



**BOSCH**

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# PAVIRO Telephone Interface

## Application Note

Turn your PAVIRO Public Address system into a fully-functioning SIP/VoIP paging solution.





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**Warning!**

Limit the use of the PRS-1AIP1 with Telephone Interface firmware to a secured LAN only and protect this LAN against unauthorized persons by using firewalls, access filters, port security, anti-virus, etc.

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**Warning!**

You are not permitted to configure the PRS-1AIP1 to use with Telephone Interface firmware for emergency calls as the SIP solution is not certified for this purpose. This SIP paging solution is intended to be used for business calls only.

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**Warning!**

Update your passwords for the Telephone Interface, the PAVIRO open interface and SIP server connection on a regular basis.

## 1. Introduction

This application note describes the configuration of the Bosch PAVIRO system and the Audio-over-IP interface (PRS-1AIP1) for making announcements into the loudspeaker zone(s) of a PAVIRO system, using a telephone. The application note is intended for PAVIRO installers and system integrators who have basic knowledge of SIP servers and clients.

Following products and software/firmware are required for the application:

- PAVIRO system (at least one controller and one amplifier)
- Audio-over-IP interface (PRS-1AIP1)
- Specific SIP firmware for PRS-1AIP1
- PBX/telephone, VoIP (soft) phones

The standard Audio-over-IP Interface (PRS-1AIP1) does not provide the required telephone interface functionality. A special SIP firmware is required for making announcements into the PAVIRO system via a VoIP telephone.

[www.boschsecurity.com](http://www.boschsecurity.com) → Product Catalog → Region/country → PA, Commercial Sound and Emergency Sound → PAVIRO → Product page → Software → “Bosch\_PRS1AIP1\_sip\_firmware\_v3\_xx.zip”

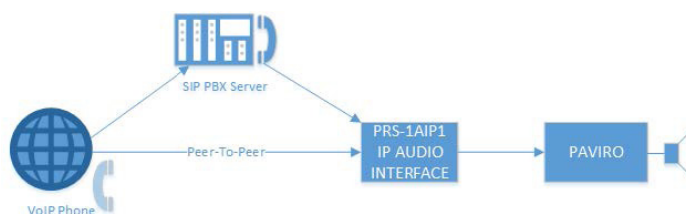
### 1.1 Solution

SIP Client firmware has been customized for PRS-1AIP1 to support telephone based Voice over IP announcement using the application-layer control protocol known as “SIP” (Session Initiation Protocol, RFC 3261).

The SIP Client firmware includes a communication module which uses the “Open Interface” module of PAVIRO to initiate calls.

The SIP Client application, based on Barix SIP Client, can be configured to work in a client-server based connection (via PBX server). Peer-to-peer mode is only supported in PRS-1AIP1 using the default firmware without SIP functionality.

The SIP client application is customized for PAVIRO system in combination with PRS-1AIP1, hence it may not work with other Voice over IP interface.



## 1.2 Features

- 90 zones or area selections
- 7 zones maximum concurrent selection for live call or playing a business message
- 8 controllable output contacts

## 1.3 Supported hardware

- The SIP Client firmware works with PRS-1AIP1 hardware version 2.0 in combination with PAVIRO system with IRIS-Net 4.0 or higher version.

## 1.4 Additional documents

This document provides sufficient information to configure PAVIRO telephone interface. For additional information following documents are recommended:

- Barix SIP Client User Manual  
[www.barix.com](http://www.barix.com) → Downloads → Firmware Download → SIP Client Application → SIP Client User Manual  
The Barix VoIP firmware does not allow the sending of call commands via “Open Interface” to PAVIRO, only via contacts. This document provides further background information but cannot be used as a manual for this PAVIRO telephone interface solution.
- Barix SIP Client and Cisco Call Manager  
[www.barix.com](http://www.barix.com) → Downloads → product family Annunicom 1000 → Calatog/brochures → SIP Client and Cisco Call Manager - How To  
This document explains how to ensure that the SIP Client firmware registers and works with the Cisco Unified Call Manager (CUCM). Barix SIP-Client-Firmware was tested with CUCM V8.6.2.)
- Bosch IP audio interface user manual  
[www.boschsecurity.com](http://www.boschsecurity.com) → Product Catalog → Region/country → Public address and voice alarm → Digital PA and emergency sound → PRAESIDEO system → PRS-1AIP1 Audio-over-IP interface Product Page -> Documents
- Bosch IRIS-Net user manual  
[www.boschsecurity.com](http://www.boschsecurity.com) → Product Catalog → Region/country → PA, Commercial Sound and Emergency Sound → PAVIRO → Product page → Documents → Configuration manual

## 1.5 SIP Client Firmware

The SIP Client application is part of the SIP firmware (abcl\_sip\_paviro\_v3.09pv\_20190110.zip). The SIP Client application works properly when the latest firmware is extracted from the original PAVIRO telephone interface installation package. SIP Client applications with a different version may not work properly.

