AETA
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SCOOPTEAM

Commentary codec

User manual

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1. Presentation – Overview

ScoopTeam is a commentary unit integrating a codec for live transmission of one or two audio signals.

The main target is TV sports reporting, but the product also fits the needs of live radio sports. The unit allows to establish communications between the stadium and the studio/MCR or the IBC (for big events). The communication includes a mono program link in high audio quality (up to 20 kHz bandwidth), and a second channel for coordination[1], optionally via a second link.

Basically, the unit is used by two journalists but it is possible to add a microphone or two for a sound technician or a guest. This unit supports up to four commentary positions in its standard version. For biggest events, more inputs are needed. To answer such needs, the "Premium+" version includes more features, including additional inputs/outputs, and supports an optional AES67/Ravenna interface.

For transmission, the unit is mainly designed for IP networks, but it can also use an ISDN line. In either case the two audio channels can be set with a single destination (transmission in "dual mono" or stereo format), or two separate destinations (double codec). With a mobile option, additional transmission modes are available: AoIP over 3G/4G, and mobile voice (including HD Voice capability).

The product is used by journalists and it aims at being as simple to use as possible, with dedicated buttons and knobs for the routine settings and volume adjustments. More sophisticated settings are typically reserved for technically-skilled operators, preparing the unit beforehand or using remote control for live adjustments. Such remote control can be from just besides, or via a remote access from the radio/TV house.

This chapter gives an **overview** of the product specifications. For comprehensive information one can refer to the rest of this document:

- Chapter 2 describes the **functions** and features of the ScoopTeam (with little detail on the operating modes)
- Chapter 3 gives a **physical description** of the unit, and describes the **main operation principles**.
- Chapter 4 details the **user interface** and **operating modes**.
- Chapter 5 provides all the **technical characteristics** of the ScoopTeam
- The annexes bring miscellaneous additional information, including an **index** you can use to look for a given information topic.

The following table shows the main features of the product.
Functions marked with P+ are available on the "Premium+" version only.
Functions marked with ● in this table are available as options.

> ⓘ *Important notice*:
> *This document describes the ScoopTeam in its "Standard" and "Premium+" versions.*
> *The "Premium" version, and its optional AES67/Ravenna interface, feature differences in the audio inputs and outputs that are not covered here. Please refer to the user manual in its edition G for adequate information on these interfaces.*
>
> *Besides, this document is relevant for units with firmware version 1.07.02.*
> *Other versions have differences in the features and the user interfaces.*

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[1] There is also a specific setup for stereo operation, wherein the coordination channel is in fact the right program channel.

Characteristics	Optional
Transmission interfaces	
Ethernet/IP transmission interfaces	
2 Ethernet interfaces, 10BaseT / 100BaseT (shared with remote control) Audio transmission (AoIP) in unicast mode: SIP or "Direct RTP" Net bit rate 12 to 256 kbit/s (depending on coding algorithm)	
ISDN transmission interface	●
S0 (BRI) interface. Protocols: ETSI, NI-1, NTT Single codec 64 or 128 kbit/s, or double 64 kbit/s codec 5AS auto configuration on incoming calls	
Mobile network access (integrated)	P+ ●
Integrated 2G/3G/3G+/LTE network access module, 2 antenna sockets Voice mode: standard telephone or "HD Voice" (7 kHz with AMR-WB) Packet data mode: IP protocol, SIP or "Direct RTP", net bit rate 12 to 256 kbit/s	
Mobile network access (external)	●
External 3G/LTE module, connection via USB socket (data mode only)	
Wireless network access	●
External Wi-Fi device, connection via USB socket	
Audio coding algorithms (audio modes)	
G711 (standard ISDN telephony, AoIP) Mono GSM, AMR (mobile telephone) Mono AMR-WB / G722.2 (mobile "HD Voice") Mono G722 SRT, H221, H242 Mono MPEG Audio Layer II Mono, Dual mono, Stereo MPEG AAC-LC , HE-AAC Mono, Dual mono, Stereo OPUS (AoIP only) Mono, Stereo	
Audio interfaces	
2 microphone inputs with phantom power, HP filter, limiter 1 mic/line input with phantom power, HP filter, limiter 1 mic/line input with HP filter, limiter 2 auxiliary line inputs or 1 auxiliary line + 1 AES3 input 4 stereo headphone sockets for monitoring, flexible routing and volume control 1 line output 1 additional line output USB audio device interface (mono) AES67/Ravenna Ethernet interface (5 input channels / 8 output channels)	P+ P+ P+ ●
User interface	
Color TFT 7" display User input: touch screen display + rotary encoder Dedicated and assignable backlit pushbuttons, potentiometers Programmable set-up/dial memories	
Remote control and supervision	
Remote control via IP interfaces for all settings GPIO: 2 inputs / 2 outputs SNMP Remote access over Internet	 ●
General	
Power: external DC 12V, or 6xAA-type rechargeable batteries (run time > 1 h), charger included Size 212 x 236 x 102 mm (WxDxH) Weight < 2.1 kg	

Table 1 – Main features of the ScoopTeam

2. Functions

ⓘ *Reminder: this chapter details the functions and capability of the ScoopTeam, but not the detailed operating modes, developed in other chapters.*

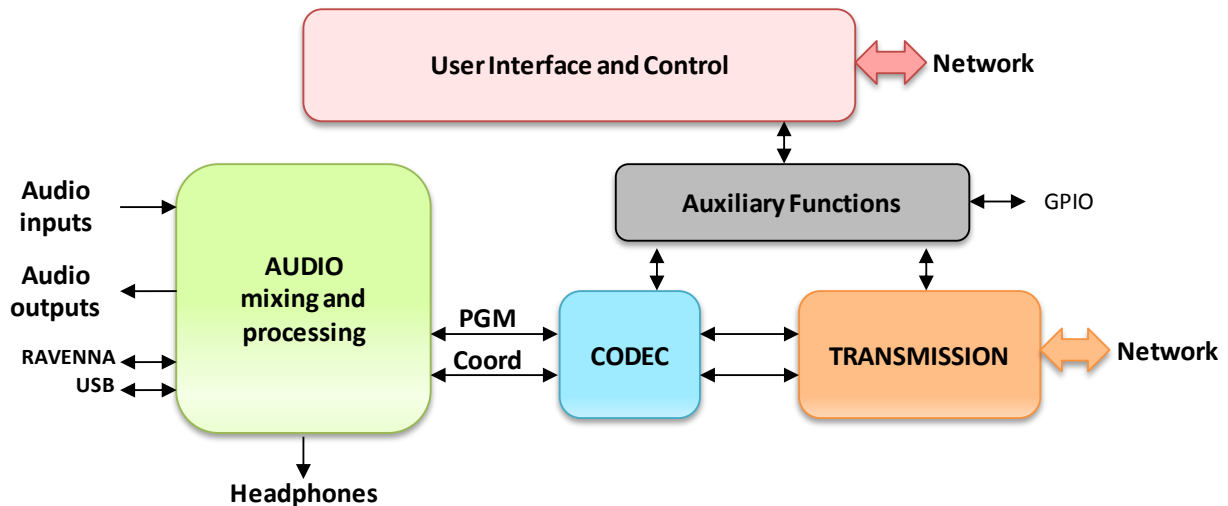The following block diagram shows the basic functions of the equipment.

Figure 1 – Functional diagram of the equipment

The AUDIO block deals with the audio interfaces and includes mixing and processing functions. Two internal mono buses "Program" and "Coordination" are generated and sent to the coding and transmission parts. Corresponding feedback signals are received from these blocks and used in the mixing part. Monitoring headphones are fed a configurable and adjustable mix of the block's input signals.

The CODEC block deals with the bit rate reduction for transmission, implementing various possible coding algorithms. Two bidirectional channels are handled.

The TRANSMISSION block handles the access and transport via the available network interface(s).

The User Interface and Control part includes the management thanks to the touch screen display and several buttons and knobs. In addition, the unit is remote controlled via the IP network interface(s), plus some GPIO.

Auxiliary functions can be optionally added, e.g. by plugging USB devices providing additional network interfaces.

2.1. Audio interfaces and functions

> ⓘ Note: this chapter does not apply to the "Premium" version and its optional AES67/Ravenna interface; these feature significant differences in their audio interfaces. Please refer to the user manual in Edition G for adequate information on these interfaces.

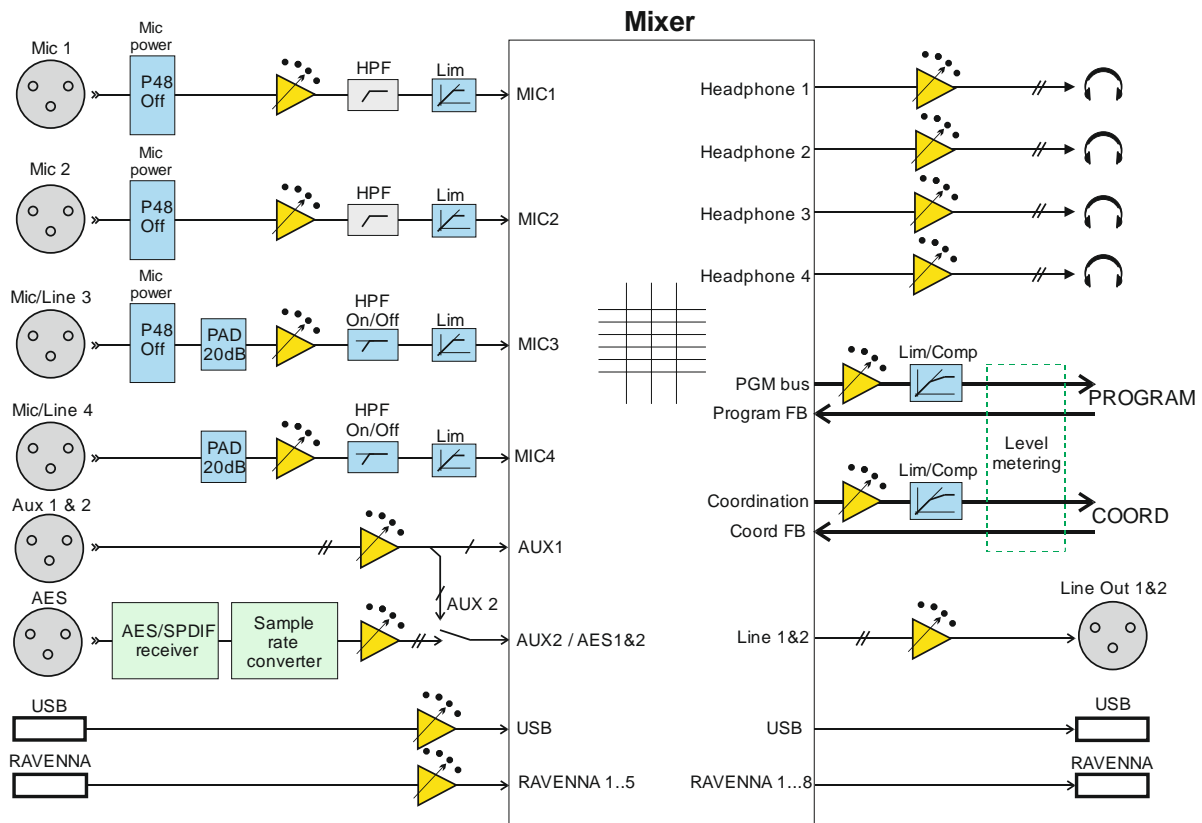The following diagram shows the main features of this functional block.

Figure 2 – Audio processing and mixing functional block

Note: in stereo operation, the "Program" and "Coord" buses become respectively "Program left" and "Program right".

2.1.1. Microphone inputs (Mic 1 and Mic 2)

Each of these two inputs has the following features:

- Balanced input, 3-pin female XLR, located on the front panel
- Phantom power (48 V, switchable)
- High-pass filter, 80 Hz, 18 dB/o (fixed, analog domain)
- Limiter (adjustable threshold, switchable, analog domain)
- Gain control +0dB to +50dB, 1 dB steps
- All parameters controllable by remote

The limiter threshold value is adjustable but common to all 4 microphone inputs.

2.1.2. Mic/Line input 3

This input has the following features:

- Balanced input, 3-pin female XLR, located on the front panel
- Phantom power (48 V, switchable)
- Mic/Line switch (pad)
- Acceptable level +22 dBu (with "line" setting, pad on)
- High-pass filter, 80 Hz, 18 dB/o (defeatable, analog domain)
- Limiter (adjustable threshold, switchable, analog domain)
- Gain control (Mic position): +0dB to +50dB, 1 dB steps
- Gain control (Line position): -30dB to +16dB, 1 dB steps
- All parameters controllable by remote

The limiter threshold value is adjustable but common to all 4 microphone inputs.

2.1.3. Mic/Line input 4

This input has the following features:

- Balanced input, 3-pin female XLR, located on the back
- Mic/Line switch (pad)
- Acceptable level +22 dBu (with "line" setting, pad on)
- High-pass filter, 80 Hz, 18 dB/o (defeatable, analog domain)
- Limiter (adjustable threshold, switchable, analog domain)
- Gain control (Mic position): +0dB to +50dB, 1 dB steps
- Gain control (Line position): -30dB to +16dB, 1 dB steps
- All parameters controllable by remote

The limiter threshold value is adjustable but common to all 4 microphone inputs.

2.1.4. Line input Aux 1

This input is only available on the "Premium+" version; it has the following features:

- Balanced input, 3-pin female XLR, located on the back
- Acceptable level +22 dBu
- Gain control -30dB to +16dB, 1 dB steps
- All parameters controllable by remote

2.1.5. AES / Aux 2 input

This input is only available on the "Premium+" version; it has the following features:

- Selectable type: analog line input or AES3 (2 channels)
- Balanced input, 3-pin female XLR, located on the back
- Acceptable level +22 dBu (for the analog mode)
- Gain control -30dB to +16dB, 1 dB steps
- All parameters controllable by remote

2.1.6. USB audio interface

This interface is a "device" interface, for a connection with a USB host such as a computer, with following features:

- Mini-B USB socket located on the back
- One input channel and one output channel
- Input gain control -30dB to +16dB, 1 dB steps, controllable by remote

2.1.7. AES67 / RAVENNA interface

This interface is only available on the "Premium+" version, and has the following features:

- Ethernet 1000BaseT, two RJ45 sockets located on the back
- Five input channels (selectable from the source streams) and eight output channels
- Input gain control -30dB to +16dB, 1 dB steps (for each channel, independently)
- All parameters controllable by remote (*some of the settings by using a separate software tool and accessing the RAVENNA/AES67 interface*)

The 8 output channels are assigned as follows:

Channel	Name	Function
1	RAV. 1	Output mix (see further)
2	RAV. 2	Output mix (see further)
3	MIC1	Direct output of Mic1 (after gain adjustment)
4	MIC2	Direct output of Mic2 (after gain adjustment)
5	MIC3	Direct output of Mic3 (after gain if Mic level)
6	MIC4	Direct output of Mic4 (after gain if Mic level)
7	AUX1	Direct output of Aux1 line input
8	AUX2	Direct output of Aux2 line input (muted if AES mode is selected for the AES/Aux2 input)

2.1.8. Line outputs

Each line output has the following features:

- Balanced output, 3-pin male XLR, located on the back
- Full scale level adjustable +4 dBu to +22 dBu, 1 dB steps, controllable by remote

Line Out 2 is only available on the "Premium+" version.

2.1.9. Headphone outputs

Four outputs are available for monitoring. Each has the following features:

- 6.35 mm jack located in the front panel
- Volume control with a dedicated knob
- Flexible routing: see further

2.1.10. Mixing capabilities

The various audio sources are mixed to produce:

- The program and coordination output signals.
- Monitoring outputs to the headphones.
- Outputs to the audio output interfaces.

The sources for mixing include:

- The signals from the local audio input interfaces, as listed previously (including the USB and RAVENNA interfaces), after the gain setting and possible channel processing.
- The feedback channels from the program link and the coordination link (Program FB and Coord FB).

Each output is a configurable mix of the available inputs. The following diagram shows the sources that are selectable in the mix for the Program and Coordination buses[1], and for the monitoring outputs.

- Each input of the ScoopTeam can be routed to the Program and/or the Coordination mix buses (except Ravenna channels 3 to 5, only used for monitoring).
- A dedicated push-button is available on the desk for the contribution of each microphone input (Mic 1, Mic 2, Mic/Line 3, Mic/Line 4) to the Program and Coordination buses.
- Each headphone monitoring output is fed with a selectable mix of all the inputs, and the Program and Coordination feedback channels. Every contribution has an adjustable gain and a routing selection Left/Center/Right ("Center"=both ears).
- For each headphone output, a dedicated knob is available on the desk for controlling the gain of the corresponding microphone in the headphone mix ("side tone" adjustment).
- A "Sound ID" can be generated internally and routed to the Program mix bus. This can provide an easy identification of the unit by the remote side when a connection is established.
  The sound ID is generated by playing back a file that can be loaded into the unit (see 4.9.10, "SETUP TRANSFER" page). The injection of this sound into the Program bus is activated via the menu (**Tools > Play Sound ID**).

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[1] in stereo operation, the "Program" and "Coord" buses become respectively "Program left" and "Program right".
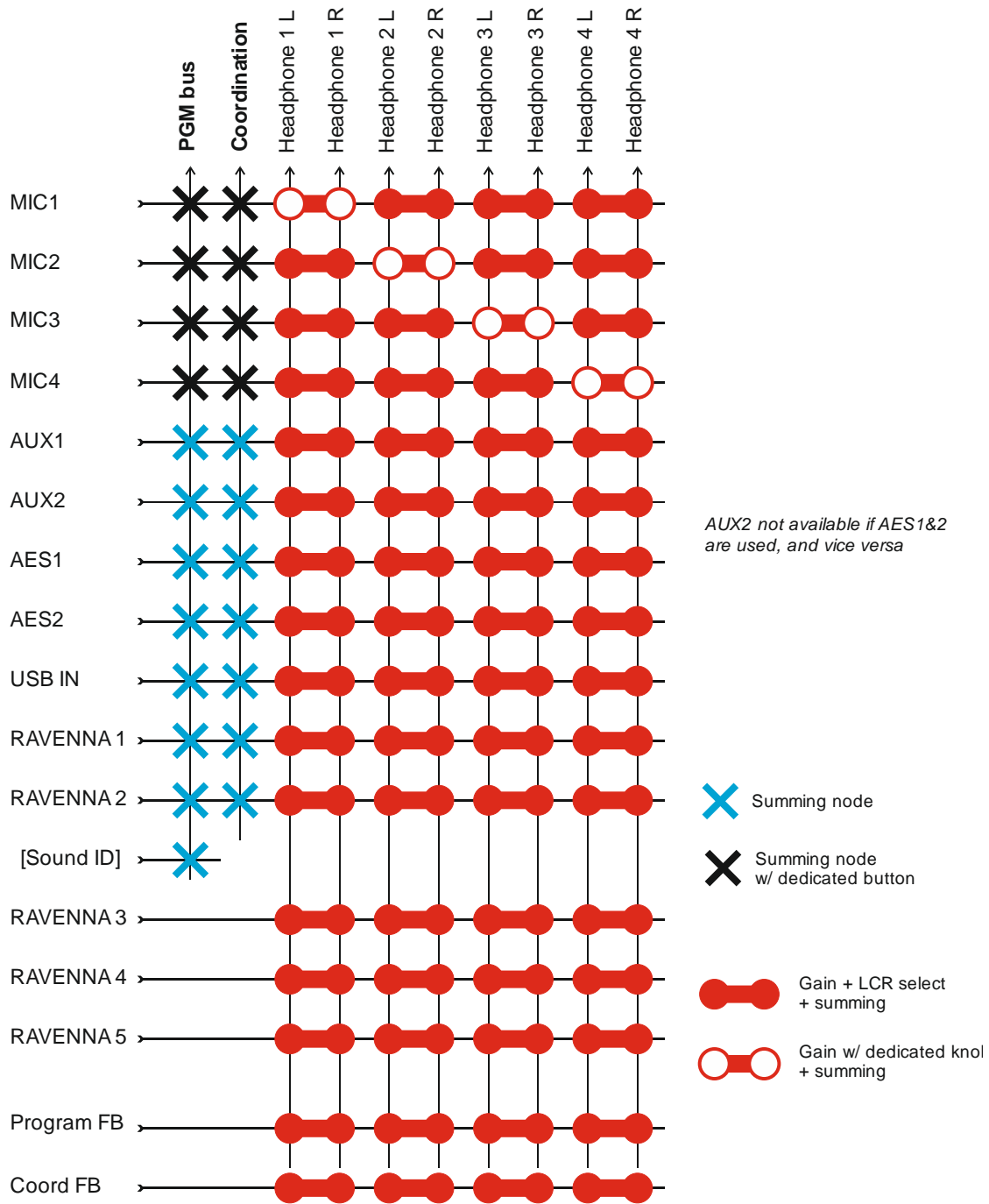
Figure 3 –Mixer diagram, main part

AUX2 not available if AES1&2 are used, and vice versa

Summing node

Summing node w/ dedicated button

Gain + LCR select + summing

Gain w/ dedicated knob + summing

A master gain and an adjustable compressor/limiter are applied on the program mix and the coordination mix. The picture below shows the adjustable parameters on the compressor/limiter. In addition to these, the attack time of the envelope detector is also adjustable.
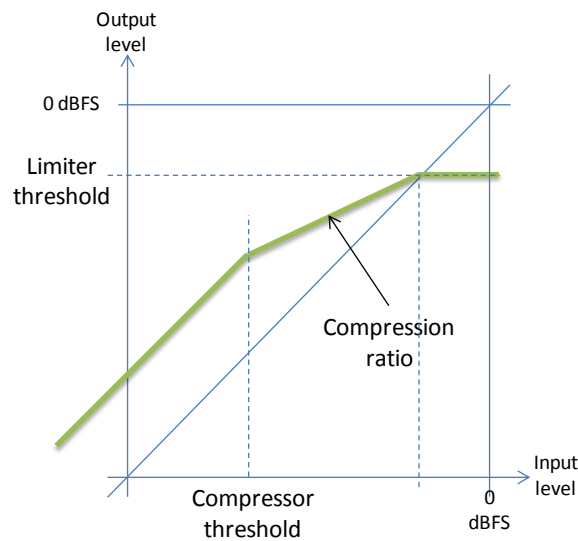
ⓘ *The compressor/limiter on the Coord bus normally runs independently from the one on the Program bus, without interchannel coupling. However, when the unit is operating in stereo mode, the two compressors/limiters are automatically linked (common gain).*

The audio output interfaces (other than headphones) get a selectable mix of the program and coordination channels and their feedback channels, as visible on the following diagram:
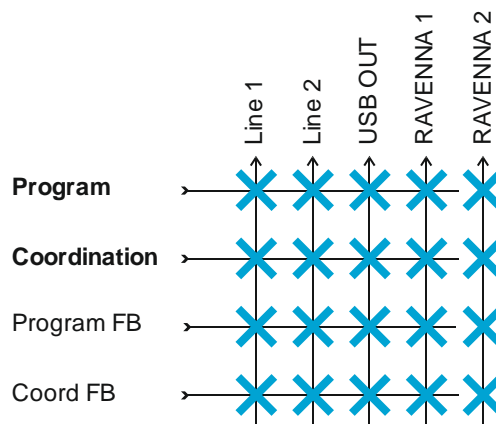

Figure 4 –Output mixer diagram

Note that the **Program** or **Coordination** signal here is the same that is forwarded to the transmission block, i.e. it is the signal picked *after the limiter/compressor*.

2.1.11. Audio level metering

The level of the following signals is monitored and displayed on the user interface:

- Program output (or "transmit") signal, picked after the compressor/limiter.
- Program feedback signal, received from the network link.
- Coordination output (or "transmit") signal, picked after the compressor/limiter.
- Coordination feedback signal, received from the network link.

The level monitoring is of PPM type, relative to full scale (full bar = 0 dBFS).

2.2. Audio encoding and decoding

ScoopTeam features several coding algorithms. Their availability depends on the configuration for transmission: network used and single/double codec mode.

ⓘ *Details about the various coding algorithms can be found in annex 6.1,"Details on the coding algorithms".*

2.2.1. Single codec mode

In this configuration, the selected network interface (ISDN or IP) is used to set up a link with a single remote device. A single encoder/decoder processes one or two mono audio channels:

- With a **mono** codec, only the program channel is processed and transmitted over the selected network.
  However, when the mobile option is installed, the mobile voice channel can be used for the Coordination channel.
- With a **dual mono** codec, two audio channels are transmitted (Program and Coordination).
- With a **stereo** codec, two audio channels are transmitted (Program left and right).

The table below synthetically describes the capabilities in the different cases:

Codec	Frequency (kHz)				Net bit rate (kbit/s)	ISDN			IP		
	16	24	32	48		Mono	Dual	Stereo	Mono	Dual	Stereo
G711					**64**	🟢			🟢	🟢	
G722	▨				**64**	🟡			🟡	🟡	
G722-H221	▨				**64**	🟡					
G722-H242	▨				**64**	🟡					
MPEG L2		▨	▨	▨	**64**	🔴	🔴	🔴	🔴	🔴	🔴
MPEG L2		▨	▨	▨	**128**	🔴	🔴	🔴	🔴	🔴	🔴
MPEG L2			▨	▨	**192**					🔴	🔴
MPEG L2				▨	**256**					🔴	🔴
AAC-LC				▨	**16 => 56**				🔴	🔴	
AAC-LC				▨	**64**	🔴	🔴		🔴	🔴	
AAC-LC				▨	**96**				🔴	🔴	
AAC-LC				▨	**128**	🔴	🔴		🔴	🔴	
AAC-LC				▨	**192**					🔴	
AAC-LC				▨	**256**					🔴	
HE-AAC				▨	**16 => 56**				🔴	🔴	
HE-AAC				▨	**64**	🔴	🔴		🔴	🔴	
HE-AAC				▨	**96**				🔴		
HE-AAC				▨	**128**		🔴	🔴	🔴		
AAC-LC LOAS				▨	**64**	🔴	🔴	🔴			
HE-AAC LOAS				▨	**64**	🔴	🔴	🔴			
OPUS					**12 => 192**				🔴		🔴

Table 2 – Single codec - Available coding depending on network

Notes:

- The "IP" case does not depend whether a wired interface (Ethernet) or a wireless interface (3G/LTE mobile, Wi-Fi) is used for the network access.

2.2.2. Double codec mode

In this configuration, two independent connections can be set over the selected network interface (ISDN or IP) towards possibly two remote devices. One mono codec is operating for each connection, one for the Program link and one for the Coordination link.

The mobile voice channel is not used in this arrangement.

The table below synthetically describes the capabilities of either codec in the different cases:

Codec	Frequency (kHz)				Net bit rate (kbit/s)	ISDN	IP
	16	24	32	48			
G711					**64**	(green)	(green)
G722	(gray)				**64**	(yellow)	(yellow)
G722-H221	(gray)				**64**	(yellow)	
G722-H242	(gray)				**64**	(yellow)	
MPEG L2		(gray)	(gray)	(gray)	**64**	(red)	(red)
MPEG L2		(gray)	(gray)	(gray)	**128**		(red)
AAC-LC				(gray)	**16 => 56**		(red)
AAC-LC				(gray)	**64**	(red)	(red)
AAC-LC				(gray)	**96**		(red)
AAC-LC				(gray)	**128**		(red)
AAC-LC				(gray)	**192**		(red)
HE-AAC				(gray)	**16 => 56**		(red)
HE-AAC				(gray)	**64**	(red)	(red)
AAC-LC LOAS				(gray)	**64**	(red)	
HE-AAC LOAS				(gray)	**64**	(red)	
OPUS					**12 => 192**		(red)

Table 3 – Double codec - Available coding depending on network

Notes:

- The "IP" case does not depend whether a wired interface (Ethernet) or a wireless interface (3G/LTE mobile, Wi-Fi) is used for the network access.

2.2.3. Mobile voice and HD voice

Application: telephony and commentaries, mobile coordination. Moderate latency.

When the unit includes the mobile option, the mobile voice channel can be used for the Coordination channel.

The standard algorithm used in the network is GSM or AMR. These algorithms are exclusively used for speech transmission over mobile telephone networks, with a 300 to 3400 Hz audio bandwidth. Whenever needed, gateways perform transcoding in order to interface with fixed PSTN, ISDN and IP networks.

When the mobile network supports the so called "HD Voice" service, the AMR-WB coding (standardised as G722.2 by the ITU-T) is used between compatible mobile terminals, and provides speech transmission with a 50-7000 Hz bandwidth. ScoopTeam automatically implements this algorithm in mobile voice mode every time it is possible, and automatically falls back to standard voice coding if not (when the network does not support HD voice, or the remote terminal is not compatible).

2.3. Audio transmission

The ScoopTeam features in all versions a couple of Ethernet interfaces for IP protocol networks. Additional interfaces are available as options: ISDN interface (BRI), mobile network access, Wi-Fi interface.

2.3.1. AoIP transmission

The ScoopTeam performs AoIP (Audio over IP) transmission in a wide range of possible bit rates, exchanging unicast streams with one (or two) remote device(s), generally bidirectional. This mode can be used on all types of networks links, LAN or WAN, including links via Internet. Audio streams are always transported under the RTP/UDP protocol. Links can be set up in three ways:

- "Peer to peer" SIP connection between two compatible units.
- Use of a SIP proxy server to set up the link, or a SIP PBX.
- "Peer to peer" connection without SIP, called "Direct RTP". This operation mode can be used for interoperation with devices that do not support SIP, or for overcoming possible network blocking that would specifically affect the SIP protocol.

The audio coding algorithm can be selected depending on the required quality and the available network bandwidth. The available algorithms are described in 2.2, Audio encoding and decoding.

SIP and SDP protocols

The SIP protocol is a signalling protocol used for live media transmission over IP connections, which allows the interoperation with IP phones and SIP compatible audio codecs, in a way similar to ISDN or telephone connections. SIP is included in the requirements of the EBU Tech3326 recommendation for audio contribution over IP, also known as "N/ACIP"".

One significant advantage is the inclusion of SDP, a protocol which allows the connecting devices to automatically negotiate and agree on the coding profile to use. Thanks to this system, it is not necessary to set the units in the same way before setting up a connection. Moreover, the calling party needs not know how the remote unit is configured before initiating a link.

ⓘ Note: the SIP protocol does <u>not mandatorily</u> imply the use of a server. Codecs can set up point-to-point links using this protocol, and benefit from some its advantages.

Conversely, for the "Direct RTP" mode, which includes no signalling, it is necessary to configure beforehand in the same way the two units to be connected.

