



Installation, Configuration & Operation

ECP-AA1 Intercom

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1 Introduction

1.1 Document Scope

This document describes the installation, configuration and operation of the ECP-AA1 Intercom.

1.2 Products

| Item Number | Item Name |
|-------------|------------------|
| 1008192011 | ECP-AA1 Intercom |

1.3 Publication Log

| Revision | Date | Author | Status |
|----------|-----------|---------|------------------------|
| 1.0 | 18.5.2016 | HVD/HKL | Published |
| 1.1 | 16.9.2019 | HKL | New doc no. A100K11789 |

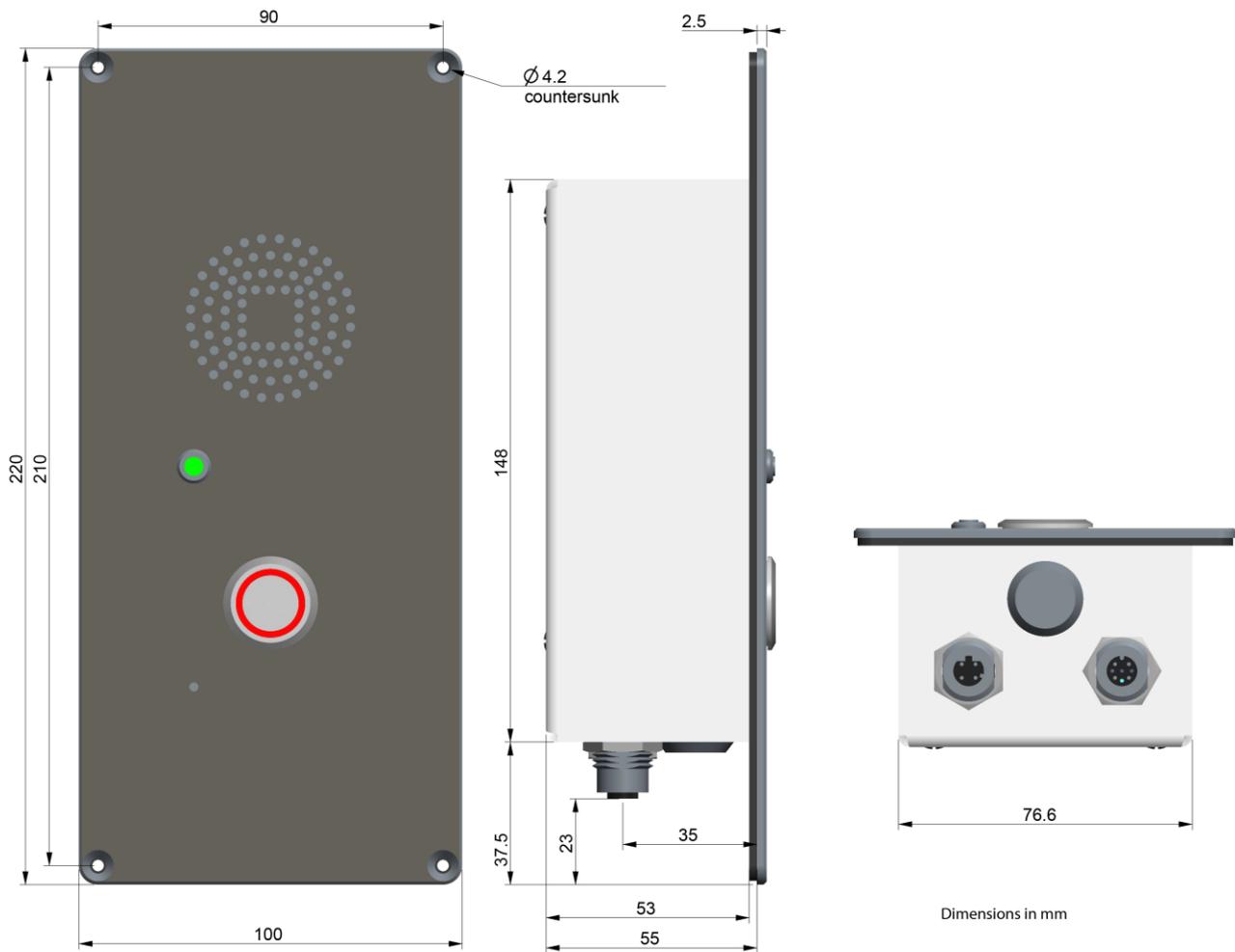
1.4 Related Documentation

| Doc. no. | Documentation |
|------------|---------------------------------------|
| A100K11194 | Turbine IP Intercoms Technical Manual |

