

LHN-UC15L-SIP | LHN-UC15W-SIP | AMN-P15-SIP



en User manual

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1 Safety

1.1 Safety information

- 1. Read and keep these safety instructions. Follow all instructions and heed all warnings.
- 2. Follow all installation instructions and observe the following alert signs:



Notice! Containing additional information. Usually, not observing a "notice" does not result in damage to the equipment or personal injuries.



Caution! The equipment or the property can be damaged, or persons can be injured if the alert is not observed.



Warning! Risk of electric shock.

- 3. System installation and servicing by qualified personnel only, in accordance with applicable local codes. No user-serviceable parts inside.
- 4. System installation for emergency sound (except for call stations and call station extensions) in a Restricted Access Area only. Children may not get access to the system.
- 5. For rack-mounting of system devices, make sure that the equipment rack is of suitable quality to support the weight of the devices. Use caution when moving a rack to avoid injury from tip over.
- 6. The apparatus shall not be exposed to dripping or splashing and no objects filled with liquids, such as vases, shall be placed on the apparatus.



Warning! To reduce the risk of fire and electric shock, do not expose this apparatus to rain or moisture.

- 7. Mains powered equipment shall be connected to a mains power outlet socket with a protective earthing connection. An external, readily operable, mains plug or all-pole mains switch shall be installed.
- 8. Only replace the mains fuse of an apparatus with a fuse of the same type.
- 9. The protective ground connection of an apparatus shall be connected to protective ground before the apparatus is connected to a power supply.
- 10. Amplifier outputs marked with A may carry audio output voltages up to 120 V_{RMS}. Touching uninsulated terminals or wiring may result in an unpleasant sensation. Amplifier outputs marked with A or ⁴ may carry audio output voltages above 120 V_{RMS}. It requires a skilled person to strip and connect the loudspeaker wires in such a way that the naked conductors are inaccessible.
- 11. The system may receive power from multiple mains power outlet sockets and backup batteries.



Warning! To prevent a shock hazard disconnect all power sources prior to system installation.

- 12. Only use recommended batteries and observe polarity. Risk of explosion if an incorrect type of battery is used.
- 13. Fiber optical converters use invisible laser radiation. To prevent injury, avoid eye exposure to the beam.
- 14. Devices for vertical (wall) mounting supporting a user interface for operation shall only be mounted below 2 m height.

- 15. Devices installed above 2 m height may cause injury when falling down. Preventive measures must be taken.
- 16. To prevent hearing damage do not listen at high volume levels for long periods.
- 17. An apparatus may use a lithium coin battery. Keep away from children. If ingested, high risk of chemical burn hazard. Seek medical attention immediately.

1.2 Disposal instructions



Old electrical and electronic appliances.

Electrical or electronic devices that are no longer serviceable must be collected
 separately and sent for environmentally compatible recycling (in accordance with
 the European Waste Electrical and Electronic Equipment Directive).
 To dispose of old electrical or electronic devices, you should use the return and
 collection systems put in place in the country concerned.

1.3 FCC statement - Wired Class B digital device (unintentional radiator)

Changes or modifications not expressly approved by Bosch could void the user's authority to operate the equipment.

Class B digital device: FCC

NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

1.4 FCC suppliers Declaration of Conformity

Product(s):

Material	Commercial code	Material description
F.01U.389.835	LHN-UC15L-SIP	Horn loudspeaker 15W, long throw, SIP
F.01U.389.865	LHN-UC15W-SIP	Horn loudspeaker 15W, wide angle, SIP
F.01U.389.866	AMN-P15-SIP	Amplifier module 15W, SIP

Compliance statement

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions:

- 1. This device may not cause harmful interference, and
- 2. this device must accept any interference received, including interference that may cause undesired operation.

Responsible party

Bosch Security Systems LLC 130 Perinton Parkway 14450 Fairport, NY, USA www.boschsecurity.us

2 Installation

This chapter shows you the steps necessary to install the IP Horn loudspeaker and the IP Amplifier module.

2.1 LHN-UC15L-SIP or LHN-UC15W-SIP IP Horn loudspeaker

2.1.1 Considerations

Before installing the IP Horn loudspeaker, consider the following:

Mounting location:

The IP Horn loudspeaker must be fixed to a stable and suitable surface to support its weight.

- The mounting surface must be able to support more than the nominal weight of a single loudspeaker.
- Choose the mounting location where there is an unobstructed line-of-sight path between the horn and the intended area of coverage.

Required tools

- M13 wrench (fasten torque: 5-6 Nm)
- T20 Torx screwdriver (fasten torque: 1.3-1.8 Nm)

2.1.2 Mounting

The IP Horn loudspeaker comes with a metal bracket attached. The mounting bracket has one central hole and two slotted holes adjacent to it. The central hole has a diameter of 6.8 mm (0.26 in.). The slotted holes are 15 mm (0.59 in.) x 7 mm (0.27 in.).



Figure 2.1: Mounting bracket



Figure 2.2: Mounting bracket hole distances

To mount the IP Horn loudspeaker unit to a wall or any other solid surface, do the following:

1. Using a M13 wrench, remove the side screws and washers that attach the bracket to the IP Horn loudspeaker.

Remove the bracket from the IP Horn loudspeaker.

- 2. Fit the bracket in its final position and secure it with one screw in the central hole.
- 3. Rotate the bracket until you achieve the position you desire.
- 4. Affix the bracket with two screws in the remaining slotted holes.
- 5. Reattach the IP Horn loudspeaker to the bracket, inserting the side screws and washers.

- 6. Make sure the IP Horn loudspeaker is facing the ground in a 10° angle.
- 7. Fasten the side screws to lock the IP Horn loudspeaker to its final position with a torque of 5-6 Nm.

2.1.3 Connections

To access the cable connections, do the following:

- 1. Using a T20 Torx screwdriver, remove the four cover screws.
- 2. Lift the cover from the chassis with care.

The back of the IP Horn loudspeaker shows as the reference view bellow.

Reference view



Figure 2.3: Back panel of the IP Horn loudspeaker (without the cover)

Connecting the network (Figure 2.3, callout 1)

The IP Horn loudspeaker supports PoE/PoE+ power supply mode. You need one Ethernet cable to connect to a PoE switch or injector for a convenient installation. For the most reliable operation, always use shielded CAT-5e or higher class cables.



Notice!

Connect the IP Horn loudspeaker using shielded network cables that are properly grounded.

To connect the network cable:

- 1. Cut an Ethernet cable to the desired length.
- 2. Thread the Ethernet cable through the cable gland on the bottom left of the cover.
- 3. Crimp the RJ45 connector to the cable.
- 4. Connect that end of the cable to the network port (Figure 2.3, callout 1).
- Verify the network connection: The network port LED turns on a few seconds after the network cable is connected, indicating that PoE/PoE+ power has been successfully applied.

There are two lights on the Ethernet jack:

- A solid green LED (right) indicates the port is operating at 1000 Mbps (1 Gbps). It flickers to indicate network activity.
- An orange/yellow LED (left) indicates a network link of 100 Mbps. It flickers to indicate network activity.

Connecting the GPIO (Figure 2.3, callout 2)

Notice!

Connect the IP Horn loudspeaker using shielded cables that are properly grounded.

The GPIO terminal on the device allows you to connect general purpose inputs and outputs.



GPI used as logical input



Figure 2.4: GPIO connection options

- Configure the GPIO behavior via *General purpose inputs and outputs, page 38*. The GPI operates in supervised or non-supervised mode. Through the toggle, you can switch between both modes.
 - Non-supervised mode (GPI used as logical input): Connect the opposite end of the cable to the voltage free relay contact or activation switch. You can specify whether the GPI activates upon making or breaking the contact.
 - Supervised mode (GPI used as supervised input): Use dedicated wiring with resistors to be able to detect open and shorted states. You can specify whether the GPI activates upon making or breaking the contact. In case of wiring shorted or opened, the GPI will keep the last valid state.