Double AoIP codec

With the SIP protocol, the ScoopTeam can operate as two mono AoIP codecs. Each of the codecs can have its own SIP registration and is managed independently from the other codec.

The available algorithms for each codec are described in 2.2.2, Double codec mode (see Table 3 – Double codec - Available coding depending on network).

Double Streaming

Audio transmission over IP is sometimes subject to data losses which can cause noise or dropouts in the audio signal. The AETA "Double Streaming" system available on the ScoopTeam is an efficient way of strongly increasing the robustness of an AoIP link, thanks to a redundant link that can compensate for losses on the main link.

Refer to chapter 6.2, "Using the Double Streaming system", for more information on the system and how to use it.

Packet duplication

ScoopTeam also proposes an RTP transmission mode with enhanced reliability, using packet replication. When enabling this mode, every packet is transmitted twice; with such system a lost packet has no effect since the receiver still gets the other copy of the packet. In this way, stable links can be obtained even with a high packet loss rate. Of course, as a disadvantage the bit rate is double; you must make sure this stays compatible with the transmission medium.

"Factory" SIP accounts

AETA has set up a public SIP server, dedicated to Audio over IP broadcasting applications. Customers can get accounts from AETA with a yearly subscription, register their audio codecs on this server and set links at will with other registered units (more details on [AETA web site](#)).

The ScoopTeam integrates a couple of "factory" SIP accounts on this server: these are built-in and ready to use. You can load them at any time into the ScoopTeam and use them free of charge.

The factory accounts are bound to a unique ScoopTeam; they cannot be used for any other physical or virtual device.

Remote control via IP

In addition, every IP interface can be used for configuring or remote controlling the unit. Refer to chapter 2.5, "User interface and control" for more information on this feature.

2.3.2. Ethernet/IP interfaces

These IP interfaces are 10BaseT/100BaseT Ethernet interfaces used for audio transmission and/or remote control.

Typically the main Ethernet interface is used for AoIP transmission. The auxiliary Ethernet interface can be used for a separate control interface, or for a redundant link (Double Streaming system).

2.3.3. ISDN interface

For access to the ISDN, the transmission interface is one S0 BRI (Basic Rate Interface), for transmission over one to two 64 kbit/s B channels. Thus, the total available bit rate ranges is 64 or 128 kbit/s.

In "double codec" mode, each of the possible links uses one B channel on this single line.

Network protocols

Available protocols:

- "Euro ISDN" (or ETSI), default protocol valid for a large number of countries, especially all over Europe.
- "NTT": valid for the Japanese network of NTT
- "NI-1": valid for numerous operators in North America. This choice is also suitable for the connection to network equipment with "NI-2" protocol".

ⓘ *In North America (USA and Canada), the available interface is often a U0 interface (instead of S0). In such case an "NT1" network adapter must be inserted between the line and the ScoopTeam. Such adapter can be found on the local market.*

5A System

Setting an ISDN connection is often difficult, at least because of the numerous coding parameters to be set. Moreover, with most proprietary algorithms, it is mandatory for the two devices to have exactly the same settings, otherwise the connection will fail, and sometimes it is not easy to find out the reason.

5A stands for Aeta Audio Advanced Automatic Adjustment. This system makes it easier to set an ISDN connection, because the codec, on receiving a call, automatically adjusts itself, following the calling party algorithm and parameters.

When the 5A System is enabled on the unit and a call is received, the unit first detects the coding algorithm used by the calling codec, and also senses its parameters: audio mode (mono, dual mono…), sampling rate, bit rate, inverse multiplexing protocol, etc. Then the unit can decode the compressed audio from the remote unit. In addition, the unit will use these same settings for encoding and sending audio to the remote unit, so that the remote unit can also decode the outgoing audio program. The whole process just takes a few seconds. Of course, all compatible coding configurations can be detected automatically by the 5A System.

Note that the 5A system is only relevant for ISDN connections.

2.3.4. Mobile network access

Units equipped with the "Mobile" option include an integrated module for access to mobile networks, and a holder for a SIM card.

The accessible networks are 2G (GSM, EDGE), 3G (UMTS), 3G+ (HSDPA, HSUPA, HPA…), and 4G/LTE.

For the operation, at least one multiband antenna (to be selected for compliance with the mobile network characteristics) must be connected on ScoopTeam. An antenna "diversity" feature gives an opportunity to improve the stability when the radio reception quality is poor, by connecting a second antenna.

Mobile IP mode

The first service available with mobile access is the data packet transmission mode, with IP protocol.

This mode brings similar capabilities as a wired IP connection via the Ethernet interface, as described above in 2.3.2, with some distinctive characteristics:

- This requires a subscription including access to the data service, with conditions compliant with the application. Among other requirements, an APN (Access Point Name) must be provided that allows this type of media stream. *Some operators provide such allowance in a "VoIP" option.*

- The available bit rate depends on various factors; first the network technology (2G/3G/3G+/4G…), but also the traffic level in the radio cell, the operator's network capacity, possibly the type of subscription. This may bring on restrictions for the usable compression algorithms.

- Setting a link implies first activating the data connection, before actually initiating an audio stream transmission link.

Mobile voice mode – HD Voice

The integrated module also allows to use the mobile phone service, for communicating with all ISDN or PSTN telephone terminals or hybrids, or with other mobile terminals. The quality is in such case that of mobile connections, with a 300-3400 Hz bandwidth and coding such as GSM, EFR, AMR...

Now many mobile networks also propose "HD Voice", an extension of this mobile telephone service. With this new capability, compatible terminals implement the AMR-WB coding algorithm (standardised as G722.2 by the ITU-T) and provide speech transmission with a 50-7000 Hz bandwidth and a quality very similar to the well-known G722. Automatic fallback to the standard coding takes place if the network does not support the service or one of the terminals does not feature this capability.

In single codec mode, the mobile voice channel can be used for coordination.

Using an external USB module

Instead of the integrated module, it is possible to plug a USB mobile module or "key", in order to access mobile IP transmission, with more or less similar conditions as described above.

ⓘ *However be aware:*

- The USB module must be from the list of devices supported by AETA. *As this list may evolve, please check our web site for up-to-date information.*

- The "Mobile Voice" mode is not available in this way.

- USB devices do not feature antenna diversity.

2.3.5. Wi-Fi interface

A specific USB device (provided with the Wi-Fi option) can be used to provide access to a Wi-Fi network.

It is possible to connect to an access point by entering (or selecting) the SSID and the password for the network. *Access control requiring a certificate and/or a login via a web browser is not supported.*

The Wi-Fi connection can be used for remote control links and/or for AoIP connections.

2.3.6. Managing audio links

The audio transmission implies a link/session setup phase.

One of the transmission interfaces is selected as the *default interface* on the ScoopTeam. A call towards a remote unit, initiated by the user of the ScoopTeam, is implicitly sent through this default interface. The default interface can also receive and accept incoming calls from a remote device. The ScoopTeam automatically picks up such calls.

On the other hand, an incoming call on an interface other than the default interface may be accepted under certain conditions. These are described more precisely in 4.4.5, "Receiving calls". In such case, the ScoopTeam first switches to the suitable interface, and then processes the incoming call. When the link is released, it will come back to its previous state (and default interface). *Note: If the unit is in double codec mode and one codec is busy, the other codec is not available for calls on the other network interface (e.g. IP link active, no ISDN call is acceptable).*

The detailed procedure for setting a link can be found in 4.4, "Setting up a link".

2.4. Auxiliary transmission: "relays"

In addition to the main function of the ScoopTeam, which is the transmission of one or two main audio channels, auxiliary functions are available.

During an audio connection, two binary states can be transmitted end-to-end along with the audio stream.

The transmitted data (also called "Contacts") can be set via the user interface, using keys or a menu, or set by input GPIs (see further in 2.5.5, GPIO).

The received states (also called "Info") can be displayed on the screen and/or can activate relays/GPO (see further in 2.5.5, GPIO).

This feature is only compatible with AETA products, because it is not covered by independent standards. It is available depending on the transmission context, as follows:

- For IP transmission and the SIP protocol, the feature is always available whatever the coding algorithm.
- For ISDN transmission, the feature is available only for G722-H242 .
- The feature is not available in other configurations, including mobile voice transmission.

In double codec mode, this feature is only available for Codec 1.

2.5. User interface and control

The control and supervision of the equipment (configuration, communication management, status monitoring), is carried out in two general ways: "local" user interface using the physical display and input devices, or "remote" control.

2.5.1. User interface

The user interface elements are as follows:

- Color 7" TFT display with touch screen
- One rotary encoder to adjust settings
- Monitoring shortcut key and Communication shortcut key
- Dedicated backlit keys for each commentator position: Program key, Coord key
- Dedicated potentiometers for volume control for each commentator position: general headphone volume, side tone volume
- For commentator positions 1 and 2, assignable potentiometers, 4 for each position. These knobs can be configured for directly controlling the gain of selected sources in the monitoring mix.
- 7 assignable backlit function keys. These keys can be assigned various functions, as detailed further in 2.5.6, Assignable function keys.

The operation is described further in the following chapters.

2.5.2. Remote control

Basically all the parameters of the unit can be remote controlled via IP, using the unit's control protocols. Several control tools are available for operating the unit by remote.

- The dedicated application **My ScoopTeam LE** provides complete control over the unit from a Windows computer, with a graphical user interface very similar to the unit's display. "My ScoopTeam LE" can control the ScoopTeam when it is connected on the same LAN as the computer. See in chapter 4.10 (Operating mode – "My ScoopTeam" remote control application) the detailed operating mode. "My ScoopTeam LE" is available as an optional accessory.

- **My ScoopTeam RE** is an application with a very similar user interface, but it differs in the fact that it can reach the ScoopTeam across the Internet, via the AETA Remote Access server (see further the remote access system in 6.3, "Remote access". Chapter 4.10 (Operating mode – "My ScoopTeam" remote control application) describes the operating mode. The ScoopTeam must include the "Remote Access +" option for being accessible by "My ScoopTeam RE".

- An embedded html server provides html pages that enable partial control using a web browser, via port 80 (default port for the HTTP protocol) or port 443 (default port for the HTTPS protocol). This control mode can be used from any computer regardless of its OS (or a mobile device with a web browser), and the embedded pages are compatible with all current browsers. No software installation is needed on the control device. However, this tool has a smaller range of features. With the "Remote Access" option (or the "Remote Access +" option), this kind of remote control is also possible via the Internet.

- For supervision needs, a management system can make use of the SNMP protocol for accessing the essential features and status of the ScoopTeam. Trap messages can be configured for the most relevant events.

2.5.3. Remote control interfaces

The unit can be remote controlled from its IP interfaces:

- The main Ethernet interface, used for audio streaming, can be used for remote control as well.
- Auxiliary Ethernet interface (10/100BaseT).
- Data connection over a mobile network; this can be used for remote control, but this requires remote access to be set up (see below, 6.3).
- Wi-Fi connection (see above in 2.3.5, Wi-Fi interface).

2.5.4. Remote access

Thanks to the AETA remote access server, it is possible to remotely access the html interface of the ScoopTeam, even through the Internet and if the ScoopTeam is behind a NAT router.

The principle of this remote access and its mode of operation are described in Annex 6.3 (Remote access). A "Remote Access" option is required on the module.

Even without this option, the function allows if necessary, and on request, remote assistance by AETA.

2.5.5. GPIO

ScoopTeam features two isolated current loops ("contacts"), and two isolated outputs ("relays").

These I/O have two classes of usage:

- The GPIO can be used in relation to the "relay transmission" function (as described in 2.4, "relays"). A GPI can be configured to define such a transmitted state, and a GPO can be set to follow a received state.
- Alternatively, a GPI can be allocated a programmable specific action (e.g. "Hang up"), or a GPO can be allocated a programmable status (e.g. "On line"). The following table shows the available features.

Interface	Available feature	Description
GPI 1	Disabled	
	Contact 1	Relay transmission: transmitted "contact" 1
	Call	Call/Recall on codec 1 / Program
	Hangup/Release	Release link on codec 1 / Program
GPI 2	Disabled	
	Contact 2	Relay transmission: transmitted "contact" 2
	Call	Call/Recall on codec 1 / Program
	Hangup/Release	Release link on codec 1 / Program
	Call 2	Call/Recall on codec 2 / Coordination
	Hangup/Release 2	Release link on codec 2 / Coordination
GPO 1	Disabled	
	INFO 1	Relay transmission: received "Info" 1
	Line status	Codec 1 / Program connected
	Sync status	Codec 1 / Program connected and synchronised
GPO 2	Disabled	
	INFO 2	Relay transmission: received "Info" 2
	Line status	Codec 1 / Program connected
	Sync status	Codec 1 / Program connected and synchronised
	Line status 2	Codec 2 / Coordination connected
	Sync status 2	Codec 2 / Coordination connected and synchronised

Table 4 – Possible uses of the GPIO

2.5.6. Assignable function keys

Each assignable backlit function key can be assigned a function among the following categories.

Cough key n

When this function is assigned, pushing the key mutes the microphone input n (n=1 to 4).

The action is "temporary" (the microphone comes back to its previous state when the key is released).

The backlight is on while the button is pressed.

Talk to speaker

This function allows the signal from the commentary position i (i=1 to 4) to be sent to the headphone of the commentary position j (j=1 to 4, but j≠i). The signal is mixed over the current mix on headphone j, while the other sources are dimmed by 12 dB (or totally muted, see setting in the "Audio" menu). Conversely, the signal is removed from the Program mix bus, even if it is active in the current settings.

The action is "temporary" (come back to the previous state when the key is released).

The backlight is on while the button is pressed.

Talk to output

This function allows the signal from the commentary position i (i=1 to 4) to be sent to an audio output of the ScoopTeam. The signal is mixed over the current mix on the selected output, while the other sources are dimmed by 12 dB (or totally muted, see setting in the "Audio" menu). Conversely, the signal is removed from the Program mix bus, even if it is active in the current settings.

The action is "temporary" (come back to the previous state when the key is released).

The backlight is on while the button is pressed.

Reminder: the possible (mono) outputs are: line output 1, line output 2, USB audio output, Ravenna output stream (channel 1), Ravenna output stream (channel 2).

Aux to Bus

This function allows to insert/mix one of the auxiliary inputs into the Program mix bus, or into the Coordination mix bus.

The button acts in a "toggle" mode (switch on/off each time the button is pressed).

The backlight is on when the mix is enabled.

Reminder: the possible (mono) auxiliary inputs are: USB input, line 1 input, line 2 input, AES input (channel 1), AES input (channel 2), Ravenna input stream (channel 1), Ravenna input stream (channel 2).

Contacts

This function allows to set on/off the "Contact" n (n=1 or 2) sent to the remote destination during the transmission, and/or show its current state.

The button acts in a "toggle" mode (switch on/off each time the button is pressed). *It is not active if the contact has been assigned to a GPI.*

The backlight reflects the state of the "Contact".

INFO

This function allows to show the current state on/off of "INFO" n (n=1 or 2).

The button is unused in this assignment.

The backlight reflects the state of the "INFO".

Shortcuts

This function allows to go directly to a specific page or menu branch of the user interface.

The backlight is unused with such key assignment.

2.5.7. Configuration and dialling memories

To ease the operation, it is possible to store configuration memories, called "profiles". These belong to four categories:

- "Call profiles", including the parameters for calling a given destination: dial number(s), coding algorithm, etc. Such profile is similar to a phonebook entry, but in addition coding parameters can be stored as well. Recalling a profile directly configures the codec and/or sets an outgoing call with the parameters previously recorded by the user in the profile.

- "Presets" which memorise the network access characteristics. Recalling a preset is a quick way to recover the configuration needed for connecting on a given line/network/site.

- "SIP Profiles" which memorise the data for registering on a SIP server. Recalling such profile allows to quickly register the unit on a new server/account.

- "Snapshots" which memorise all the settings for the audio interfaces and processing.

These various profiles can be used locally and also via remote control, and they can be imported/exported from/to a computer.

2.5.8. Access control

The unit provides selective access control: it is possible to configure differentiated access to the unit's functions.

ScoopTeam provides 3 access levels:

- "Administrator" level: full control granted.

- "Expert" level: the most critical parameters are not accessible but all audio settings are granted control with this access level.

- "Operator" level: only the "live operation" settings are accessible in this mode.

The user selects the desired access level, and a password must be entered to access it. Each level is accessed with a dedicated password. Only the "Administrator" can set the passwords.

A rescue procedure is available in case the administrator password is lost.

A blank password can be set as well for each level; in such case the concerned access level is granted without having to "log in".

Details on the allowed actions for each access level can be found in chapter 4, Detailed operating mode – User interface.

3. Physical description and operation principles

ⓘ *This chapter **physically** describes the device, shows the main operation **principles**. Details on the operation modes are provided in chapter 4.*

3.1. General principles – Control means

The equipment control and supervision (configuration, status monitoring) is possible either in "Local" mode (using the unit's keys, knobs and display), or "Remote control" mode, thanks to an IP interface.

As a general rule, the configuration parameters are saved in non-volatile memory, and restored when the unit is powered on.

Local mode operation is described in detail in chapter 4 (Detailed operating mode).

Remote control operation using a computer and a dedicated application is detailed in chapter 4.10: Operating mode – "My ScoopTeam" remote control application. Alternatively, most of the functions can be controlled from an html browser, using the embedded HTML pages.

The unit includes some GPIO; their assignable functions include control functions, e.g. call trigger or release, etc. This is described in 2.5.5, GPIO.

In this document we use the following naming conventions for control and status elements:

- **Knob** / **potentiometer** / **pot**: rotating knob ("real" physical device) used as input device. Normally used for entering/adjusting a continuous parameter.

- **Virtual knob / Virtual pot:** virtual potentiometer on the touch screen. Normally linked to a continuous gain parameter. The linked parameter is adjusted by tapping the virtual pot (to select the parameter) and using the *rotary encoder* for the adjustment.

- **Key** / **pushbutton**: "real" physical push-button.

- **Backlit key** / **backlit pushbutton**: "real" physical push-button, including a backlight.

- **Button** / **Soft button** / **Touch button** / **Virtual button**: virtual button on the touch screen, triggering an action when tapped. May include a virtual indicator

- **Virtual indicator**: indicator on the screen showing a state.

- **Assignable** device: input/output device that can be assigned a programmable action, parameter, or status.

- **Dedicated** device: input/output device that is permanently linked to a unique action, parameter, or status.

3.2. Physical description of the equipment

The ScoopTeam is housed in a compact console-shaped chassis. It is powered from a 12V DC source, or from a set of rechargeable batteries (charging circuit included in the product).

3.2.1. Front panel

The front side provides sockets for all the connections needed by the commentators using the device (except the 4th microphone input).

1. Microphone/Line input (3-pin female XLR), for commentary position 1
2. Headphone socket (1/4" or 6.35 mm stereo jack), for commentary position 1
3. Headphone socket (1/4" or 6.35 mm stereo jack), for commentary position 3
4. Microphone/Line input (3-pin female XLR), for commentary position 3
5. Headphone socket (1/4" or 6.35 mm stereo jack), for commentary position 4
6. Headphone socket (1/4" or 6.35 mm stereo jack), for commentary position 2
7. Microphone/Line input (3-pin female XLR), for commentary position 2

3.2.2. Top panel: control surface

All the elements for local control are located on the top panel:
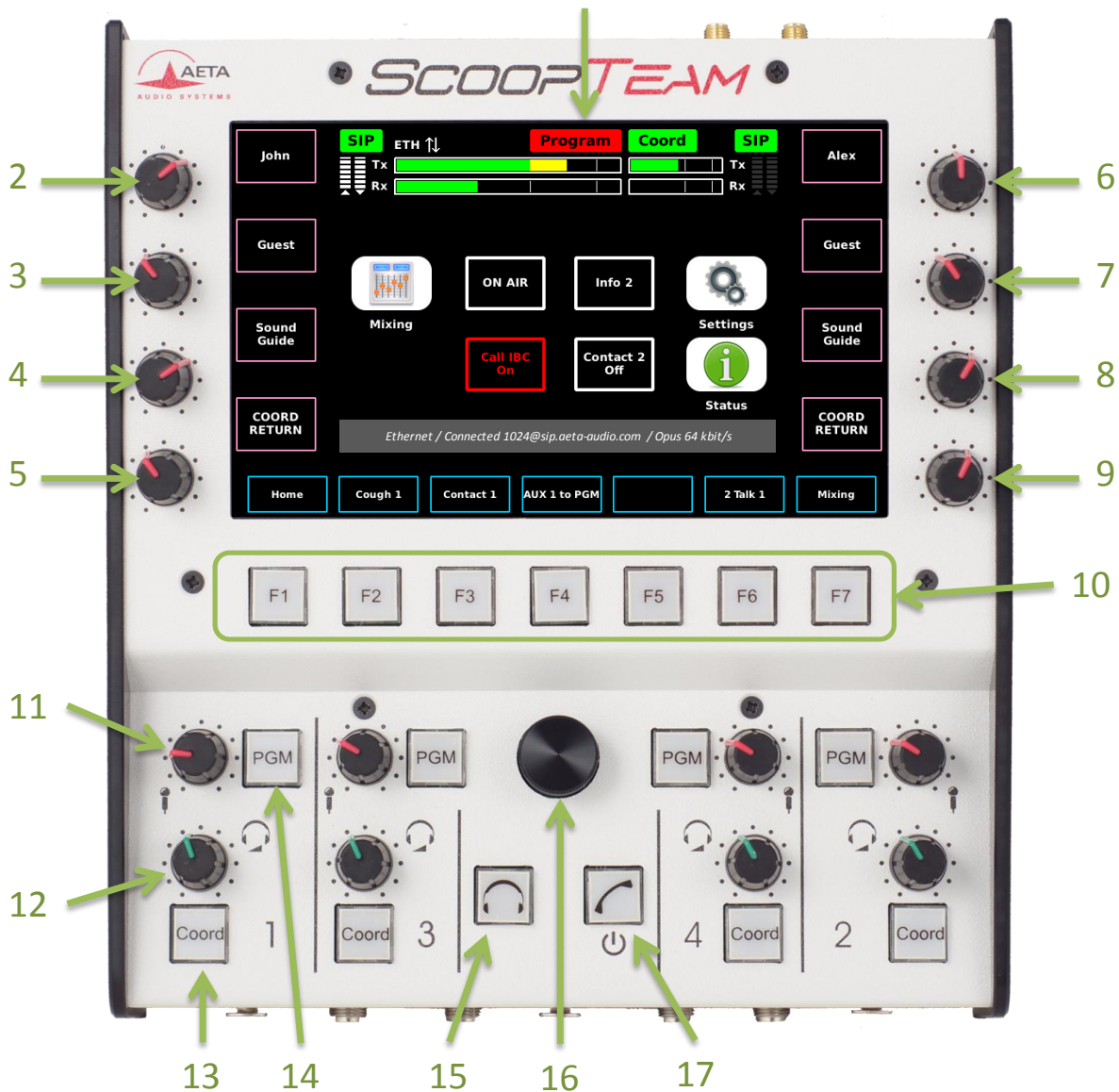
This control desk is divided in two areas. The upper part includes the following elements:

1. Touch screen display
2. Assignable knob (reserved for commentary position 1)
3. Assignable knob (reserved for commentary position 1)
4. Assignable knob (reserved for commentary position 1)
5. Assignable knob (reserved for commentary position 1)
6. Assignable knob (reserved for commentary position 2)
7. Assignable knob (reserved for commentary position 2)
8. Assignable knob (reserved for commentary position 2)
9. Assignable knob (reserved for commentary position 2)
10. Assignable backlit pushbuttons (x7)

The lower part includes several dedicated elements:

11. Volume knob, controlling the "side tone" on position n°1: amount of input n°1 that is fed to the headphone output n°1. *Note that this contribution to the headphone output may be muted as long as the input is not routed to a mixing bus, depending on the mode selected for "side tone". Please refer to 4.7.4, "Audio sub-menu", about this topic.*
3 other similar knobs are also available for the other commentary positions.

12. Volume knob for the headphone on position n°1.
3 other similar knobs are available for the other commentary positions.

13. "Coord" key for commentary position n°1.
3 other similar keys are available for the other commentary positions.

14. "PGM" key for commentary position n°1.
3 other similar keys are available for the other commentary positions.

15. "Monitoring" key: pushing this key directly calls the "Monitoring" page on the display.

16. Rotary encoder.

17. "Power/Connections" key: pushing this key directly calls the "Connections" page on the display.

The following describes some details about the various control elements.

Display [1]

The display is a 7" TFT color display, with touch screen. The brightness of the display is adjustable.

Assignable knobs [2] to [9]

The four knobs on the left hand side are used for commentary position 1. Each of these knobs can be configured for directly controlling the gain of one of the sources in the monitoring mix for this position. Once configured, the knob is permanently linked to the selected gain, regardless of the current page or menu displayed on the screen.

The four knobs on the right hand side are used for commentary position 2. Each of these knobs can be configured for directly controlling the gain of one of the sources in the monitoring mix for this position. Once configured, the knob is permanently linked to the selected gain, regardless of the current page or menu displayed on the screen.

Assignable backlit pushbuttons [10]

Each of these 7 keys includes a red backlight. The switch and the light can be assigned a configurable function. The available functions are described in 2.5.6, Assignable function keys.

For operation convenience, some functions are not available on all the assignable keys. The following table shows the allowed assignments.

Category	Function	F1	F2	F3	F4	F5	F6	F7	Button	Light
Cough key	Cough 1	■	■	■					Temporary	State
	Cough 2					■	■	■		
	Cough 3		■	■	■					
	Cough 4				■	■	■			
Talk to Speaker	1 Talk 2	■	■	■					Temporary	State
	1 Talk 3	■	■	■						
	1 Talk 4	■	■	■						
	2 Talk 1					■	■	■		
	2 Talk 3					■	■	■		
	2 Talk 4					■	■	■		
	3 Talk 1		■	■	■					
	3 Talk 2		■	■	■					
	3 Talk 4		■	■	■					
	4 Talk 1				■	■	■			
	4 Talk 2				■	■	■			
	4 Talk 3				■	■	■			
Talk to Output	1 Talk USB	■	■	■					Temporary	State
	1 Talk LINE1	■	■	■						
	1 Talk LINE2	■	■	■						
	1 Talk RAV1	■	■	■						
	1 Talk RAV2	■	■	■						
	2 Talk USB					■	■	■		
	2 Talk LINE1					■	■	■		
	2 Talk LINE2					■	■	■		
	2 Talk RAV1					■	■	■		
	2 Talk RAV2					■	■	■		
	3 Talk USB		■	■	■					
	3 Talk LINE1		■	■	■					
	3 Talk LINE2		■	■	■					
	3 Talk RAV1		■	■	■					
	3 Talk RAV2		■	■	■					
	4 Talk USB				■	■	■			
	4 Talk LINE1				■	■	■			
	4 Talk LINE2				■	■	■			
	4 Talk RAV1				■	■	■			
	4 Talk RAV2				■	■	■			
Aux to Bus	USB to PGM	■	■	■	■	■	■	■	Toggle	State
	USB to Coord	■	■	■	■	■	■	■		
	AUX1 to PGM	■	■	■	■	■	■	■		
	AUX1 to Coord	■	■	■	■	■	■	■		
	AUX2 to PGM	■	■	■	■	■	■	■		
	AUX2 to Coord	■	■	■	■	■	■	■		
	AES1 to PGM	■	■	■	■	■	■	■		
	AES1 to Coord	■	■	■	■	■	■	■		
	AES2 to PGM	■	■	■	■	■	■	■		
	AES2 to Coord	■	■	■	■	■	■	■		
	RAV1 to PGM	■	■	■	■	■	■	■		
	RAV1 to Coord	■	■	■	■	■	■	■		
	RAV2 to PGM	■	■	■	■	■	■	■		
	RAV2 to Coord	■	■	■	■	■	■	■		
Contacts	Contact 1	■	■	■	■	■	■	■	Toggle	Status (On=Closed)
	Contact 2	■	■	■	■	■	■	■		
Info	Info 1	■	■	■	■	■	■	■	-	Status
	Info 2	■	■	■	■	■	■	■		
Shortcut	Home	■	■	■	■	■	■	■	Pulse	-
	Mixing	■	■	■	■	■	■	■		
	Inputs	■	■	■	■	■	■	■		
	Network	■	■	■	■	■	■	■		
	Coding	■	■	■	■	■	■	■		
	Audio	■	■	■	■	■	■	■		
	Book	■	■	■	■	■	■	■		
	Tools	■	■	■	■	■	■	■		
	Status	■	■	■	■	■	■	■		

Table 5 – Available functions for the assignable keys

Dedicated volume knobs [11] *and* [12]

One headphone volume knob [12] is available for each commentary position.

One "Side tone" knob [11] is also available for each commentary position; this knob controls the amount of the commentator's voice mixed into the commentator's headphone.

"PGM" and "Coord" buttons [14] *and* [13]

Each commentary position features:

- A "PGM" backlit pushbutton [14] that controls the insertion of the commentator's microphone (or line signal) into the program send mix. The key is in a "toggle" mode. The red backlight is on when the signal is injected into the program.

- A "Coord" key [13] allows inserting the commentator's microphone into the coordination send mix. The push-button is in "temporary mode": the signal is sent as long as the key is depressed, and muted as soon as the key is released. Conversely, the possible contribution to the program mix is muted while "Coord" is pressed; when it is released the routing of the input comes back to its previous state. The yellow backlight is on when the signal is injected into the coordination channel (this may also be permanent if the channel is set for permanent injection into the coordination send mix).

"Monitoring" key [15]

Pushing this key directly calls the "Monitoring" page, that deals with the setup and mix for the headphone monitoring signals.

ⓘ *Conversely, if the* Monitoring *page is already displayed, pressing this key again brings you back to the previous page (that was active before switching to the monitoring page).*

Rotary encoder [16]

This knob is used as an input device when setting parameters in the menus. It includes a switch and it can also be depressed to confirm or switch a selection. Typical uses are:

- Adjusting gains
- Scrolling through lists

"Power/Connections" key [17]

In operation, pushing this key (briefly) directly calls the "Connections" page.