## 2. System setup

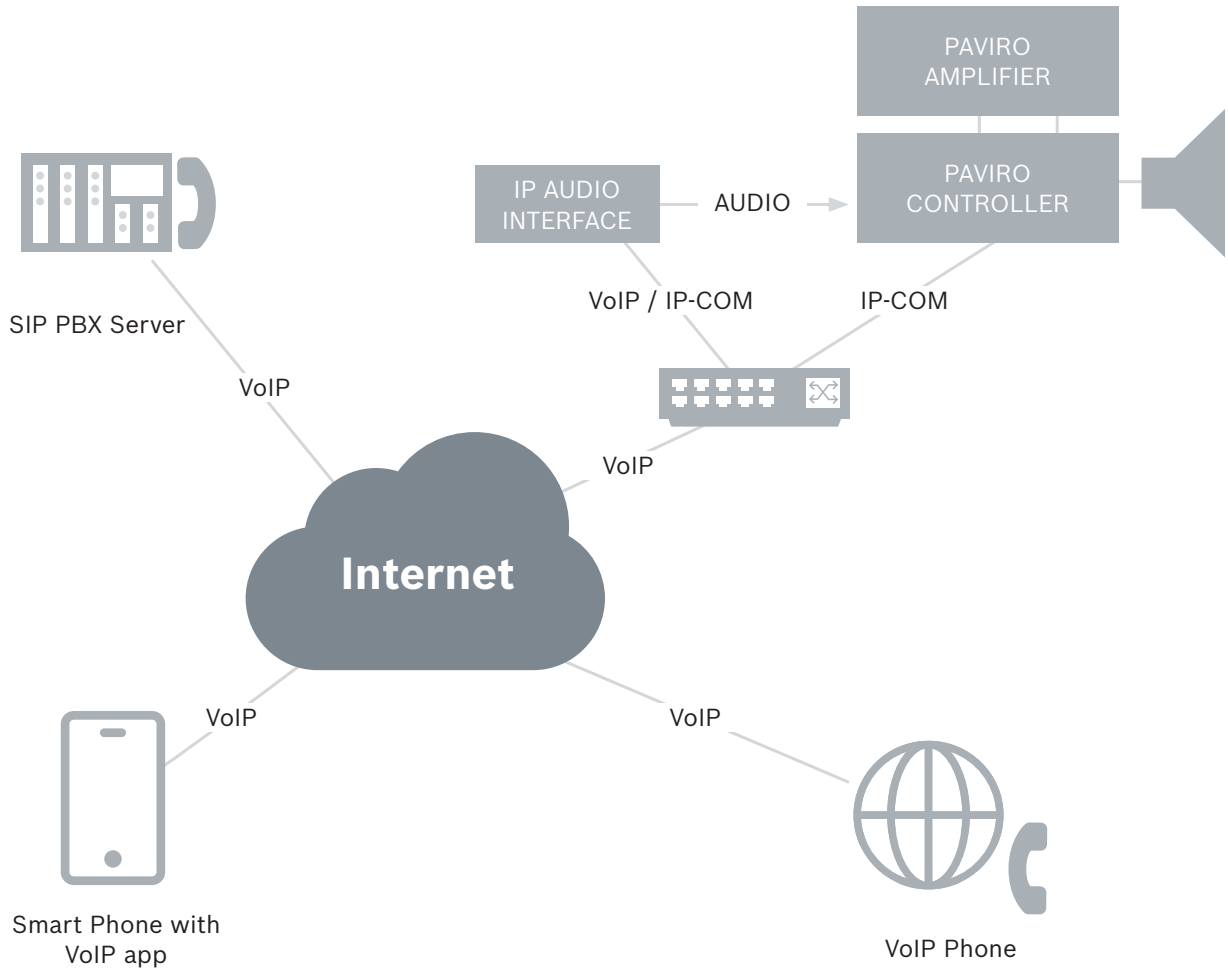


Figure 1 System setup

A 3rd party SIP PBX Server is required for being able to call the IP Audio Interface from multiple telephones. For reference, chapter 3.4 provides some notes about how to configure an Asterisk PBX server. PBX server manual can be referred for further information.

The PAVIRO Telephone Interface software is based on the Barix SIP Client software which has been tested with the following list of SIP servers/PBX:

- Asterisk 1.4.29
- OpenSER 1.3.2-notls (i386/linux)
- FreeSWITCH 1.0.head (git-00207ce 2010-09-30 02-37-57 -0400)

The following SIP servers have been confirmed by partners to work (with some limitations):

- Cisco UCM 8.x/9.1
- Panasonic NCP
- Mitel
- 3CX
- 3Com/HP VCX 7210 IP Call processor
- Shoretel

The customer is responsible for the support and protection of the PBX against any security or fraud threats.

The Audio-over-IP Interface PRS-1AIP1 from Bosch will be loaded with release v3.08 or later of the Bosch SIP PAVIRO GW firmware (sip\_paviro\_v3.09pv\_20190524.bin). This unit can subscribe as a SIP Client to the SIP PBX Server. It will have a phone number which can be called like any other VoIP client.

In the system setup example above, a Smart Phone with VoIP app, a standard VoIP Phone and the Audio-over-IP Interface are all SIP clients. They connect to the SIP PBX Server system where they get their phone number. After “calling” the Audio-over-IP Interface from one of the other phones, the PBX will set up a telephone connection. Based on the digits received from the caller, the Audio-over-IP Interface will trigger calls in the PAVIRO system via Ethernet. The Audio-over-IP Interface receives the audio stream from the caller and provides it via Line output to the Line input of the PAVIRO system to page this audio, or a prerecorded message, into requested zones.

In case of a having only one source from a fixed location a Peer-to-Peer connection between Phone and IP Audio Interface could be set up without using a PBX server.



### 3. Quick Start Guide

#### 3.1 Connecting the Audio-over-IP Interface from Bosch

This chapter illustrates how to use the BARIX SIP Client FW running on an Audio-over-IP Interface. For more details refer to Audio-over-IP Interface PRS-1AIP1 Operating Manual.

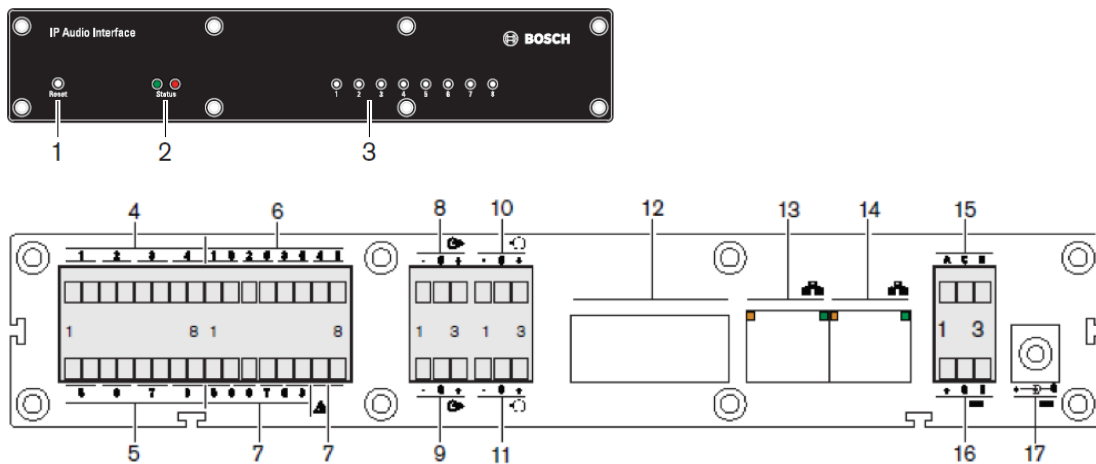


Figure 2 Front and Rear view of PRS-1AIP1

##### STEP 1 (connect to network)

Plug a standard (straight-through) network cable into the LAN port (13) of the Audio-over-IP Interface and the other end into your switch connected to your LAN.

##### STEP 2 (connect audio)

When the device is first powered up (in the next step), it will announce its IP address in English on both Audio Outputs (9 and 8). E.g. 192.168.0.1 (voice: “one nine two dot one six eight dot zero dot one”). Connect a headphone to these Audio Outputs in order to hear this SonicIP announcement.

##### STEP 3 (power up)

Connect the power supply to the power supply input connector (17). Optionally you can connect a back-up power supply to the power supply socket (16). The device will start-up and the eight green Control I/O LEDs (3) go ON and the green and red status LEDs (2) will flash.

The device will be powered from the voltage source with the highest voltage. Make sure that the battery back-up power supply has a lower voltage than the mains power supply to avoid battery drain during normal operation.

### No DHCP server?

If no DHCP server is found then the unit will search for a free IP address in the network.

This can take up to 5 minutes. If the IP address is not announced, check the following:

- If the green LED of the used network port (13 or 14) is OFF, then check your network cabling and connection between this port and the network switch.
- If the green status LED (2) is OFF, then check the power supply cabling.

If the device still fails, revert to the factory default settings by pressing the reset button (1) between 5 and 10 seconds while it is powered and repeat STEPs 1, 2 and 3 (above).

### STEP 4 (open a web browser to setup a static IP address)

Wait for the eight green Control I/O LEDs (3) and the one red status LED (2) to go OFF and the green status LED (2) to switch ON. Open a web browser and enter the IP address obtained in STEP 2 to get into the configuration page of the Audio-over-IP Interface. It is recommended that you configure the device with a static IP address. If you are accessing the unit from outside the local area network, you should also set the subnet mask and default gateway.

After assignment of the static IP address, you can switch off the SonicIP feature (described in STEP 2) via the web browser, as in Figure 3. This will make device startup slightly quicker.

STATUS	CONFIGURATION	CONFIG VIEW	RESTART	UPDATE	MAC: 0008E1024112 FW VB2.03
--------	---------------	-------------	---------	--------	-----------------------------

**IP-Audio Interface**

CONFIGURATION

- ▶ Network
- ▶ Streaming
- ▶ Partner Table
- ▶ Outgoing Route Table
- ▶ Control
- ▶ Audio Settings
- ▶ Supervision
- ▶ Control I/O mapping
- ▶ Serial
- ▶ SNMP
- ▶ Security
- ▶ Default Settings

Unit name	<input type="text"/>
IP Address	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Subnet mask	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Default gateway	<input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Type of Service/DSCP	<input type="text" value="0"/>
Use SonicIP®	<input checked="" type="radio"/> Yes <input type="radio"/> No
<input type="button" value="Submit"/> <input type="button" value="Cancel"/>	

Figure 3 Set a static IP address

**STEP 5 (load SIP firmware)**

The Telephone Interface installation package contains this Application Note, an IRIS-Net example configuration and the SIP firmware, which needs to be loaded into the PRS-1AIP1.