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Connecting the analog Audio (Figure 2.3, callout 3)

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Notice!

Connect the IP Horn loudspeaker using shielded cables that are properly grounded. Cables for Line-In / Line-Out should not exceed a length of 3 m (118 in.).

The Line-in allows you to route an external analog audio signal into the device. The Line-out allows you to route an audio signal to an external device.

Both audio connections are analog unbalanced.



Analog audio used as Line-in

Analog audio used as Line-out

Figure 2.5: Audio connection options

To connect the analog Audio:

- 1. Remove the cap on the bottom right of the cover.
- 2. Attach the cable gland (supplied) to the bottom right of the cover.
- 3. Thread the cable through the cable gland.
- 4. Connect the wires to the Audio terminal on the device (Figure 2.3, callout 3).

Reassembling the cover

After you make all the relevant connections to the IP Horn loudspeaker, reassemble the cover.

To reassemble the cover:

- 1. Fit the rubber seal on the cover. Make sure that the seal is properly placed.
- 2. Align the cover with the screw holes, with the cable glands facing downwards.
- Using a T20 Torx screwdriver, tighten the four cover screws with a torque of 1.3 -1.8 Nm.

2.1.4 Microphone

The integrated microphone provides talkback capability, ambient noise level trigger, and automatic volume control.

It is possible to switch on/off the integrated microphone on the IP Horn loudspeaker via the physical switch inside the IP Horn loudspeaker:



Figure 2.6: Integrated microphone physical switch

On/Off switch

Switches the microphone on and off (Figure 2.6, callout 1).



Notice!

By default, the microphone is enabled.

2.1.5 Physical reset button

Reference view



Figure 2.7: Physical reset button and LED

Test/Reset button (Figure 2.7, callout 1)

There is a physical button for test and reset purposes. This button will perform different actions depending on how long you press it:

- 1. Press and hold for 1 5 seconds: the IP Horn loudspeaker will play a default message to test the loop between the speaker and the microphone. The IP Horn loudspeaker performs a check that the speaker is operating and therefore the microphone is used. When the microphone physical switch is On, the tone lasts around 2 seconds; when it is turned Off, you hear a brief beep (approximately ½ second).
- 2. Press and hold for 6 10 seconds: the IP address of the IP Horn loudspeaker will reset to factory default (DHCP);
- 3. Press and hold for 11 20 seconds: the IP Horn loudspeaker will reset to factory default.

Indication LED (Figure 2.7, callout 2)

The LED next to the Test/Reset button serves as a time indicator of how long the Test/Reset button is pressed. Once you click the Test/Reset button once, the LED lights up every five seconds for as long as you keep the button pressed, and once more when you let go of the button to indicate the reset is applied.

The LED next to the test button flashes slowly (1 Hz) if the test is successful and quickly (4 Hz) when it is unsuccessful.

2.2 AMN-P15-SIP IP Amplifier Module

2.2.1 Considerations

Before installing the IP Amplifier module, consider the following:

Required tools

- Slotted screwdriver (0.4 x 2.5 mm) (0.015 x 0.09 in.)
- 4x ≤5 mm (0.19 in.) Allen head screws
- T20 Torx screwdriver (fasten torque: 2.0-2.5 Nm)

2.2.2 Mounting

The IP Amplifier module has two installation options:



Surface mount (A)

To mount the IP Amplifier module to a solid surface:

 Remove the cover by pressing the two latches on the top of the cover (1).

The cover disengages from the IP Amplifier module (2).



2 Pre-drill the marked holes on the chassis of the IP Amplifier module.





Using four screws with less than
 5 mm (0.19 in.) diameter (not supplied), attach the IP Amplifier module to the surface.





4 Align the cover tabs with the clamp holes on the IP Amplifier module.

5 Gently push the cover until you hear a click.



DIN rail mount (B)

To mount the IP Amplifier module in a DIN rail:

1 Using the two TX20 screws (supplied), attach the DIN rail mounting plate to the back of the IP Amplifier module with a torque of 2.0 - 2.5 Nm.





2 After the mounting plate is secured in place, clip the IP Amplifier module to the DIN rail.



NOTE: To remove the IP Amplifier module, push it downward until the bottom slot disengages from the rail.

2.2.3 Connections

Once the IP Amplifier module is mounted, you can proceed making the cable connections. The interior of the IP Amplifier module shows as the reference view bellow.



Figure 2.8: Front panel of the IP Amplifier module

Connecting to the network (Figure 2.8, callout 1)

The IP Amplifier module supports PoE/PoE+ power supply mode. You need one Ethernet cable to connect to a PoE switch for a convenient installation. For the most reliable operation, always use shielded CAT-5e or higher class cables.



Notice!

The IP Amplifier module shall be connected using shielded network cables that are properly grounded.

To connect the network cable:

- Using an Ethernet cable, connect one end of the cable to the PoE/PoE+ port (Figure 2.8, callout 1).
- Connect the other end of the Ethernet cable to the switch or the injector. The network port light turns on a few seconds after the network cable is connected, indicating that PoE/PoE+ power has been successfully applied.

There are two lights on the Ethernet jack:

- A solid green LED (right) indicates the port is operating at 1000 Mbps (1 Gbps). It flickers to indicate network activity.
- A orange/yellow LED (left) indicates a network link of 100 Mbps. It flickers to indicate network activity.

Connecting the GPIO (Figure 2.8, callout 2)

i

Notice!

The IP Amplifier module shall be connected using shielded cables that are properly grounded.

The GPIO terminal on the device allows you to connect general purpose inputs and outputs.



GPI used as logical input

GPI used as supervised input GPO used as direct logical outputs

Figure 2.9: GPIO connection options

- Configure the GPIO behavior via *General purpose inputs and outputs, page 38.* The GPI operates in supervised or non-supervised mode. Through the toggle, you can switch between both modes.
 - Non-supervised mode (GPI used as logical input): Connect the opposite end of the cable to the voltage free relay contact or activation switch. You can specify whether the GPI activates upon making or breaking the contact.
 - Supervised mode (GPI used as supervised input): Use dedicated wiring with resistors to be able to detect open and shorted states. You can specify whether the GPI activates upon making or breaking the contact. In case of wiring shorted or opened, the GPI will keep the last valid state.

Connecting the analog Audio (Figure 2.8, callout 3)

Notice!

The IP Amplifier module shall be connected using shielded cables that are properly grounded. Cables for Line-In / Line-Out should not exceed a length of 3 m (118 in.).

The Line-in allows you to route an external analog audio signal into the device. The Line-out allows you to route an audio signal to an external device. Both audio connections are analog unbalanced.



Analog audio used as Line-in

Analog audio used as Line-out

Figure 2.10: Audio connection options

To connect the analog Audio:

- Connect the wires to the Audio terminal on the device (Figure 2.8, callout 3).

Connecting the passive loudspeaker (Figure 2.8, callout 4)

The IP Amplifier module allows you to connect passive loudspeakers from an impedance of 4 Ohms to an impedance of 8 Ohms.



Passive loudspeaker basic connection

Figure 2.11: Passive loudspeaker connection option



2.2.4 Physical reset button

Reference view

Figure 2.12: Test/Reset button and LED

Test/Reset button (Figure 2.12, callout 1)

There is a physical button for test and reset purposes (Figure 2.12, callout 1). Use a paper clip or similar object to press and hold the Test/Reset button. This button will perform different actions depending on how long you press it:

- 1. Press and hold for 6 10 seconds: the IP address of the IP Amplifier module will reset to factory default;
- 2. Press and hold for 11 20 seconds: the IP Amplifier module will reset to factory default.