2 Installation & Mounting

The ECP-AA1 is flush-mounted using 4 screws of type:

- M4x16 DIN 7991 A4 – Flat Head Socket Cap Screw, Stainless Steel A4 DIN7991
- Length as required



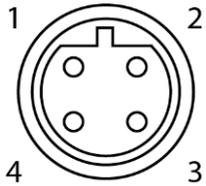
3 Connectors & Power

3.1 Connectors

The ECP-AA1 has 2 connectors located at the bottom of the unit.

3.1.1 Ethernet Connector

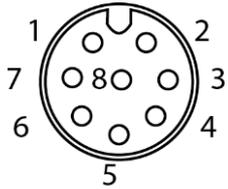
Type: M12, D-coded, Female, 4-pin



| | |
|---|-----|
| 1 | TD+ |
| 2 | RD+ |
| 3 | TD- |
| 4 | RD- |

3.1.2 I/O Connector

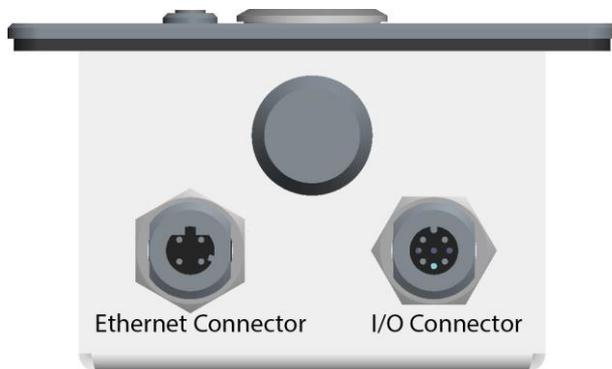
Type: M12, A-coded, Female, 8-pin



| | |
|---|----------------|
| 1 | +5V DC |
| 2 | GND |
| 3 | I/O 1 |
| 4 | I/O 2 |
| 5 | I/O 3 |
| 6 | Relay – Common |
| 7 | Relay – NC |
| 8 | Relay – NO |

3.2 Power

The ECP-AA1 is powered through the Ethernet connector via Power over Ethernet (PoE). The Ethernet connector is located at the bottom of the unit.



3.3 Connection to Emergency Break

The emergency break provides a clean contact to the ECP-AA1. Its purpose is so that an emergency call is made as soon as the emergency break handle is operated.

Procedure

- Connect the clean contact between pins 2 and 3 on the I/O connector.

3.4 Connection to CCTV System

The ECP-AA1 provides a clean contact to the CCTV system. Its purpose is to send a trigger to the CCTV system when a call is made from the ECP-AA1.

Procedure

- Connect the CCTV trigger input between pins 6 and 8 on the I/O connector.

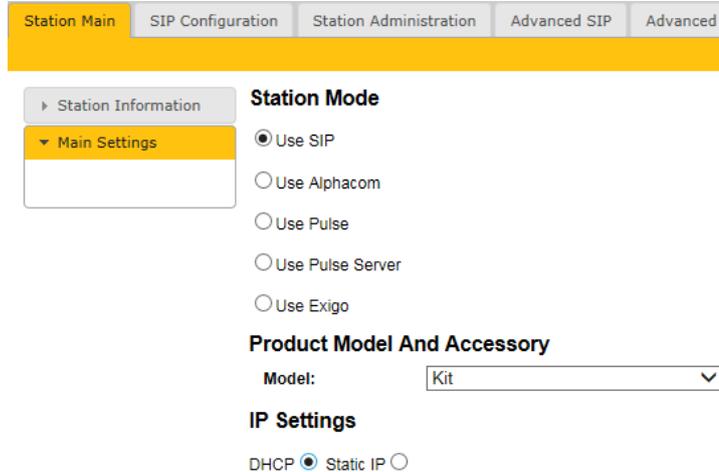
4 Configuration

4.1 Initial Setup

The ECP-AA1 is delivered with factory default settings. By default, the unit's IP settings are configured according to DHCP.