Unpack the zip file “Bosch\_PRS1AIP1\_sip\_firmware\_v3\_08.zip” (or later version) and save the sip\_paviro\_v3.08pv\_20180719.bin to a local drive.

Click UPDATE on top of the page. The update page appears and shows the current device HW/SW version and a link. See Figure 4 below.

Please click here to start the update. A countdown screen appears and then the next update page.

Click UPDATE on top of the web browser page to check and/or update the device firmware and CONFIGURATION pages.



Figure 4 Firmware update

Click Browse and select the file sip\_paviro\_v3.08pv\_20180719.bin (or later version).

After a successful upload, the following text “...\bin successfully loaded” appears.

Then Reboot. Finally reload the main page in your web browser. The STATUS page appears.

### STEP 6 (load default settings)

Clicking the “Back to factory defaults” button (last option in Configuration menu) removes remaining components of the old firmware.

### Hard default settings (only in case of a network connectivity issue)

To revert all settings (including the Network configuration) to factory defaults, the Reset button (1) should be pressed for 5 to 10 seconds while it is powered. Repeat configuration STEPs 4 and 5 (above).

### Configuring the SIP IP gateway

The Audio-over-IP Interface with PAVIRO SIP IP gateway firmware will (depending on your previous configuration) try to get an IP address via DHCP, or already has a statically-assigned IP address. After boot, open your browser and enter the IP address. You will see the empty home page of the PAVIRO SIP IP gateway.

In Basic Settings:

- Where using a PBX, set Peer to Peer to “No”
- Enter the IP address of the PBX.
- Enter the SIP ID (username) that has been created on the PBX for the gateway.
- Enter the SIP Password that has been created on the SIP Server for the gateway. You may leave it empty if the PBX does not require authentication.

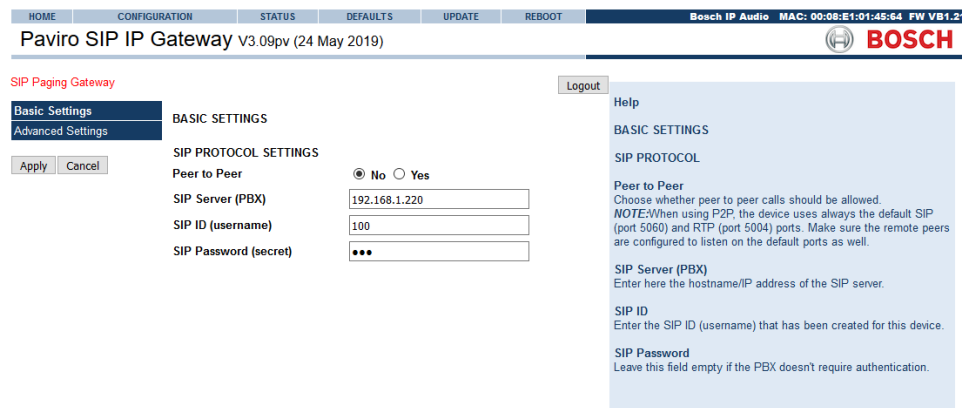
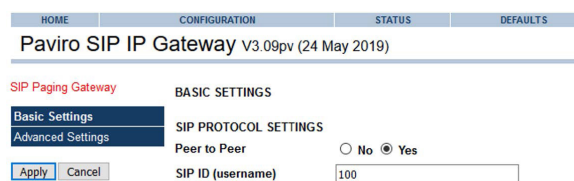


Figure 5 SIP IP Gateway Basic Settings

- Where using a direct connection between Phone and IP Audio Interface, set Peer to Peer to “Yes”
- Enter the SIP ID (username) that has been created on the PBX for the gateway.




**In Advanced Settings → System:**

HOME	CONFIGURATION	STATUS	DEFAULTS	UPDATE	REBOOT
------	---------------	--------	----------	--------	--------

**Paviro SIP IP Gateway V3.09pv (24 May 2019)**

SIP Paging Gateway Logout

Basic Settings	
Advanced Settings	
<b>System</b>	<b>SYSTEM SETTINGS</b>
Network	PAVIRO IP Address <input type="text" value="192"/> . <input type="text" value="168"/> . <input type="text" value="1"/> . <input type="text" value="200"/>
SIP Protocol	PAVIRO TCP Port <input type="text" value="49152"/>
Inbound Calls	PAVIRO Controller ID <input type="text" value="1"/>
Audio	PAVIRO Interface Password <input type="text"/>
Security	Paging Station ID <input type="text" value="19"/>
	Maximum Zones <input type="text" value="12"/>
	"All Zones" Command ID <input type="text" value="13"/>
	"Paging Status" Command ID <input type="text" value="1"/>
	"Paging OK" Status Value <input type="text" value="9"/>
	"Trigger" Command ID <input type="text" value="14"/>
	Busy Zone Timeout <input type="text" value="3500"/> ms
	LEDs control <input type="text" value="Relays"/>

Apply Cancel

Figure 6 Advanced system settings

- Enter the IP address of the PAVIRO controller
- Enter the PAVIRO TCP communication port of the Open Interface (default 49152)
- Enter the PAVIRO controller ID (default 1)

If you add one controller after the other and do not delete a controller, the controller ID is corresponding to the object name (e.g. PVACON\_1 -> Controller ID 1).

But if you delete a controller in a multi-controller system, the ID can change.

Example:

- o Adding two controllers
- o PVACON\_1 -> Controller ID 1
- o PVACON\_2 -> Controller ID 2
- o Deleting controller 1
- o PVACON\_2 -> Controller ID 1

Therefore if you delete a controller, you have to set up a telnet connection via Open Interface or the PA Event log in IRIS-Net to identify the controller ID.

- Enter the password which is used by the SIP gateway for authorized connection to the PAVIRO Controller. Using a password for the PAVIRO interface is optional.
- Enter the Paging request ID  
The Paging request ID is used by the SIP IP gateway to get feedback on paging requests. This ID is included in the Open Interface response of PAVIRO but is not automatically known by the SIP IP gateway. A valid IRIS-Net configuration and active connection to the PAVIRO system is required to obtain it. The ID can be retrieved based on the name configured in IRIS-Net for the SIP IP gateway pagings.

In IRIS-Net press <Ctrl> + double left-click on the PVACON\_1 to get a list of available Controller Request IDs.

In case Pagings are configured e.g. for “ANNOUNCE\_1”, the ID for the Pagings\_ANNOUNCE\_1 should be listed here. The “Pagings” name and its ID could change after alterations in the IRIS-Net configuration, so they should be verified after every configuration update to be in line with the SIP IP gateway Paging request ID.

Please note this ID retrieval feature is only supported in IRIS-Net version V4.0 or higher.