ⓘ *Conversely, if the Connections page is already displayed, pressing this key again brings you back to the previous page (that was active before switching to the connection page).*

This key is also used for switching the unit on/off:

- The key is backlit (blue) when the unit is off and a DC power source is connected.
- Depress the key for switching on the unit.
- For switching off, hold the key down for 2 seconds; a pop-up message asks for confirmation. Hit the "OK" button to switch off the unit, or touch the screen anywhere else to cancel.
- *If you keep the key depressed for more than 5 seconds, the ScoopTeam is forcefully rebooted; this procedure can be used in case the unit hangs.*

3.2.3. Rear panel

All connections are done on the rear panel of the codec. The characteristics of the interfaces and layout of the sockets are detailed in chapter 5.1, Characteristics of audio interfaces.
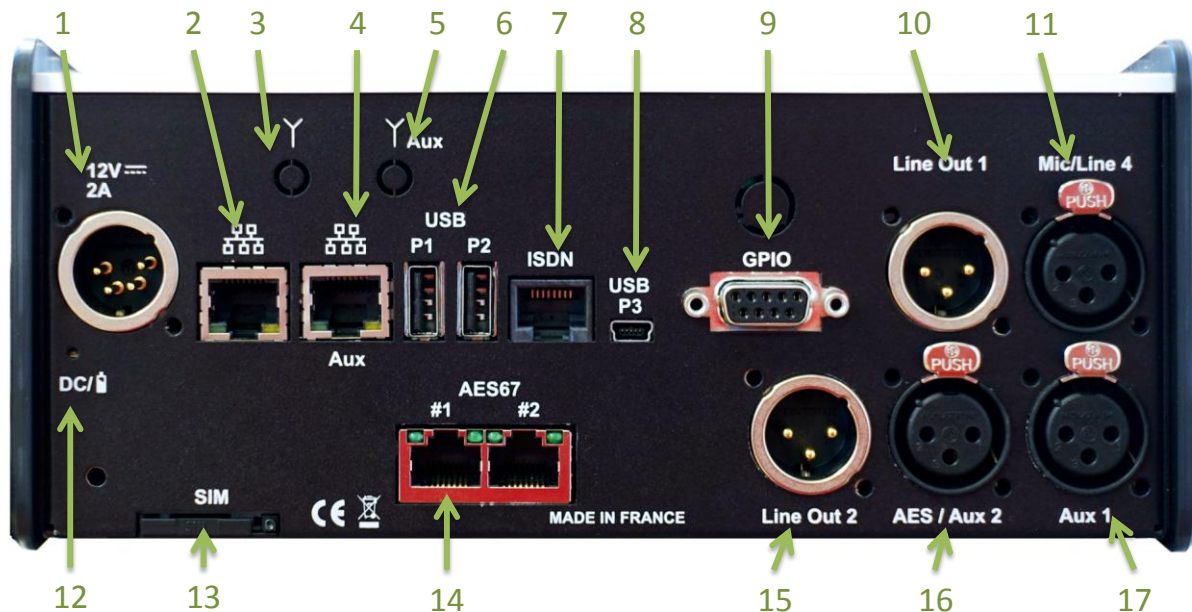
The following elements are available (*numbers such as [16] refer to the above picture)*:

DC 12V power socket [1]

4 pin male XLR. When power is applied, the unit stays off, but the internal charger circuits recharge the batteries if there are in the battery compartment. To power it you must use the Power/Connection pushbutton. The LED indicator [12] is green when DC is connected and red when charging the batteries. *See details about the power supply in chapter 5.4, "Power supply".*

Ethernet interfaces [2] and [4]

These interfaces are 100BaseT/10BaseT ports, used for audio transmission over IP and/or for remote controlling the unit. These RJ45 socket are devised for normal "straight" cables to an Ethernet hub or switch. The two integrated LEDs show the presence (green LED) and activity (yellow LED) of the network.

The primary interface [2] is normally used for the audio transmission over IP. Remote control is also possible through this interface.

The auxiliary interface [4] can be used in addition, typically for remote control on a separate IP network.

The configuration of the interfaces is described in 3.3, Initial setup of the Ethernet interface.

Antenna sockets [3] and [5]

These interfaces are only available when the unit is in "premium" version with the integrated mobile access option.

On the products fitted with the mobile network access option, two SMA sockets allow to connect one or two antennas (two multiband antennas are included with the mobile network option).

At least an antenna must be plugged on the main socket [3]. A second antenna on the "Aux" socket [5] is optional, but it allows to improve the reception quality in less favourable areas.

The antennas must cover the band(s) used for the operator and network services. In doubt, refer to the operator. The provided antennas cover bands 1, 2, 3, 7, 8, 20 (800 MHz, 900 MHz, 1800 MHz, 1900 MHz, 2100 MHz, 2600 MHz). They are compatible with almost all the 2G/3G/4G networks in Europe.

USB host sockets [6]

These "host" sockets allows the connection of peripheral devices, such as a mobile access USB module or "key" in order to access mobile IP transmission, or a Wi-Fi adapter key, etc.

Refer to chapter 2.3.4 about using a USB mobile access device.

ISDN socket [7]

This RJ45 socket allows the connection to the ISDN. The wiring is for connecting an S0 (BRI) ISDN line using a standard RJ45 cable.

USB device socket [8]

This interface is a mini-B USB socket that allows the connection to a host computer. In such case the ScoopTeam is linked as an audio interface. The host computer gets one audio input and one audio output from this interface.

"GPIO" socket [9]

This 9-pin female sub-D socket is the interface for the GPIO, as described in 2.5.5, GPIO.

Its wiring is described in:5.3.2 - GPIO interface.

Audio inputs/outputs

- Line Out 1 [10]: male XLR socket.
- Microphone/Line 4 input [11]: 3-pin female XLR, for commentary position 4.

Drawer for SIM card [13]

This is only available when the unit is in "premium" version.

In the units fitted with the mobile network access option, this drawer houses the SIM card enabling the access to the mobile network and services.

ⓘ *The SIM card must be inserted while the unit is off (or in standby).*

To remove the drawer and the card, use a pen or a pencil and push the small button shown by the arrow on the picture below:

To set the SIM card in place, first insert it in the cavity inside the drawer. Make sure to have the cut corner on the right place, with the SIM chip visible. Then insert the whole assembly into the slot, with the SIM card chip facing up. Check that the drawer is well in its guides before pushing it completely in place.

ⓘ *If you have to use a smaller format µSIM or nanoSIM card, you can use a SIM/µSIM adapter (available on request from AETA): first set the small card inside this adapter, afterwards use this assembly like a regular SIM card.*

***RAVENNA/AES67 interfaces* [14]**

These interfaces are only available when the unit is in "Premium+" version.

These interfaces are 1000BaseT ports. The unit can get five audio inputs from this AES67 interface, and transmits eight audio outputs via this interface.

Extension audio inputs/outputs

These audio interfaces are only available when the unit is in "Premium+" version.

- Line Out 2 **[15]**: male XLR socket.

- Auxiliary line input: "Aux 1" **[17]**

- "AES / Aux2" **[16]**: female XLR socket switchable as either a digital input in AES/EBU format, or an auxiliary line input.

3.2.4. Bottom: battery compartment

The battery compartment is accessible on this side of the unit. It houses 6 AA-type (R6) batteries.

Only fast charge NiMH batteries are allowed, such as those proposed by AETA as accessories (see further, 5.8).

An internal charger recharges the batteries when the external DC source is connected (regardless if the unit is on or not). It is possible to disable it. To do so, slide towards the front the switch located on the bottom of the unit near the battery compartment. The picture below shows its normal position ("CHARGE").
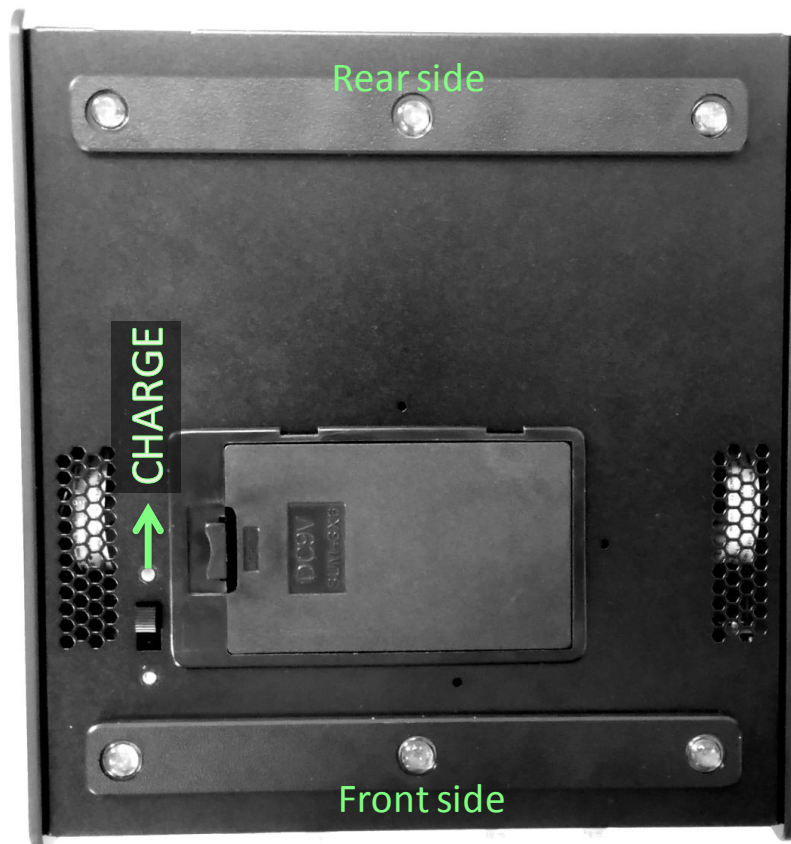
ⓘ *It is possible to use disposable batteries for an emergency, but in such case you MUST disable the charger circuits.*

3.3. Initial setup of the Ethernet interfaces

An initial set up is needed for using an Ethernet interface. For setting into operation, first connect the Ethernet interface to the network, using CAT5 wiring. Connections to 10BaseT or 100BaseT interfaces are both suitable, as the ScoopTeam automatically switches to the adequate 10 Mbit/s or 100 Mbit/s mode.

As a very first step, the Ethernet interface must be assigned an IP address, and related parameters. This phase is very simple when a DHCP server is available in the network.

The following is applicable to either the normal or the "Aux" Ethernet interface.

3.3.1. DHCP server available

This is the simplest case, because the DHCP server will allocate a suitable IP address and give the unit the right settings. Activate "DHCP" in the **Network**/**LAN1** (or **LAN2**) menu. The unit will then automatically find the DHCP server and automatically set the parameters. You can read the IP address (allocated to the unit by the DHCP server) in the same menu, or in the **Status** menu, **Ethernet** or **Ethernet Aux** section.

Note that, as an additional advantage with DHCP, you do not need to change this setting later, even if you move the codec to another network, as long as it is still connected to a DHCP server.

3.3.2. "Static" IP configuration

ⓘ *"Expert" access level is required for this setup.*

When there is no DHCP server, you have to enter the settings manually. The IP address must be "available", i.e. not already assigned to other equipment. Ask support from the network administrator(s) as needed.

First select the manual mode, menu **Network**/**LAN1 settings**[1]/**DHCP***: **Disabled**. Then, still in this menu, you must enter the following parameters:

Parameter	Notes
IP Address	Must be unique on the network
Subnet Mask	A typical value is 255.255.255.0
Default Gateway	
Preferred DNS	Domain Name Server (main)
Secondary DNS	Domain Name Server (secondary)

All addresses are in "dot-decimal" format, such as e.g.: 192.168.0.12, 10.0.54.123.

ⓘ *Note: in contrast to the configuration with DHCP, the "static" setting has to be reviewed each time you move the unit to a new physical site/network, as the previous IP addressing is probably not valid for the new location.*

3.3.3. Checking the IP configuration

The above configuration is kept in the unit's memory, and reloaded at each start.

To check the setting, you can read the IP address in the **Status** menu, **Ethernet** or **Ethernet (Aux)** section.

You may also check that the unit is seen on the network and at the right address: from a computer connected to the same network, enter (in the command mode, or console mode depending on the OS) "ping *ipaddr*", where *ipaddr* is the IP address of the ScoopTeam.

If the response is positive, then you can proceed with the rest.

---

[1] Or "**LAN2 settings**" if using the secondary Ethernet interface

3.3.4. Optional configuration of the "link" Ethernet layer

With factory settings, the Ethernet interface is set for an auto negotiation of the "link" mode: speed (10 or 100 Mbit/s) and half-duplex or full-duplex.

ⓘ *This setting is suitable for almost all situations and usually you don't need to change it.*

However, in the rare cases where this is needed, it is possible to set the desired mode manually: use the menu **Network**/**LAN1 settings**/**Link Mode**.

Whenever this setting is needed, it must be done before the IP configuration described above.

3.4. Managing links

3.4.1. Setting up and releasing links

Generally speaking, once the codec is set up and the transmission interface(s) to be used is (are) configured, it is possible to manage audio links with remote devices.

Transmission links/sessions have to be set, which can be done in two ways:

- "Outgoing call" launched towards a remote device: the procedure is to select a transmission interface, a coding configuration, "dial" the destination to call and then set the call to the remote unit. The detailed operating mode is described further, for the various transmission media, in 4.4 (Setting up a link).

- "Incoming call" received from a remote device: on receiving a call on one of the connected and active interfaces, the codec switches to this interface and processes the call.

In a similar way, ending/releasing a link is either initiated by the remote unit (remote release), or by the operator of the ScoopTeam (local release).

3.4.2. Auto-redial feature

When it is necessary to hold the link on permanently, outgoing calls may be backed up by using the auto-redial function (for each codec in double codec mode). When it is active and the codec is the initiator of the link, the codec automatically tries to re-set the link in case of an initial failure, or if an established link is dropped for another reason than a local release (i.e. hanging up by the user). The redial capability applies in the following situations:

- If the initial call fails for any reason (e.g. called party is busy); the codec then redials and retries to establish the link.

- The codec can also redial if the link is already established and the link is lost, for any reason else than "local release" (e.g. the remote unit mistakenly dropped the line).

- After a power failure, after rebooting the codec will automatically redial and set-up the link back.

ⓘ *Note that, while "auto redial" is active, an established link can be definitively stopped only by releasing the line on the <u>calling codec side</u>. Otherwise, every time the called party will hang up, the calling codec will redial and reset the link.*

It is possible to program the time period that the unit will wait before redialling after a failed trial, and it is also possible to program the maximum number of times the codec will redial before giving up.

The activation of this function and the configuration of its parameters can be found in 4.7.6, Tools sub-menu.

In double codec mode, the function can be activated separately for each codec.

3.4.3. GPIO control

In normal operation, outgoing calls are sent or released using the menus and/or the remote control interface. It is also possible to assign functions to the GPIO so that these can partially manage the connections: "Call" action or "Release" action. This capability is described in 2.5.5, GPIO.

3.5. First level maintenance

3.5.1. Clearing / Saving / Restoring settings

Clearing all settings

In some cases, it can be necessary to come back to the initial "factory" configuration.

To clear up the whole configuration and come back to the factory settings, normally you should go to the sub-menu item **Tools** / **Troubleshooting** / **Reset** and select "Full Factory Reset", or "Restore Default Settings" (which does not delete the memorized profiles and snapshots).

The default passwords are blank.

ⓘ *If it is not possible to access the menu to perform the factory reset (administrator password lost), the erasure should be done using the backup procedure described below in 3.5.2 (Backup reset).*

Saving and restoring the configuration

It is possible to export/save all the equipment settings into a file, and conversely to import/restore a complete configuration from a file previously recorded in such way. Such transfers can be done using the embedded html interface; refer to 4.9.10 ("SETUP TRANSFER" page) for the operating mode.

Partial exports are also possible, selecting elements to include in the export: call profiles, presets, snapshots, settings…

3.5.2. Backup reset

This procedure should be applied to recover full control over the unit if it is in a status, in which it is not possible to access all the functions (this might happen e.g. if the administrator's password has been lost, and the html access password as well).

After this reset, the ScoopTeam will be brought back to its "factory" configuration with all parameters in their default value, especially: blank passwords, Ethernet interface set for auto link mode, DHCP client enabled.

Go through the following steps:

- Get the *reset_scoopteam.bin* file (to be downloaded from our www.aeta-audio.com web site, a link can be found on the page dedicated to ScoopTeam).
- Copy this file *without changing its name* on a USB memory stick.
- Plug this stick into one of the USB host sockets on the rear panel of the unit (while it is off).
- Power on the ScoopTeam.
- After the initialization phase, ScoopTeam will go back to its "factory" settings.
- Remove the USB stick (no matter with or without power on), <u>before restarting</u> the ScoopTeam, otherwise all your possible new settings will be deleted again.

ⓘ *Note: existing profiles are not deleted by this procedure.*

3.5.3. Firmware update

There are two methods for updating the firmware of the unit:

- Use the embedded html server: the procedure is described further in chapter 4.9.10, ""SYSTEM UPDATE" page".

- Use a USB memory device: copy the update file into the root of the USB stick; switch the ScoopTeam on and wait for full startup; plug the USB stick into the rear panel of the unit, tap the <OK> button to acknowledge the "Update Unit from USB?" message and initiate the firmware update. Updating the firmware will last for several minutes, and the unit will be automatically restarted at the end. Remove the USB device when the update is over.

4. Detailed operating mode – User interface

In local mode, the unit is operated thanks to the top panel control surface.

The user interface is in English language, and only in English.

The unit provides selective access control. See further (4.8, Access control) the configuration of this feature.

In the following, the access level that grants control to a given element (menu/page/parameter...) is abbreviated AL (Administrator level), EL (Expert level), or OL (Operator level, default when not specified).

4.1. Equipment start-up

When the unit is ready with batteries inside and/or a DC power source connected, depress the "Power/Connections" key [icon]. During start-up, the unit displays a splash screen for a few seconds. Afterwards, the "login" page is displayed, as shown below: select the desired access level (Administrator/Expert/Operator) and tap the "Log in" button.
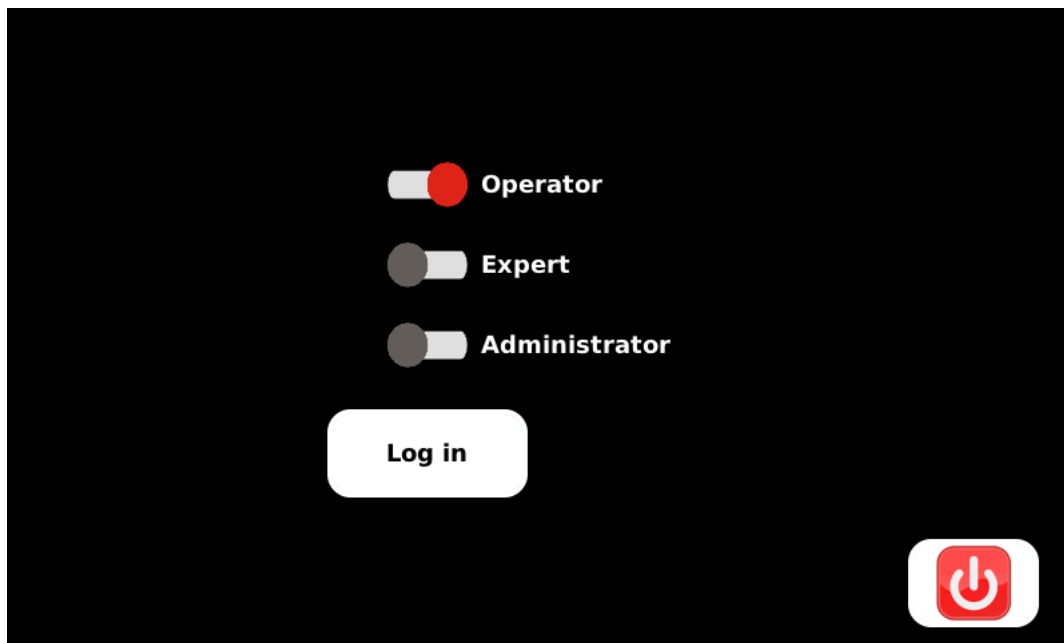
If a password is needed for further access, you must enter it at this time. A "virtual" keyboard is displayed; enter the password and press the "OK" button.

The "Home" page is then displayed. This is the default page for regular operation.

To access the other pages or the menu, tap one of the icons, or hit one of the two shortcut keys (lower part of the control panel).

4.2. Principles for navigating and setting parameters

The user interface is organized in specialised *pages* and a *Menu*, as shown by the diagram further (Figure 5 – User interface pages: overview, page 38).

Starting from the **Home page**, navigating among the pages and menus is done by tapping the soft buttons on the screen. In addition, there are two pages which are reached by using keys on the control surface (lower part):

- The ⬚ key leads to the "Monitoring" pages; these deal with the settings for the headphone monitoring, starting from the settings for the commentary position 1.

- The ⬚ key leads to the "Connections" page, for handling the network links. *A long press triggers the shutdown process.*

If one of these keys is pushed a second time, the display goes back to the menu that was active before hitting the key the first time.

Each page on screen typically includes a "Home" soft button ⬚ , leading directly to the default home page, and/or a "Back" button ⬚ , leading back to the upper level page and/or the previous page.

Other soft buttons and icons are displayed and used for switching to other pages, selecting among lists for discrete parameters, etc.

For **continuous parameters** such as gains, there are two cases:

- A parameter linked to a *physical knob* is adjusted by directly using the knob at any time (whatever the current page displayed).

- For a parameter linked to a *virtual knob*, first select the knob by tapping it (the knob turns to blue instead of white), and adjust the gain/volume using the rotary encoder. In some cases, additional settings are done by pushing down the rotary encoder. When you are done, tap the virtual knob again to release control (the knob turns back to white).

4.3. Description of the user interface pages

4.3.1. User interface structure
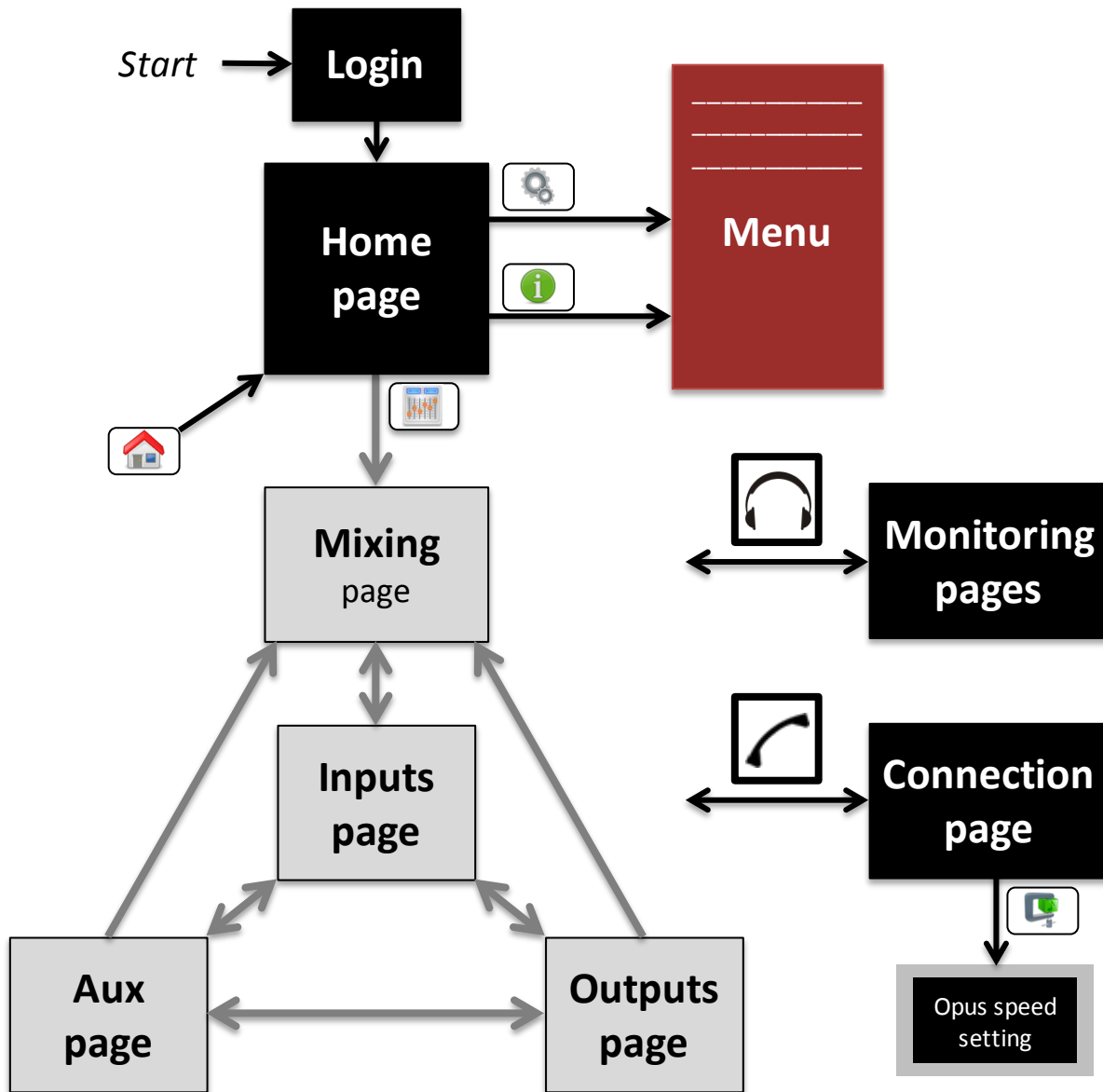
Figure 5 – User interface pages: overview

Pages with a light grey background are not accessible from the Operator access level.

4.3.2. Home page

Access level: Operator

The "Home" page is the default page for regular operation.
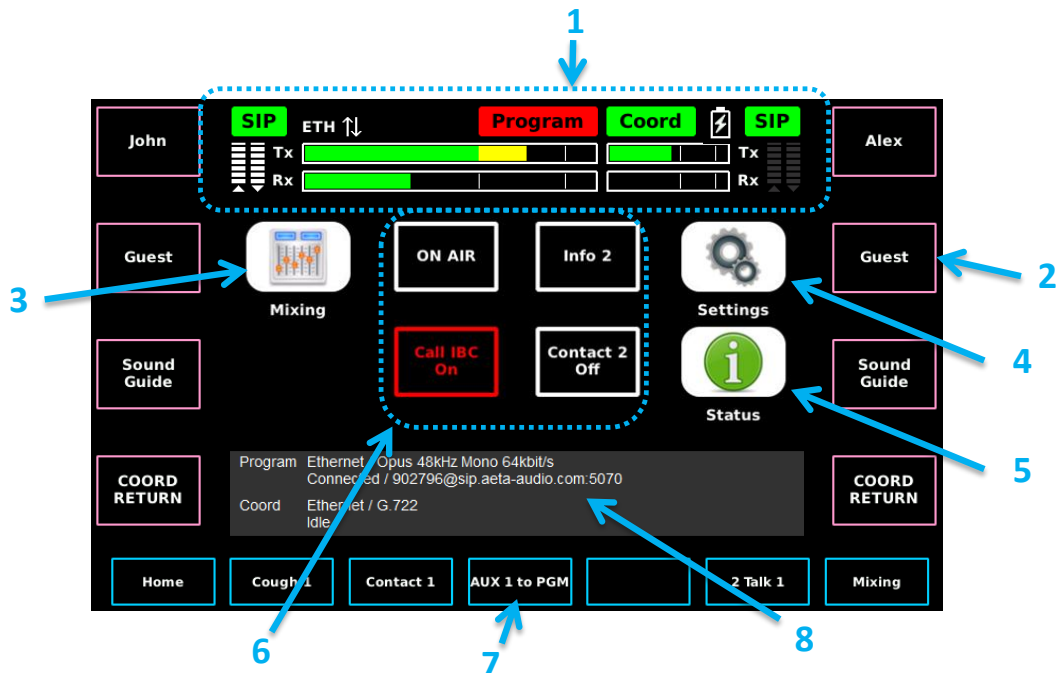
The home screen includes the following areas:

1. **Status panel**, including the audio level (send and receive) for the **Program** channel/line and the **Coordination** channel/line. Refer to chapter 4.5.1, Status panel for details about the status panel.

2. Labels of the signals assigned to the rotating knobs on the sides of the display (on the above picture, the arrow points towards the label of the second knob on the right side).

3. "Mixing" button: *this button is not shown for the Operator access level*. Tap the button to switch to the "Mixing" page (EL access needed), and further to the "Inputs", "Aux" and "Outputs" pages. See further the description of these pages.

4. "Settings" button: tap the button to switch to the menu and access all the settings of the unit.

5. "Status" button: tap the button to switch to the menu and access the status information on the unit.

6. Current state of the "Contacts" and "Info" data (regarding the function/meaning of these data, refer to 2.4, Auxiliary transmission: "relays").

7. Assignment of the pushbuttons below the screen (on the above picture, the arrow shows the name for the 4th button from the left).

8. Information banner.

You can also hit one of the two shortcut keys (lower part of the control panel) and access respectively the "Monitoring" pages, or the "Connection" page.

4.3.3. "Mixing" page

Access level: Expert

This page allows setting the gain of each input source in the **Program** and **Coordination** mix buses, and also provides access to the pages dedicated to the audio input/output settings.
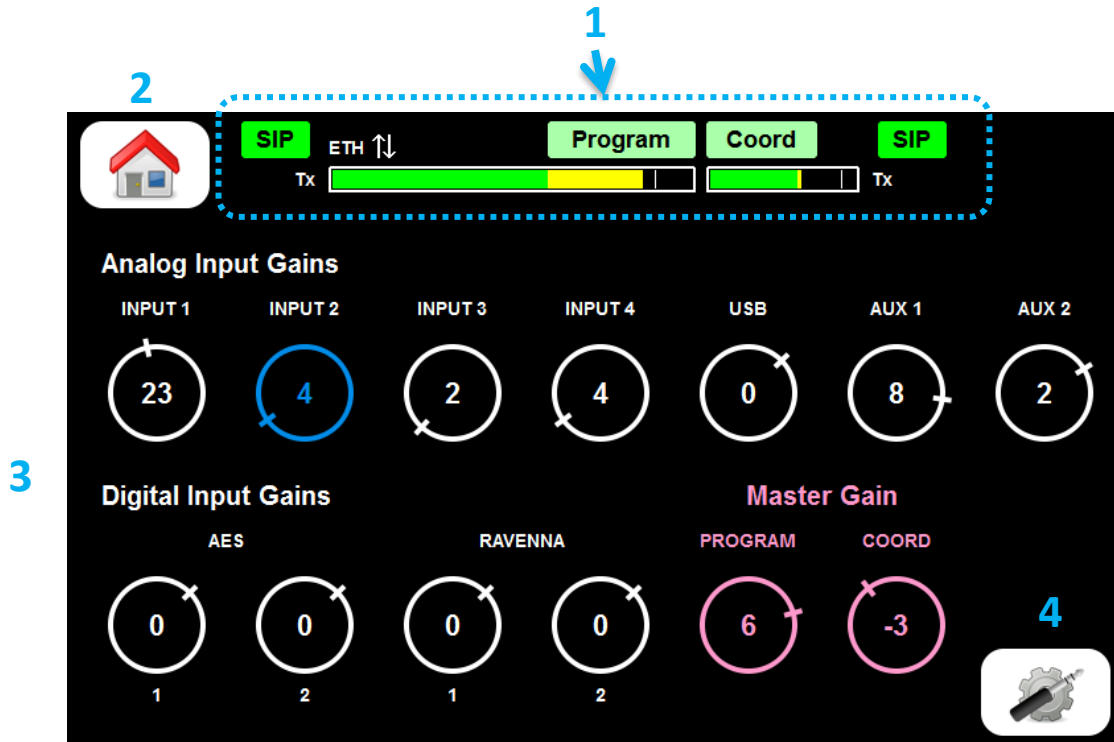
The page includes the following areas and items:

1. Status panel, in a compact variation, without the bargraphs for the level of the received audio signals (Program and Coord feedback). Refer to chapter 4.5.1, Status panel for details about the status panel.

