

The amplifier will now try to contact the TFTP server. If the connection cannot be established or the

tftp_test.txt file is missing from the folder, the message **TFTP_CONN_ERROR** is displayed. If the response is **TFTP_CONN_OK** the settings are saved, and the **Upgrade** button will appear.



7. Click the Upgrade button to upgrade the software on the amplifier.



The upgrade procedure takes about 3 minutes. The process can be monitored by clicking the **Log viewer** tab in the TFTP server program.

6 Example Configuration File

```
[sip]
sip_id=0203
sip_domain=10.5.11.75
nick_name=CCP03
auth_user=0203
auth_pwd=Ashley77

[call]
# Use 3 GPI as key matrix for DAK1-7
input_as_key_matrix=3
io_pin1=0
io_pin2=0
io_pin3=0
io_pin4=1
io_pin5=1
io_pin6=1
fast_blink_pattern=1011111
slow_blink_pattern=0000001000000
# handset w/offhook - normally closed
accessory=6
# Allow speech mode to be overriden
override_remote_ptt=1
# use DTMF 9 go to open duplex
open_duplex_dtmf=9
# Allow maximum audio output
poe_audio=1
# Use RFC2833 to send DTMF
dtmf_style=1
# Disable tones
tone_volume=-1
# Use PTT as default speech mode
speech_mode=1
# auto answer enabled
auto_answer_mode=1
# reduced mic sensitivity
mic_sensitivity=4
# onhook send dtmf 8 in call
onhook_in_call_function=2
onhook_dtmf_on=8
# Call 301
dak1_value=0401
dak1_in_call_function=0
dak2_value=401
dak2_in_call_function=0
dak3_value=501
dak3_in_call_function=0
dak4_value=203
dak4_in_call_function=0
dak5_value=510
dak5_in_call_function=0
dak6_value=502
dak6_in_call_function=0

[relays]
gpio3_dtmf_activate=2
gpio3_dtmf_deactivate=0
gpio3_dtmf_flashing_slow=1
gpio4_dtmf_activate=5
gpio4_dtmf_deactivate=3
gpio4_dtmf_flashing_slow=4
gpio5_dtmf_activate=7
gpio5_dtmf_deactivate=6
```

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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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