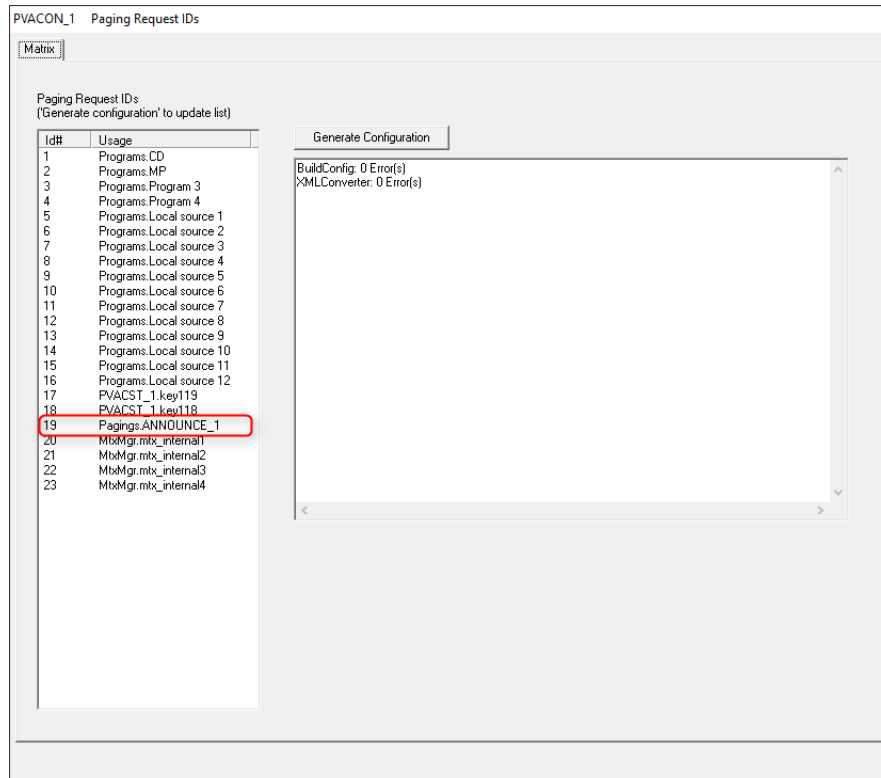


Figure 7 Retrieving the paging status ID

- Go back to the PRS-1AIP1 web page to enter the Paging station ID which was discovered as explained above.
- Maximum zones  
Zones and zone-groups in PAVIRO need OpenInterface Logic values (LVals) to be triggered. The total number of available LVals is configured in IRIS-Net OpenInterface setting “NR LOGIC VALUES” (for example ‘15’ as indicated in Figure 14).

The Maximum zones setting in the SIP IP Gateway is related to this and should be at least one number below this “NR LOGIC VALUES” (in this example ‘12’).

Numbers above the “Maximum zones” can be configured as special command ID for “All zones” paging to “Sysgroup\_500” and as command ID for triggers as shown in Figure 10.

NR LOGIC VALUES	Lval 15		
	Lval 14	Trigger	
	Lval 13	All_zones	Sysgroup_500
Maximum zones	Lval 12	Zonegroup 1	Sysgroup_1
	Lval 11	Zone 11	
	Lval 10	Zone 10	
	Lval 9	Zone 9	
	Lval 8	Zone 8	
	Lval 7	Zone 7	
	Lval 6	Zone 6	
	Lval 5	Zone 5	
	Lval 4	Zone 4	
	Lval 3	Zone 3	
	Lval 2	Zone 2	
	Lval 1	Zone 1	

Figure 8 Logic values and maximum zones

- Enter the “Paging Status” command ID (default 1)
 

This is the analog value (Aval1) that is sent by the PAVIRO controller to signal the current status of the paging process. In order to send the status of the audio connection (e.g. call/announcement is active) from the controller to the telephone interface, the following must be configured in the Task Engine of the controller:

  - o Go offline
  - o Add an Input Analog block from the Values-Analog segment
  - o Add an Output Analog block from the Values-Analog segment
  - o Link the added blocks
  - o Connect the Input Analog block with the Action Status of the Paging, used for the audio connection to the telephone interface



Example below: PVIACON\_1.PagingRequestID.Pagings.ANNOUNCE\_1.ActionStatus

- Connect the Output Analog block with an analog value of the Open Interface to send the call status to the telephone interface

Example below: PVIACON\_1.OpenInterface.Aval1

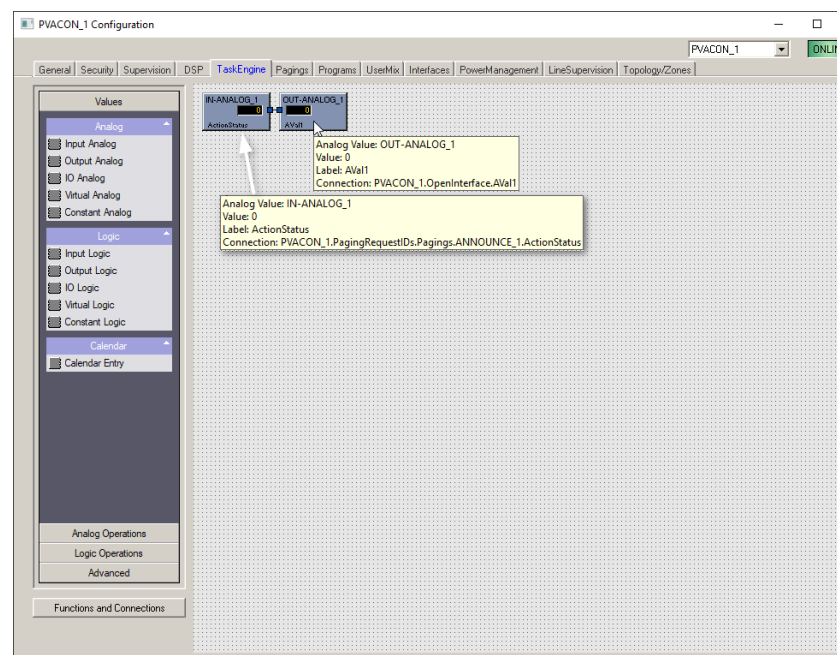


Figure 9 Call status feedback configuration

- Enter the “Paging Ok” status value (default 9)  
This is the value provided by the PAVIRO controller via the OpenInterface.Aval1 to indicate that the paging process on the PAVIRO controller has been successfully set up and that the call is running.
- Enter the “Trigger” command ID (e.g. 14)  
The Trigger command ID is included in the PAVIRO Pagings-configuration to trigger the start and stop of a call, see Figure 10. After all paging destinations are sent, this LVal14 will simultaneously trigger the start and end call for all dialed destinations. If not configured, the trigger will not be recognized, calls will not be started and the system will report zones as busy. This ID has to be > “Maximum zones” number.

PVACON\_1 Configuration

General Security Supervision DSP TaskEngine **Pagings** Programs UserMix Interfaces PowerManagement LineSupervision Topology/Zones

TYPE	CONDITION	INVERT	START/ADD TRIGGER	STOP/SUB TRIGGER	PAGING	DESTINATIONS	REQUEST
STATIC	OpenInterface.LVal1	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_1	INACTIVE
STATIC	OpenInterface.LVal2	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_2	INACTIVE
STATIC	OpenInterface.LVal3	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_3	INACTIVE
STATIC	OpenInterface.LVal4	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_4	INACTIVE
STATIC	OpenInterface.LVal5	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_5	INACTIVE
STATIC	OpenInterface.LVal6	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_6	INACTIVE
STATIC	OpenInterface.LVal7	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_7	INACTIVE
STATIC	OpenInterface.LVal8	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_8	INACTIVE
STATIC	OpenInterface.LVal9	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_9	INACTIVE
STATIC	OpenInterface.LVal10	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_10	INACTIVE
STATIC	OpenInterface.LVal11	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	PVACON_1.Zone_11	INACTIVE
STATIC	OpenInterface.LVal12	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	SysGroup_1	INACTIVE
STATIC	OpenInterface.LVal13	<input type="checkbox"/>	OpenInterface.LVa14	OpenInterface.LVa14	ANNOUNCE_1	SysGroup_500	INACTIVE

DELETE UPDATE

CONDITION

FILTER:

- GPIDigital1
- GPIDigital2
- GPIDigital3
- GPIDigital4
- GPIDigital5
- GPIDigital6
- GPIDigital7
- GPIDigital8
- OpenInterface.LVal1
- OpenInterface.LVal10
- OpenInterface.LVal11