2. "Home" button: tap the button to go to the "Home" page.

3. Gain setting and display area, where the gain for each audio input is accessible on a virtual knob. The master gain for each mixing bus is also available.
For setting a gain, tap the desired virtual knob to select it: the knob turns blue (as "Input 2" on the picture above). Rotate the rotary encoder to adjust the gain as desired (the gain is displayed in dB). *The gain is locked whenever you tap another knob or leave the page.*

4. "Audio I/O settings" button: tap the button to switch to the "Inputs" page, described in the following chapter ("Input Settings" page).

You can hit one of the two shortcut keys (lower part of the control panel) and access respectively the "Monitoring" pages, or the "Connection" page.

4.3.4. "Input Settings" page

Access level: Expert

This page gives access to the settings for the audio inputs 1 to 4.
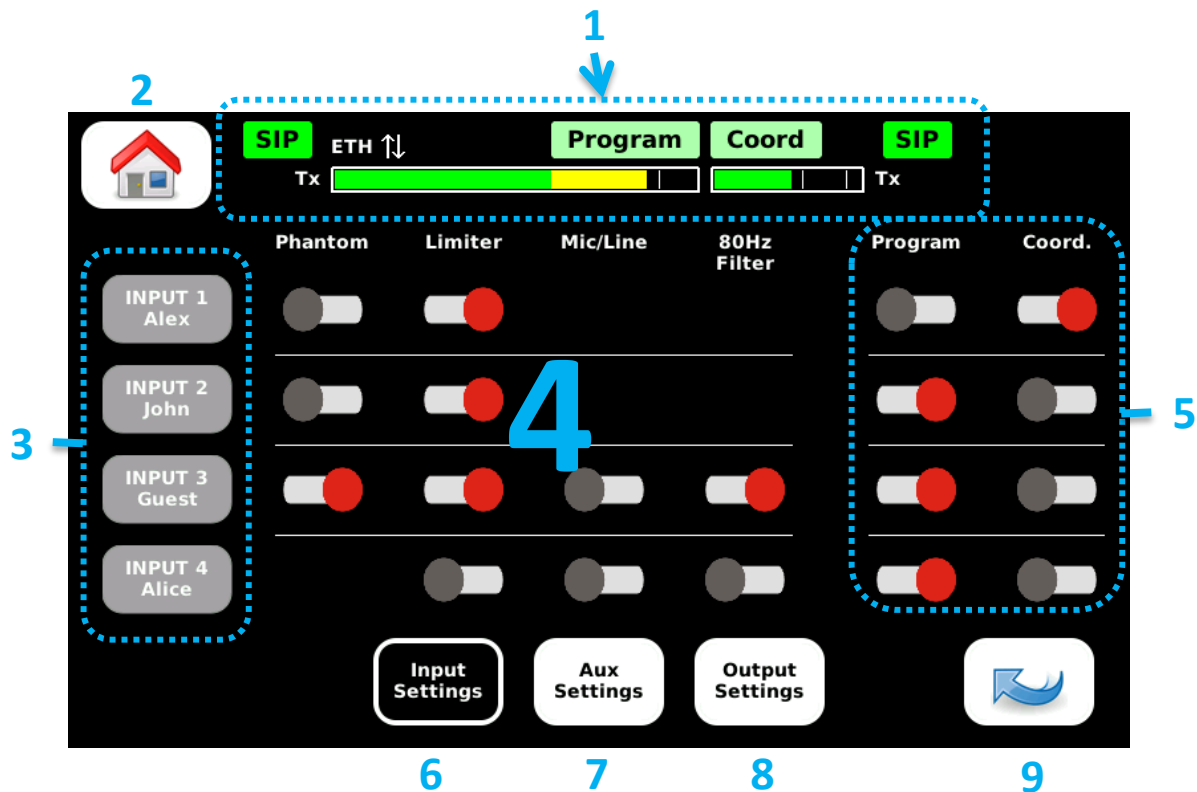
The page includes the following areas and items:

1. Status panel, in a compact variation, without the bargraphs for the level of the received audio signals (Program and Coord feedback). Refer to chapter 4.5.1, Status panel for details about the status panel.

2. "Home" button: tap the button to go to the "Home" page.

3. Buttons corresponding to inputs 1 to 4, and showing the labels assigned to the sources. These buttons are grey to indicate that they can be used for configuring the labels.
    To do this, tap the button to be configured. A virtual keyboard pops up as on the picture below:

There you can edit the current label and/or enter the new label, and tap "OK". This label is associated to the selected input all over the user interface. *If the label is left blank, only the source name is displayed everywhere.*
The labels can be edited via the menu too (see further the description of the menu).

4. Switches for activating or defeating the features selectable on the input channels: microphone phantom power (except input 4), input limiter, mic/line switch (only inputs 3 and 4 can be set for line level), and high-pass filter (the high-pass filter is always active on inputs 1 and 2, thus no switch is shown). Tap a switch to toggle the feature active/disabled.

5. Routing switches: on each input channel:
   - A virtual switch selects the signal for sending to the **Program** bus; tap the switch to toggle it on/off (*note that this is the same as pressing the input's dedicated PGM key on the control panel*).
   - Another virtual switch selects the signal for sending to the **Coordination** bus (Coord.); tap the switch to toggle it on/off. *This is **not** the same as pressing the input's dedicated Coord key on the control panel. Here the switch is static, i.e. when active it **permanently** sends the signal to the Coordination bus, and the dedicated Coord button on the control panel has no effect.*
   - For stereo operation, the "Program" and "Coord" buses become respectively "Program left" and "Program right". By routing an input appropriately to Program and/or Coord you can set the signal on a left, right or center mapping.

6. "Input Settings" button: this button has a black background to remind that this is the current page.

7. "Aux Settings" button: tap the button to go to the "Aux Settings" page, described in the following ("Aux Settings" page).

8. "Output Settings" button: tap the button to go to the "Output Settings" page, described in the following ("Output Settings" page).

9. "Back" button: tap the button to come back to the "Mixing" page.

You can hit one of the two shortcut keys (lower part of the control panel) and access respectively the "Monitoring" pages, or the "Connection" page.

4.3.5. "Aux Settings" page

Access level: Expert

This page allows selecting the auxiliary inputs which contribute to the **Program** and **Coordination** mix buses.
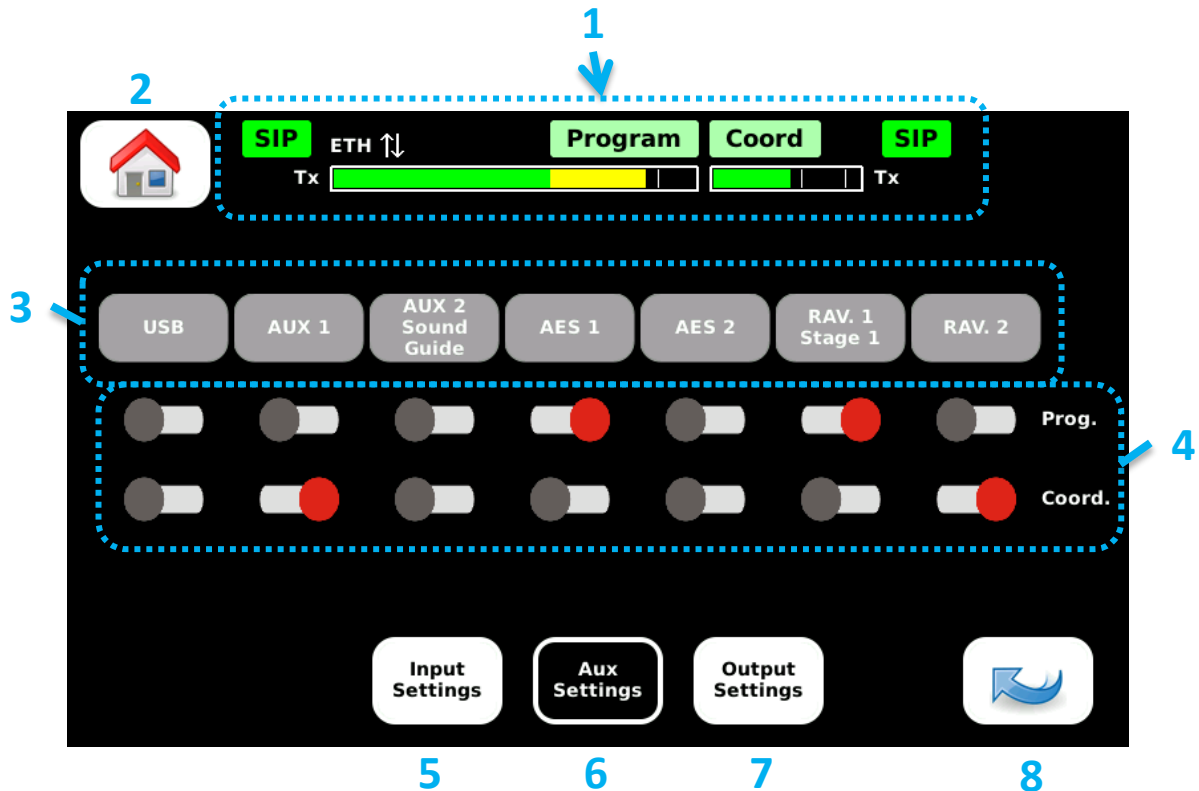
The page includes the following areas and items:

1. Status panel, in a compact variation, without the bargraphs for the level of the received audio signals (Program and Coord feedback). Refer to chapter 4.5.1, Status panel for details about the status panel.

2. "Home" button: tap the button to go to the "Home" page.

3. Buttons corresponding to the aux inputs, and showing the labels assigned to the sources. These buttons are grey to indicate that they can be used for configuring the labels.
   To do this, tap the button to be configured. A virtual keyboard pops up; there you can edit the current label and/or enter the new label, and tap "OK". This label is associated to the selected aux input all over the user interface. If the label is left blank, only the source name is displayed everywhere.
   The labels can be edited via the menu too (see further the description of the menu).

4. Routing switches: on each input channel:
   - A virtual switch selects the signal for sending to the Program bus; tap the switch to toggle it on/off. *It is possible to assign one of these switches to a function key, and in this way get instant control over the switch. See chapters 2.5.6 and 3.2.2.*
   - Another virtual switch selects the signal for sending to the Coordination bus (Coord.); tap the switch to toggle it on/off.
   - For stereo operation, the "Program" and "Coord" buses become respectively "Program left" and "Program right". In the same way as for the microphone inputs, you can set each signal on a left, right or center mapping by appropriately routing to Program and/or Coord.

5. "Input Settings" button: tap the button to go to the "Input Settings" page, described above ("Input Settings" page).

6. "Aux Settings" button: this button has a black background to remind that this is the current page.

7. "Output Settings" button: tap the button to go to the "Output Settings" page, described in the following ("Output Settings" page).

8. "Back" button: tap the button to come back to the "Mixing" page.

You can hit one of the two shortcut keys (lower part of the control panel) and access respectively the "Monitoring" pages, or the "Connection" page.

4.3.6. "Output Settings" page

Access level: Expert

This page mainly allows selecting the signals which contribute to the output mix buses (USB output, Line outputs, RAVENNA output channels), among the following: Program mix and Program return, Coordination mix and Coordination return.
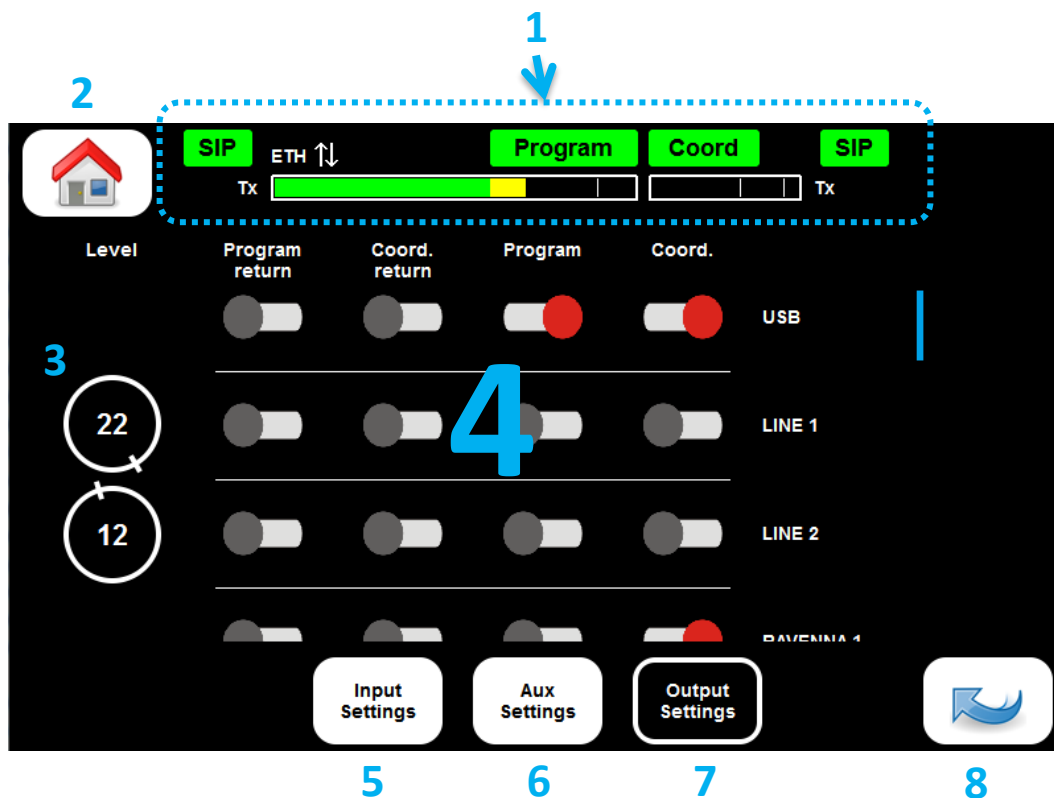
The page includes the following areas and items:

1. Status panel, in a compact variation, without the bargraphs for the level of the received audio signals (Program and Coord feedback). Refer to chapter 4.5.1, Status panel for details about the status panel.

2. "Home" button: tap the button to go to the "Home" page.

3. Gain setting for the line outputs. For setting a gain, tap the virtual knob to select it: the knob turns blue. Rotate the rotary encoder to adjust the gain as desired. The value displayed is the maximum output level in dBu. *The gain is locked whenever you tap another item or leave the page.*

4. Routing switches: for each output channel, virtual switches configure the selected sources:
   - A virtual switch enables the **Program** bus into the output mix; tap the switch to toggle it on/off.
   - Another virtual switch enables the **Coordination** bus into the output mix; tap the switch to toggle it on/off.
   - A couple of switches enable the Program and Coordination return signals into the output mix.
   - For stereo operation, the "Program" and "Coord" buses become respectively "Program left" and "Program right".

5. "Input Settings" button: tap the button to go to the "Input Settings" page, described above ("Input Settings" page).

6. "Aux Settings" button: tap the button to go to the "Aux Settings" page, described in the following ("Aux Settings" page).

7. "Output Settings" button: this button has a black background to remind that this is the current page.

8. "Back" button: tap the button to come back to the "Mixing" page.

You can hit one of the two shortcut keys (lower part of the control panel) and access respectively the "Monitoring" pages, or the "Connection" page.

ⓘ *When all the output buses cannot be seen at once (Premium+ version with Ravenna option), you can scroll the page: either drag the scroll bar (blue bar on the right hand side), or simply touch and drag the switch area to reach the desired part.*

4.3.7. "Monitoring" pages

Access level: Operator

These 4 pages allow the commentators to adjust the monitoring mixes in the headphone outputs. One page is available for each commentator position, where one can adjust the gain of each audio source into the mix.

The first page (monitoring for position 1) is reached directly from any page or menu by pressing the dedicated [🎧] shortcut key on the control panel. Conversely, pressing again the key brings the interface back to the previous location (where it was beforehand).

Monitoring 1
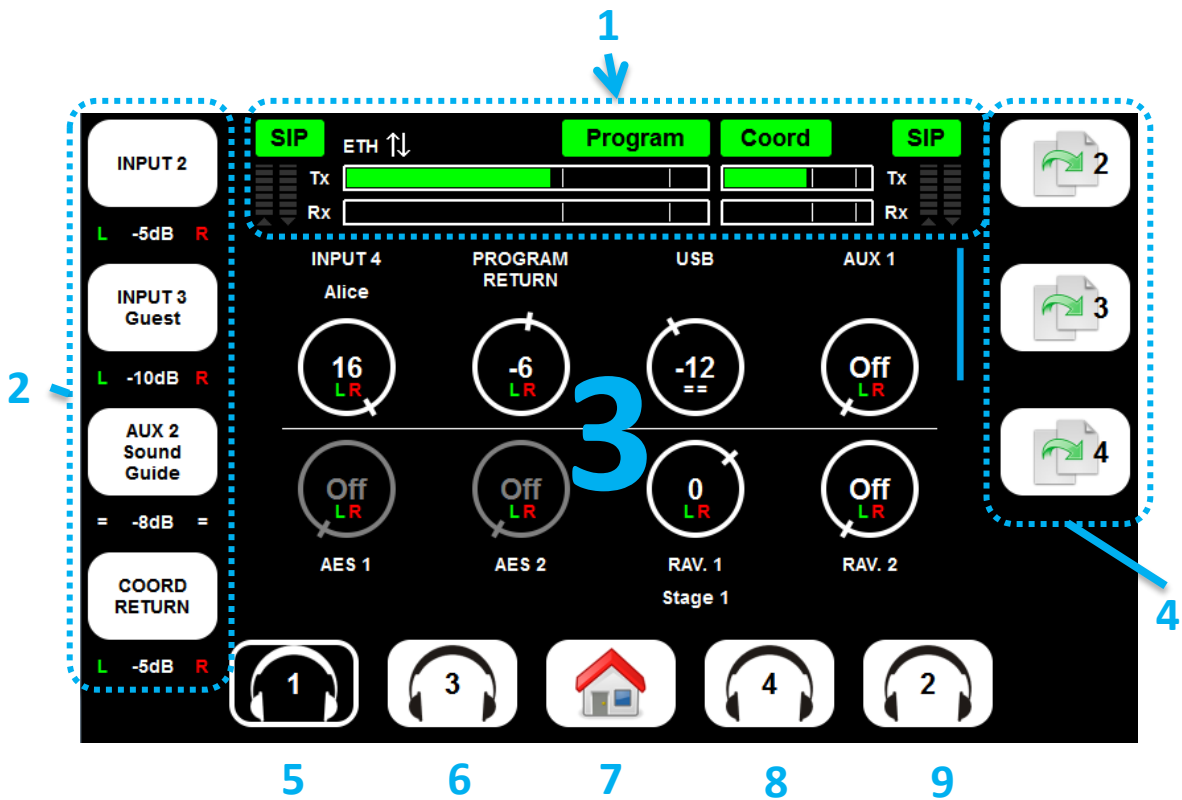
The page includes the following areas and items:

1. Status panel, as on the home page, including the audio level (send and receive) for the **Program** channel/line and the **Coordination** channel/line. Refer to chapter 4.5.1, Status panel for details about the status panel.

2. Gain and routing for the sources that are assigned to knobs on the left side of the screen. Besides each knob you can see a button showing the name of the assigned signal, its current gain (in dB) and the routing to the ears (Left, or Right, or both, or muted). The gain is adjusted directly with the knob (below -30 dB gain, the source is muted, "Off"). For the other settings, tap the button: a pop-up box appears as shown on the picture below:
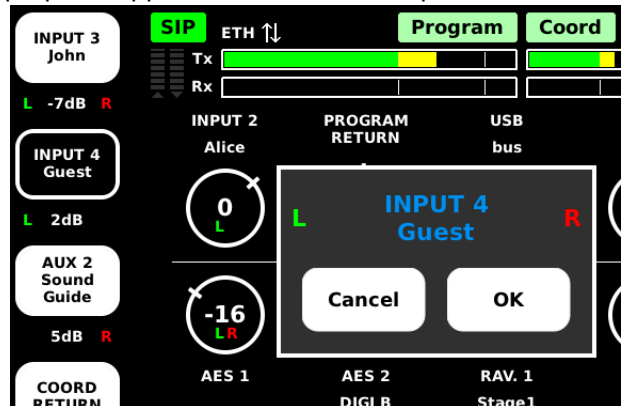
- The selected button is reminded by a black background.
- If you wish to change the assignment of the knob, rotate the rotary encoder and scroll through the list of available sources. *This setting is not available for the Operator level.*
- Push down the rotary encoder to change the routing, in the following sequence: L+R, L, R, "==", etc.

N*ote: the last choice for routing "==" mutes the source in the mix; as a difference with the "Off" (minimum gain) setting, you can quickly restore the source to its previous gain if desired.*
- Tap OK to commit the changes, or touch anywhere else to cancel and stay with the existing assignment and routing.

3. Virtual knobs for the sources which are not assigned to the physical knobs on the side. For adjusting a source, tap the virtual knob to select it: the knob turns blue.
- Rotate the rotary encoder to adjust the gain (value in dB) as desired.
- Push down the rotary encoder to change the routing, in the following sequence: L+R, L, R, "==", etc.

The gain is locked whenever you tap another item or leave the page.
Note: the last choice for routing "==" mutes the source in the mix; as a difference with the "Off" (minimum gain) setting, you can quickly restore the source to its previous gain if desired.

ⓘ *When all the available sources cannot be seen at once (Premium+ version with Ravenna option), a blue bar is visible on the right and an additional row of knobs is available: touch and drag the knob area to reach it.*

4. "Copy" buttons (*not active for the Operator level*), allowing to easily reuse the monitoring mix for another commentator position. If you tap the ⬚ button, a pop-up box asks for confirmation; tap **OK** to confirm. The gain and routing settings are all copied onto the monitoring mix for position 2 (except the gain for Mic 2, because it is controlled by the dedicated "side tone" knob on the front panel).
The other copy buttons have similar function, for copying the settings onto respectively position 3 and position 4.

5. "Monitoring 1" button: black background to remind that this is the current page.

6. "Monitoring 3" button: tap the button to go to the settings for position 3, described further.

7. "Home" button: tap the button to go to the "Home" page.

8. "Monitoring 4" button: tap the button to go to the settings for position 4, described further.

9. "Monitoring 2" button: tap the button to go to the settings for position 2, described further.

If you hit the ⌂ shortcut key on the front panel, the display goes back to the page or menu that was active before entering the monitoring pages. If you hit the ◠ shortcut key, the unit directly jumps to the "Connection" page.

Monitoring 2
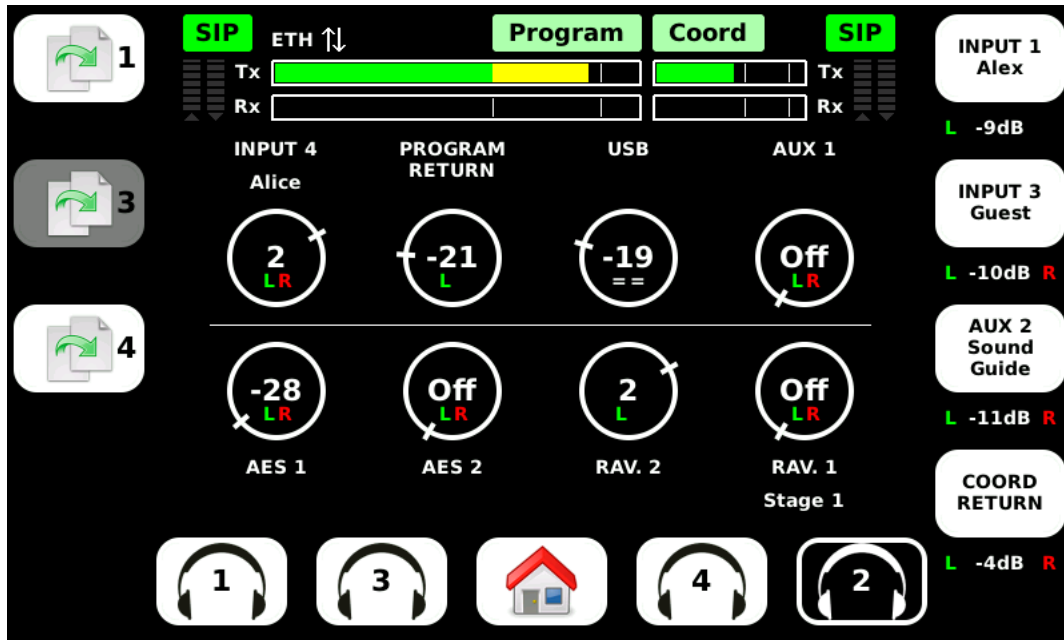
This page is very similar to the one for position 1, and the operating mode is the same (see above, Monitoring 1).

The "Copy" buttons (*not active for the Operator level*) allow here to reproduce the same mix onto (respectively) positions 1, 3 and 4.

ⓘ *Note the shaded "Copy to 3" button: this means that the button is disabled and copying settings onto position 3 is not allowed. This is the case whenever position 3 is set in a "follow" mode: see further the possible cases.*

If you hit the ⌂ shortcut key on the front panel, the display goes back to the page or menu that was active before entering the monitoring pages. If you hit the ◠ shortcut key, the unit directly jumps to the "Connection" page.

Monitoring 3
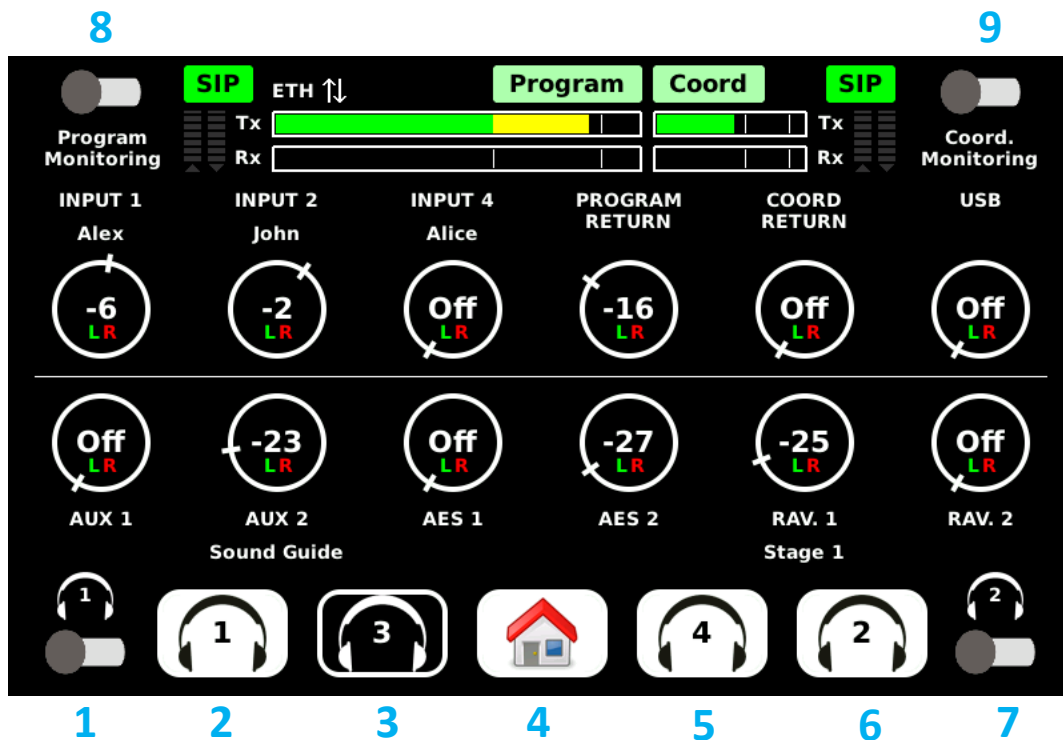
This page is very similar to the other monitoring pages (see above, Monitoring 1), with however some differences.

1. "Follow 1" switch: if this switch is active, the monitoring mix is permanently identical to the one for position 1. To remind this, all the virtual knobs are shaded (see picture below). However, the gain for the microphone on position 1 can still be adjusted.
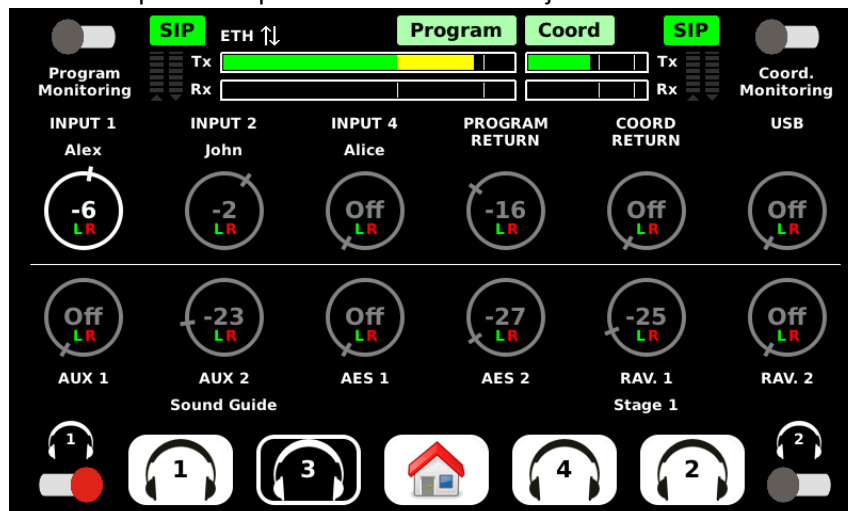
   While this switch is on, it is not possible to copy the settings from positions 1 or 2 ("Copy to 3" buttons are disabled on the monitoring pages). If the switch is disabled, the monitoring mix comes back to its previous settings.