To configure the unit:

1. Press the call button on the unit for it to speak its IP address
2. Ensure that the IP address of your PC is in the same range as that of the station IP address.
3. Access the station by logging into the web interface using a standard web browser on your PC
4. In the browser's address bar, enter the station IP address and press the ENTER key
 - The station login page will be displayed.
5. Click **Login**
6. Enter the default User name: **admin**
7. Enter the default Password: **alphaadmin**
8. Select **Station Main > Main Settings**



Station Main SIP Configuration Station Administration Advanced SIP Advanced I

Station Information

Main Settings

Station Mode

Use SIP

Use Alphacom

Use Pulse

Use Pulse Server

Use Exigo

Product Model And Accessory

Model:

IP Settings

DHCP Static IP

9. Under **Station Mode** select **Use SIP**
10. Under **Product Model And Accessory** select **Kit**
11. Depending on system requirements, set the IP Settings to **DHCP** or **Static IP**. In the latter case, fill in the network details as required
12. Click **Save**
13. On the new page, click **Apply**

The unit will now reboot.

4.2 Audio Settings

Normally no changes are required to be made to the audio settings, but depending on the specific circumstances where the unit is installed, minor adjustments may be needed.

- Select **SIP Configuration > Audio Settings**

The screenshot shows a web interface for configuring audio settings. At the top, there are navigation tabs: Station Main, SIP Configuration (selected), Station Administration, Advanced SIP, and Advanced Network. Below the tabs is a sidebar with a tree view containing: SIP Settings, Audio Settings (selected), Direct Access Key Settings, Relay Settings, Time Settings, I/O Settings, Keyboard Settings, Script Configuration, Script Events, and Script Upload. The main content area is titled 'Audio Settings' and contains a table with two columns: 'Description' and 'Configuration'. The table lists various settings such as Speaker Volume, Volume Override Level, Microphone Sensitivity, Remote Controlled Audio Direction, SIP Message Controlled Audio Direction, Automatic Volume Control, Debug Automatic Volume Control, Conversation Mode, Audio Profile, Noise Reduction Level, Tone Volume, Tftp Server For Audio Files, Outgoing Ringing Audio File, and Incoming Ringing Audio File. Each setting has a corresponding input field (dropdown, checkbox, or text box) and a brief description of its function. A 'Save' button is located at the bottom right of the configuration area.

| Description | Configuration | |
|---|--------------------------|--|
| Speaker Volume: | 0 | |
| Volume Override Level: | 8 | Sets the volume during volume override. |
| Microphone Sensitivity: | 5 | Default value 5 |
| Remote Controlled Audio Direction: | <input type="checkbox"/> | (DTMF * to talk, DTMF # to listen, DTMF 0 for open duplex) |
| SIP Message Controlled Audio Direction: | <input type="checkbox"/> | (SIP MESSAGE controls audio direction) |
| Automatic Volume Control: | <input type="checkbox"/> | Volume depends on noise level |
| Debug Automatic Volume Control: | <input type="checkbox"/> | Shows current volume level on OLED display |
| Conversation Mode: | Full Open Duplex | |
| Audio Profile: | Normal | |
| Noise Reduction Level: | 0 | 0 = disabled. |
| Tone Volume: | 0 | (-1)=disabled, 0=default, [1..4]=[-22..-1]dB |
| Tftp Server For Audio Files: | | Tftp server used for downloading audio files. If empty, the station will use the configuration for TFTP remote provisioning. |
| Outgoing Ringing Audio File: | | Wav file to be played during outgoing calls. The wav file must be 16 kHz, 16 bit, singel channel wav file and be below 1 mb file size. It might take several minutes before the wav file is downloaded from tftp server and applied. |
| Incoming Ringing Audio File: | | Wav file to be played during incoming calls. The wav file must be 16 kHz, 16 bit, singel channel wav file and be below 1 mb file size. It might take several minutes before the wav file is downloaded from tftp server and applied. |

On this page it is possible to set:

- Speaker Volume
- Microphone Sensitivity
- Noise Reduction Level

Furthermore, it is possible to select different algorithms for the **Conversation Mode**.

Note that due to the close proximity of the microphone and speaker it may be necessary to select **Robust Duplex** as Conversation Mode if either the speaker level or the microphone sensitivity is increased from that of the default setting.