Indication LED (Figure 2.12, callout 2)

The LED (Figure 2.12, callout 2) next to the Test/Reset button serves as a time indicator of how long the Test/Reset button is pressed. Once you click the Test/Reset button once, the LED lights up every five seconds for as long as you keep the button pressed, and once more when you let go of the button to indicate the reset is applied.

3 Getting started

The configuration of the IP Horn loudspeaker and the IP Amplifier module is done via WEB based GUI.

Before starting the configuration and operation of the IP Horn loudspeaker and the IP Amplifier module, it is advised to do the following:

- 1. Download and update firmware, page 21
- 2. IP address detection, page 21
- 3. Logon with the web browser, page 22

3.1 Download and update firmware

You perform firmware updates via the Firmware Upload Tool (FWUT) version 9.10 or above. You can get the latest firmware and required Firmware Upload Tool (FWUT) from the product page on <u>www.boschsecurity.com</u>.

The firmware is available to you as a digitally signed installer. After that, the firmware file is placed centrally and shows up in the FWUT.

To update the firmware of the device, do the following:

- 1. Open the **OMNEO Firmware Upload Tool**.
- From the File menu, select **Options**.
 Make sure the checkbox **Use secure connection** is unchecked.
- 3. Click the **Change** button.
- 4. Click **OK**, again.
- 5. On the tab **Devices**, select the device to update.
- 6. Click the **Upload** button.
- 7. From the list of firmware, select the firmware to download.
- 8. Click the **Start** button.

When the upload is complete, the **State** tab displays **Finished** and your device is ready to use.



Notice!

Do not disconnect the device while the update is running. If the update fails, disconnect the device and connect it again. Repeat the update process.

3.2 IP address detection

You need the IP address of the device to be able to do its configuration via a web browser. There are different ways to obtain the IP address.

Below are two options on how you can get the IP address of the device.

With the Bosch Configuration Manager

 You can quickly discover the IP address of your device using the Bosch Configuration Manager. You can download this software from <u>https://</u> downloadstore.boschsecurity.com.

To discover the IP address using Bosch Configuration Manager:

- 1. Connect your device to the same network of your configuration PC.
- 2. In the navigation bar, select the Network Scan tab. The Network Scan screen appears.
 - Every network device that is supported and detected on the network appears in the screen.

The information about a device is shown in bold if the device is newly detected since the last network scan.

The information about a device is shown in red if the device has an IP address or a MAC address that is already used by another device in the system. This might be the case, for example, if several devices that have not yet been configured are connected directly after one another.

With the Firmware Upload Tool

Another form of discovering the IP address of the device is via the Firmware Upload Tool. To discover the IP address of the device using FWUT:

- 1. Connect your device to the same network of your configuration PC.
- 2. Launch the FWUT.
 - The FWUT discovers the device and its IP address appears in the IP address tab.

(i)

Notice!

Make sure the IP address column shows up in the FWUT window. To add the IP address column on the FWUT go to **View, Select columns...** and check the box of IP address.

3.3 Logon with the web browser

The device is configurable via a web browser. Launch your web browser and type the string "https://" followed by the hostname printed on the label or the IP address, e.g. "https://169.254.100.180".

To log on with the web browser, do the following:

 Open a browser and enter the IP address (https://[ip address]) or host name (https:// [hostname]) of the device. Accept the risk for the self-signed certificate.

If you access the device for the first time, you will get a register setup screen. This requires you to register the administrative account for the device:

- Enter a unique username. The username must be 4 64 characters long.
- Enter a unique strong password for the user. The password must be 8 64 characters long.
- Choose the language of the interface. Later you can change your preference in *Generic settings, page 24*.
- Register the administrative account.
- 2. Enter your username and password.
- 3. Login to the web interface.

You can also access the web-GUI of the device via the Bosch Configuration Manager. To do so:

- 1. On the navigation bar, click the **Network Scan** button.
- 2. Right-click the selected device, then click **Show in Web Browser...**. The web-GUI of the device opens in your default browser.

There is no default password; it is required to set a user and a password to logon to the device the first time during the register account setup.

Supported browsers

- Chrome
- Safari
- Firefox
- Microsoft Edge.

Notice!

Avoid using the same password for multiple devices. Setting multiple devices with the same password poses a high security risk.

Password creation tips

- Contain a combination of uppercase letters, lowercase letters, special characters and digits.
- Avoid trivial character strings (e.g., 1234567890, ABCDEFGHIJ, QWERT).
- Avoid manufacturer-related names, general names, brand names, words from a dictionary or word combinations, including foreign words.
- Avoid public or easily identifiable personal information (e.g., telephone number, date of birth, ID number).
- Any combinations with incremental counters (e.g., month, year).

4 Web-GUI configuration

4.1 Overview

Use the Overview page to view four functions (Users, SIP, Messages and Rules) of the web interface and to navigate to those functions in the system using the informative graphical tiles. Each tile displays configuration data and real-time status information of the device. Navigate directly to these functions by clicking on the tile. To access all the functions of the device, use the left navigation menu.

Identify device button

Use the Identify device button to play a default message with a priority of 1 (further information on priority in *Rules, page 31*), in order to identify the device.

Left Navigation Overview		
Users, page 25	To manage user account and permissions.	
SIP, page 26	To manage SIP settings and accounts.	
Audio, page 28	To configure and manage audio settings.	
Messages, page 31	To manage and configure messages.	
Rules, page 31	To manage and configure rules.	
Certificates, page 35	To manage certifications for the device.	
Security, page 36	To manage security for the device.	
Maintenance, page 37	To manage and maintain the device.	
Logging, page 39	To view and manage logging events.	
Logout, page 39	To log out of the web application.	

4.2 Generic settings

The Generic settings page contains information about name settings, time settings, network settings, ONVIF interface settings and user interface settings.

4.2.1 Device settings

- Change the device name.
- Select the time zone.
- Define an NTP (Network Time Protocol) server address (where to obtain clock synchronization). The NTP address can be an IP address or domain name.
 To add an IP address for NTP synchronization, input the desired address and press the Enter key. The device will attempt to establish a connection with the specified NTP server. If the connection is successfully established, the device internal clock will automatically synchronize with the NTP server.



Notice!

In many cases, IP devices such as routers or switches provide NTP by DHCP. Changing the NTP Address overrides the NTP given by the DHCP. After the NTP IP is changed, the system tries to retrieve the latest time from the WebUI. The sync occurs every 15 minutes.

4.2.2	 Network settings Define the IP and DNS settings of the device. Default settings configured by DHCP. 		
	 Option to set static DNS and IP address. 		
í	Notice! In order to apply all changes performed in this menu, you need to reboot the device.		
4.2.3	 ONVIF Interface settings Change ports (HTTP, HTTPS, RTSP and UDP) used for the ONVIF interface. Enable/disable device support for G.711 audio codec used for ONVIF audio streaming (Talk-down). The device supports AAC and G.711 audio codec, where AAC has the preference since higher audio quality resulting in higher speech intelligibility. Therefore the G.711 audio codec is by default disabled. In case a connected VMS is not supporting the AAC codec the G.711 audio codec can be enabled assuming this is supported by the VMS. 		
í	Notice! Ports 80 and 443, commonly associated with web interfaces, are unavailable as they are already assigned to the current web interface.		
í	Notice! Press the save button to apply any changes that were made.		
4.2.4	 User interface settings Select Dark mode for UI appearance. Language dropdown menu for UI language selection. Activate or deactivate the Advanced view for access to advanced audio settings. 		
4.3	Users		
	The User page displays a list of current users in the system. From this page, you can - Create a user account - Modify a user account - Delete a user account		
	 User accounts include a username (4 - 64 characters), a password (8 - 64 characters), and a user group. There are three user groups available: API: grants read/write access to other devices via http or https. Create login credentials for accessing things like the virtual GPIOs using this group. The user can also access further services. The API is described on https://<ip_address>/build/api/index.html.</ip_address> 		

- ADMIN: grants access to configuration and control options in the web interface.