2. "Monitoring 1" button: tap the button to go to the settings for position 1, described above.

3. "Monitoring 3" button: black background to remind that this is the current page.

4. "Home" button: tap the button to go to the "Home" page.

5. "Monitoring 4" button: tap the button to go to the settings for position 4, described further.

6. "Monitoring 2" button: tap the button to go to the settings for position 2, described above.

7. "Follow 2" switch: if this switch is active, the monitoring mix is permanently identical to the one for position 2. To remind this, all the virtual knobs are shaded. However, the gain for the microphone on position 2 can still be adjusted. While this switch is on, "Copy to 3" buttons are disabled on the monitoring pages 1 and 2. If the switch is disabled, the monitoring mix comes back to its previous settings.

8. "Program Monitoring" switch: if this switch is active, the Program mix replaces the current mix in the headphone, thus enabling to listen and check the current program mix. To remind this, all the virtual knobs are shaded and inactive, and all switches are inactive too. While this switch is on, "Copy to 3" buttons are disabled on the monitoring pages 1 and 2. If the switch is disabled, the monitoring mix comes back to its previous settings.
Note: the program mix is picked up after the limiter/compressor.

9. "Coordination Monitoring" switch: if this switch is active, the Coordination mix replaces the current mix in the headphone, thus enabling to listen and check the current coordination mix. To remind this, all the virtual knobs are shaded and inactive, and all switches are inactive too. While this switch is on, "Copy to 3" buttons are disabled on the monitoring pages 1 and 2. If the switch is disabled, the monitoring mix comes back to its previous settings.
Note: the coordination mix is picked up after the limiter /compressor.

If you hit the [headphone icon] shortcut key on the front panel, the display goes back to the page or menu that was active before entering the monitoring pages. If you hit the [connection icon] shortcut key, the unit directly jumps to the "Connection" page.
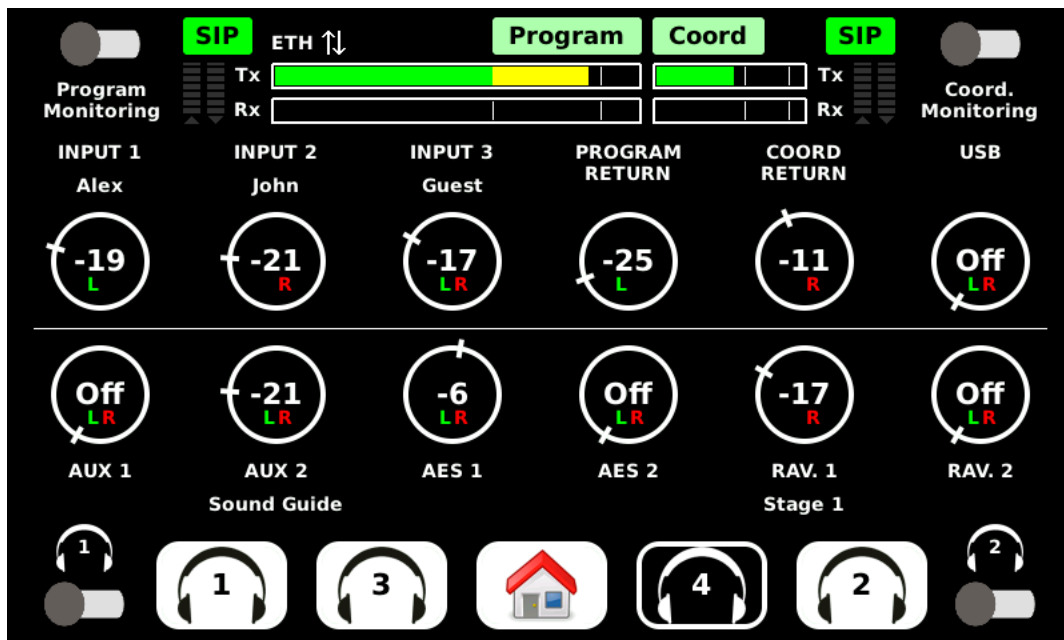
Monitoring 4

This page is very similar to the "Monitoring 3" page (see above), with same operating mode.

If you hit the [headphone icon] shortcut key on the front panel, the display goes back to the page or menu that was active before entering the monitoring pages. If you hit the [connection icon] shortcut key, the unit directly jumps to the "Connection" page.

4.3.8. "Connection" page

Access level: Operator

This page is used for managing connections with remote devices.
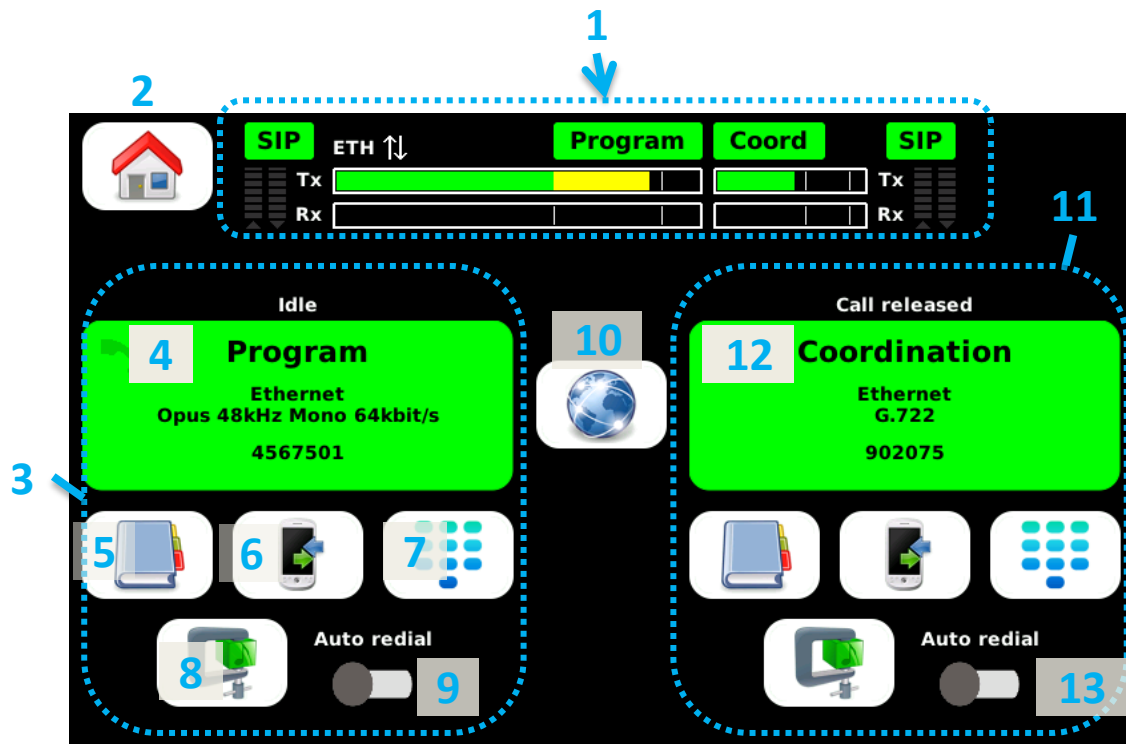
The page includes the following areas and items:

1. Status panel, as on the home page, including the audio level (send and receive) for the **Program** channel/line and the **Coordination** channel/line. Refer to chapter 4.5.1, Status panel for details about the status panel.

2. "Home" button: tap the button to switch back to the "Home" page.

3. This area on the left side is dedicated to the management of the connections for Codec 1. This is the only codec in the "single codec" mode, and it carries in this case both the **Program** channel and the **Coordination** channel (when available, with dual mono coding). In the "double codec" mode, this is the codec used for the **Program** transmission.

4. "Connection pad" for codec 1: this area is both a call/release button and an information area for the prepared or running connection. Tap the pad when the codec is "ready" (no running connection), and the unit calls the destination that is preselected and displayed in the pad. If the codec is on line, tap the pad and the unit hangs up (after confirmation). The color of the pad also reflects the state of the connection. The detailed behaviour of this element is described further in 4.5.3, Connection pads.

5. "Book" shortcut button: tap the button to call the menu, and go directly to the **Book** / **Call Profiles** sub-menu. There you can pick up a call profile and load it. The loaded coding profile and destination number/address are then shown in the connection pad, ready for launching an outgoing call. You can find further (4.7.5) the detailed description of this branch in the menu.

6. "Call history" button: tap the button and the "Call history" pop-up box opens, where you can pick a destination from the past called numbers.
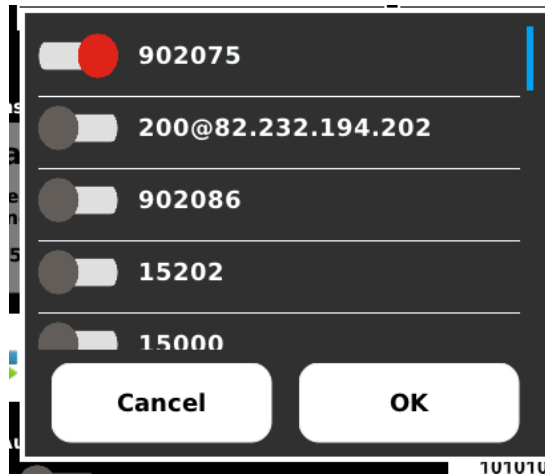
Once confirmed with the "OK" button, the selected number is set inside the connection pad, ready for calling out.

Note that only relevant numbers are shown: previous ISDN numbers called are not displayed if the current network is an IP interface, and previous AoIP destinations called are not displayed if the current network interface is ISDN.

7. "Dial" button: tap the button and the dial keypad pops up.
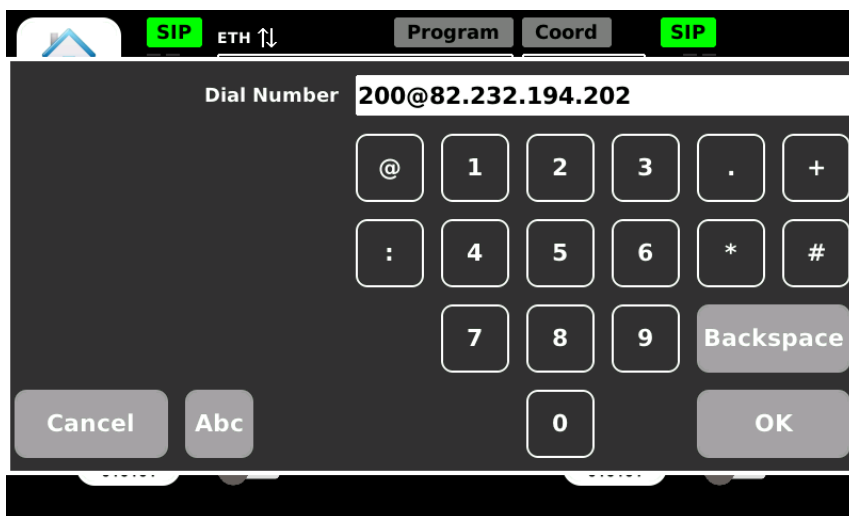
You can edit the last number or enter a new number (it is possible to switch to the full alphanumeric keypad if necessary: tap the "Abc" button). Once you validate with OK, the edited number is set inside the connection pad, ready for calling out.

8. "Coding" shortcut button: tap the button to call the menu, and go directly to the **Coding** branch, for the selected codec. There you can set up the coding algorithm and parameters for the codec. The resulting configuration is shown in the connection pad, ready for launching an outgoing call. You can find further (4.7.3) the detailed description of this branch in the menu.
When the codec is busy, the button is inactive, because you are not allowed to change the coding parameters when a connection is active. There is an **exception to this if the coding is Opus**: when you tap this button, a pop-up box allows you to change the bit rate "on the fly".

9. "Auto redial" switch: when this feature is activated, codec 1 restarts a call to the destination in case the connection is dropped undesirably. The feature is described in detail in 3.4.2, Auto-redial feature, and its parameters can be set in the **Tools** / **Miscellaneous** branch of the menu, **Auto Redialing** section (see further 4.7.6).

10. "Network" shortcut button: tap the button to call the menu, and go directly to the **Network** main branch: see further (4.7.2) the description of this branch in the menu.

11. This area on the right side is dedicated to the management of the connections for Codec 2. In the "single codec" mode this area is inactive and signalled as such. Conversely, this is the area used for managing the Coordination transmission when it involves a separate connection: operation using the "double codec" mode, or with mobile voice used for coordination.

12. "Connection pad" for codec 2: this area is both a call/release button and an information area for the prepared or running connection. Tap the pad when the codec is idle (no running connection), and the codec calls the destination that is preselected and displayed in the pad. If the codec is on line, tap the pad and the unit hangs up (after confirmation). The color of the pad also reflects the state of the connection. The detailed behaviour of this element is described further in 4.5.3, Connection pads.

13. "Auto redial" switch for codec 2: when this feature is activated, codec 2 restarts a call to the destination in case the connection is dropped undesirably. The feature is described in detail in 3.4.2, Auto-redial feature, and its parameters can be set in the **Tools** / **Miscellaneous** branch of the menu, **Auto Redialing** section (see further 4.7.6).

If you hit the shortcut key on the front panel, the unit directly jumps to the "Monitoring" pages. If you hit the shortcut key, the display goes back to the page or menu that was active before entering the "Connection" page.

4.4. Setting up a link

Generally you should go through the following steps:

1. Select the **basic options** that will essentially determine the operation: network interface and main codec mode. This is described below (4.4.1, Basic options).

2. Prepare or check the settings for the **network interface**(s) you will use. See below (4.4.2, Preparing the network interface(s)).

3. Select the suitable **coding configuration** for the link(s). See below (4.4.3, Setting the coding configuration).

4. Select or dial the destination address/number and **launch calls** to the desired destination(s). This is described below (4.4.4, Launching calls).

5. Alternatively, the unit can also **receive calls** from remote units. See below (4.4.5, Receiving calls).

4.4.1. Basic options

Before setting up links with the unit, you should select the basic operating mode. For this, the main choices to do are:

- What will be the main network for the transmission? The type of network is ISDN, Ethernet, mobile IP or Wi-Fi, depending on the available resources. In addition, there may be alternate interfaces: for instance, the mobile access may be using the internal module, or an external USB device.

- Will you use a Coordination channel in addition to the main Program transmission?

- Is it necessary to use a separate network link for this coordination channel?

Whatever the network interface selected as main interface, the table below summarises the possible combinations available[1]:

Case	Codec mode	Coding channel mode	Program	Coord	Links
1	Single codec	Mono	Network	*unused*	1
2		2-channel	Network	Network	1
3	Double codec	Mono + Mono	Network	Network	2
4	Single codec + mobile voice	Mono + Voice	Network	Voice	2

Table 6 – Basic connection management modes

- **Case 1:** only one link is set, and only the program mix is transmitted.

- **Case 2:** only one link is set, but the codec is set for dual mono coding, and both the program and coordination are transmitted/received on the same connection. *In the "stereo" variant, the "Program" and "Coord" buses become respectively "Program left" and "Program right".*

- **Case 3**: the double codec mode allows setting two independent links for respectively the program and the coordination channels.

- **Case 4**: on one side the program is transmitted over the network on a mono connection, and on the other side a mobile voice link (using the integrated mobile access module) is used for a transmitting the coordination.

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[1] The table assumes all possible options are available; of course your specific unit may not provide all these network interfaces.

These main choices are selected via the **Network** menu branch, reached by tapping the "Settings" soft button visible on the "Home" page, or the "Network" soft button found on the "Connection" page:
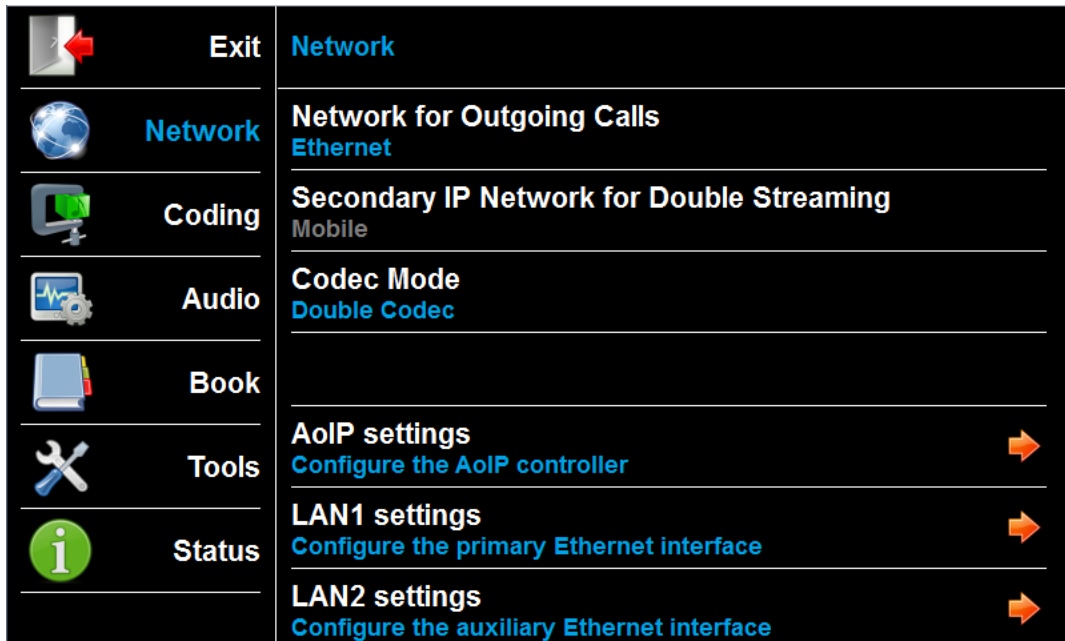
You can also reach this branch from any place in the menu tree by tapping the "Network" icon on the left column.

First tap the "**Network for Outgoing Calls**" row: a pop-up box allows you to pick the desired network interface. This interface is used for all outgoing calls (*except for coordination calls in case 4 above*).

The following row is for selecting the secondary interface that will be used for a redundant stream, if Double Streaming is active. Refer to chapter 6.2, "Using the Double Streaming system", for the detailed operation of this system.

On the third row, tap "**Codec Mode**" and select the desired mode. Once selected, this choice applies to any network, even if you switch to another network interface. *Note: selecting the double codec mode automatically disables Double Streaming.*

4.4.2. Preparing the network interface(s)

This part describes how to prepare the network interfaces for the subsequent connections, in order to be ready to set calls. This should also be done to get the unit ready to possibly receive calls from a remote device. Each interface potentially involved in links must be prepared[1].

ISDN interface

Access level: Expert

Use an RJ45 cable to connect the ISDN line to the ISDN connector on the back of the unit (see3.2.3, Rear panel).

The network interface must be configured depending on the local ISDN line that is used, thanks to the **ISDN settings** sub-menu (**Network** / **ISDN settings**).

The default settings which are suitable for most public ISDN lines are the following:

- 5AS : Active
- Protocol : Euro ISDN (exception: NI-1 in North America, NTT in Japan)
- Local addresses and sub-addresses: leave blank
- SPID: only relevant for North America

More details about the parameters of the ISDN interface can be found in 4.7.2, Network sub-menu, Network > ISDN settings section.

AoIP settings

Access level: Expert

These settings apply to **all IP protocol connections**, hence all network interfaces other than the ISDN interface.

These settings are found in the **Network** / **AoIP settings** menu; details on the various parameters are described in 4.7.2, "Network sub-menu", "**Network > AoIP settings** sub-menu" section.

The usual and most convenient operating mode uses the SIP protocol: select "SIP" for the first parameter (**Default Protocol for Outgoing Calls**). If a SIP server is used, the ScoopTeam must be registered on this server. Use the **SIP** section of the **Network** / **AoIP settings** sub-menu for entering the SIP account data and activate the SIP registration. *Check that the codec is successfully registered by looking for the indicators on the status panel ("SIP" bar on the top line, green or dark green when registered).*

Besides, for links over the Internet, most often the link passes through a NAT router which "hides" to the codec the characteristics of the public access: public IP address and RTP ports actually used. This can make the SIP transactions fail, or get a non-optimal route through the network. To avoid this, a STUN server is often used; the corresponding settings are also found in the **Network** / **AoIP settings** sub-menu.
Check that the public address discovery has succeeded: **Status** / **AoIP controller** *sub-menu:* **Public IP** *information (its mere presence shows the discovery has succeeded). You can also see there the type of NAT that the codec has detected (***NAT Type***).*

If you want to use Direct RTP instead of SIP for outgoing calls, select "Direct RTP" for the first parameter (**Default Protocol for Outgoing Calls**). This mode does not include any signaling protocol as SIP does, but the operation will be somewhat similar: one unit "calls" the other one, in fact initiates the link by directly sending a stream to the other end. The "receiver" of the call is configured to accept this incoming call and feed back its own stream to the initiator of the link.

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[1] This includes the possible secondary IP interface if Double Streaming is used.

Ethernet settings

Access level: Expert

Use an RJ45 cable (CAT5 grade at least) to connect the unit to a network switch or router. Plug the cable on the back of the unit (see 3.2.3, Rear panel), either in the primary Ethernet interface or the auxiliary Ethernet interface (marked "Aux" on the rear panel). A couple of LEDs on the socket show the state and activity of the connection.

The network interface may have to be configured for the local installation, especially if there is no DHCP server available. The relevant settings are accessed in the **Network** / **LAN1 settings** sub-menu for the primary Ethernet interface, or in the **Network** / **LAN2 settings** sub-menu for the auxiliary Ethernet interface. *The following applies either to the primary Ethernet interface or the auxiliary Ethernet interface.*

Details about these settings can be found in 4.7.2, Network sub-menu, Network > LAN1 settings / LAN2 settings section.

In the most frequent case:

- A DHCP server is available: set the **DHCP** switch "Active"; the allocated IP address can be seen in the following parameters.
- Check that the **Link Mode** is "Auto Negotiation".
- Select "Middle" for the **Network Quality** setting.

Mobile settings

A mobile network may be used as the main interface for AoIP connections. It can also be used in voice mode (regular telephone connections) for the coordination channel (case 4 in 4.4.1 above).

As a prerequisite, you must have a subscription and a SIM card granting access to mobile data transmission. *In case 4 as described above the subscription must include access to the basic telephone service.*

First insert the SIM card in the ScoopTeam (the unit must be **powered off** for this), and connect at least one antenna to the unit. See how to proceed on page 29, "Drawer for SIM card [13]". Power up the codec, enter the sub-menu **Network** / **Mobile settings (Internal)** and tap **PIN Code**. Enter the PIN code of the SIM card using the virtual keypad and tap **OK**. Tap the ✅ icon to confirm the entry. *This is useless, of course, if the card's PIN code is blank or disabled. Conversely, if you enable **PIN Save** you will not need to enter the PIN code again later, even if the unit is switched off.*

The network access must be configured using the **Network** / **Mobile settings (Internal)** menu. You can find details in 4.7.2, Network sub-menu, Network > Mobile settings (Internal) section; the following settings are suitable in most occasions:

- Leave **Preferred Radio Access technology** in the "Auto" setting.
- Leave **Preferred Network Operator** in the "Auto" setting, except if there is a definite reason not to do so.
- **Mode** must be set "Audio over IP".
- In the **Audio over IP** section, DATA service should be active, and the other parameters are set automatically.
- However, if the **Access Point Name** is not set, or its value is not appropriate, you can enter a new value. *Afterwards, tap the ✅ icon to confirm the entry.*
- Select "Middle" for the **Network Quality** setting.

After having performed these settings, you can check the status of the mobile registration in the **Status** / **Mobile** sub-menu: operator, radio access technology and signal strength, status of the data connection. In addition, in the operation pages the status bar includes status icons related to the mobile network.

Mobile settings (USB device)

Using a USB device for mobile access is an alternative method[1] to access a mobile network, with for example the following applications:

- Access to mobile networks on a unit not equipped with the (integrated) mobile option.
- Use of a second SIM card for mobile access, with the capability to switch quickly from one network to the other (switching internal module / USB device, instead of switching SIM cards).

As a prerequisite, you must have a subscription and a SIM card granting access to mobile data transmission. First insert the SIM card in the USB device, and plug the device into a USB socket of the ScoopTeam ([6] in 3.2.3, Rear panel); you may do that "hot", while the unit is already on. After a while, you should see the device in the sub-menu **Network**, with the **Mobile settings (External 1)** heading.

Enter this sub-menu, and the rest is just the same as for setting up the internal module, as described above.

After having performed these settings, you can check the status of the mobile registration in the **Status** / **Mobile (ext 1)** sub-menu: operator, radio access technology and signal strength, status of the data connection.

Wi-Fi settings

A Wi-Fi network can be used for AoIP transmission, and/or as an interface for remote controlling the unit. A USB Wi-Fi device is provided with the option.

Plug the device into one of the USB host sockets ([6] in 3.2.3, Rear panel); you may do that "hot", while the unit is already on. After a while, you should see the device in the sub-menu **Network**, with the **Wi-Fi settings (External 1)** heading. Enter this sub-menu and configure the Wi-Fi network access. Refer to 4.7.2, Network sub-menu, Network > Wi-Fi settings section for more details:

- Tap **Network List** and select the desired Wi-Fi network from the list that is detected by the unit. Doing so fills up automatically the following setting **Name (SSID)**.
- Alternatively, enter manually the appropriate SSID if necessary.
- Leave **Security** with "Auto" setting (let the ScoopTeam find the appropriate security scheme on the network), except if there is a definite reason not to do so.
- Enter the password to get access to the Wi-Fi network.
- As an initial setting, select "Middle" for the **Network Quality** setting. *As a Wi-Fi network is often crowded, you might have to change this later for a more "pessimistic" setting.* Monitor the transmission quality in further connections, and check if the setting is appropriate.
- Lastly, switch the Wi-Fi interface to "Active", and validate the whole set up (tap the ✅ icon).
- You can check the status of the Wi-Fi link in the **Status** menu, **Wi-Fi (ext 1)** row: network name (SSID), connection status and/or IP address, signal strength. In addition, in the operation pages the status bar includes a status icon related to the Wi-Fi network.

ⓘ *Note: if you have already set up the unit for Wi-Fi access but removed the device (or switched off the unit), and you just re-plug the device, the ScoopTeam will automatically reconnect to the network as before, with no need to remake or re-activate this whole setup.*

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[1] However, less powerful than using the integrated mobile access module; the differences are described in 2.3.4, Using an external USB module.

4.4.3. Setting the coding configuration

Access level: Expert

ⓘ *When preparing an AoIP link with SIP, and if you expect to receive an* incoming call*, you don't need to go through this, because the unit will simply "follow" the proposed setting from the calling device. Just skip this part in such case.*
Conversely, without SIP ("Direct RTP" mode), in any case you must set precisely the same configuration on both ends of the link ("caller" and "receiver").

The coding configuration is done via the **Coding** menu.

The **Coding** / **Codec 1** sub-menu deals with the settings for Codec 1.

The **Coding** / **Codec 2** sub-menu deals with the settings for Codec 2. **It is disabled when the codec mode is "Single Codec" (see above in 4.4.1, case 1 and case 2).** When it is active, it provides the same parameters as the menu for Codec 1, but the settings are independent from each other.

Each codec on the "Connection" page includes a "Coding" shortcut button that leads directly to the appropriate **Coding** / **Codec n** sub-menu when tapped.

The main setting is of course the **Algorithm**: tap this row and the list of available algorithms is displayed; pick the desired one. Proceed with the other parameters as needed. You can find more details in 4.7.3, Coding sub-menu.

You will notice that for every setting, the proposed list depends on the current network, and depends on the values for the other parameters. *As a result, it is not possible to build an invalid combination of coding parameters.*

When the configuration is complete, **make sure to tap the** ✅ **icon** to confirm the entry. Conversely, you can cancel the whole configuration by exiting the sub-menu without doing this.

4.4.4. Launching calls

Once the preliminary settings are done, the procedure for a call on a codec (Codec 1 or Codec 2, depending on the situation) is straightforward: dial a destination "number" and launch the call.

First of all, hit the [icon] shortcut key and you reach the "Connection" page. Refer to the above chapter describing this page and its elements (4.3.8, "Connection" page).

Preliminary note about the "number"

What we call here for simplicity a "number" is more generally a destination identifier. Its type depends on the transmission network and, for audio over IP, on the protocol used. The possible types are:

- Classical **phone number**: this is used for an ISDN destination, or a mobile voice call. A leading "+" sign is used if the international prefix in included. If a sub-address if necessary, append a "*" sign and the sub-address number.
  Example "+33141361268*12"

- **IP address**: for an AoIP connection (SIP protocol or Direct RTP). The address is in the classical "dot-decimal" format. If a port number must be specified, append ":" and the port number.
  Example: "192.168.24.65:5080"
  For the SIP protocol, the port number to specify is the SIP port, if it differs from the default ("SIP / SIP Port" in the AoIP parameters). For "Direct RTP", it is the RTP port, if different from the default ("Direct RTP / RTP port" in the AoIP parameters).

- **SIP URI** (Uniform Resource Identifier): this is an identifier with a general format *username@domain_name*, using alphanumeric characters.
  Example: John@myserver.com
  Actually *username* most often only includes numbers, for easier dialling on VoIP terminals. In addition, the *@domain_name* part may be omitted if the caller is registered on the same server (as the called unit).
  Eventually, a SIP number may be purely numeric, and very similar to a telephone number.

Direct dialing

ⓘ *In case of a SIP call, remember that in most cases the unit must be registered on a SIP server! Check the* SIP *bar in the status panel on top of the screen.*

On the area of the codec you want to call with, tap the "Dial" button [icon]: the dial keypad pops up. Enter/edit here the desired number, and tap the "OK" button.

ⓘ *Whenever an ISDN connection requires two numbers (128 kbit/s coding with two different numbers for the remote device), enter the two numbers separated by a comma, as in this example:*
0123467810,0123467820
Similarly, when using Double IP Streaming, enter the two SIP URIs separated by a coma.

You can see now the number set inside the "Connection pad", ready for subsequent calls.

Tap the connection pad: the outgoing call is started towards the number. The pad changes colour to show the various phases, and ends up red when the connection is established with the remote device. A text line above the pad provides an explicit status as well. Chapter 4.5.3, "Connection pads" describes the detailed behaviour of this element.

Releasing a connection

For releasing the connection, just tap the connection pad again: a dialog box pops up, asking for confirmation. Confirm by tapping the "OK" button.