4.3 I/O Settings

- Select **SIP Configuration > I/O Settings**

The screenshot shows a web interface with a navigation bar at the top containing 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. Below the navigation bar is a sidebar menu with options: 'SIP Settings', 'Audio Settings', 'Direct Access Key Settings', 'Relay Settings', 'Time Settings', and 'I/O Settings' (which is highlighted). The main content area is titled 'I/O Settings' and contains a table with two columns: 'Description' and 'Configuration'. The table lists six I/O pins, each with a dropdown menu set to 'Input'. A 'Save' button is located at the bottom right of the table. Below the table, a note states: 'Note: Changes requires application reboot'.

| Description | Configuration |
|-------------|---------------|
| I/O Pin 1: | Input 1 ▾ |
| I/O Pin 2: | Input 2 ▾ |
| I/O Pin 3: | Input 3 ▾ |
| I/O Pin 4: | Input 4 ▾ |
| I/O Pin 5: | Input 5 ▾ |
| I/O Pin 6: | Input 6 ▾ |

Save

Note: Changes requires application reboot

Under **Configuration** set the following values for the I/O Pins:

- I/O Pin 1: Input 1
- I/O Pin 2: Input 2 or Output 2 – depending on the application for which this I/O will be used
- I/O Pin 3: Input 3 or Output 3 – depending on the application for which this I/O will be used
- I/O Pin 4: Input 4
- I/O Pin 5: Output 5
- I/O Pin 6: Output 6

- Click **Save**

Although the ECP-AA1 will not use these settings until after a reboot, it is not necessary to reboot at this stage.

4.4 Output Settings

- Select **SIP Configuration > Relay Settings**

4.4.1 CCTV Interface

For **Choose Relay To Configure**:

- Select **Relay 1**

| Description | Configuration |
|-------------------------------------|---------------|
| Remote Digit For Relay On: | - ▾ |
| Remote Digit For Relay Off: | - ▾ |
| Remote Digit For Relay Slow Flash : | - ▾ |
| Remote Digit For Relay Fast Flash: | - ▾ |
| Remote Digit For Relay Toggle: | - ▾ |
| Remote Digit For Timed Relay On: | - ▾ |
| Timed Relay Duration: | 0 seconds. |
| Outgoing Ringing: | On ▾ |
| Incoming Ringing: | - ▾ |
| Outgoing Call: | - ▾ |
| Incoming Call: | - ▾ |
| Group Call (Pulse mode only): | - ▾ |
| Idle: | Off ▾ |
| Error (Not Registered): | - ▾ |

Under **Configuration** set the following values:

- Outgoing Ringing: **On**
- Idle: **Off**
- Click **Save**

4.4.2 General Settings

For **Choose Relay To Configure**:

- Select **Output 5**

The screenshot shows a web-based configuration interface for SIP settings. The top navigation bar includes 'Station Main', 'SIP Configuration' (highlighted), 'Station Administration', 'Advanced SIP', and 'Advanced Network'. On the left, a sidebar menu lists various settings categories, with 'Relay Settings' selected and highlighted in yellow. The main content area is titled 'Relay Settings' and features a dropdown menu for 'Choose Relay To Configure' set to 'Output 5'. Below this is the 'Output 5 Settings' section, which contains a table with two columns: 'Description' and 'Configuration'. The table lists various relay-related settings, most of which are currently set to a default value of '-'. The 'Timed Relay Duration' is set to '0 seconds'. The 'Outgoing Ringing' is set to 'Slow flash', 'Outgoing Call' is set to 'Off', and 'Idle' is set to 'Off'. A 'Save' button is located at the bottom right of the settings table.

| Description | Configuration |
|-------------------------------------|---------------|
| Remote Digit For Relay On: | - |
| Remote Digit For Relay Off: | - |
| Remote Digit For Relay Slow Flash : | - |
| Remote Digit For Relay Fast Flash: | - |
| Remote Digit For Relay Toggle: | - |
| Remote Digit For Timed Relay On: | - |
| Timed Relay Duration: | 0 seconds. |
| Outgoing Ringing: | Slow flash |
| Incoming Ringing: | - |
| Outgoing Call: | Off |
| Incoming Call: | - |
| Group Call (Pulse mode only): | - |
| Idle: | Off |
| Error (Not Registered): | - |

Under **Configuration** set the following values:

- Outgoing Ringing: **Slow flash**
- Outgoing Call: **Off**
- Idle: **Off**

- Click **Save**

For **Choose Relay To Configure**:

- Select **Output 6**

The screenshot shows a web-based configuration interface for SIP settings. The top navigation bar includes 'Station Main', 'SIP Configuration' (highlighted), 'Station Administration', 'Advanced SIP', and 'Advanced Network'. On the left, a sidebar lists various settings categories, with 'Relay Settings' selected and highlighted in yellow. The main content area is titled 'Relay Settings' and features a dropdown menu for 'Choose Relay To Configure' set to 'Output 6'. Below this is the 'Output 6 Settings' section, which contains a table with two columns: 'Description' and 'Configuration'. The table lists various relay-related settings, most of which are set to a default value of '-'. The 'Timed Relay Duration' is set to '0 seconds'. The 'Outgoing Call' is set to 'On' and the 'Idle' setting is set to 'Off'. A 'Save' button is located at the bottom right of the configuration area.

| Description | Configuration |
|-------------------------------------|---------------|
| Remote Digit For Relay On: | - |
| Remote Digit For Relay Off: | - |
| Remote Digit For Relay Slow Flash : | - |
| Remote Digit For Relay Fast Flash: | - |
| Remote Digit For Relay Toggle: | - |
| Remote Digit For Timed Relay On: | - |
| Timed Relay Duration: | 0 seconds. |
| Outgoing Ringing: | - |
| Incoming Ringing: | - |
| Outgoing Call: | On |
| Incoming Call: | - |
| Group Call (Pulse mode only): | - |
| Idle: | Off |
| Error (Not Registered): | - |

Under **Configuration** set the following values:

- Outgoing Call: **On**
- Idle: **Off**
 - Click **Save**

4.5 Call Button Settings

Both the call button on the unit and the emergency break input are set up as a call button.

- Select **SIP Configuration > Direct Access Key Settings**

The screenshot shows a web interface for SIP Configuration. The top navigation bar includes 'Station Main', 'SIP Configuration' (highlighted), 'Station Administration', 'Advanced SIP', and 'Advanced Network'. A left sidebar contains a menu with 'Direct Access Key Settings' selected. The main content area is divided into two sections: 'Direct Access Key Settings' and 'Direct Access Key Settings (In Call)'. The first section contains a table with columns 'Function (idle)' and 'Value', and a 'Save' button. The second section contains a table with columns 'Function (in call)' and 'Activated', and a 'Save' button.

| | Function (idle) | Value |
|---------|-----------------|----------|
| Input 1 | Call To | 12345678 |
| Input 4 | Call To | 12345678 |

Save

| | Function (in call) | Activated |
|---------|--------------------|-----------|
| Input 1 | Do Nothing | |
| Input 4 | Do Nothing | |

Save

Assuming that the call button should call a fixed number:

- Enter the number to be called in the field **Value** for both **Input 1** and **Input 4**
- Make certain that the **Function (in call)** settings for both inputs are set to **Do Nothing**
- Click **Save**

4.5.1 Ringlist Settings

Optionally it is possible to make the ECP-AA1 call numbers in sequence until the call is accepted. If this is the requirement, make the settings as shown below:

The screenshot shows the SIP Configuration interface with the following sections:

- Navigation:** Station Main, SIP Configuration (selected), Station Administration, Advanced SIP, Advanced Network.
- Left Menu:** SIP Settings, Audio Settings, Direct Access Key Settings (selected), Relay Settings, Time Settings, I/O Settings, Keyboard Settings, Script Configuration, Script Events, Script Upload.
- Direct Access Key Settings:**

| Input | Function (idle) | Value | Option |
|---------|-----------------|-------|------------|
| Input 1 | Call To | | Ringlist 1 |
| Input 4 | Call To | | Ringlist 1 |
- Direct Access Key Settings (In Call):**

| Input | Function (in call) | Activated | Deactivated |
|---------|--------------------|-----------|-------------|
| Input 1 | Do Nothing | | |
| Input 4 | Do Nothing | | |
- Ringlist Settings:**

| | Ringlist 1 | With Previous | Ringlist 2 | With Previous | Ringlist 3 | W P |
|----------|------------|--------------------------|------------|--------------------------|------------|-----|
| Value 1 | 12345678 | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 2 | 98765432 | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 3 | 10203040 | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 4 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 5 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 6 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 7 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 8 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 9 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 10 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 11 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 12 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 13 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|
| Value 14 | | <input type="checkbox"/> | | <input type="checkbox"/> | | [|

Call Until Answer (loops the ringlist)
Ringing Time seconds, (0=unlimited)
Max Conversation Time seconds, (0=unlimited)

- Under the parameter **Option** select **Ringlist 1** for both **Input 1** and **Input 4**
- Under **Ringlist Settings** and **Ringlist 1** define up to 14 numbers which will be called in sequence
- Check the **Call Until Answer** box
- In the **Ringing Time** field enter a maximum time in seconds before the next number in the list should be called
- Click **Save**

4.6 SIP Configuration

For the ECP-AA1 to be able to register to a SIP server, the correct SIP settings must first be done.