 ONVIF operator: serves the purpose of facilitating the seamless connection between video management systems and other applications utilizing the ONVIF protocol to interface with the device. Even in the absence of a user with the ONVIF operator group, the ONVIF interface remains accessible. However, due to the lack of authentication, the interface is effectively inactive.

When you add a user account, the device's ONVIF processes will restart. This may cause a short delay of a few seconds before the Users page updates.

í	Notice! Once the user account is created, you can only modify the password. It is not possible to change the user group. If the user needs to have a different user group, delete the account and create a new profile with the new user group assignment.
4.4	SIP
	Use the SIP page to configure the system for SIP operations. By default, the SIP is active using a pre-configured P2P SIP account.
	From this page you can access: - SIP settings - SIP accounts
4.4.1	SIP settings Use the SIP settings page to view, modify, enable and disable SIP service settings.
	SIP service enabled checkbox The SIP service enabled checkbox determines whether the SIP client should be running or not. The type of transport protocols the SIP client is allowed to use is set in the dropdown menu upon account creation. By default, SIP supports TCP, UDP and TLS protocols.
	Server certificate Server certificates are digital authorizations that allow secure transmissions between the SIP server and the IP Horn loudspeaker/IP Amplifier module. When you select the Verify server certification checkbox, the device verifies the SIP server is authorized to transmit and receive audio and data by checking the digital certificate. Select the proper certificate from the Certificate for this device drop down menu.

Manage the available certificates on the Certificates, page 35.

Advanced SIP configuration settings

Further configuration settings can be adapted to match the SIP server requirements.



Notice!

The device is unable to establish a direct connection with an online SIP proxy. Use SIP trunking to connect to the public telephone network via a SIP provider.

4.4.2 SIP accounts

Use the SIP accounts page to provide information on the current existing accounts for this device. From this page, you can enable and disable existing SIP accounts and add, modify, or delete accounts.

You can create two type of accounts:

- P2P account applicable for direct SIP phone to SIP device (softphone, USB phone, SIP-based telephone hardware, ATA phone adaptors, etc.) communication. To set up a P2P account, two parameters are mandatory: Username (Label) and the Transport protocol.
- Registrar account applicable if the device connects to a SIP server.
 To set up a Registrar account:
 - The Username and Password need to match to the SIP server's dedicated account for the device.
 - You must select the Transport protocol.
 - You must assign an IP address of the SIP server in the Registrar.

In-house / on premise



Figure 4.1: Connection to IP-PBX system



Notice!

For a correct functionality, every SIP account requires at least one rule referencing the account. For more information, see *Rules, page 31*.

4.5

Audio

Use the Audio page to adjust and configure audio preferences. Different tabs are shown depending on the type of device connected (IP Horn loudspeaker or IP Amplifier module). There are basic audio settings and advanced audio settings.

By default, the device comes with the Advanced view disabled. Only the basic audio settings are displayed unless you enable the Advance view.

Basic audio settings:

- Audio
- User Equalizer
- Microphone Equalizer
- Advanced audio settings:
- Noise gate
- Speaker equalization
- ANC (Ambient Noise Control for automatic volume adaption)
- Sine generator



Figure 4.2: DSP signal flow diagram for IP Horn loudspeaker





To display the advanced audio settings, enable the Advanced view on *Maintenance, page* 37.

4.5.1 Basic audio settings

Audio page

There are different audio settings available according to the type of device you are configuring.

User level	The User level field controls the amount of gain, in dB, for the loudspeaker (IP Horn loudspeaker) or for the amplifier output (IP Amplifier module).
Line out level	The Line out level field controls the amount of gain, in dB, of the analog line output.
Mic talkback level (IP Horn loudspeaker only)	The Mic talkback level field controls the gain, in dB, of the integrated microphone.
User delay	The User delay field allows you to delay the loudspeaker output (IP Horn loudspeaker) or the amplifier output (IP Amplifier module) up to 170 ms. For more information, see the DSP signal flow in figure 4.2 and 4.3.
Noise trigger threshold (IP Horn loudspeaker only)	The Noise trigger threshold field sets the noise level threshold before triggering an action. For more information, see <i>Rules, page 31</i> .
Noise trigger RMS time (IP Horn loudspeaker only)	The Noise trigger RMS time field defines the amount of time, in milliseconds, to evaluate a noise before triggering an action. For example, if a specific noise level occurs for a set amount of time, then a message plays. Having a short RMS value triggers an action for a quick noise, such as a clap of thunder. Having a longer RMS value needs to have a longer noise before triggering an action, such as footsteps, conversations, etc.). For more information, refer to <i>Rules</i> , <i>page 31</i> .

User equalizer page

The User equalizer page allows you to configure the Type, Gain, Frequency, Quality, and Bypass of the output from the IP Horn loudspeaker or the IP Amplifier module. Use the 3-band PEQ (parametric equalizer) to set parameters by entering values in the specific fields, using the + or - to increase or decrease the values, or by moving a point in the plot graph.

A Reset to default button, located at the top of the page resets the values to the default settings.

Microphone equalizer page (IP Horn loudspeaker only)

The Microphone equalizer page allows you to configure the Type, Gain, Frequency, Quality, and determine the Bypass status for the talkback microphone audio.

Use the 3-band PEQ (parametric equalizer) to set parameters by entering values in the specific fields, using the + or - to increase or decrease the values, or by moving a point in the plot graph.

A Reset to default button, located at the top of the page resets the values to the default settings.

4.5.2 Advanced audio settings

Advanced audio settings, when active, allow the user to set parameters for noise gate, speaker equalization, ANC (automatic volume control) and to configure sine wave generator for measurement testing.

To display the advanced audio settings, enable Advanced mode on *Maintenance, page* 37.

Noise gate page

The Noise gate page defines the point at which noise suppression activates based on the below parameters:

- configure the audio level of activation (Threshold)
- configure the speed at which the device gets to the activation threshold (Attack)
- configure how long suppression is active (Hold) after the threshold met
- configure the speed at which the gain transitions to silence during release (Release)
- configure the depth of the audio when noise gate is active (Depth)

By default, the noise gate is active with factory settings. If needed, you can modify the parameters you can disable the noise gate by setting the bypass option.

Speaker equalizer page

The Speaker equalizer page allows you to configure the Type, Gain, Frequency, Quality, and Bypass of the output from the IP Horn loudspeaker or the IP Amplifier module.

Use the 6-band PEQ (parametric equalizer) to set parameters by entering values in the specific fields, using the + or - to increase or decrease the values, or by moving a point in the plot graph.

A Reset to default button, located at the top of the page resets the values to the default settings.

Alternatively, it is possible to pick from a set of factory presets made especially for the IP Horn loudspeakers.

ANC page (IP Horn loudspeaker only)

Use the Ambient Noise Control (ANC) page to configure when using an automatic adaption of the volume depending of background noise. The integrated microphone measures the audio level. You can set a threshold that automatically adjusts the volume of the device depending on the amount of ambient noise.

The automatic volume adjustment is done at the moment an action starts and remains the same during a call or message playback. This will affect the build-in loudspeaker volume but not the line-out. For more information, see Figure 4.2: DSP signal flow. This means you can:

- configure the audio level at which adjustments start (Threshold),
- configure the amount of time, in milliseconds, to evaluate a noise before triggering the adjustment (RMS time),
- configure ratio of adjustment increases (Ratio),
- configure the maximum level of gain in adjustments (Gain max),
- activate or deactivate the ANC (factory default) via the Bypass option.