ⓘ *The ScoopTeam features an automatic release for AoIP connections: whenever a connection is established, but the unit does not receive a stream for more than 30 seconds, it automatically releases the link and comes back to the "idle" state. In this way the unit cannot stay "stuck" in a busy state while for some reason the connection is permanently broken.*
*It is possible to amend this feature for Direct RTP connections: see "**Network > AoIP settings** sub-menu" section in 4.7.2, "Network sub-menu".*

Recalling the same number

Once the desired number is set in the connection pad, you just have to tap the connection pad to recall it at any time.

Recalling a previous number

Tap the "Call history" button : a pop up box shows a list of the numbers previously called with the unit, in chronological order (most recent on top). Scroll the list and pick up a number, then accept by tapping "**OK**".

The connection pad is now "loaded" with the number you have selected, and you can call it by tapping the pad.

Using the directory

Tap the "Book" button : the menu is called, directly into the **Book** / **Call profiles** sub-menu. Select the suitable item in the list, and load it into the codec.

The connection pad is now "loaded" with the number you have selected, and you can call it by tapping the pad.

ⓘ *With a Call Profile you load not only the destination number, but also (depending on the profile) a network type and coding configuration that are linked with the profile.*

4.4.5. Receiving calls

Actually there is no additional setting to do for receiving calls. Incoming calls are automatically accepted by the ScoopTeam, with some differences depending on the type of network involved.

ISDN incoming calls

When the 5A System is active, receiving calls is simple. When a call is received, the codec automatically "unhooks" and recognises the coding algorithm and protocol used, and finally sets the link automatically. In simple words, the unit "follows" the calling unit.

When the 5A System is not active, you should first configure the codec for the desired coding algorithm and configuration. When a call is received, the unit will synchronise with the calling device, but the link will usually fail if the calling party has used another coding configuration than expected.

Dealing with the **double codec** mode: the default rule is that Codec 1 answers first if available. If it is already busy with a connection, then the call is accepted by Codec 2. There are two methods for getting a more deterministic behaviour:

- Use a distinctive ISDN number on each codec: this is possible only if the ISDN line supports more than one subscriber number. However this is not always possible with a public ISDN line.

- Use a distinctive sub-address on each codec.

In either case the calling device can select the codec that should answer by calling for the appropriate number and/or sub-address.

ⓘ *If an ISDN line is connected but this line is not the current network interface, the ScoopTeam will however accept incoming ISDN calls, provided that it is not already busy with an AoIP connection. In such event the unit will switch to the ISDN Interface when receiving the ISDN call. Once the connection is released, the ScoopTeam will come back to its previous default interface.*

AoIP incoming calls

Such calls are accepted if they are presented on the interface that is selected as default for outgoing calls. However, there is an exception: while the current interface is the ISDN line, the unit can still register on a SIP server via the primary Ethernet interface, and it can accept calls presented on this interface (either directly or via a SIP server).

Receiving **SIP incoming calls** is very simple, regardless if it is a direct peer to peer link or a call via a SIP server. There is nothing to do… When a call is received, the units negotiate automatically a commonly acceptable coding algorithm, and set the link automatically. On the receiving side, ScoopTeam "follows" the calling unit.

Direct RTP incoming calls are less straightforward to set up, because the unit must be set up beforehand with exactly the same coding configuration as the calling device. In addition, the caller must be aware, in addition of the IP address of the ScoopTeam to call, of the port used for the RTP transmission. *More explanation about the operation with Direct RTP is provided in the section related to the AoIP settings: 4.7.2, Network sub-menu, "*Network > AoIP settings *sub-menu" section.*

Dealing with the **double SIP codec** mode: the best for receiving calls in such mode is to use a couple of SIP accounts on a server, one for each codec. In this case, the two codecs are unambiguously identified with distinctive numbers, and a caller device can address specifically either codec. Otherwise, if there is no SIP registration, or only one, all incoming calls are directed to the same single identifier (IP address or SIP URI). Then the rule is that Codec 1 answers first if available. If it is already busy with a connection, then the call is accepted by Codec 2.

ⓘ *Details about receiving calls with Double Streaming can be found in 6.2, "Using the Double Streaming system".*

4.4.6. "Network quality" setting (IP networks)

Depending on the quality of service provided on the IP network, especially its jitter performance, it is possible to change the stability/latency compromise used by the ScoopTeam. For this purpose, a "**Network Quality**" setting is available among the settings of the network interface (example for the primary Ethernet interface: sub-menu **Network** / **LAN1 settings**). Six choices are proposed:

- "Very High": suitable for a very good quality network, with very low jitter and packet loss rate; latency is the smallest, but the codec will be very sensitive to possible jitter.
- "High": suitable for a good quality and low jitter network; latency is minimal, but the codec will have little tolerance to possible jitter.
- "Middle": intermediate (and default) setting, suitable for a moderate transmission jitter.
- "Low": to be preferred when the network has low QoS, especially for residential ADSL lines. This setting ensures a safer operation, at the cost of a high latency.
- "Bad": to be used when the network is very disturbed, e.g. on mobile networks. The latency is significantly higher than the above setting.
- "Very Bad": to be used in worst case, very poor quality mobile networks. This choice also gives the highest latency.

On a LAN and/or private network with a controlled quality, the "Very High" quality setting is recommended, as it yields minimum latency.

On the contrary, it should be avoided for a link via the Internet, as it can only tolerate a low jitter. One solution can be to start with a "Middle" setting, and move to the "Low" setting, or further, if too much audio disturbance is heard.

ⓘ *This parameter is interface-specific, i.e. a different choice can be set for each network interface.*

The resulting latency depends on this setting, but also on the coding algorithm which is used for the transmission. The table below shows for each combination (coding and "Network quality" setting) the end-to-end latency, from the audio input of a codec to the audio output of the codec at the other end of the link. One should add to this amount of time the possible latency of the transmission network: time from the transmission of an IP packet at one end to its reception at the other end.

| Coding	Very High	High	Middle	Low	Bad	Very Bad	Tolerance
G711	85 ms	120 ms	220 ms	320 ms	420 ms	780 ms	± 10 ms
G722	85 ms	120 ms	215 ms	315 ms	415 ms	775 ms	± 10 ms
MPEG L2 @48kHz	95 ms	140 ms	260 ms	355 ms	450 ms	815 ms	± 10 ms
MPEG L2 @32kHz	155 ms	179 ms	285 ms	395 ms	505 ms	827 ms	± 10 ms
MPEG L2 @24kHz	190 ms	190 ms	320 ms	415 ms	510 ms	850 ms	± 10 ms
AAC-LC @48kHz	135 ms	175 ms	275 ms	385 ms	470 ms	830 ms	± 10 ms
HE-AAC @48kHz	245 ms	275 ms	355 ms	485 ms	570 ms	910 ms	± 15 ms
OPUS	85 ms	125 ms	220 ms	320 ms	420 ms	780 ms	± 10 ms

Tableau 7 – Latency as a function of the coding algorithm and the "Network Quality" setting

4.5. Details on specific objects

4.5.1. Status panel

Layout

The status panel is an area on the top of the screen, visible on most pages of the user interface.
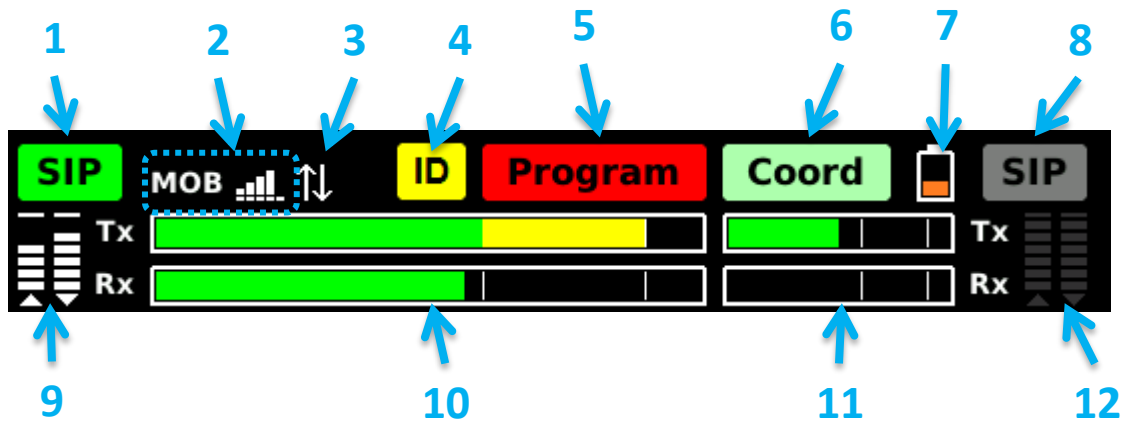
1. SIP registration status for the Account n°1 / Program line (on the picture: registered)
2. Current network interface information: type, and additional info (on the picture: mobile, and radio signal strength meter)
3. Network interface connection status (on the picture: mobile data connection active and OK)
4. "Sound ID" indicator: when displayed, shows that the sound ID is being played (see 2.1.10, Mixing capabilities)
5. Program line/connection state (on the picture: link established)
6. Coordination line/ connection state (on the picture: link idle, no number is preset)
7. Battery state, when batteries are inserted
8. SIP registration status for the Account n°2 / Coordination line (on the picture: not registered)
9. IP transmission quality for the program connection
10. Audio level (send and receive) for the **Program** channel/line: PPM, ref 0 dBFS
11. Audio level (send and receive) for the **Coordination** channel/line.
12. IP transmission quality for the coordination connection (here irrelevant, as the link is idle)

The above applies to the "double codec" configuration, or the "single codec + mobile voice" configuration (respectively case 3 or case 4 in 4.4.1 above).

In the case of a single codec with mono coding (case 1 in 4.4.1 above), the panel has the following layout:

Only the "Program" status indicator is shown, because only a program channel is supported in this configuration.

In the single codec case with dual mono or stereo coding (case 2 in 4.4.1 above), the panel has the following layout:
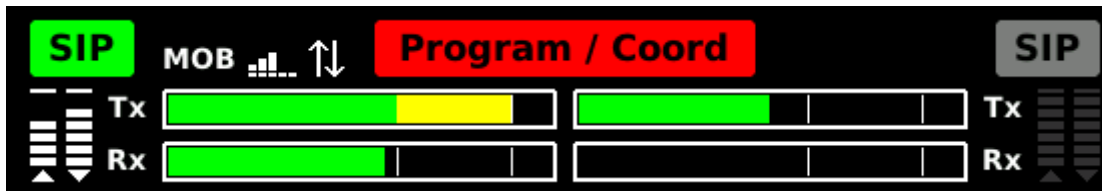
The "Program" and "Coord" status indicators are merged, because the coordination channel is transmitted at the same time as the program channel, inside the same (dual mono) connection.

The following describes more details about the various indicators found in the status panel.

Program and Coord connection state

Each indicator shows the current state of respectively the Program connection and the Coordination connection, with a color code as follows:

Light green: "Idle" state: no connection, no destination is preset

Green: "Ready" state: line is idle, and a call destination is preset

Yellow: Connection in progress

Red: Connection established

Light brown: Connection release in progress

Pink: Connection problem

Red: Connection established (single codec, dual mono or stereo)

As can be seen on the last example, the indicators are merged as a "Program / Coord" indicator when the unit is set as a single codec, with one dual mono connection carrying both the Program and Coordination channels.

SIP registration status

For each SIP account (1 or 2), an indicator shows the SIP registration status with a color code.

SIP (grey)	Grey: SIP registration not set / not active
SIP (green)	Green: Registered on SIP proxy/server, current network is IP protocol, ready to send or receive calls via the SIP server
SIP (dark green)	Dark green: Registered on SIP proxy/server and ready to receive calls via the SIP server, but current network is ISDN
SIP (red)	SIP registration failure

Notes:

- If the unit is set as single AoIP codec, there is possibly a single SIP registration, using the account n°1. There is no separate AoIP link, and the account n°2 is unused; its indicator is never active (irrelevant). This is also true when using the configuration with a single codec and mobile voice coordination.
- If the unit is set as a double AoIP codec, each line has got a distinctive registration and the two indicators are independent from each other (except that both accounts are on the same SIP server).

Network interface information

This area provides information on the current network interface: type of network and additional information, as shown by the table below.

Interface	Type displayed	Additional information
Ethernet, primary	**ETH**	(None)
Ethernet, auxiliary	**ETH**	(None)
Mobile, internal	**MOB**	• Radio strength (5-bar indicator) • SIM alerts : missing, locked
Mobile, USB device	**MOB**	• Radio strength (5-bar indicator) SIM alerts : missing, locked
Wi-Fi, USB device	**WiFi**	• Radio strength (5-bar indicator)

Audio level meters

These are peak program meters (PPM), referenced to 0 dBFS. The bars are:

- Red between -3 dBFS and 0 dBFS.
- Yellow between -12 dBFS and -3 dBFS;
- Green under -12 dBFS ;

IP Transmission quality meters

When an IP transmission link is running, these indicators show the quality of the link (the higher the bar, the better the quality). If the line/channel is idle or is not IP transmission, the indicator is dark grey.

The left bar shows the "transmit" direction (ScoopTeam towards the remote device), and the right bar shows the "receive" direction.

ⓘ *The left bar may not be relevant for connections with codecs or devices from other brands than AETA, if they do not support the RTCP protocol.*

The quality meters provide a global indication at a glance. If desired, much more detailed information is available in the "Status" section of the menu: **Status** > **Audio Connections**. Refer to chapter 4.7.7, "Status sub-menu".

Battery state indicator

If no batteries are installed, the icon is not visible. When batteries are inserted in the compartment, the indicator can show the following states:
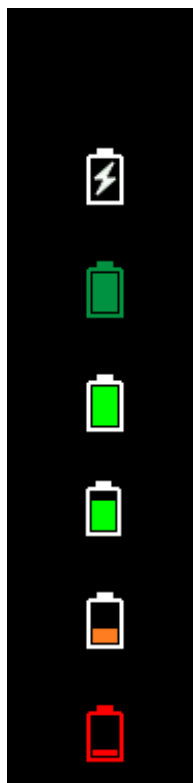
Blank area: no battery inside the unit

Running from DC, batteries charging

Running from DC, batteries fully charged

Running from batteries (DC disconnected), batteries full

Running from batteries (DC disconnected), high battery level

Running from batteries, low battery level (remaining capacity < 40 %)

Running from batteries, batteries at critical level (less than 20 % remaining). Connect DC before the unit switches off automatically!
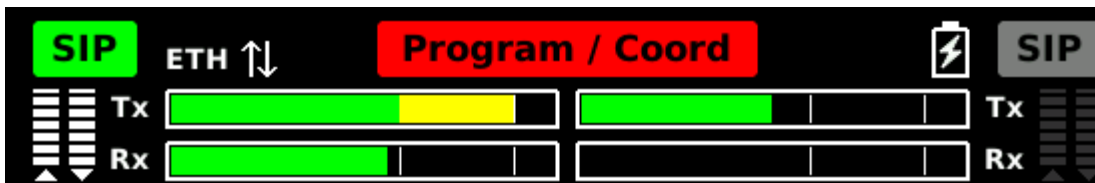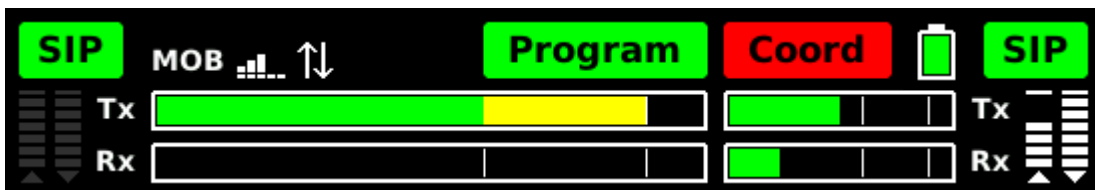
Example 1:
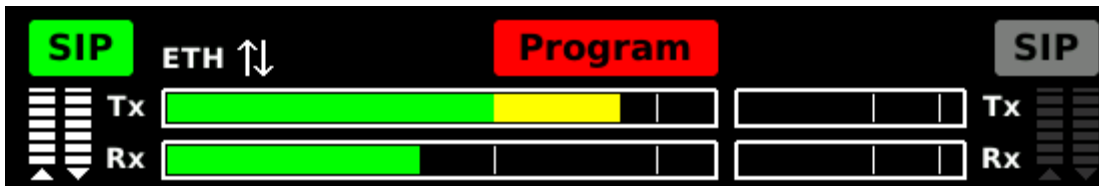
The unit is set as a single mono codec, and a SIP connection is active through an Ethernet interface. The unit runs from DC, with no battery inserted.
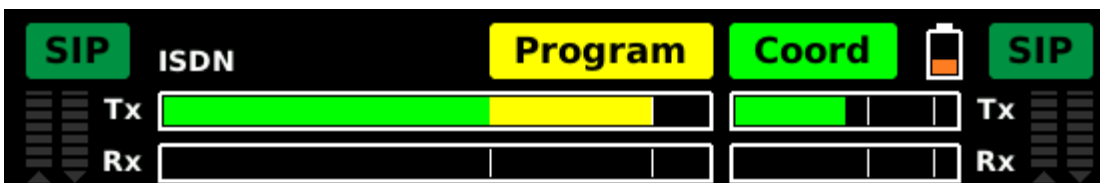
Example 2:

The unit is set as a single *dual mono* codec, and a SIP connection is active through an Ethernet interface. Batteries are being charged.

Example 3:

The unit is set as a double codec on a mobile network and running from batteries (full charged). A SIP connection is active for the coordination channel. The program channel is ready with a preset call destination.

Example 4:

The unit is set as a double codec. The two SIP accounts are registered, but the current interface is ISDN. A call is being set for the program channel.

Compact status panel

Some pages (example: "Mixing" page) show a "compact" variation of the status panel, with a lower height and a few elements less, as on the picture below:

This compact version does not include audio meters for the received signals, and does not include transmission quality meters. *Note the "empty battery" warning!*

4.5.2. Information banner

The information banner is located at the bottom of the home page under the buttons. It shows a summary of the current configuration of the codec(s), and of the current state of the audio link(s).

The available information can be easily understood from the two examples below.

```
Program  Ethernet / Opus 48kHz Mono 64kbit/s
         Connected / 902796@sip.aeta-audio.com:5070

Coord    Ethernet / G.722
         Idle
```

In this first example, the unit is set as a double AoIP codec using the Ethernet interface for transmission. Codec 1 (Program transmission) is currently connected to the device 902786@sip.aeta-audio.com, and the coding configuration is Opus at 64 kbit/s. Meanwhile, Codec 2 (Coordination) is configured for G722, with no audio link currently (idle).

The following example shows the banner's aspect when ScoopTeam is set as a single codec:

```
Program  Ethernet / Opus 48kHz Mono 48kbit/s
         Idle / 902796

   - - -
```

The "- - -" mark reminds that there is no second codec for Coordination.

4.5.3. Connection pads

The "Connection" page includes two such areas, one for each codec (Program and Coordination). In the single codec mode, the second codec (Coordination) is not used. Apart from this difference, both connection pads have the same layout and behaviour:
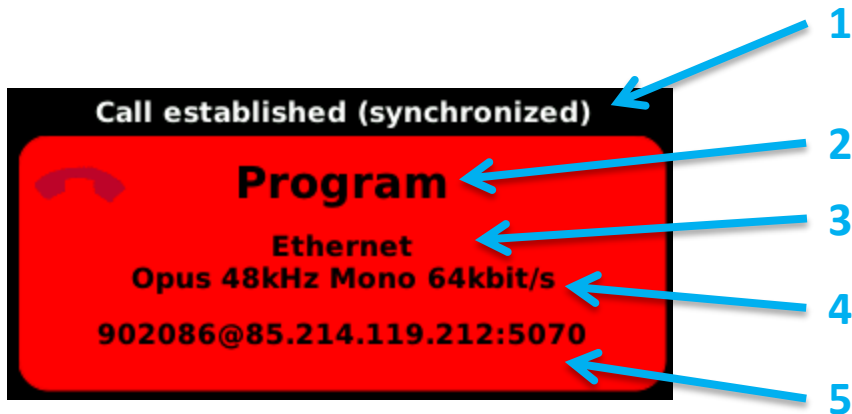

1. Current status or step. This text provides more details than the color code of the area (see further). *When the codec is idle, this text holds the last release cause.*
2. Codec : Program or Coordination.
3. Current network interface. When idle, this is the (main) interface for outgoing calls.
4. Current coding configuration.
5. Number or address of remote device. When in the "ready" state, this is the destination prepared for calling out. When on line, this is the identification of the remote device.

The color of the area shows the current state of the connection:

Light green: Idle (no connection)
Green: Ready (no connection, a destination is preset)
Yellow: Connection in progress
Red: Connection established
Light brown: Connection release in progress
Pink: Connection problem
Grey: irrelevant *(used for the Coord pad in single codec mode)*

4.5.4. "Opus live bit rate" box

While a connection runs with the Opus coding algorithm, it is possible to change the bit rate "on the fly": tap the "Coding" button on the connection page, and the following box pops up:
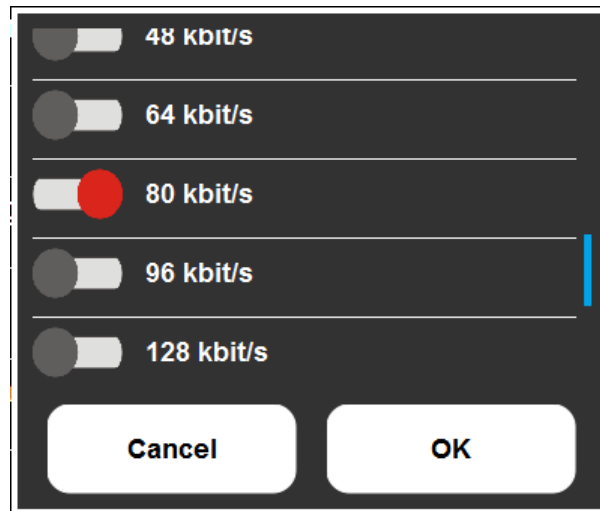
Swipe the screen to scroll the list of available bit rates. Select the desired bit rate and tap "OK".

4.6. Management of the configuration profiles

The configuration profiles ease configuration changes and the setting up of routine links. There are four types of profiles:

- "**Call profiles**", which include call parameters for a given **destination** (number(s), coding, etc.). Such a profile is similar to a directory entry, but in addition it can memorize the coding parameters.

- "**Presets**" memorize the **network access** parameters. These are the settings found in the **Network** menu. Recalling a *preset* is a quick way to restore the configuration needed for a given line or network.

- "**SIP Profiles**" memorise the data for registering on a SIP server. These are settings found in the **Network > AoIP settings > SIP > Account n** menu. Recalling a SIP profile allows to quickly register the unit on a new server/account.

- "**Snapshots**" memorize the settings for the **audio** interfaces and processing. These are settings done via the various pages of the user interface ("Mixing", "Inputs", "Aux", "Outputs", "Monitoring"), and those found in the **Audio** menu.

These various memories are usable locally, but in addition they can be imported/exported from/to a computer, using the remote control tools.

Each call profile includes the following elements:

- *Name* assigned to the profile.

- *Remote Number(s)*: ISDN or telephone call number(s), or IP URI(s), IP address, depending on the transmission mode.

- *Coding configuration* associated with the profile.

A call profile may have no number included; in such case it is used for quick and safe recall of a given coding configuration.
Conversely a call profile may have no associated coding configuration included; in such case it is just like a directory entry, used for instant dialling a known destination.

The profiles are managed and used via the **Book** sub-menu (refer to details in 4.7.5, Book sub-menu).
For call profiles, the available possibilities are:

- Create a *New* profile, assigning it a name, possibly call number(s), and possibly an associated coding configuration.

- *Load* a profile: the ScoopTeam then loads the number memorized in the profile, and configures itself with the coding configuration included in the profile.

- *Delete* an existing profile.

- *Update* a profile, after having made changes in its parameters, or after renaming it (in this latter case a new profile is created with the new name).

For presets or snapshots, the available possibilities are:

- Create a *New* profile, which will memorize all the current network settings (for a preset) or all the current audio settings (for a snapshot), assigning it a name.

- *Load* a preset or snapshot: the ScoopTeam then restores the settings included in the profile.

- *Update* a preset or snapshot: the ScoopTeam then overwrites the preset or snapshot with all the current network settings (for a preset) or all the current audio settings (for a snapshot).

- *Delete* a preset or snapshot.

For SIP profiles, the available possibilities are:

- Create a *New* SIP profile, which will memorize the registration data currently set on one of the accounts (selectable), assigning it a name.

- *Load* a SIP profile: the ScoopTeam then restores the data included in the profile and applies them to one of the accounts (selectable).

- *Delete* a SIP profile.

ⓘ *Note: you are not allowed to create a SIP profile from a "factory" SIP account. However, it is easy and straightforward to reload these accounts, as they are built-in: menu **Tools > Troubleshooting > Reset** and select "**Reload the Factory SIP Accounts**".*

4.7. Description of the menu

The menu of the ScoopTeam is a tree-structured menu that allows viewing and editing many parameters which are not accessible from the various pages on the user interface. This menu is accessed through various shortcut buttons on the various pages, and especially the "Settings" page.

The following describes all details for the various sub-menus and their parameters.

4.7.1. Menu organisation and operation principles

The first level of the menu structure is shown in the diagram below.
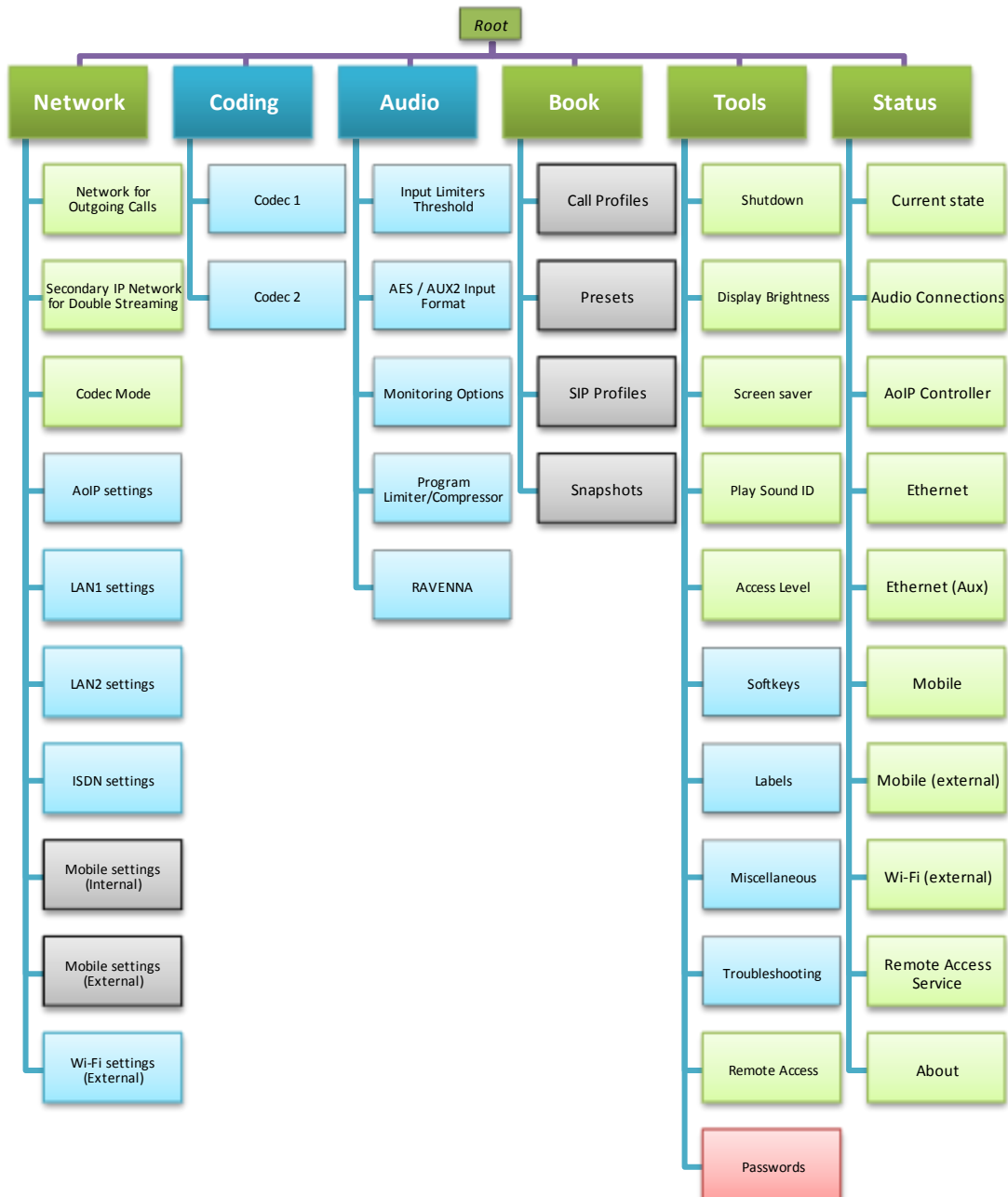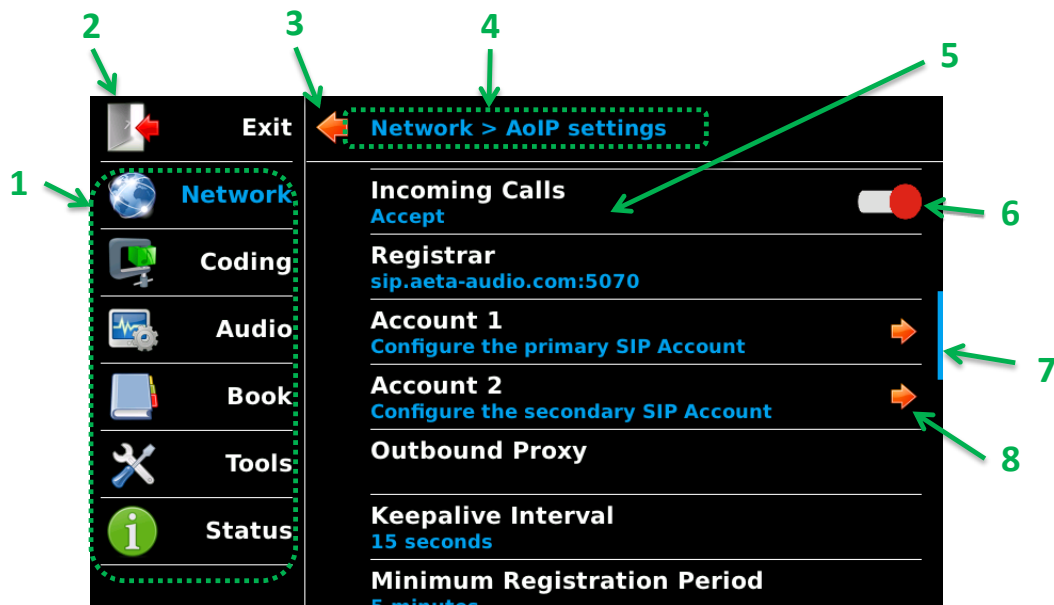

**Figure 6 – User interface pages: overview**

The general layout of a menu screen is shown on the picture below:

1. Shortcuts to the top level menu branches; tap one to reach directly a top level sub-menu.

2. "Exit" icon: tap to exit the menu and go back to the last page (before entering the menu).