START/ADD TRIGGER

IMMEDIATE  
 TRIGGERED  
 NEXT SEQUENCE

FILTER:

- GPIDigital1
- GPIDigital2
- GPIDigital3
- GPIDigital4
- GPIDigital5
- GPIDigital6

STOP/SUB TRIGGER

IMMEDIATE  
 TRIGGERED  
 NEXT SEQUENCE

FILTER:

- GPIDigital1
- GPIDigital2
- GPIDigital3
- GPIDigital4
- GPIDigital5
- GPIDigital6

DESTINATION

STATIC  
 DYNAMIC

- SysGroup\_1
- SysGroup\_500
- PVACON\_1.Zone\_1
- PVACON\_1.Zone\_2
- PVACON\_1.Zone\_3
- PVACON\_1.Zone\_4
- PVACON\_1.Zone\_5
- PVACON\_1.Zone\_6
- PVACON\_1.Zone\_7
- PVACON\_1.Zone\_8
- PVACON\_1.Zone\_9
- PVACON\_1.Zone\_10
- PVACON\_1.Zone\_11

PAGING

EXISTING  NEW/EDIT

TYPE: ALARM      ALARM: DIN Alarm

LABEL: ALARM\_1      INPUT CHANNEL: AUX\_1

PRIORITY: 90

Figure 10 Call start and stop triggers

**In Advanced settings → Network:**

- You may disable “SonicIP” to prevent the IP address being announced over the PRS-1AIP1 line out after reboot. Disabling will also speed up future reboots of this device.
- Entering a Syslog IP Address of a PC running WireShark (filter on “syslog”) will provide more operational information during setup and testing.
- For more details refer to the Help text at the right-hand side of the web page.

HOME	CONFIGURATION	STATUS	DEFAULTS	UPDATE	REBOOT
------	---------------	--------	----------	--------	--------

**Paviro SIP IP Gateway V3.09pv (24 May 2019)**

SIP Paging Gateway Logout

Basic Settings	
<b>Advanced Settings</b>	<b>NETWORK SETTINGS</b>
System	Use SonicIP® <input type="radio"/> Yes <input checked="" type="radio"/> No
<b>Network</b>	IP Address <input type="text" value="192"/> . <input type="text" value="168"/> . <input type="text" value="1"/> . <input type="text" value="215"/>
SIP Protocol	Netmask <input type="text" value="255"/> . <input type="text" value="255"/> . <input type="text" value="255"/> . <input type="text" value="0"/>
Inbound Calls	Gateway IP Address <input type="text" value="192"/> . <input type="text" value="168"/> . <input type="text" value="1"/> . <input type="text" value="254"/>
Audio	Primary DNS <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Security	Alternative DNS <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
Apply <input type="button" value="Apply"/> <input type="button" value="Cancel"/>	Syslog Address <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/> . <input type="text" value="0"/>
	DHCP Host Name <input type="text"/>
	Web Server Port <input type="text" value="80"/>
	QoS/DSCP <input type="text" value="0"/>
	SNMP System Name <input type="text"/>
	SNMP System Location <input type="text"/>
	SNMP System Contact <input type="text"/>

Figure 11 Network settings

In Advanced SIP Protocol settings, more details of the PBX Server can be configured. For more details refer to the Help text at the right-hand side of the web page.

HOME	CONFIGURATION	STATUS	DEFAULTS	UPDATE	REBOOT
------	---------------	--------	----------	--------	--------

Paviro SIP IP Gateway V3.09pv (24 May 2019)

SIP Paging Gateway

Basic Settings	
<b>Advanced Settings</b>	<b>SIP PROTOCOL SETTINGS</b>
System	Peer to Peer <input checked="" type="radio"/> No <input type="radio"/> Yes
Network	SIP Server (PBX) <input type="text" value="192.168.1.220"/>
<b>SIP Protocol</b>	Backup SIP Server (PBX) <input type="text"/>
Inbound Calls	SIP ID (username) <input type="text" value="100"/>
Audio	SIP Password (secret) <input type="password" value="..."/>
Security	SIP Display ID <input type="text"/>
	Authentication ID <input type="text"/>
	Listen SIP Port <input type="text" value="0"/>
	Listen RTP Port <input type="text" value="0"/>
	Default REGISTER Time <input type="text" value="1800"/>
	Send NAT-keepalives <input checked="" type="radio"/> No <input type="radio"/> Yes
	Periodically Renew DNS <input checked="" type="radio"/> No <input type="radio"/> Yes
	Registration Fail Timeout <input type="text" value="10"/> seconds
	Call Timeout <input type="text" value="2"/>
	Debug Mode <input checked="" type="radio"/> Off <input type="radio"/> On

Apply Cancel

Figure 12 SIP Protocol settings

HOME	CONFIGURATION	STATUS	DEFAULTS
------	---------------	--------	----------

Paviro SIP IP Gateway V3.09pv (24 May 2019)

SIP Paging Gateway

Basic Settings	
<b>Advanced Settings</b>	<b>SIP PROTOCOL SETTINGS</b>
System	Peer to Peer <input type="radio"/> No <input checked="" type="radio"/> Yes
Network	SIP ID (username) <input type="text" value="100"/>
<b>SIP Protocol</b>	SIP Display ID <input type="text" value="100"/>
Inbound Calls	Authentication ID <input type="text" value="100"/>
Audio	Listen SIP Port <input type="text" value="0"/>
Security	Listen RTP Port <input type="text" value="0"/>
	Default REGISTER Time <input type="text" value="1800"/>
	Send NAT-keepalives <input checked="" type="radio"/> No <input type="radio"/> Yes
	Periodically Renew DNS <input checked="" type="radio"/> No <input type="radio"/> Yes
	Registration Fail Timeout <input type="text" value="10"/> seconds
	Call Timeout <input type="text" value="2"/>
	Debug Mode <input checked="" type="radio"/> Off <input type="radio"/> On

Apply Cancel

In case Peer-to-Peer is configured, the Listen SIP and RTP ports are not configurable (0 means default which is 5060 for SIP and 5004 for RTP.)

**In Advanced Audio settings:**

- Select the g.711 Encoding type, aLaw or uLaw to indicate how the voice is compressed.  
aLaw (a-law) is used mainly in European systems  
uLaw (μ-law) is used in American systems
- Best experience on Volume settings is about 80%, but this depends on PAVIRO system configuration.

HOME	CONFIGURATION	STATUS	DEFAULTS	UPDATE	REBOOT
------	---------------	--------	----------	--------	--------

Paviro SIP IP Gateway V3.09pv (24 May 2019)

SIP Paging Gateway Logout

Basic Settings	
<b>Advanced Settings</b>	<b>AUDIO SETTINGS</b>
System	Encoding <input type="text" value="uLaw / 8 kHz (G.711)"/>
Network	Volume <input type="text" value="90"/> %
SIP Protocol	
Inbound Calls	
<b>Audio</b>	
Security	

Apply Cancel

Figure 13 Audio settings

### 3.2 Configuring PAVIRO

This manual does not include instructions on initial set-up of a PAVIRO system, nor how to configure it in IRIS-Net. This will require dedicated training. A demo IRIS-Net configuration is included in the installation package to quickly set up a test environment for the Audio-over-IP Interface. This chapter explains that configuration.

The PAVIRO system should at least consist of a PAVIRO controller, a PAVIRO amplifier, power supply and a loudspeaker. Ensure the audio output of the Audio-over-IP Interface from Bosch is balanced connected to one of the PAVIRO MIC/LINE inputs.