Set these parameters by entering values in the specific fields or by using the + or - to increase or decrease the values.

When the sine generator is activated, the value Calculated gain (dB) indicates the current volume adaption.

- Practical example: Starting at a noise of -40 dBFS, the adaption shall raise the volume equally to the noise getting louder, not more than 20 dB.

- Set Threshold to -40 dBFS.
- Keep Ratio at 1.0.
- Set Gain max (dB) to 20.

Sine generator page

Use the Sine generator page to configure a test signal that tests the signal strength of the device.

Set the Frequency and Gain fields to the desired parameters for the environment.

4.6 Messages

The Messages page displays the current user messages and default factory messages in the system.

The system comes with a set of factory messages. You cannot edit, download, or delete these messages, and they are hidden by default. Click the Show factory messages button, located in the top right corner of the window, to make the factory messages visible. Also located in the top right corner of the messages page is the available storage space field. This field displays the amount of storage left for user generated messages. You can add messages by either uploading a file or pointing to a url (audio streams and audio files delivered by http url, online if the device has a properly configured network). Acceptable file types for new messages include .wav, .mp3, .ogg, and .opus.

To find specific messages quickly, use the search field at the top of the page. Simply enter the name of the message and click the $^{\circ}$ button.

From this page, you can:

- Add new messages
- Play existing messages
- Download existing messages
- Edit existing messages
- Delete messages

4.7 Rules

Use the Rules page to configure triggers that initiate actions within the system. The primary component around which the system should be set up is its set of Rules. Using the + button, you may add new rules. You can also edit or delete existing ones.

There are three sections on the rules page:

- Name and priority
- Trigger settings
- Action settings

There are two default rules: SIP Thru and ONVIF talk-down. The ONVIF talk-down rule is preconfigured to facilitate quick setup with a Video Management System (VMS) for talkdown functionality to the device (ONVIF backchannel). Once you add an ONVIF user and activate this rule with the correct credentials, the device receives audio from the VMS when triggered by the VMS. Depending on project requirements, the preconfigured rule can be edited, or a new rule can be created to specify ONVIF talk-back behavior, such as priority, schedule, and GPO activation. The other preconfigured rule is SIP Thru, which allows easy setup either via the preconfigured P2P account or by editing the rule and selecting another SIP account. Enabling this rule enables SIP audio communication triggered by an external SIP system. It's important to note that the device employs a priority-based verification process before allowing routed audio to be played back.

To find an existing rule quickly, use the search field at the top of the page. Simply enter the name of the rule and click the \smallsetminus button.

4.7.1 Name and priority

Label Field

Use the Label field to enter a unique name for the rule.

Enabled checkbox

Use the Enabled checkbox to activate or deactivate a rule. You can also use the slider button on the main Rules page to activate and deactivate a rule.

Priority field

Use the Priority field to define the importance of the rule. Enter the priority in the field or use the + and - to increase or decrease the value.

The range for this field is 1 (lowest) to 100 (highest).

The rule with the highest priority will start if there are active triggers on both rules. If a rule with a lower priority is already in effect, it is stopped.

4.7.2 Trigger settings

Trigger settings define the event that starts an action. You can initiate actions by a schedule, by ambient noise, by an incoming SIP call, by a physical GPI, or by using a virtual GPI.

Schedule only

Use the Schedule only trigger type when an action is time-controlled. For example, an alarm sounds at the same time every day. You can schedule actions to perform minutely, hourly, daily, weekly, monthly, yearly. Select the option Always to bypass or take out the time-controlled settings.

SIP

Use the SIP trigger type to configure an action to start when a call comes into the system. The most common application of the SIP trigger type is using a SIP call (Action type - Route call). Another possibility for the SIP trigger type is to initiate an action using DTMF (Dual Tone Multi-Frequency).

You can combine this trigger type with a schedule via Schedule type, if applicable. Otherwise, keep the default setting of Always.

There are two additional parameters needed when using the SIP trigger type:

- You must select the appropriate SIP account. For more information on SIP account configuration, see the SIP page.
- You must select the appropriate DTMF code, if applicable.

GPI

Use the GPI trigger type to initiate an action via a physical GPI (General Purpose Input). The device offers one physical GPI. For information on installing the GPI, see *Installation, page 8*. You can combine this trigger type with a schedule via Schedule type, if applicable. Otherwise, keep the default setting of Always.

Configure the physical GPI on Maintenance, page 37, using the GPI active range parameter.

VGPI (Virtual GPI)

Use the VGPI trigger type to initiate an action via a virtual GPI. The device offers 32 virtual GPIs. You can manually configure these GPIs from the user interface via *Maintenance, page 37*, under Virtual GPIs or by using the API.

You can combine this trigger type with a schedule via Schedule type, if applicable. Otherwise, keep the default setting of Always.

ONVIF output

This trigger type is designed for compatibility with Video Management Systems (VMS). The ONVIF output can be used to directly trigger a rule for different scenarios. The device support up to 32 ONVIF outputs. To use the ONVIF outputs it is essential that the connected VMS is equipped to support them. Refer to the VMS documentation to learn whether and how this support is supported.

ONVIF audio

The ONVIF audio type triggers the rule when the device receives an ONVIF backchannel audio stream. This usually happens in a talk-down scenario, commonly initiated by a Video Management System (VMS). Once the rule is activated, the live audio from the VMS becomes audible as long as the rule remains active.

Noise trigger (IP Horn loudspeakers only)

Use the Noise trigger type when an action starts after an ambient noise reaches a specific level or the noise level exceeds a specified limit.

You can combine this trigger type with a schedule via Schedule type, if applicable. Otherwise, keep the default setting of Always.

There are two noise parameters necessary for operation. You can find these parameters on the Audio page:

- Noise trigger threshold
- Noise trigger RMS time

"Trigger-end stops the action instantly" checkbox

If enabled, the action will stop immediately the trigger finishes. If disabled, it allows an action to run its course completely depending on the settings, even when the trigger is no longer present. For instance, a message plays to the end or for a specified number of times as determined by the repeat count setting, as illustrated in Figure 4.4.



Notice!

The scheduled types only work properly if the device is synchronized with an NTP server.

4.7.3 Action settings

Triggers start actions, such as playing a message, testing the device, accepting and routing an incoming SIP call, or configuring the predefined audio routes.

Start message

To define the message played when the rule is activated, use the Start message feature. This can be a custom uploaded or default factory message, or audio streams/files accessed via an HTTP URL. For more information about messages, see *Messages, page 31*. The Gain (dB) setting allows you to adjust the volume for each message, either increasing or decreasing it if necessary. A value of 0 dB implies no adjustment to the general audio settings, and this is the default setting.

The Repeat count option becomes available when the "Trigger-end stops the action instantly" setting is disabled. It allows you to play the message multiple times based on a single trigger activation. The repeat count value determines how many times the message will play (message cycle), assuming the trigger is a short pulse or shorter than the length of the message cycle playback. For instance, if the repeat count is set to 2 and the trigger condition is only present for a short period (pulse), the message plays twice. If the trigger condition is still present after the message cycle finishes, it repeats itself. This cycle continues until the trigger condition is no longer present. Refer to Figure 4.5 for visual representation. A zero value repeats the message indefinitely.