3. "Backwards" icon: tap to move one level up in the tree structure.

4. "Path": location of the current page in the tree structure.

5. Elementary setting: name of setting/parameter on the first line, and below the current setting or value.
   The second line is **blue** for a switchable/editable setting or **grey** when the setting is "read-only": status information, or non-editable parameter (e.g. bit rate for G.722 coding).

6. "Switch" icon: tap to toggle a "binary" parameter on/off, or active/disabled, etc.

7. Scroll bar; shows the current position in a long list of items. This bar is shown when the sub-menu does not fit all inside the screen. To access the rest of the sub-menu, "swipe" the screen: touch somewhere in the sub-menu and scroll up/down while holding your finger down.

8. "Enter" icon: tap to enter a sub-menu. *On a sub-menu row like this, the blue sub-title is a complementary description for the sub-menu.*

For switching/editing a setting, tap the item: the suitable entry tool pops up, depending on the parameter to set:

- **Virtual keyboard** for entering text or numeric data :
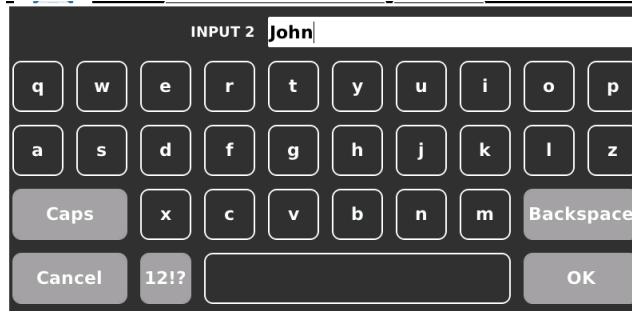
- **List box** for a selection among a number of possible choices:
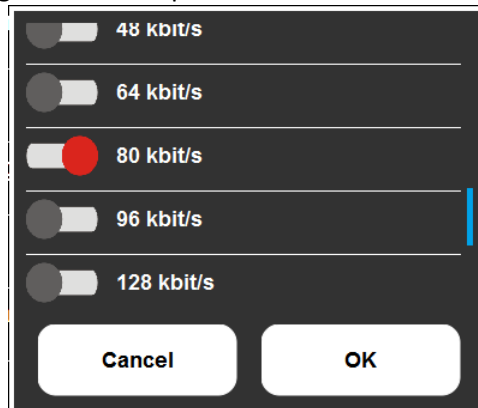
For some parameters you have a specific icon on the right end of the item, in which case you must tap this icon:

- "Switch icon" as shown above: tap to toggle a "binary" parameter on/off, or active/disabled, etc. The result is reflected in the blue sub-title of the parameter.

- "Virtual knob" icon as below ("Display brightness" setting):
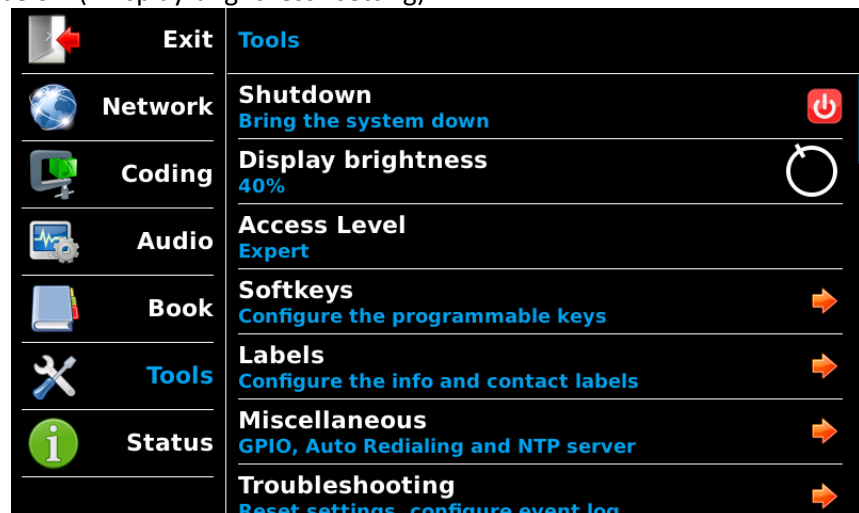
  Tap the icon: it turns **blue**, while the displayed value (40%) turns **red**. Rotate the rotary encoder to reach the desired setting. You can tap the icon again and it turns back to white: the setting is locked. *It is also locked if you touch another item, leave the sub-menu, or exit the menu.*

- "Shutdown" icon: also visible on the example above. Tap it to switch off the unit. You must confirm with OK (otherwise touch anywhere else).

After changing a parameter, in some cases a "check" icon will appear in the top right corner, as in the example below:
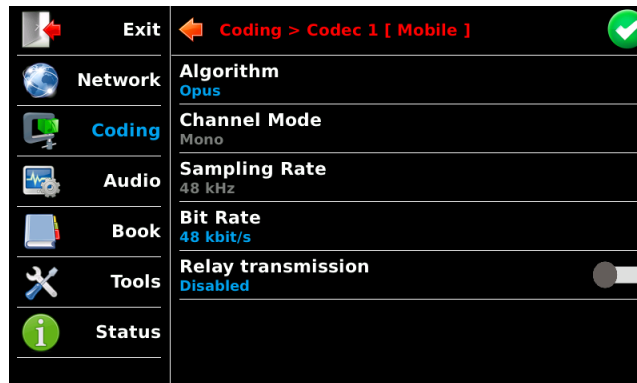
This means you must not leave the sub-menu without first tapping the icon for applying the changes. However, before that you may edit/change several settings inside the current sub-menu. *If you don't confirm, all the changes are cancelled.*

Lastly, in the **Status** sub-menus, you will see a "refresh" icon in the top right corner, as on this picture:
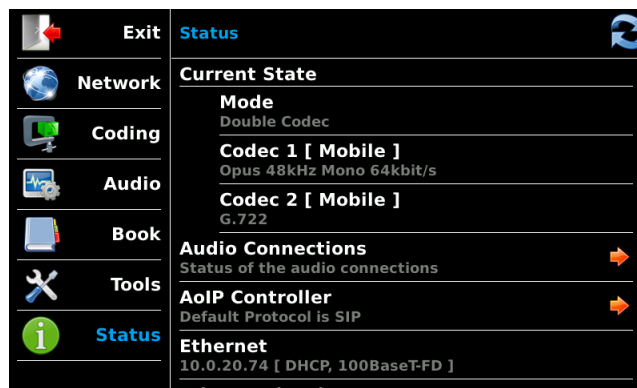
Some of the information items may not be refreshed automatically; tap this refresh icon to update their state on the display.

The following chapters describe the sub-menus and the parameters they provide access to.

4.7.2. Network sub-menu

Item / Label	Type / Choices / Range / Description
Network for outgoing calls Ethernet	Ethernet | Ethernet (aux) | ISDN | Mobile | Mobile (Ext) | Wi-Fi Selection of the network interface that will be used when an outgoing call is dialled and launched. Only existing interfaces are listed.
Secondary IP Network for Double Streaming Mobile	Ethernet | Ethernet (aux) | Mobile | Mobile (Ext) | Wi-Fi Selection of the interface to be used for the redundant link when using Double Streaming. Only existing IP interfaces are listed. *This selection is locked unless Double Streaming is activated (see "AoIP settings").*
Codec Mode Double Codec	Single Codec | Double Codec | Single Codec and Mobile Voice coordination This is a major parameter for the operation of the unit; it is described before in the "Basic options" chapter.
AoIP settings Configure the AoIP controller	*Tap to enter the sub-menu (Expert level required)* See "AoIP settings" table below
LAN1 settings Configure the primary Ethernet Interface	*Tap to enter the sub-menu (Expert level required)* See "LAN1 settings / LAN2 settings" table further
LAN2 settings Configure the auxiliary Ethernet Interface	*Tap to enter the sub-menu (Expert level required)* See "LAN1 settings / LAN2 settings" table further
ISDN settings Configure the ISDN controller	*Tap to enter the sub-menu (Expert level required)* See "ISDN settings" table further. *This sub-menu is not available if the ISDN option is not installed.*
Mobile settings (Internal) Configure the Mobile controller	*Tap to enter the sub-menu* See "Mobile settings (Internal)" table further. *This sub-menu is not available if the ISDN option is not installed.* "SIM Missing" is displayed if no SIM card is inserted in the unit. "PIN Required" displayed if the SIM card is locked.
Mobile settings (External) Configure the Mobile controller	*Tap to enter the sub-menu* See "Mobile settings (External)" section further. *This sub-menu is not available if the suitable option is not installed or no USB mobile access device is connected.* "SIM Missing" is displayed if no SIM card is inserted in the unit. "PIN Required" displayed if the SIM card is locked.
Wi-Fi settings (External) Configure the Wi-Fi controller	*Tap to enter the sub-menu* *See "Wi-Fi settings (External)" table further.* *This sub-menu is not available if the Wi-Fi option is not installed or the USB Wi-Fi device is not plugged.*

Table 8 – Network sub-menu description

The sub-menus are described below.

"Expert" access level required.

This sub-menu groups all settings related to the Audio over IP (AoIP) transmission. It is relevant for Ethernet interfaces but also for all other IP interfaces: integrated module for mobile access, or external mobile access USB module, or Wi-Fi access USB device.

Item / Label	Type / Choices / Range / Description
Default Protocol for Outgoing Calls **SIP**	SIP | Direct RTP Selection of the protocol that will be used for an outgoing call (more precisely, an AoIP session initiated by the operator of the unit). SIP is the standard choice; "Direct RTP" is a mode without signaling used for instance for compatibility with devices that do not support SIP. *This selection is not relevant for accepting either mode for an "incoming call".*
Double Streaming for SIP Calls **Disabled**	Active | Disabled Activate or disable Double Streaming. *Activation is not allowed if "Codec Mode" is set as "Double Codec".*
STUN	*Section header* This protocol, used along with SIP, allows ScoopTeam to discover its public IP address when it accesses the Internet through a router with NAT. You can use AETA's server: stun.aeta-audio.com.
State Disabled	Active | Disabled Enable or disable STUN. In this way you can disable STUN without erasing the STUN server address.
Server stun.aeta-audio.com	< enter IP address or domain name > Address of the STUN server, either in numeric form or with a symbolic name.
SIP	*Section header* This section groups all settings related to the operation when using the SIP protocol, with or without a SIP server.
Registration Active	Active | Disabled Enable or disable the registration on a SIP registrar server. You can disable the registration and hence stop using the SIP server, without having to erase account parameters; they stay ready for later use.
Incoming Calls Accept	Accept | Disabled "Accept": the codec accepts incoming SIP calls (regardless whether the codec is registered or not on a SIP server). "Disabled": incoming SIP calls are rejected with a "busy" indication.
Registrar sip.aeta-audio.com	< enter IP address or domain name > URL of the SIP server, in case one is used. You can also enter a numeric IP address. If a non-standard port is used, specify the port after a colon like in "mySIPserver:5070".

Table 9 – Network > AoIP settings sub-menu description (1 of 4)

Item / Label	Type / Choices / Range / Description
Account 1 Configure the primary SIP account	*Tap to enter the sub-menu (expanded below)* This menu is used to enter the parameters of the account n°1 on the SIP server, when using such service. These parameters should be provided by the administrator of the SIP server.
User Name 902099	< enter alphanumeric name > Enter the user ID as provided with the account. This field corresponds to a number that a remote party can call to initiate a link with the codec when it is registered on the server. If this field is left empty, codec 1 does not register on the SIP server.
Display Name ScoopTeam	< enter alphanumeric name > This (optional) parameter may be displayed by the remote agent during a link; you can enter any text you wish. *If this field is left empty, the actual value is the SIP URI of the unit.*
User Authentication	< enter alphanumeric name > Enter here the authentication ID as provided with the account. *If this field is left empty, the actual parameter is the same as the User Name, which is appropriate in most cases.*
Authentication Password ********	< enter alphanumeric password > Password provided with the SIP account.
SIP Port 5060	< enter numeric value > The default port for SIP signalling is 5060 (UDP), you can enter a different value here. *Note: this port number is valid for the ScoopTeam itself, not the server. For the server side, refer to the "Registrar" setting above.*
RTP Port 5004	< enter numeric value > The default port for RTP/SIP audio streams is 5004 (UDP), but you can enter a custom value here if desired

Table 10 – Network > AoIP settings sub-menu description (2 of 4)

Item / Label	Type / Choices / Range / Description
Account 2 Configure the secondary SIP account	*Tap to enter the sub-menu (expanded below)* Here you can set the parameters for the account n°2, used for the second link (coord.) in double AoIP configuration, or with Double Streaming. The parameters have same definition as for account n° 1.
User Name 9020992	< enter alphanumeric name > Enter the user ID as provided with the account. If this field is left empty, codec 2 does not register on the SIP server.
Display Name ScoopTeam_2	< enter alphanumeric name > This (optional) parameter may be displayed by the remote agent during a link. *If left empty, the actual value is the SIP URI of the unit.*
User Authentication	< enter alphanumeric name > Enter here the authentication ID as provided with the account. *If left empty, the actual parameter is the same as the User Name.*
Authentication Password ********	< enter alphanumeric password > Password provided with the SIP account.
Use alternate SIP Port Disabled	Active | Disabled If disabled (default), the SIP port for the secondary account is the same as the primary account.
SIP Port 5060	(Read only) SIP port for the secondary account. If you select the "alternate" port, the SIP port is [port for account 1] + 2.
RTP Port 5006	(Read only) The RTP port for the secondary account is [port for account 1] + 2.
Outbound Proxy	< enter IP address or domain name > IP address or domain name of a proxy server through which SIP transactions and audio streams are relayed. This parameter is optional, to be filled only if such server is used in the organization. *Note: contrary to the SIP registration data, this parameter is active even if « SIP Registration » is disabled.*
Keepalive Interval 15 seconds	5 seconds| … | 180 seconds Whenever necessary, the codec performs transactions to "keep alive" a translation through a NAT router. This setting defines the maximum time left between two transactions with the SIP server. It should be set on a value lower than the NAT translation time out of the router.
Minimum Registration Period 5 minutes	30 seconds| … | 60 minutes Sets the interval between two re-registrations on the SIP server. Actually the server always imposes a maximal period; therefore this setting is optionally used to make the registration renewal more frequent, if its value is lower than the time set by the SIP server.

Table 11 – Network > AoIP settings sub-menu description (3 of 4)

Item / Label	Type / Choices / Range / Description
Direct RTP	*Section header*
	This section groups all settings related to the operation without the SIP protocol.
Incoming Calls Accept	Accept | Disabled "Accept": the codec accepts incoming RTP streams. "Disabled": incoming RTP streams are ignored.
Send Only Disabled	Active | Disabled If activated and the operator sets an outgoing call, the unit does not expect to receive an RTP stream and rather decodes its own stream. The device keeps sending data until the user releases the link on purpose. In this use case the "Release Timeout" parameter (see further) is irrelevant. *This parameter should not be active if Direct RTP "Incoming Calls" are accepted.*
RTP Port 9000	< enter numeric value > Port allocated to the transmission in Direct RTP mode; *must not be the same as the RTP port for SIP.*
Release Timeout 30 seconds	1 second| … | 30 seconds (default 30 seconds) After a link is established, if no data is received within this time period, the device "releases" the link (i.e. stops sending its own RTP stream) and comes back to the idle state (Direct RTP Incoming Calls Accept is unchanged).
Quality of Service	*Section header*
RTP DSCP 0	< enter numeric value > If the transmission network supports DiffServ to manage the media priority, you can set here the type of service which is assigned to the stream transmitted by the codec. You should enter here a decimal value, e.g. 46 for the maximal priority code "Expedite Forwarding". The default value is 0, "Best Effort".
Packet Replication None	None | Short Delay | Interleaved (recommended) (See also xxx regarding this feature) You can select here the setting for packet replication: **None** = standard mode without replication (default setting). **Short Delay** = packet replication without interleaving. **Interleaved** = packet replication with interleaving: the second packet is delayed; robustness is improved but latency is increased.

Table 12 – Network > AoIP settings sub-menu description (4 of 4)

Network > LAN1 settings / LAN2 settings

"Expert" access level required

LAN1 is the primary Ethernet interface, LAN2 is the auxiliary Ethernet interface. These two sections have the same structure, with settings described below.

ⓘ *If you make any change in this sub-menu, the green "check" icon appears on top right: this is to remind that you must confirm the changes (tap the icon) before leaving the menu.*

Item / Label	Type / Choices / Range / Description
DHCP Active	Active | Disabled Enable or disable DHCP. If DHCP is active, most fields below are grey because they are provided automatically by the DHCP server; they cannot be edited.
IP Address 192.168.1.45	< enter IP address in dot-decimal format > IP address of the interface
Subnet Mask 255.255.255.0	< enter mask in dot-decimal format > Subnet mask
Default Gateway 192.168.1.254	< enter IP address in dot-decimal format > Address of the default gateway
Preferred DNS 90.91.92.93	< enter IP address in dot-decimal format > Main domain name server. Not required if only numeric addresses are used for all IP parameters.
Auxiliary DNS	< enter IP address in dot-decimal format > Auxiliary domain name server (optional).
Link Mode Auto Negotiation	Auto Negotiation | 100BaseT-FD | 100BaseT-HD | 10BaseT-FD | 10BaseT-HD This is related to the physical Ethernet interface. The default setting "Auto Negotiation" is recommended in almost all situations.
Network Quality High	Very High | High | Middle | Low | Bad | Very Bad Select here the *expected* quality of the transmission via the Ethernet interface. The unit switches to internal settings suitable for this quality level. For instance, if you select "Bad", the codec sets a large size for the reception buffer, in order to stand a higher jitter. As a counterpart, the latency will be higher as well. Conversely, if you select "High", the buffer is smaller and the latency is lower, but the system is more vulnerable to possible jitter (fluctuations of the network transmission latency). In doubt, select "Middle".
MAC Address 00:15:97:01:32:53	(hexadecimal 12-digit address, read-only) MAC address of the Ethernet interface. This parameter cannot be edited.

Table 13 – Network > LANx settings sub-menu description

Network > ISDN settings

"Expert" access level required. Sub-menu only available if the ISDN option is installed.

ⓘ *If you make any change in this sub-menu, the "check" icon appears on top right, as a reminder that you must confirm the changes (tap the icon) before leaving the menu.*

Item / Label	Type / Choices / Range / Description
5AS Active	Active | Disabled By default 5AS for ISDN is active (see 2.3.2, 5A System). If required, it can be disabled here.
Protocol Euro ISDN	Euro ISDN | NI-1/NI-2 | NTT NI-1 is used in North America, NTT is the protocol of the Japanese operator NTT, and Euro ISDN (or ETSI) can be used in the majority of the other countries (more details in 2.3.2, Network protocols).
Local Address 1	< enter numeric value > In some cases, it may be necessary to set the local address (or local ISDN number) of the line, and/or it is possible to assign a sub-address to the codec. The local number allows "multiple subscriber numbering" or MSN. This number is usually the number that remote equipment must dial to call your equipment. Configuring this number in the equipment is not mandatory if the equipment is directly connected to the public network. On the other hand, if the equipment is connected to a PABX, the number(s) may be required. The PABX may also impose a unique number for each B channel within the same BRI interface. In such a case, refer to the characteristics and configuration of the PABX. *Proper configuration of the local numbers is essential, and many problems in setting up links originate from mistakes or misunderstandings regarding this configuration. In doubt, leave this number blank! This is usually appropriate for public lines.*
Local Address 2	< enter numeric value > This number applies to codec 2 when operating as a double codec. It follows same rules as the first local address described above.
Sub-Address 1 10	< enter numeric value > This number differentiates several terminals connected to the same ISDN bus, which are allocated the same call number(s). Thus it can be useful in case other devices are connected with the ScoopTeam on the same line, or in double codec mode (see 4.4.5, ISDN incoming calls).
Sub-Address 2 20	< enter numeric value > This number applies to codec 2 when operating as a double codec. It can be useful to discriminate incoming calls for codec 1 or codec 2 (see 4.4.5, ISDN incoming calls).
SPID	< enter numeric value > This "Service Profile Identifier" is only relevant for the NI-1 protocol. This number identifies the services of the ISDN line, and usually includes the subscriber number or part of it. This number is mandatory to operate the line, and is provided by the network operator. *Note: the ScoopTeam only supports one SPID, therefore setting connections beyond 64 kbit/s may not be possible.*

Table 14 – Network > ISDN settings sub-menu description

This sub-menu is only available if the integrated mobile option is installed. It cannot be entered if no SIM car is inserted in the ScoopTeam.

Item / Label	Type / Choices / Range / Description
PIN Code ****	< enter numeric value > *This setting is visible when the SIM card is locked.* Enter the PIN code allowing to use the SIM card and access the network. Entering this code is the first step necessary to operate the mobile access, except if the code for the card is empty or disabled. After entering the PIN, tap the "check" icon (top right) to unlock the SIM card and proceed. The usual safety rules apply to this setting, so be careful not to block the SIM card with repeated mistakes!
PIN Save Disabled	Active | Disabled If this is active, the unit saves the PIN code and unlocks the SIM card automatically when the unit is switched on. It is recommended to disable this feature before changing for another SIM card (and possibly another PIN). However, the unit will not make more than one trial if the memorized code is wrong.
Preferred Radio Access Technology Auto	Auto | 2G | 3G | 4G/LTE *Not visible when the SIM card is locked.* "Auto" is the normal choice, which provides operation on 4G/LTE as long as the conditions allow it, but fallback to 3G/3G+ or GSM/2G if the radio link is too poor (or for dealing with a telephone link, not supported on the LTE network). Alternatively it is possible to force a specific technology. *Do not force 4G/LTE when using the single codec + voice coordination mode, as the voice/HD voice service is not supported on 4G.*
Preferred Network Operator Auto	Auto | [List of operators] *Not visible when the SIM card is locked. Expert access level required.* Normally the unit automatically selects a suitable network. If you want to force a specific operator (which can make sense when using the unit abroad), this menu presents the list of available network so you can manually select one. But not all networks are actually allowed; this depends on the policy of the SIM card's provider!

Table 15 – Network > Mobile settings (Internal) sub-menu description (1 of 2)

Item / Label	Type / Choices / Range / Description
Audio over IP	*Section header*
	This section deals with the mobile data service.
	It is not visible if the SIM card is locked.
Data Service Active	Active | Disabled
	Activating the mobile data service is necessary for mobile AoIP transmission (but useless for mobile voice). This is similar in function to connecting an Ethernet cable on a wired network: by activating the connection the unit is linked to the Internet and it can send and receive IP data.
	Note: once this setting is active, it is memorized: the data connection is performed automatically by the ScoopTeam at startup, when it registers on the mobile network.
	Why disabling the data service? For avoiding unwanted data consumption. For instance, if you only intend to use the voice / HD voice service for coordination, mobile data are useless.
Network Quality Middle	Very High | High | Middle | Low | Bad | Very Bad
	Select here the *expected* quality of the transmission via the mobile interface. This is the same setting as for an Ethernet interface (see above, LAN1/LAN2 settings). *Expert access level only.*
	In doubt, select **Middle**.
Access Point Name internet.apn	< enter alphanumeric chain >
	This and the 3 following settings are parameters for the mobile data service. The essential one is the APN (Access Point Name), mandatory for accessing the mobile IP network. When it is left blank, ScoopTeam automatically sets the APN for most operators. However you may have to enter this yourself.
	The other parameters usually can stay empty; otherwise they should be provided by the network operator, along with the APN.
Authentication Mode PAP	PAP | CHAP
	Only change this if you know what you are doing! Expert level only.
Authentication User Name	< enter alphanumeric chain >
	Leave blank unless you know what you are doing. Expert level only.
Authentication Password	< enter alphanumeric chain >
	Leave blank unless you know what you are doing. Expert level only.
Information	*Section header*
	This section is information about the integrated module and the SIM card. *It is visible even when the SIM card is locked.*
Manufacturer Huawei Technologies	(read-only) Manufacturer of the module.
Model MB907-123	(read-only) Model reference of the module.
Revision 12.1324.65	(read-only) Firmware revision of the module.
IMEI 860461025225090	(read-only) IMEI of the module.
IMSI 208103698521547	(read-only) IMSI of the SIM card (if it is unlocked).
Suscriber Number +33611866818	(read-only) Telephone number of the SIM card (if it is unlocked).

Table 16 – Network > Mobile settings (Internal) sub-menu description (2 of 2)

Network > Mobile settings (External)

When a mobile USB device is inserted, it appears in the list of network interfaces at the end of the menu. Its structure and parameters are exactly the same as for the internal device (see above).

Network > Wi-Fi settings (External)

When the option is installed and the provided Wi-Fi USB device is inserted, it appears in the list of network interfaces at the end of the menu. The sub-menu features the following settings:

Item / Label	Type / Choices / Range / Description
Active Active	Active | Disabled When this is active, the unit tries to connect to the Wi-Fi network as configured by the following items. Before editing the following fields/selections, you must disable this, and validate by tapping the "check" icon.
Network List Select a Wi-Fi network from list	[List of available networks] | Tap to see a list of the detected networks, and select the desired one. This will automatically fill up the two following fields: Name (SSID) and Security.
Name (SSID) WiFi_Net5	< enter alphanumeric chain > SSID of the wireless network to connect to, automatically filled if you have used the list above. You may override the current value of the SSID if needed.
Security Auto	Auto | None | WEP | WPA-PSK | WPA2-PSK Security scheme for the wireless network, handled automatically if you have used the list above. You may override the setting if (and only if) needed.
Passphrase ***********	< enter alphanumeric chain > Password or sentence for accessing the wireless network.
Network Quality High	Very High | High | Middle | Low | Bad | Very Bad Select here the *expected* quality of the transmission via the Wi-Fi interface. The unit switches to internal settings suitable for this quality level. For instance, if you select "Bad", the codec sets a large size for the reception buffer, in order to stand a higher jitter. As a counterpart, the latency will be higher as well. Conversely, if you select "High", the buffer is smaller and the latency is lower, but the system is more vulnerable to possible jitter (fluctuations of the network transmission latency). In doubt, select "Middle".

Table 17 – Network > Wi-Fi (External) sub-menu description

ⓘ *Note: access to a network requiring a certificate is not supported, neither to a network requiring a login on a browser.*

4.7.3. Coding sub-menu

"Expert" access level required.

This sub-menu allows to select the desired configuration for an outgoing call, using the current transmission interface (as selected in the "**Network**" menu). It has two branches, one for each codec; Codec 2 makes sense only when using the double codec mode.

Item / Label	Type / Choices / Range / Description
Codec 1 [*current network*] Opus 48 kHz Mono 64 kbit/s	*Tap to enter the sub-menu (expanded below)* *This menu is used to configure Codec 1.*
Algorithm Opus	G.711 A-Law | ... | Opus Select the coding algorithm. The list depends on the network type (IP or ISDN) and the single/double codec operation. The primary selection done here impacts the available choices in the following settings. (refer to 2.2, "Audio encoding and decoding" about the allowed configurations)
Channel Mode Mono	Mono | Dual Mono | Stereo Dual Mono and Stereo are not available for all algorithms; in addition they are not available in the "double codec" operation mode.
Sampling Rate 48 kHz	8 kHz | 16 kHz | 24 kHz | 32 kHz | 48 kHz In most cases this parameter is mainly informative, and imposed by the coding algorithm.
Bit Rate 80 kbit/s	12 kbit/s | ... | 256 kbit/s The allowed setting range highly depends on the coding algorithm and the network. Many algorithms impose a fixed value at 64 kbit/s.
Relay transmission Disabled	Active | Disabled The auxiliary "relay transmission" feature can be activated here. With ISDN, it is not available in all situations: refer to 2.4 for more.
Codec 2 [*current network*] MPEG L2 48kHz Mono 64 kbit/s	*Tap to enter the sub-menu (expanded below)* *This menu is used to configure Codec 2. It is active only in double codec mode.*
Algorithm MPEG L2	G.711 A-Law | ... | Opus Select the coding algorithm. The list depends on the network type. The primary selection done here impacts the available choices in the following settings. (refer to 2.2, "Audio encoding and decoding" about the allowed configurations)
Channel Mode Mono	(read-only) Reminder, as only Mono is allowed for Codec 2.
Sampling Rate 48 kHz	8 kHz | 16 kHz | 24 kHz | 48 kHz In most cases this parameter is mainly informative, and imposed by the coding algorithm.
Bit Rate 64 kbit/s	12 kbit/s | ... | 192 kbit/s The allowed setting range highly depends on the coding algorithm and the network. Many algorithms impose a fixed value at 64 kbit/s.

Table 18 – Coding sub-menu description

4.7.4. Audio sub-menu

"Expert" access level required.

This sub-menu includes advanced audio settings which are not available on the dedicated audio pages ("Mixing", "Inputs", "Aux", "Outputs"). Refer to 2.1 (Audio interfaces and functions) about the concerned audio functions. The sub-menu also allows configuring the optional RAVENNA/AES67 interface.