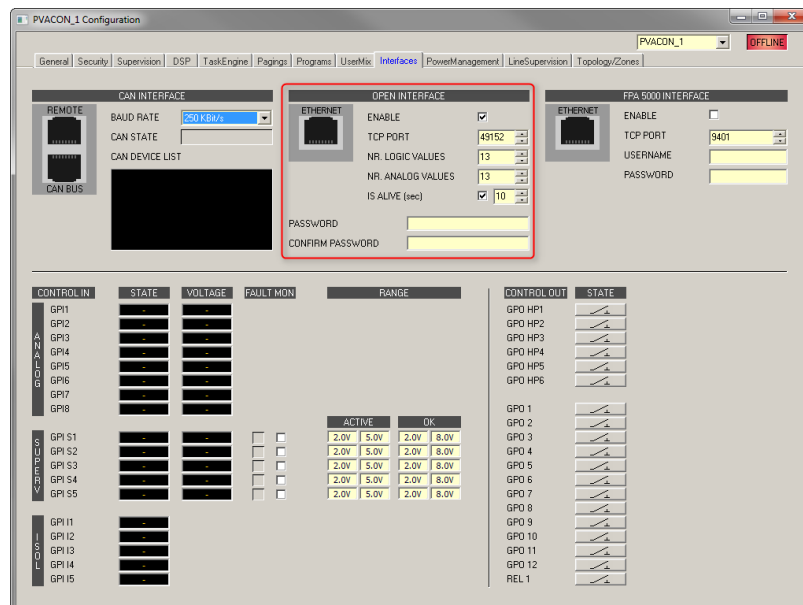


Figure 14 PAVIRO Open Interface settings

Start IRIS-Net and load the configuration provided in the installation package for reference.

Configure Open Interface according to Figure 14:

- Enter TCP Port 49152
- Enter the maximum number of logic values being used (in our example 13; 12 zones and 1 for “all call”).
- Enter the maximum number of analog values being used (should at least include the AVall number configured for the call status feedback).
- Keep alive messages should be sent every 10 seconds to monitor connectivity.
- In case authentication is required, fill in a secret password.

Configure topology and zones:

- In the Topology/Zones Tab (refer to example in Figure 15) the connection between the Internal Router and the Amplifier needs to be defined to be in line with the actual setup (in our example 1\_IN\_N but this is not a requirement for the telephone interface).
- One zone group is default available (All). In the Pagings tab this “All group” can be selected as destination SysGroup\_500 and will be activated when dialing \*\* on the phone.
- Any other zone group could be configured (e.g. Group\_1). In the Pagings tab this zone group can be selected as destination SysGroup\_1 and will be activated when dialing the corresponding OpenInterface.LVal number.

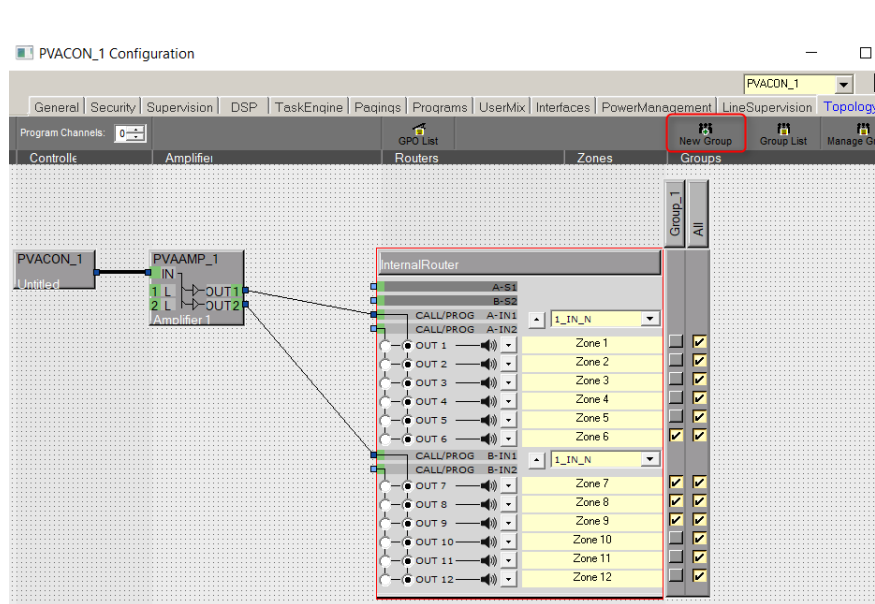


Figure 15 Topology and zone groups configuration

- In the Pagings tab the OpenInterface logic values must be defined. In our example in Figure 16, paging condition LVal 1-11 is configured for 11 zones, LVal 12 for SysGroup\_1 (zone 6-9) and LVal13 for SysGroup\_500 (all zones).

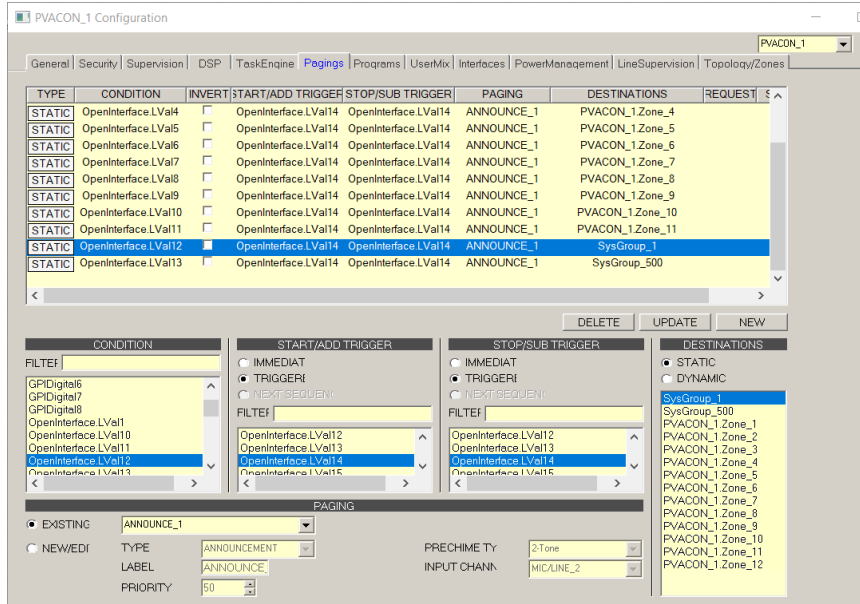


Figure 16 Pagings condition configuration

Pagings configuration (refer to Figure 16):

- Create a new paging and select the CONDITION (OpenInterface.LVal1..13).
- Change START/ADD & STOP/SUB TRIGGER from “Immediate” to “Trigger” and select as trigger OpenInterface.LVal14 (14 = the trigger value defined in the PRS-1AIP1 web page Figure 6 Advanced system settings).
- Optionally a PRECHIME can be selected here to be generated by PAVIRO.
- If you select a pre-chime other than the default 2-Tone chime, you will need to extend the zone-busy timeout (currently set to 3,500 ms; refer to Figure 6). This is the time needed to set up the call, play the pre-chime and wait until paging feedback is received.
- Select a priority (1-69). **Paging conditions for the telephone interface should never have a priority above 69 as the default EVAC paging priority of 70 and beyond will put the system into a voice alarm status which is not allowed in an EVAC (EN54-16) certified system.**
- INPUT CHANNEL should be “MIC/LINE\_1” or “MIC/LINE\_2” depending on where the line out of the telephone interface is connected to.