Trigger-ends stops the action instantly OFF Repeat count: 1	Trigger-ends stops the action instantly OFF Repeat count: 1
(V)GPI	(V)GPI
Message Play	Message Play
্যা Message	্ব্য) ব্য) ব্য) Message Message Message
Trigger-ends stops the action instantly OFF Repeat count: 2	Trigger-ends stops the action instantly OFF Repeat count: 2
(V)GPI	(V)GPI
Message Play	Message Play
්ා ් Message Message	্বাগ বিগ বিগ বিগ Message Message Message
Figure 4.4: Message play behavior with	h "Trigger-end stops the action instantly" checkbox disabled
Trigger - ends stops the action instantly ON	Trigger - ends stops the action instantly ON
(V)GPI	(V)GPI
Message Play	Message Play
طائ الله المعالم المعال Message	ব্য ব্য প্র Message Message

Figure 4.5: Message play behavior with "Trigger-end stops the action instantly" checkbox enabled

Device test

Use the Device test action to define a test to see if the device is performing correctly. Select the type of device test to run when a trigger initiates.

NOTE: The Loopback horn mic is only applicable when using the IP Horn loudspeaker.

Notice!

Device tests run to completion no matter what state the "Trigger-end stops the action instantly" checkbox shows.

The results of the device test can be provided via the API. The result can be read out via <ip>/api/ext/v1/status - value "speakertest" (TRUE = passed, FALSE = failed).

Route call

Use the Route call action type for the device to accept and route an incoming SIP call (this setting works only if the trigger is a SIP call). You can define the call routing to be full duplex (audio goes both ways) or speaker only (audio goes one-way).

Full duplex audio mode (IP Horn loudspeaker only) utilizes the integrated microphone of the device establishing a two-way audio connection with the SIP caller. This means that a person standing near the device can converse with the person at the other end of the SIP call.

Speaker only audio mode means the SIP call is the only audio heard over the device. Speaker only is a one-way call/announcement to the device.



Notice!

There is no possibility to initiate a SIP call from the device to another SIP device.

Set routing

Use the Set routing action type to configure the device's audio output. There are two audio output channels: line out and speaker out. You can configure each channel independently with its own input source which gives greater flexibility in routing audio.

Available input source: no audio (silence), audio line input, and sine generator. If using the sine generator, you may adjust the sine signal in the Audio page.

SIP calls and playing messages are automatically routed to both line out and speaker out.

GPO Action

In addition to a running action, you can configure an additional physical GPO or virtual GPO to occur. For example, every time a live call or message is transmitted, a GPO is used to activate a separate light (an optical signal).

The device has 1 physical GPO which may be connected using a separate cable and 16 virtual GPOs, which can been seen on *Maintenance, page 37*, under API interface.

4.8 Certificates

Use the Certificates page to create and manage the certificates the device uses for secure transmissions within the system. Certificates are digital authorizations that allow devices to communicate with each other over secure communication lines.

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Notice!

Certificates must be Base64-encoded X.509.

There are two types of certificates available:

Server client certificate

The Server client certificate allows secure data transmission between the IP Horn loudspeaker/IP Amplifier module and the PC or mobile device browser. Selection of this certificate is done from the *Security, page 36*.

To add this certificate, select the Server client certification; enter a public key and upload the certificate file via the select file button. Add the private key file via the Select private key file button.

CA cert

The CA certificate allows secure SIP communication (transport protocol TLS) between a SIP server and the IP Horn loudspeaker/IP Amplifier module, including authorization. This certificate has to be identical to the one on the SIP server. For more information, see the SIP page.

To add this certificate, select CA cert and upload the certificate file via the select file button.

To find specific certificates quickly, use the search field at the top of the page. Simply enter the name of the certificate and click the \bigcirc button.

4.9 Security

Use the connection settings for the secure transmission of data between the IP Horn loudspeaker/IP Amplifier module and a browser on a PC or mobile device. Select the connection settings for this web interface, the REST API and ONVIF. To provide higher security levels, this page allows you to select specific ciphers suites.

Connection policy

By default, the connection policy is set to HTTPS (a secure data transmission policy). If you prefer using an unsecure transmission, select one of the other options available: HTTP or HTTP and HTTPS.



Notice!

The Connection policy affects both User and ONVIF interface.

HTTPS certificate

The HTTPS certificate allows you to select a specific certificate to increase the security of data transmission. To select from the certificates available, first generate them in *Certificates, page 35*.

Firmware update

The firmware update toggle allows you choose between updating or maintaining the current firmware state. It is disabled by default.

Click Save to save and apply your preferences.
Notice!

This can lead to a restart of the User interface

4.10 Maintenance

Use the Maintenance page to access detailed device settings. There is an overview of the product and various options for the device configuration, including:

- Device tests
- User interface settings
- General purpose inputs and outputs
- Open Source info and support

At the top of the Maintenance page are four buttons with different functions:

Project save

The Project save button allows you to save a file with your current project settings (project.yapf file) into your local downloads folder. This will save all your current configurations including all the messages. All your credentials, even the ones for SIP, and also the certificates are encrypted for additional security.

Project load

The Project load button allows you to upload a file with existing project settings (.yapf file) from your local PC.

Here's a quick overview of compatibility when saving and loading project files across various firmware versions.

		Load		
Save	FW 0.3.4	FW 1.0.233	FW 2.0.800	
FW 0.3.4	✓	Х	Х	
FW 1.0.233	Х	1	✓	
FW 2.0.800	Х	Х	✓	



Notice!

Uploading a project file overwrites your current settings and requests a new login.

Reboot device

The Reboot device button allows you to restart the device. This action does not affect any of the current settings.

Factory reset

The Factory reset button allows you to return all settings to the factory default settings and values.

4.10.1 Device tests

The Device tests field allows you to perform three different types of tests to your device:

- Power supply tests the actual power supply consumption
- Ethernet tests the speed of the Ethernet
- Loopback (IP Horn loudspeaker only) tests the speaker and the microphone (IP Horn loudspeaker only)

To run any of these tests, click the button Start test. The table displays the results.

Notice!

You can only run the tests if all device rules are inactive.

4.10.2 General purpose inputs and outputs

Use the General purpose inputs and outputs page to

- Verify the microphone's state. You cannot change the microphone settings from this page. Turning the microphone on or off is done from the physical switch on the device. For more information, see *Microphone, page 12*.
- Control and indication of GPI and VGPI status.
- Indication of GPO status.

Use the Hardware interface settings see the status of GPIs and GPOs.

- Configure the GPI behavior depending on the external wiring. You can specify whether the GPI activates upon making or breaking the contact. The GPI operates in supervised and non-supervised mode, selectable by the toggle. For more information, see IP Horn loudspeaker *Connections, page 9* or IP Amplifier module *Connections, page 17*.
 - Non-supervised mode (GPI used as logical input).
 - Supervised mode (GPI used as supervised input).
 A blue tag in the interface column shows the current voltage of the GPI.
 A yellow tag appears in the interface column if the device detects that the connection to the GPI is open/shorted

NOTE: In supervised mode, you can check remotely if the line is shorten or open.

- Enable/disable the GPO by toggling the GPO button.

Virtual GPIs

The Virtual GPIs page allows you to verify the status and enable/disable the 32 virtual GPI triggers.

Virtual GPOs

The Virtual GPOs page allows you to verify which of the 16 virtual GPOs are currently triggered.

ONVIF outputs (visible with Advanced view enabled)

The ONVIF outputs page displays the present configuration of ONVIF outputs as established and configured by ONVIF clients or Video Management Systems (VMSs), reflecting the parameters and settings determined by these external entities rather than the current operational state.

4.10.3 Open Source Info & Support

The product makes use of open source software. You can download the open source licenses by clicking the corresponding button.

4.11 Logging

Use the Logging page to access the list of events logged by the device. The logs appear with the most recent logging event at the top of the list.

Logging is an important tool for debugging and monitoring your device. You can view the type/level of the event, the time and date of the occurrence and the message displayed. To find a specific log entry quickly, use the search field at the top of the page. Simply enter the log entry and click the \bigcirc button.

Users can download the comprehensive logging list for in-depth analysis and troubleshooting.



Notice!

Notice!

A time control either via NTP by DHCP or via a separate NTP server is required to properly use the Logging function. Otherwise the logs may display random timestamps.