- Select **SIP Configuration > SIP Settings**

The screenshot shows a web interface for SIP Configuration. At the top, there are tabs for 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. The 'SIP Configuration' tab is active. On the left, there is a sidebar with a tree view containing 'SIP Settings' (expanded), 'Audio Settings', 'Direct Access Key Settings', 'Relay Settings', 'Time Settings', 'I/O Settings', and 'Keyboard Settings'. The main content area is titled 'Account Settings' and contains a table with two columns: 'Description' and 'Configuration'.

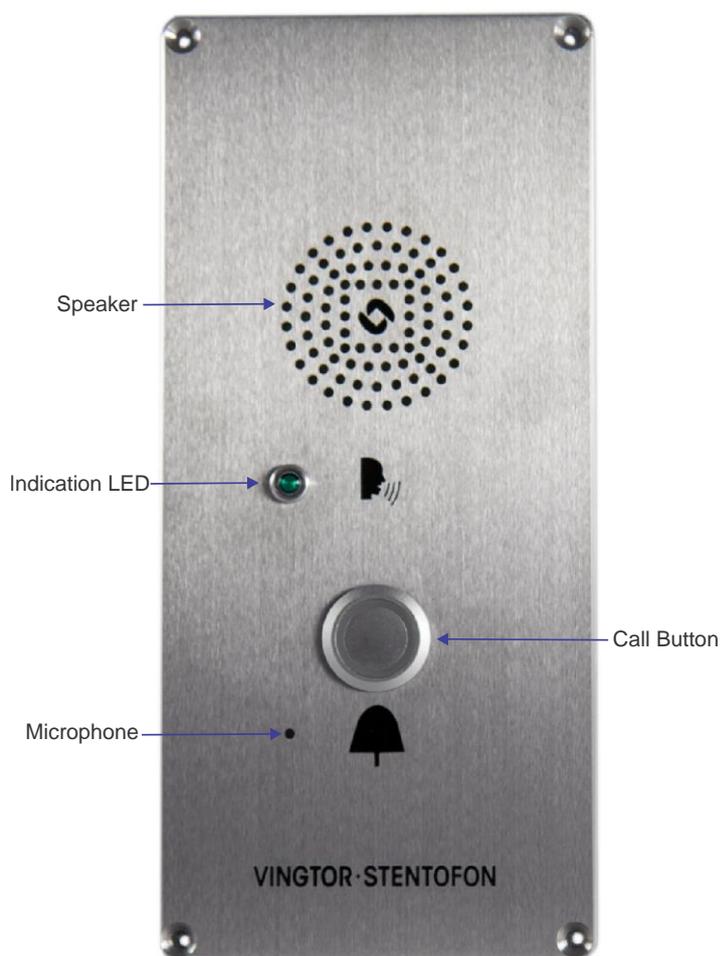
| Description | Configuration |
|----------------------------|---------------|
| Display Name: | ECP1 |
| Directory Number (SIP ID): | 5001 |
| Server Domain (SIP): | 10.5.112.111 |
| Backup Domain (SIP): | |
| Backup Domain 2 (SIP): | |
| Registration Method: | Parallell |
| Authentication User Name: | 5001 |
| Authentication Password: | •••••••• |

- Enter values for : **Display Name, Directory Number (SIP ID), Server Domain (SIP), Authentication User Name**
- Enter values in other fields as may be required by the SIP server
- Click **Save**
- Click **Apply**

The ECP-AA1 will now reboot and all settings that have been made in the previous sections will take effect when the unit is up and running again.

5 Operation

The ECP-AA1 Intercom is a user-friendly Emergency Call Point with a single call button.



To make a call:

- Press the call button



- The red illumination ring around the call button will start flashing
- A ring-back tone will be audible from the speaker

When the operator accepts the call:



- The green LED will be lit, indicating that 2-way communication is now possible
- The ring-back tone will cease
- Speak into the microphone to communicate with the operator

When the operator terminates the call:

- The unit will revert to its idle state



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.