### 3.3 Verify connectivity

Status screen in the SIP IP Gateway Status web page (refer to Figure 17) will show:

- the APPLICATION status
  - o Application Mode
    - in SIP Mode it shows
      - to which SIP PBX it is connected to
      - using which SIP ID
      - current registration time-out
    - in Peer-to-Peer it shows
      - to which P2P Remote Party it is connected to
  - o Call status (idle or busy)
  - o Remote Party (in case the call is active)
- the PAVIRO system status
  - o Is PAVIRO controller connected or disconnected?
  - o Paging within PAVIRO active / inactive
  - o Received DTMF characters
  - o Last SIP messages sent
  - o Selected / busy zones
- Device health (power / temperature / fault relay)
- the Audio Status
  - o During a call the Audio Output peak levels

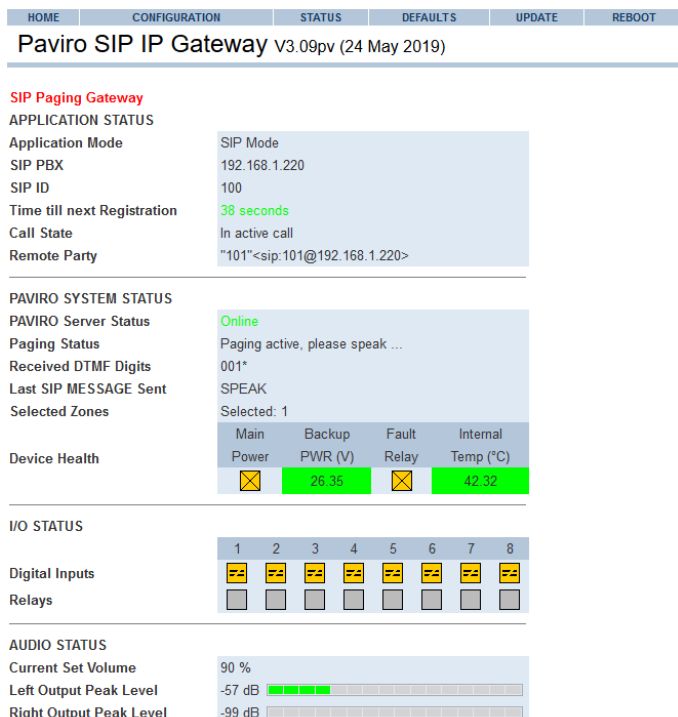
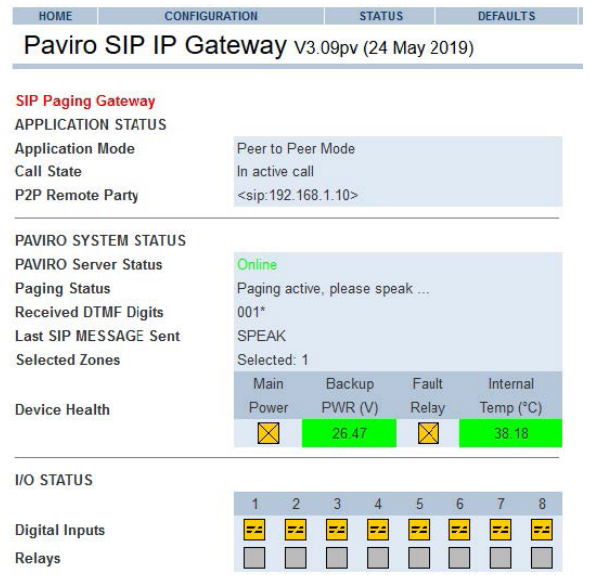


Figure 17 SIP IP gateway system status





## 3.4 Notes on Asterisk SIP server configuration

### SIP registration (REGISTER method)

In “Peer to Peer” mode, a basic SIP dialog can be established by setting up “Remote Peer” and “SIP Id” fields.

To establish a SIP dialog through a SIP PBX/Server, “SIP Server”, “SIP Id” and “SIP Password” must be set, and the registration process must succeed.

A corresponding entry for this account must be created at the server side. Server configuration depends upon the brand and model of the SIP PBX. Please consult the server-side documentation for this.

Shown below is a server configuration example for the Asterisk open-source PBX, supposing that both the Audio-over-IP Interface and the Asterisk PBX are in the same local network, that both are in the same network subnet (192.168.1.XXX) and that the server network address is 192.168.1.220.

In “sip.conf” configuration file:

```
[ann01]
type=friend
username=ann01
secret=1234
host=dynamic
```

In “extensions.conf” configuration file:

```
exten => 10001,1,Dial(SIP/ann01)
```

After the Asterisk server is configured and restarted, Audio-over-IP Interface should successfully register to the PBX using the following settings:

```
SIP Server (PBX)/ Remote Peer : 192.168.1.220
SIP Id : ann01
SIP Password : 1234
Listen SIP Port : 0
Listen RTP Port : 0
Send NAT Keepalives : No
Debug mode : On
```

The “Debug mode” option set to “On” allows verification that the Bosch IP audio device has correctly registered to the SIP server. Using a common Syslog application such as “Kiwi Syslog Daemon” ([www.kiwisyslog.com](http://www.kiwisyslog.com)) or a similar tool, after a successful registration process the following debug messages will be displayed:

```
[OUT] REGISTER
[in ] 100
[in ] 401
[OUT] REGISTER
[in ] 100
[in ] 200
```

200 (Ok) status message received (in) confirms the SIP Client application has correctly registered to the server, and is ready to issue or receive calls.

### **SIP calls (INVITE method)**

Note: before performing any call through a PBX server, please be sure the registration process has succeeded, as shown in the Status web page (Figure 17).

Calls can be issued as preconfigured via the web browser. Until the remote peer answers or rejects the call, a bell ring sound will be audible.

Once the call has been answered, audio communication starts immediately.

## 4. Call handling (PAVIRO)

### 4.1 Features

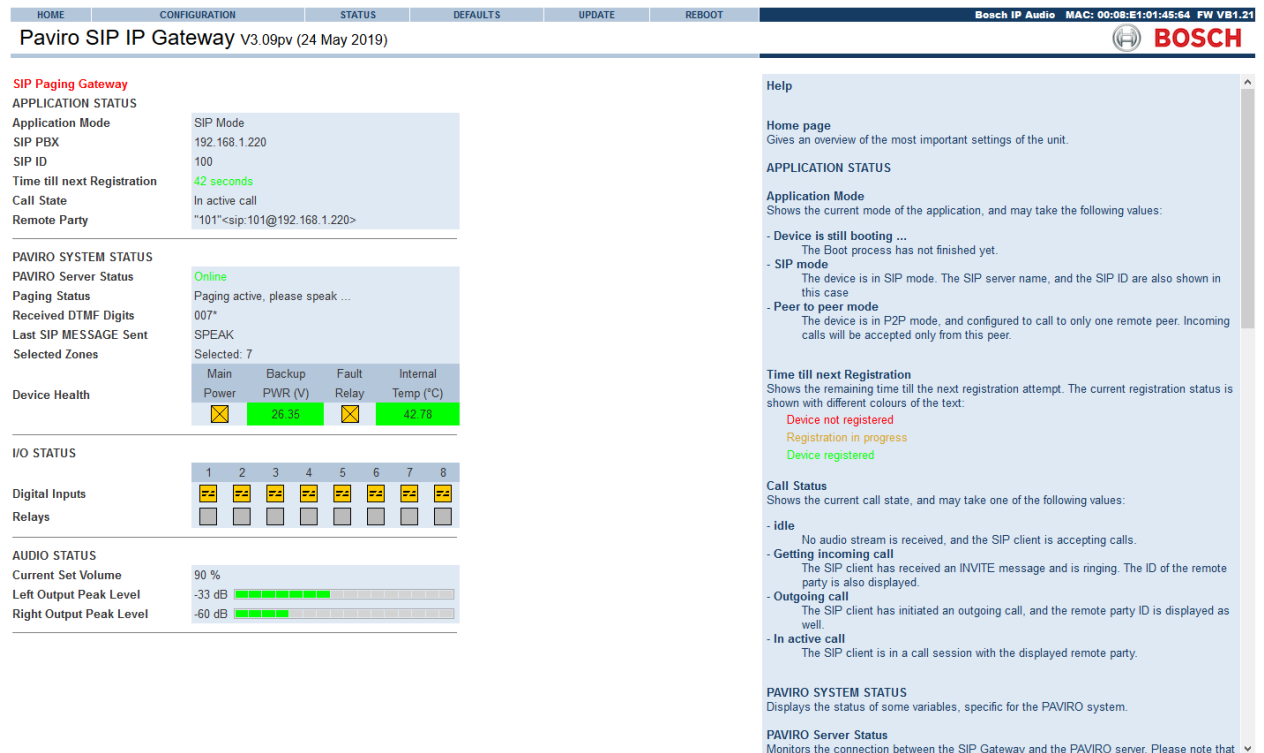
- Selection of paging conditions from 001 to maximum 090
- Each paging condition can have zones assigned which are available in the system
- 7 paging conditions maximum concurrent selection
- Paging condition selection via DTMF
  - o Paging condition numbers must always contain 3 digits
  - o Multiple paging conditions can be selected (max 7, from low to high value)
  - o Enter paging conditions in 3-digit format (e.g. 002004008\* to activate paging conditions 2, 4 and 8)
  - o After completing digit entry, press \* to start paging
  - o Note: pressing \*\* will immediately start an 'all call'
  - o The PRS-1AIP1 has 8 relay contacts available at the back of the unit which can be activated (closed) to trigger 3rd party equipment. The relays can be activated by entering a # immediately after the paging condition entry, followed by a decimal number (1-255). Next, enter the \* to start and activate the selected relays during this paging.
  - o The relay activation code 1-255 is a binary representation of the 8 relays which makes it possible to select all different 255 combinations. See Table 1.