4.12 Logout

Click the Logout button to log out of the current session.

Inactive sessions will be logged off after one hour of inactivity.

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5 HTTPS REST API

In a system with many devices, an operator may want to centrally supervise all devices and start actions. A quick and easy way to do this is to share system states over an HTTPS API. An HTTPS API communicates between the devices in a system via the Hypertext Transport Protocol. This provides an easy way to collect and process requests.

There is a complete list of states and services available via the HTTPS API documented on https://<IP_Address>/api/index.html.

The HTTPS API provides states for:

- Getting and setting virtual GPIOs;
- Getting systems states;
- Getting and setting gains.
- Retrieving the log file.



Notice!

To gain access to the API, you need to authenticate the according program with the user credentials of an API user.

The IP Horn loudspeaker/IP Amplifier module provides status information to be read using the GET command.

- 16 virtual GPOs to display the status information for VGPO 4 use the command: GET https://<IP_Address>/api/ext/v1/vgpos/4
- 16 virtual GPIs to display the status information for VGPI 16 use the command: GET https://<IP_Address>/api/ext/v1/vgpis/16
- An overview of system states (BOOLEAN), such as message playing, power states, memory usage, etc, use the command: GET https://<IP_Address>/api/ext/v1/status

The IP Horn loudspeaker/IP Amplifier module provides the following information to be written (e.g. via PUSH command):

16 virtual GPIs - use the command:
 PUT TRUE https://<IP_Address>/api/ext/v1/vgpis/16

Notes:

- You must have a configured API user profile for access to the APIs. For more information, see the Users page.
- Set the physical GPO to follow any actions defined on the Rules page.

You can obtain more information on GPIs, GPOs, and further parameters in detail on https://<IP_Address>/api/index.html, where you can also test these functionalities.

Direct Bosch camera integration via ATSL

Using the camera's Alarm Task Scripting Language (ATSL) you can create a direct integration between a Bosch camera and an IP Horn loudspeaker or IP amplifier module. The VGPI of the IP Horn loudspeaker/Amplifier module controls the action, often a message playback, starting or stopping it using this scripting.

To trigger an action on the IP Horn loudspeaker or the IP amplifier module, the scripting can leverage the Alarm output and/or the VCA task alarms of the camera.



Figure 6.1: Direct camera integration principle

To link a camera with the IP Horn loudspeaker/IP Amplifier module, the following steps are necessary:

At the IP Horn loudspeaker/IP Amplifier module:

- Create an API account or use an existing one.
- Create a Rule or use an existing one with VGPI trigger type to start an action, such as an activation of a message.

At the camera:

Add a script at the Alarm Task Editor.



6

Notice!

The username and password for both the Alarm Task Editor Script (camera) and the API user account (IP Horn loudspeaker/IP Amplifier module) must match.

Example Script

In the example, the camera defines two areas that activate two different messages via VGPIs at the IP Horn loudspeaker/IP Amplifier module.

```
HttpCommand sendHttpOn:={
Command("api/ext/v1/vgpis/1")SSL(true)Port(443)IP("10.30.201.236")
Password("pwd12345")UserName("API_username")Method(POST)ForceBasicAuth(true)
ContentType("application/json")
Payload("true")
Name("Http Command 1")
};
HttpCommand sendHttpOff:={
Command("api/ext/v1/vgpis/[")SSL(true)Port(443)IP("10.30.201.236")
Password("pwd12345")UserName("API_username")Method(POST)ForceBasicAuth(true)
ContentType("application/json")
Payload("false")
Name("Http Command 1")
};
HttpCommand sendHttp_2On:={
Command("api/ext/v1/vgpis/2")SSL(true)Port(443)IP("10.30.201.236")
Password("pwd12345")UserName("API_username")Method(POST)ForceBasicAuth(true)
ContentType("application/json")
Pavload("true")
Name("Http Command 2")
};
HttpCommand sendHttp_2Off:={
Command("api/ext/v1/vgpis/2")SSL(true)Port(443)IP("10.30.201.236")
Password("pwd12345")UserName("API_username")Method(POST)ForceBasicAuth(true)
ContentType("application/json")
Pavload("false")
Name("Http Command 2")
if(VCARule(1,2)) then sendHttpOn else sendHttpOff;
if(VCARule(1,1)) then sendHttp_2On else sendHttp_2Off;
};
        Number of VGPI
        IP address of the device
        Username and Password of the API user
```

7 Technical data

7.1 LHN-UC15W-SIP Horn loudspeaker 15W, wide angle, SIP

Electrical

Power transfer			
Power over Ethernet		PoE IEEE 802.3af Class 3 PoE+ IEEE 802.3at Class 4	
Power consumption	er consumption PoE <5 W in Idle <7 W at 1/8 th of rated power <13 W at rated power	<7 W at 1/8 th of rated power	
	PoE+	<6 W in Idle <9 W at 1/8 th of rated power <26 W at rated power	
Speaker*			

Rated power	7 W with PoE 15 W with PoE+
Maximum sound pressure level (500 Hz — 8 kHz, 1 m)	115 dB with PoE 118 dB with PoE+
Effective frequency range (-10 dB)	370 Hz — 11 kHz
Coverage angle HxV (-6 dB, 1 kHz)	140°x180°
Coverage angle HxV (-6 dB, 4 kHz)	40°x40°
	/

*Technical performance data acc. to IEC 60268-21

Amplifier	
Туре	15 W class D amplifier
Microphone	
Туре	Integrated omnidirectional Electret Condenser Microphone (can be disabled via hardware switch)
Network	
Ethernet	100BASE-TX, 1000BASE-T
Protocols	IPv4, SIP, NTP, TCP, UDP, HTTP, HTTPS, IPv4 link local, UPnP
SIP audio codecs	G.711 (u-law and a-law), G.722, Opus
Ports	1x RJ45
Analog audio input/output	
Туре	1 line-level input, 1 line-level output; unbalanced
Connector	3-pin screw terminals

Analog audio input/output	
Wire gauge	AWG 28 – AWG 14
Maximum level line input	1 V
Maximum level line output	1 V
Input impedance	>10 kΩ
Output impedance	<100 Ω
Reliability	
MTBF (active part calculated according to Telcordia SR-332 Issue 3, passive part based on field data)	1.000.000 h
Stored messages	
Uploadable	Via web-GUI
Capacity	300 MB
Supported file formats	WAV, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz MP3, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz Ogg Vorbis, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz Opus, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz
GPIO	
Туре	Terminal block with screw terminals
Connector	3-pin screw terminals
Wire gauge	AWG 28 – AWG 14
Ports and operating modes	1x supervised/non-supervised GPI, 1x GPO
Digital inputs	Non-supervised Normal: On: =<0.75 V Off: >0.75 V Inverted: On: =>2 V Off: <2 V
	Supervised Normal: Shorted: =<0.75 V Open: >2 V

On: 0.75 V - 1.25 V Off: 1.25 V - 2 V

Inverted:

GPIO	
	Shorted: =<0.75 V Open: >2 V On: 1.25 V — 2 V Off: 0.75 V — 1.25 V
Digital outputs	On: Output switched to GND, max. 48 V/500 mA Off: Open collector (>10 MΩ to GND)

Mechanical

Horn	
Material	Acrylonitrile Styrene Acrylate (ASA)
Dimension (HxWxD) (mm)	144 mm x 188 mm x 207 mm
Dimension (HxWXD) (in)	5.67 in x 7.40 in x 8.15 in
Weight (kg)	1.6 kg
Weight (lb)	3.53 lb
IP rating	IP66
Color in RAL	RAL 7035 Light gray
Cable gland (standard supplied)	M20 Polyamide (Nylon)
Cable diameter (mm)	5 mm – 12 mm
Cable diameter (in)	0.20 in – 0.47 in

Bracket Material

Stainless steel (grade 316)

Environmental

Climatic conditions	
Operating temperature (°C)	-40 °C – 55 °C
Operating temperature (°F)	-40 °F — 131 °F
Storage temperature (°C)	-40 °C - 70 °C
Storage temperature (°F)	-40 °F — 158 °F
Operating relative humidity, non-condensing (%)	5% — 95%

7.2 LHN-UC15L-SIP Horn loudspeaker 15W, long throw, SIP

Electrical

Power transfer	
Power over Ethernet	PoE IEEE 802.3af Class 3
	PoE+ IEEE 802.3at Class 4

Power transfer		
Power consumption	PoE	<5 W in Idle <7 W at 1/8 th of rated power <13 W at rated power
	PoE+	<6 W in Idle <9 W at 1/8 th of rated power <26 W at rated power
Speaker*		
Rated power		7 W with PoE 15 W with PoE+
Maximum sound pressu 8 kHz, 1 m)	re level (500 H	Iz – 119 dB with PoE 122 dB with PoE+
Effective frequency rang	ge (-10 dB)	380 Hz — 11 kHz
Coverage angle HxV (-6	dB, 1 kHz)	100°x130°
Coverage angle HxV (-6	dB, 4 kHz)	30°x30°
*Technical performance	data acc. to IE	C 60268-21
Amplifier		
Туре		15 W class D amplifier
Microphone		
Туре		Integrated omnidirectional Electret Condenser Microphone (can be disabled via hardware switch)
Network		
Ethernet		100BASE-TX, 1000BASE-T
Protocols		IPv4, SIP, NTP, TCP, UDP, HTTP, HTTPS, IPv4 link local, UPnP
SIP audio codecs		G.711 (u-law and a-law), G.722, Opus
Ports		1x RJ45
Analog audio input/out	put	
Туре		1 line-level input, 1 line-level output; unbalanced
Connector		3-pin screw terminals
Wire gauge		AWG 28 – AWG 14
Maximum level line inpu	ıt	1 V
Maximum level line outp	out	1 V
Input impedance		>10 kΩ
Output impedance		<100 Ω

Reliability	
MTBF (active part calculated according to Telcordia SR-332 Issue 3, passive part based on field data)	1.000.000 h
Stored messages	
Uploadable	Via web-GUI
Capacity	300 MB
Supported file formats	WAV, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz MP3, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz Ogg Vorbis, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz Opus, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz
GPIO	
Туре	Terminal block with screw terminals
Connector	3-pin screw terminals
Wire gauge	AWG 28 – AWG 14
Ports and operating modes	1x supervised/non-supervised GPI, 1x GPO
Digital inputs	Non-supervised Normal: On: =<0.75 V Off: >0.75 V Inverted: On: =>2 V Off: <2 V
	Supervised Normal: Shorted: =<0.75 V Open: >2 V On: 0.75 V - 1.25 V Off: 1.25 V - 2 V Inverted: Shorted: =<0.75 V Open: >2 V On: 1.25 V - 2 V Off: 0.75 V - 1.25 V
Digital outputs	On: Output switched to GND, max. 48 V/500 mA Off: Open collector (>10 MΩ to GND)

Mechanical

Horn	
Material	Acrylonitrile Styrene Acrylate (ASA)
Dimension (HxWxD) (mm)	200 mm x 274 mm x 348 mm
Dimension (HxWXD) (in)	7.87 in x 10.79 in x 13.70 in
Weight (kg)	2 kg
Weight (lb)	4.41 lb
Ingress Protection (IP)	IP66
Color in RAL	RAL 7035 Light gray
Cable gland (standard supplied)	M20 Polyamide (Nylon)
Cable diameter (mm)	5 mm – 12 mm
Cable diameter (in)	0.20 in - 0.47 in
Bracket	
Material	Stainless steel (grade 316)

Environmental

Climatic conditions	
Operating temperature (°C)	-40 °C – 55 °C
Operating temperature (°F)	-40 °F — 131 °F
Storage temperature (°C)	-40 °C - 70 °C
Storage temperature (°F)	-40 °F — 158 °F
Operating relative humidity, non-condensing (%)	5% — 95%

7.3

AMN-P15-SIP Amplifier module 15W, SIP

Electrical

Power transfer			
Power over Ethernet		PoE IEEE 802.3af Class 3 PoE+ IEEE 802.3at Class 4	
Power consumption	PoE	<5 W in Idle <7 W at 1/8 th of rated power <13 W at rated power	
	PoE+	<6 W in Idle <9 W at 1/8 th of rated power <26 W at rated power	
Network			

Network

Ethernet	100BASE-TX, 1000BASE-T

Network	
Protocols	IPv4, SIP, NTP, TCP, UDP, HTTP, HTTPS, IPv4 link local, UPnP
SIP audio codecs	G.711 (u-law and a-law), G.722, Opus
Ports	1x RJ45
Amplifier	
Туре	Class D amplifier
Rated power (W)	7 W with PoE (4-8 Ω) 15 W with PoE+ (4-8 Ω)
Maximum output level	10.95 V
THD+N (3 dB below max., @1 kHz)	<0.03%
Signal-to-Noise Ratio (A-weighted)	>95 dB
Analog audio input/output	
Туре	1 line-level input, 1 line-level output; unbalanced
Connector	3-pin screw terminals
Accepted Wire gauge	AWG 28 – AWG 14
Maximum level line input	1 V
Maximum level line output	1 V
Line input impedance	>10 kΩ
Line output impedance	<100 Ω
Digital signal processing (DSP)	
Sample rate	48 kHz
Signal latency (typically)	<45 ms
Processing	User PEQ (3 bands), Speaker PEQ (6 bands), Compressor, Noisegate, RMS- limiter, Peak limiter, Level, Mute, Delay
Reliability	
MTBF (active part calculated according to Telcordia SR-332 Issue 3, passive part based on field data)	1.000.000 h
Stored messages	
Uploadable	Via Web-GUI
Capacity	300 MB

Stored messages	
Supported file formats	WAV, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz MP3, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz Ogg Vorbis, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz Opus, channels: mono, stereo; sampling rates: 44.1 kHz, 48 kHz

GPIO

GPIO	
Туре	Terminal block with screw terminals
Connector	3-pin screw terminals
Wire gauge	AWG 28 – AWG 14
Ports and operating modes	1x supervised/non-supervised GPI, 1x GPO
Digital inputs	Non-supervised Normal: On: =<0.75 V Off: >0.75 V Inverted: On: =>2 V Off: <2 V
	Supervised Normal: Shorted: =<0.75 V Open: >2 V On: 0.75 V - 1.25 V Off: 1.25 V - 2 V Inverted: Shorted: =<0.75 V Open: >2 V On: 1.25 V - 2 V Off: 0.75 V - 1.25 V
Digital outputs	On: Output switched to GND, max. 48 V/500 mA Off: Open collector (>10 MΩ to GND)

Mechanical

Enclosure	
Material	Acrylonitrile Butadiene Styrene (ABS)
Dimension (HxWxD) (mm)	92 mm x 146 mm x 40 mm
Dimension (HxWXD) (in)	3.62 in x 5.75 in x 1.57 in
Weight (g)	215 g
Weight (lb)	0.474 lb

Enclosure	
Ingress Protection (IP)	IP30
Color	Black
DIN rail adapter	
Material	Aluminum

Environmental

Climatic conditions	
Operating temperature (°C)	-40 °C – 55 °C
Operating temperature (°F)	-40 °F — 131 °F
Storage temperature (°C)	-40 °C - 70 °C
Storage temperature (°F)	-40 °F — 158 °F
Operating relative humidity, non-condensing (%)	5% — 95%

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