entry	Relay							
	8	7	6	5	4	3	2	1
#1*	0	0	0	0	0	0	0	1
#2*	0	0	0	0	0	0	1	0
#3*	0	0	0	0	0	0	1	1
#4*	0	0	0	0	0	1	0	0
#5*	0	0	0	0	0	1	0	1
#8*	0	0	0	0	1	0	0	0
#224*	1	1	1	0	0	0	0	0
#255*	1	1	1	1	1	1	1	1

Table 1 Relay activation codes

- o E.g. select 001002004#8\* to start paging condition 1, 2 and 4 & activate relay 4

- Chime is played via PAVIRO if the Prechime is configured (refer to the PAVIRO IRIS-Net paging configuration example in Figure 16)
- SIP Method Message provides the following commands / status information (where supported by the SIP phone):
  - o “SELECT”: Zone selection can be entered
  - o “SPEAK”: Caller can speak once chime is complete
  - o “BUSY”: A zone is busy
  - o “INVALID”: Zone entry invalid
- The homepage of the SIP IP gateway provides an overview of the most important settings of the unit and indicates a live view of the actual system status (Figure 18)



**PAVIRO SIP IP Gateway V3.09pv (24 May 2019)**

**SIP Paging Gateway**

**APPLICATION STATUS**

Application Mode: SIP Mode  
 SIP PBX: 192.168.1.220  
 SIP ID: 100  
 Time till next Registration: 42 seconds  
 Call State: In active call  
 Remote Party: \*101<sip:101@192.168.1.220>

**PAVIRO SYSTEM STATUS**

PAVIRO Server Status: Online  
 Paging Status: Paging active, please speak ...  
 Received DTMF Digits: 007\*  
 Last SIP MESSAGE Sent: SPEAK  
 Selected Zones: 7

Device Health	Main Power	Backup PWR (V)	Fault Relay	Internal Temp (°C)
	26.35	26.35	42.76	42.76

**I/O STATUS**

Digital Inputs: 8 buttons (1-8)  
 Relays: 8 buttons (1-8)

**AUDIO STATUS**

Current Set Volume: 90 %  
 Left Output Peak Level: -33 dB  
 Right Output Peak Level: -60 dB

**Help**

**Home page**  
 Gives an overview of the most important settings of the unit.

**APPLICATION STATUS**

**Application Mode**  
 Shows the current mode of the application, and may take the following values:

- **Device is still booting ...**  
 The Boot process has not finished yet.
- **SIP mode**  
 The device is in SIP mode. The SIP server name, and the SIP ID are also shown in this case
- **Peer to peer mode**  
 The device is in P2P mode, and configured to call to only one remote peer. Incoming calls will be accepted only from this peer.

**Time till next Registration**  
 Shows the remaining time till the next registration attempt. The current registration status is shown with different colours of the text:  
 Device not registered  
 Registration in progress  
 Device registered

**Call Status**  
 Shows the current call state, and may take one of the following values:

- **idle**  
 No audio stream is received, and the SIP client is accepting calls.
- **Getting incoming call**  
 The SIP client has received an INVITE message and is ringing. The ID of the remote party is also displayed.
- **Outgoing call**  
 The SIP client has initiated an outgoing call, and the remote party ID is displayed as well.
- **In active call**  
 The SIP client is in a call session with the displayed remote party.

**PAVIRO SYSTEM STATUS**  
 Displays the status of some variables, specific for the PAVIRO system.

**PAVIRO Server Status**  
 Monitors the connection between the SIP Gateway and the PAVIRO server. Please note that

Figure 18 SIP IP Gateway homepage

## 4.2 Call procedure

- When calling the Audio-over-IP Interface by dialing its SIP ID, it answers the incoming call by sending the SIP Method Message “SELECT” to the dialing SIP phone and playing a prerecorded message “Please select zones”.
- For dialing (max 7) paging conditions, enter them in a 3-digit format and terminate with a \* to start the paging process in PAVIRO. (e.g. dial 002005010\* to start paging into zones 2, 5 and 10)
- If an “All call” is requested, enter \*\* which will immediately start the paging process into all zones of the PAVIRO system.
- If configured, a pre-chime will be played by the PAVIRO system and the call will be activated. Next the SIP Method Message “SPEAK” and audio message “You can speak now” will be sent to the SIP Phone to indicate that the caller may start speaking.
- Next, the audio from the SIP phone will be directly broadcast into the selected zones.
- In case the zone entry is invalid (non-existing values), the SIP Method Message “INVALID” will be provided and the Audio-over-IP Interface will hang up the call without further audio messages.
- It is not possible to select multiple paging conditions which overlap destination zones.
- A call time-out (in minutes) can be configured in the Advanced SIP protocol settings to automatically abort the paging process in PAVIRO (refer to Figure 12). This is needed to avoid accidentally unclosed calls. To determine time-out length, the entire SIP call duration should be considered, including time required to select the zones and not only the time needed for paging.
- To close the paging process, end the call by “hanging up” the phone.

## 5. Common issues

Refer to the Common issues chapter 7 of the Barix SIP Client User Manual (location indicated in chapter 1.4) for SIP server compatibility, SIP Registry, audio reception, port connectivity and DTMF command execution.

Good technical knowledge about PBX and PAVIRO configuration is required set up a working system! For additional information consult the respective configuration manuals and supplier.

Known issues and limitations are indicated in the release notes provided together with the PRS-1AIP1 SIP Client firmware (for download location refer to chapter 1 of this document).

## 6. Additional information

Chapter 8.2 of the Barix SIP Client User Manual indicates how to update a device using the RS-232 serial port. This should be consulted in case the flash memory is corrupted for some reason and serial rescue may be needed to reset the device to factory defaults.

**Bosch Security Systems, B.V.**

Torenallee 49  
5600 JB Eindhoven  
The Netherlands

**[www.boschsecurity.com](http://www.boschsecurity.com)**

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**Bosch Sicherheitssysteme GmbH**

Robert-Bosch-Ring 5  
85630 Grasbrunn Germany