Item / Label	Type / Choices / Range / Description
Input Limiters Threshold -9 dBFS	Off | -3 dBFS | -6 dBFS |-9 dBFS |-12 dBFS |-16 dBFS Inputs 1 to 4 each include a limiter in the analog domain. The threshold of this limiter is selected here, applicable to all 4 inputs. The reference 0 dBFS is the clipping level of the input analog/digital converter. Setting "Off" globally disables the limiter for all inputs, whatever the on/off setting on each channel ("Input settings" page).
AES / AUX2 Input Format Analog (AUX2)	Digital (AES 1/2) | Analog (AUX2) Select the format of the input: analog or 2-channel digital
Monitoring Options Configure the monitoring options	*Tap to enter the sub-menu.*
Side Tone Permanently available	Permanently available | Only when routed to Program or Coord. Select the operation mode for the side tone. With the second option, the input of a given position is not heard in the headphone unless the input is routed to the Program or Coordination bus.
Intercom Functions Fade non-relevant signals	Fade non-relevant signals | Mute non-relevant signals Select the behaviour for intercom operation: the non relevant signals are either faded or simply muted while the intercom button is pushed.
Program Limiter / Compressor	*Section header* This section deals with the settings for the limiter/compressor that processes the program and the coordination signals before transmission.
Dynamic Processing Disabled	Active | Disabled Switch on/off the limiter/compressor with this setting.
Limiter Threshold -2 dBFS	[-20 dBFS … -2 dBFS] *Tap the "virtual knob" icon and rotate the rotary encoder to adjust the value by 1 dB steps. You cannot set the threshold below the compressor threshold (next setting).* The limiter prevents the signal from going beyond this threshold.
Compressor Threshold -16 dBFS	[-40 dBFS … -3 dBFS] *Tap the "virtual knob" icon and rotate the rotary encoder to adjust the value by 1 dB steps. You cannot set the threshold beyond the limiter threshold (above setting).* The compressor is active for signals above this level (but below the limiter threshold).
Compressor Ratio 2:1	1:1 (Off) | 1.5:1 | 2:1 | 3:1 | 6:1 Compression ratio; for instance 2:1 means that for 2 dB change at its input, the compressor provides a 1 dB change at its output.
Attack Time Fast	Slow | Medium | Fast Select here the attack time for the side chain (signal amplitude detection).
RAVENNA Configure the RAVENNA interface	*Tap to enter the sub-menu.* *Only visible if the RAVENNA option is installed.* See "RAVENNA" tables further.

Table 19 – Audio sub-menu description

Audio > RAVENNA

> ⓘ *Note: the sub-menu as described here only applies to the "Premium+" version with its optional AES67/Ravenna interface. For a "Premium" version with a 2 channel Ravenna interface, please refer to the previous user manual (Edition G).*

The AES67 interface of the ScoopTeam can be seen as five audio inputs and eight audio outputs. The configuration of the interface is mainly for:

- Selecting signals available on the network for "connecting" them at the inputs of the ScoopTeam.
- Activating or not the outputs of the ScoopTeam to the network, and naming the output stream.

Most of the settings are accessible through the menu of the ScoopTeam, as described below. Advanced parameters such as e.g. PTP parameters or receiver delays, must be managed using an external tool such as the *ANEMAN software* from Merging Technologies.

The **Audio > RAVENNA** sub-menu includes the following settings:

Item / Label		Type / Choices / Range / Description
Device Name ScoopTeam_110024		< enter alphanumeric chain > This name can be modified only with "Administrator" access level. Using the "space" character is not allowed in this field. *After editing this name, you must restart the unit for the change to be applied.*
Sampling Frequency **48 kHz**		44.1 kHz | 48 kHz | 88.2 kHz | 96 kHz | 176.4 kHz | 192 kHz
Sample Resolution 24 bits		16 bits | 24 bits Resolution of the audio samples (24 bits recommended)
PTP Domain Number 0		< enter numeric value > *Change this only if you are sure to know the adequate value.*
IP Addressing 192.168.1.85 [DHCP]		*Tap to enter the sub-menu (expanded below)* *This sub-menu configures the address of the embedded html server of the AES67/RAVENNA interface.*
	DHCP Active	Active | Disabled Enable or disable DHCP. If DHCP is active, most fields below are grey because they are provided automatically by the DHCP server; they cannot be edited.
	IP Address 192.168.1.85	< enter IP address in dot-decimal format > IP address of the interface
	Subnet Mask 255.255.255.0	< enter mask in dot-decimal format > Subnet mask
	Default Gateway 192.168.1.254	< enter IP address in dot-decimal format > Address of the default gateway
	MAC Address 00:07:F5:00:AB:56	(hexadecimal 12-digit address, read-only) MAC address of the Ethernet interface. This parameter cannot be edited.

Table 20 – Audio > RAVENNA sub-menu description (1 of 3)

Item / Label	Type / Choices / Range / Description
Input Streams **Stream 1 [Receiving]** **Stream 2 [Receiving]** **Stream 3 [Idle]** **Stream 4 [Idle]** **Stream 5 [Idle]**	*Tap to enter the sub-menu (expanded below)* *This sub-menu allows to select audio signals from the available RAVENNA streams for feeeding the inputs called "RAVENNA 1" to "RAVENNA 5" in the user interface.*
Input Stream i Extdevice_stream [Channel i]	*Tap to enter the sub-menu (expanded below)* *This sub-menu selects the stream and channel for the input signal "RAVENNA i".*
Stream ID Extdevice_stream	None | Detected_Stream_1 | … | Detected_Stream_n The RAVENNA streams found on the network are listed here; select the stream carrying the signal desired for the input "RAVENNA 1". *After selecting the stream, you must click the "check" icon before proceeding with the channel selection below.* Select "None" for leaving the input "open".
Channel 3	1 | 2 | … The channels available in the stream selected above are listed here. Select the channel to use for the input "RAVENNA i".
Mode Multicast	(Read only) Protocol of the input stream.
Receiver Delay (0..2048 samples) Automatic	(Read only) Network to audio output latency setting. *Fixed "automatic".*

Table 21 – Audio > RAVENNA sub-menu description (2 of 3)

Item / Label	Type / Choices / Range / Description
Output Stream **ScoopTeam_110024_1 [Transmitting]**	*Tap to enter the sub-menu (expanded below)* *This sub-menu configures the 8-channel audio stream produced by the ScoopTeam. This stream carries in its channel 1 the output called "RAVENNA 1" in the user interface, in its channel 2 the output called "RAVENNA 2", and the 6 other channels are direct outputs for the MIC and AUX inputs.*
Stream ID ScoopTeam-00-AB-56	< enter alphanumeric chain > This name can be modified only with "Administrator" access level. It identifies the output stream on the network.
State Active	Active | Disabled Activate if you wish to deliver a stream on the network. If you disable, the ScoopTeam transmits no stream, and is only possibly receiving audio from the network.
Auto Configuration Active	Active | Disabled If you select "Active", the two following parameters are set automatically by the ScoopTeam (recommended).
Multicast Address 239.1.1.85	< enter address in dot-decimal format > Multicast address of the stream output by the ScoopTeam.
Multicast Port 5004	< enter numeric value > Multicast RTP port number
Channel Count 8	(Read-only) Reminder of the number of channels in the output stream.
Packet Time 1 ms	(Read-only) Reminder of the interval between packets. Packets have a size of 48 samples (for 44.1 or 48 kHz) or 96 samples (for 88.2 or 96 kHz).
STATUS	*Section header* This section shows information on the current operation of the AES67/RAVENNA interface.
Link State Active	(read only) State of the Ravenna interface.
Sync State OK	(read only) State of the stream synchronization.
PTP State Master	(read only) Master/Slave state of the interface regarding clock synchronization.
Information	*Section header* This section shows information on the AES67 board.
Manufacturer Merging technologies	(read only) Manufacturer of the Ravenna interface card.
Model ZMAN Family	(read only) Reference of the Ravenna interface card.
Software revision 1.0.7b38341-1.01	(read only) Version of the Ravenna interface firmware.
Hardware Revision ZMAN_Z010	(read only) Version of the Ravenna interface card.

Table 22 – Audio > RAVENNA sub-menu description (3 of 3)

4.7.5. Book sub-menu

"Profiles" are configuration memories that can be recorded and later loaded/recalled at will (see their description in 4.6 above, "Management of the configuration profiles"). For each profile category, a sub-menu provides the list of profiles already recorded in the ScoopTeam memory, each with its name.

Item / Label	Type / Choices / Range / Description
Call Profiles Manage the call profiles	*Tap to enter the sub-menu* See this sub-menu further.
Presets Manage the presets	*Tap to enter the sub-menu* See this sub-menu further.
SIP Profiles Manage the SIP profiles	*Tap to enter the sub-menu* See this sub-menu further.
Snapshots Manage the snapshots	*Tap to enter the sub-menu* See this sub-menu further.

Book > Call Profiles

The first part of this sub-menu allows creating a new profile. *It is only shown for the "Expert" access level.*

Item / Label	Type / Choices / Range / Description
New Call Profile Create a new call profile	*Tap to enter the sub-menu (expanded below)* After editing the data below, for confirming the creation of the new profile you must tap the "Check" icon that appears on top right. If you don't do so, all edits are lost when leaving the sub-menu.
Call Profile Name noname	< enter text > Enter a name for the new profile.
Network Ethernet	Ethernet | Ethernet (aux) | ISDN | Mobile | Mobile (Ext) | Wi-Fi Selection of the network interface that will be used with this profile.
Algorithm G.711 A-Law	None | G.711 A-Law | … | Opus Select the coding algorithm. The list depends on the type of network above (IP or ISDN). *With "None", the current coding configuration is unchanged when loading the profile. In this case, the profile is similar to a classical phone book entry.*
Channel Mode Mono	Mono | Dual Mono | Stereo Dual Mono and Stereo are not available for all algorithms
Sampling Rate 8 kHz	8 kHz | 16 kHz | 24 kHz | 32 kHz | 48 kHz In most cases this parameter is mainly informative, and imposed by the coding algorithm.
Bit Rate 64000 bit/s	12 kbit/s | … | 256 kbit/s The allowed setting range depends on the coding algorithm and the network type. Many algorithms impose a fixed value at 64 kbit/s.
Remote Number 452145	< enter alphanumeric value > ISDN or telephone call number(s), or IP URI, IP address, depending on the transmission mode. *If left blank, no number is pre-loaded when loading the profile.The profile in this case is a convenient way to set up quickly the ScoopTeam with a given coding configuration.*
Second Remote Number 452146*23	< enter numeric value > This setting only exists for a profile on ISDN. For a profile using two B channels, the number to call for the second B channel can be entered here.
From here the list of existing and allowed profiles is listed in alphabetical order (see following table).	

Table 23 – Book > Call Profiles sub-menu description (1 of 2)

Within the list of profiles, some cannot be loaded depending on the current configuration.

- If the current network is an IP network in double codec mode, ISDN and/or dual mono profiles cannot be loaded.
- If the current network is ISDN in double codec mode, AoIP and/or dual mono cannot be loaded.

Item / Label	Type / Choices / Range / Description
AA [AoIP] 5415@15.12.45.4 Ethernet - G.722	*Tap to enter the sub-menu (Expert) or load the profile (Operator).* Name of the first profile and summary of its contents. *Expert level:* Details are shown if you tap the item. *Operator level:* summary is grey if loading the profile is not allowed.
Bks [ISDN] 0123456 ISDN - HE-AAC 48kHz Mono 64kbit/s	*Tap to enter the sub-menu (Expert) or load the profile (Operator).* Name of the profile and summary of its contents.
	….
Studio 56 [AoIP] 40998547 Ethernet - Opus 48kHz Mono 80kbit/s	*Expert level: Tap to enter the sub-menu (expanded below).* *Operator level: Tap to load profile (if allowed).* Name of the profile and summary of its contents. *Expert level:* After editing the data below, you must tap the "Check" icon that appears on top right. If you don't do so, all edits are lost when leaving the sub-menu. *Operator level:* summary is grey if loading the profile is not allowed.
Call Profile Name Studio 56	< enter text > Enter a new name for the profile. If you do so, when you tap the "check" icon you create a new profile with the edited name, while the original profile is kept the same.
Network Ethernet	Ethernet | Ethernet (aux) | ISDN | Mobile | Mobile (Ext) | Wi-Fi Selection of the network interface that will be used with this profile.
Algorithm Opus	None | G.711 A-Law | … | Opus Select the coding algorithm.
Channel Mode Mono	Mono | Dual Mono | Stereo Dual Mono and Stereo are not available for all algorithms.
Sampling Rate 48 kHz	8 kHz | 16 kHz | 24 kHz | 32 kHz | 48 kHz
Bit Rate 80 kbit/s	12 kbit/s | … | 256 kbit/s
Remote Number 40998547	< enter alphanumeric value > ISDN or telephone call number(s), or IP URI, or IP address.
Load Load the profile	Tap to load the profile into the codec (grey if not allowed). *Operator level: this is the only action accessible in this sub-menu.* *Whenever needed, a pop-up box asks you to select Codec 1 (Program) or Codec 2 (Coordination). In any case, a confirmation is asked before loading the profile.*
Delete Delete the profile	Tap to delete the profile. *A pop-up box asks you for confirmation.*
	….
ZZ [AoIP] 1258@15.12.45.4 Mobile - AAC-LC 48kHz Mono 40kbit/s	*Tap to enter the sub-menu (Expert) or load the profile (Operator).* Name of the profile and summary of its contents.

Table 24 – Book > Call Profiles sub-menu description (2 of 2)

Book > Presets

Item / Label	Type / Choices / Range / Description
New Preset Create a new preset	Tap to save the current network settings in a preset. *Expert level only.* The virtual keyboard pops up so you can enter a name for the preset.
	From here the list of existing presets is listed in alphabetical order:
AABB Load, update or delete the preset	*Tap to enter the sub-menu (Expert) or load the preset (Operator).* First Preset.
CC Load, update or delete the preset	*Tap to enter the sub-menu (Expert) or load the preset (Operator).*
	….
Config 1 Load, update or delete the preset	*Expert level: Tap to enter the sub-menu (expanded below).* *Operator level: Tap to load the preset.*
Preset Name Config 1	(read only) Name of the preset (reminder, not editable).
Load Load the preset	Tap to load the preset into the ScoopTeam. *Only this action is available for the Operator level.* *A pop-up box asks for a confirmation.*
Update Update the preset	Tap to overwrite the preset with the current network settings. *A pop-up box asks you for confirmation.*
Delete Delete the preset	Tap to delete the preset. *A pop-up box asks you for confirmation.*
	….
Z config Load, update or delete the preset	*Tap to enter the sub-menu (Expert) or load the preset (Operator).* Last Preset.

Table 25 – Book > Presets sub-menu description

Book > SIP Profiles

Item / Label	Type / Choices / Range / Description
New SIP Profile Create a new SIP Profile	Tap to save the current registration data in a profile. *Expert level only.* The virtual keyboard pops up so you can enter a name for the profile; afterwards you can select which account you want to save (primary or secondary).
	From here the list of existing presets is listed in alphabetical order:
AABC Load or delete the SIP profile	*Tap to enter the sub-menu (Expert) or load the profile (Operator).* First profile.
CC Load or delete the SIP profile	*Tap to enter the sub-menu (Expert) or load the profile (Operator).*
	….
Profile x Load or delete the SIP profile	*Expert level: Tap to enter the sub-menu (expanded below).* *Operator level: Tap to load the profile.*
	SIP Profile Name Profile x	(read only) Name of the profile (reminder, not editable).
	Load Load the SIP profile	Tap to load the profile into the ScoopTeam. *Only this action is available for the Operator level.* *A pop-up box asks for a confirmation; afterwards you can select which account you want to load (primary or secondary).*
	Delete Delete the SIP profile	Tap to delete the SIP profile. *A pop-up box asks you for confirmation.*
	….
Z config Load or delete the SIP profile	*Tap to enter the sub-menu (Expert) or load the profile (Operator).* Last profile.

Table 26 – Book > SIP Profiles sub-menu description

Item / Label	Type / Choices / Range / Description
New Snapshot Create a new snapshot	Tap to save the current audio settings in a snapshot. *Expert level only.* The virtual keyboard pops up so you can enter a name for the snapshot.
	From here the list of existing snapshots is listed in alphabetical order:
AABB Load, update or delete the snapshot	*Tap to enter the sub-menu (Expert) or load the snapshot (Operator).* First snapshot.
CC Load, update or delete the snapshot	*Tap to enter the sub-menu (Expert) or load the snapshot (Operator).*
	....
Config 1 Load, update or delete the snapshot	*Expert level: Tap to enter the sub-menu (expanded below).* *Operator level: Tap to load the snapshot.*
Snapshot Name Config 1	(read only) Name of the snapshot (reminder, not editable).
Load Load the snapshot	Tap to load the snapshot into the ScoopTeam. *Only this action is available for the Operator level.* *A pop-up box asks for a confirmation.*
Update Update the snapshot	Tap to overwrite the snapshot with the current network settings. *A pop-up box asks you for confirmation.*
Delete Delete the snapshot	Tap to delete the snapshot. *A pop-up box asks you for confirmation.*
	....
Z config Load, update or delete the snapshot	*Tap to enter the sub-menu (Expert) or load the snapshot (Operator).* Last snapshot.

Table 27 – Book > Snapshots sub-menu description

4.7.6. Tools sub-menu

Item / Label	Type / Choices / Range / Description
Shutdown Bring the system down	Tap the icon to switch off the unit. *Confirmation required. To cancel, touch anywhere else than the OK button.*
Display brightness 40%	Tap the virtual knob and use the rotary encoder to adjust the display backlight intensity. *Tap and deselect the knob, or touch anywhere else when you are done with the adjustment.*
Screen saver Dim the screen saver after 5 minutes of inactivity	Tap the virtual knob and use the rotary encoder to adjust the time before the display switches to low backlight intensity. *Tap and deselect the knob when you are done with the adjustment.*
Play Sound ID **Disabled**	Disabled | Active Activate or disable the injection of the Sound ID into the Program bus. The sound is an audio sample played back in a loop. The audio sample can be imported into the unit using the html pages (see chapter 4.9).
Access Level Expert	Operator | Expert | Administrator You can change the current access level here. When the selected level is an upgrade (e.g. "Operator" to "Expert"), you must enter the appropriate password in the dialog box that pops up.
Softkeys Configure the programmable keys	*Tap to enter the sub-menu (Expert level required)* See "Softkeys" table further
Labels Configure the info, contact and input labels	*Tap to enter the sub-menu (Expert level required)* See "Labels" table further
Miscellaneous Auto login, GPIO, Auto Redialling and NTP server	*Tap to enter the sub-menu (Expert level required)* See "Miscellaneous" table further
Troubleshooting Reset settings, configure event log	*Tap to enter the sub-menu (Expert level required)* See "Troubleshooting" table further
Remote Access Configure the remote access service	*Tap to enter the sub-menu* *If a "Remote Access" option is not installed, the sub-menu is labelled "Remote Assistance". See "Remote Access / Remote Assistance" table further*
Passwords Configure the passwords	*Tap to enter the sub-menu (Administrator level required)* See "Passwords" table further

Table 28 – Tools sub-menu description

Tools > Softkeys

"Expert" access level required.

Here the current assignment of the soft keys is listed. Tap the row corresponding to a given key to access the assignment for this key.

Item / Label	Type / Choices / Range / Description
Programmable Key 1 Talf to Speaker [1 Talk 2]	*Tap to enter the sub-menu and configure the key.* Reminder of the current assignment of the key: Category [Function]
Programmable Key 2 Talf to Speaker [1 Talk 3]	*Tap to enter the sub-menu and configure the key.* Reminder of the current assignment of the key: Category [Function]
Programmable Key 3 Aux to Bus [USB to PGM]	*Tap to enter the sub-menu (expanded below) and configure the key.* Reminder of the current assignment of the key: Category [Function]
Category Aux to Bus	None | Cough key | Talk to Speaker | Talk to Output | Aux to Bus | Contacts | Info Select the category of function to be assigned. The functions are described in 2.5.6, "Assignable function keys".
Function USB to PGM	[List depending on the selected category] Select the function to be assigned. The functions are described in 2.5.6, "Assignable function keys".
Programmable Key 4 Coughkey [Cough 3]	*Tap to enter the sub-menu and configure the key.* Reminder of the current assignment of the key: Category [Function]
Programmable Key 5 None	*Tap to enter the sub-menu and configure the key.* Reminder of the current assignment of the key: Category [Function]
Programmable Key 6 Talf to Speaker [2 Talk 1]	*Tap to enter the sub-menu and configure the key.* Reminder of the current assignment of the key: Category [Function]
Programmable Key 7 Talf to Speaker [2 Talk 3]	*Tap to enter the sub-menu and configure the key.* Reminder of the current assignment of the key: Category [Function]

Table 29 – Tools > Softkeys sub-menu description

Tools > Labels

"Expert" access level required.

On the home page, four boxes display the state of the sent/received "Contacts" and "Info" auxiliary data. Using this sub-menu, it is possible to set labels that are displayed inside these boxes, instead of their default names ("Contact 1", "Info 1", etc.). *A blank entry restores the default name.*

This sub-menu also allows to set the labels for all the inputs: inputs 1 to 4 (commentator positions), auxiliary inputs and AES67/Ravenna input channels 1 to 5.

Item / Label	Type / Choices / Range / Description
Info and Contact Labels	*Section header*
Info 1 Label ON AIR	< enter text > The label entered here is displayed on the home page inside the corresponding box, instead of the default "Info 1"
Info 2 Label	< enter text > The label entered here is displayed on the home page inside the corresponding box, instead of the default "Info 2"
Contact 1 Label Alert	< enter text > The label entered here is displayed on the home page inside the corresponding box, instead of the default "Contact 1"
Contact 2 Label	< enter text > The label entered here is displayed on the home page inside the corresponding box, instead of the default "Contact 2"
Audio Input Labels	*Section header* The labels entered here are displayed in buttons or boxes in addition to or instead of the default labels.
INPUT 1 John	< enter text >
INPUT 2 Bob	< enter text >
INPUT 3 Guest	< enter text >
INPUT 4	< enter text >
USB Laptop	< enter text >
AUX 1 Guide	< enter text >
AUX 2	< enter text >
AES 1	< enter text >
AES 2	< enter text >
RAV. 1	< enter text >
RAV. 2	< enter text >
RAV. 3 Cue	< enter text >
RAV. 4 Intercom	< enter text >
RAV. 5 Feedback	< enter text >

Table 30 – Tools > Labels sub-menu description

"Expert" access level required.

Item / Label	Type / Choices / Range / Description
Auto Operator Login Disabled	Active | Disabled If active, the unit directly logs in with the "Operator" level when it starts (the password for this level must be blank). *Reserved for "Administrator" level*
GPIO	*Section header* This section deals with the assignment of the GPIO.
Input 1 Info 1	Disabled | Contact 1 | Redial | Release call Select the assignment for Input 1. The available choices are described in 2.5.5, GPIO.
Input 2 Redial	Disabled | Contact 2 | Redial | Release | Redial 2 | Release 2 Select the assignment for Input 2. The available choices are described in 2.5.5, GPIO.
Relay 1 Line status	Disabled | Info 1 | Line status | Sync status Select the assignment for Output 1. The available choices are described in 2.5.5, GPIO.
Relay 2 Info 2	Disabled | Info 2 | Line status | Sync status | Line status 2 | Sync status 2 Select the assignment for Output 2. The available choices are described in 2.5.5, GPIO.
Auto Redialing	*Section header* This section allows to configure the auto-redialing function. The function is described in 3.4.2, "Auto redial feature".
Auto Redial Codec 1 Active	Active | Disabled Switch on/off the function for Codec 1 with this setting. *This setting is also accessible directly on the "Connections" page.*
Auto Redial Codec 2 Disabled	Active | Disabled Switch on/off the function for Codec 2 with this setting. *This setting is also accessible directly on the "Connections" page.*
Number of Attempts 3	1 | … | 20 | Infinite After an undesired release, the unit tries to recall the remote unit. Select the number of times the unit will try to re-establish the link. If this number of attempts is reached unsuccessfully, the unit gives up ands stays idle. "Infinite" means the unit will never stop retrying. *Tap the virtual knob and use the rotary encoder for the adjustment.*
Time Before Redialling 5 seconds	1 second | … | 30 seconds Select the time to wait before trying to recall after a loss of link or a failed attempt to re-establish the link. Be aware that a too short time here may lead to repeated (or endless) failures instead of restoring the link faster. The best policy is to leave enough time to that the units are both ready to reset the link. *Tap the virtual knob and use the rotary encoder for the adjustment.*
NTP	*Section header* The ScoopTeam can use a time server to set its internal clock and time-stamp accurately its event logs. The NTP server is selected in this section.
NTP Server pool.ntp.org	Custom | [list of public NTP servers] Select among a list of NTP servers, or select "Custom" if you prefer to specify another server.
Custom Server	< enter IP address or domain name > Domain name or IP address of the NTP server. *Only relevant if "Custom" is selected in the list above.*

Table 31 – Tools > Miscellaneous sub-menu description

Tools > Troubleshooting

"Expert" access level required.

This sub-section includes a few maintenance or debugging functions.

Item / Label	Type / Choices / Range / Description
Log Level Debug	Coarse | Normal | Debug You can select the detail level for the history of events that is logged in the unit. *Refer to chapter 4.11.9 (Maintenance) for more details on this parameter.*
Syslog Server	< enter IP address in dot-decimal format > The ScoopTeam can also send event messages to a SYSLOG server if available: enter here the address of this server . *Refer to chapter 4.11.9 (Maintenance) for more details on this parameter.*
Reset Select the category of settings to reset	Call History | Call Profiles | Presets | SIP Profiles | Snapshots | Restore Default Settings | Full Factory Reset | Reload the Factory SIP Accounts You can reset the unit completely or clear only a category of data; the available options are: • Call History: reset the list of previous calls displayed when you tap the "Call History" button on the "Connections" page. • Call Profiles, Presets, SIP Profiles, Snapshots: delete all the corresponding configuration profiles. • "Restore Default Settings": all parameters are brought back to their default setting, but all memories are kept (call profiles, presets, SIP profiles and snapshots). • "Full Factory reset": all parameters are brought back to their default setting, and all memories (call profiles, presets, snapshots) are deleted. Afterwards, the unit automatically reboots to ensure the parameters are properly reset. • "Reload the Factory SIP Accounts": reload the SIP account data integrated in the unit. The selected action is performed after you tap the "OK" button.

Table 32 – Tools > Troubleshooting sub-menu description

Tools > Remote Access (or Remote Assistance)

This sub-section allows to configure the connection of the ScoopTeam to the AETA remote access server. When the unit is connected, AETA can access the unit remotely for providing assistance. In addition:

- If the unit includes the "remote access" option, one can access its html pages remotely, by logging onto the AETA remote access server (https://cloud.aeta.com/rac).

- If the unit includes the "remote access +" option, in addition to the above the ScoopTeam can be remote controlled via the Internet from the "My ScoopTeam RE" application.

Item / Label	Type / Choices / Range / Description
Remote Access / Remote Assistance Active	Active | Disabled Switch on/off the connection to the remote access server.
Interface for Control Auto	Auto | Ethernet | Ethernet (aux) | ISDN | Mobile | Mobile (Ext) | Wi-Fi Selection of the network interface that will be used for remote access. "Auto" selects the same interface as the one selected for AoIP transmission.
Remote Access Server cloud.aeta.com	(read only) Domain name or IP address of the remote access server.
Serial Number 01165	(read only) Serial number of the unit. This is part of the codec ID below. It is used with the password below for accessing the embedded html pages through the remote access server.
Codec Identifier scoopteam_1165	(read only) This ID is built from the product type and the unit's serial number.
Codec Password mypass87sfds	(read only except Administrator) Only the Administrator level allows to create/edit this password. *The password must be minimum 8 characters long (otherwise a random password is automatically generated).*

Table 33 – Tools > Remote Access / Assistance sub-menu description

You can check the connection state in the Status sub-menu (Remote Access Service).

Tools > Passwords

"Administrator" access level required.

This sub-section allows the administrator to define the password for each user level. *You may leave some passwords blank.*

Item / Label	Type / Choices / Range / Description
Administrator Password *****	< enter text > Enter here the desired password for the "Administrator" level. *Be careful not to lose this password!*
Expert Password ***	< enter text > Enter here the desired password for the "Expert" level.
Operator Password	< enter text > Enter here the desired password for the "Operator" level.

Table 34 – Tools > Passwords sub-menu description

4.7.7. Status sub-menu

This sub-menu is all "read only". On one hand it reminds some essential settings, and on the other hand it monitors the operation and state and the unit.

ⓘ *The "Refresh" icon appears on top right; tap it when needed to get the most up-to-date picture.*

Item / Label	Type / Choices / Range / Description
Current State	*Section header* This section reminds the current audio transmission configuration.
Mode Double Codec	(read only) Current setting for the "Codec Mode"
Codec 1 [Ethernet] Opus 48kHz Mono 64kbit/s	(read only) *Between brackets: current network for outgoing calls.* Current coding configuration for Codec 1.
Codec 2 [Ethernet] MPEG L2 48kHz Mono 64kbit/s	(read only) *Between brackets: current network for outgoing calls.* Current coding configuration for Codec 2, if the "Mode" above is "Double codec".
Audio Connections Status of the audio connections	*Tap to enter the sub-menu (expanded below)*
Connection 1	*Section header* Information on the current connection on Codec 1.
State Call established (synchronized)	(read only) Current state of the connection, here active.
Network Ethernet	(read only) Network interface used by this connection.
Remote Address 452145	(read only) Address of the remote device.
Quality	*Section header* This section is only visible during an active AoIP connection. It provides an assessment of the link quality that takes account of various elements: packet loss rate, jitter, jitter buffer variations…
Transmission 93%	(read only) 100% corresponds to the best link quality. Global assessment of the quality on the "transmit" direction, from ScoopTeam to remote unit. *This measurement may not be available, as it relies on the support of RTCP by the remote device.*
Reception 88%	(read only) 100% corresponds to the best link quality. Global assessment of the quality on the "receive" direction, from the remote unit to the ScoopTeam.
Detailed Statistics Show detailed statistics	*Tap to enter the sub-menu.* This leads to additional quality data. See "Detailed Statistics" table further.
Connection 2	*Section header* Information on the current connection on Codec 2.
State Idle	(read only) Current state of the connection, here no running connection.
Network Ethernet	(read only) Network interface used by this connection.
Latest AoIP Statistics Show detailed statistics of latest AoIP call	*Tap to enter the sub-menu.* This sub-menu is available when the "State" is "Idle". It has the same definition as the detailed statistics found above for a running connection, but related to the latest completed AoIP connection. See "Detailed Statistics" table further.

Table 35 – Status sub-menu description (1 of 2)

AoIP Controller Default Protocol is SIP		Tap to enter the sub-menu (expanded below)
	Default Protocol SIP	(read only) This is a reminder of the current "Default Protocol for Outgoing Calls" set in the "Network>AoIP" sub-menu.
	SIP User 1 902994	(read only) Reminder of the identifier for the first SIP account.
	SIP User 2 90305794	(read only) Reminder of the identifier for the second SIP account.
	Public IP 82.232.194.202	(read only) IP address on the public side of the Internet access router, as detected thanks to STUN (and only visible if STUN is active).
	NAT Type Full Cone	(read only) Type of network address translation performed by the access router, as detected with STUN. This is only visible if STUN is active.
Ethernet 10.0.20.74 [DHCP, 100BaseT-FD]		(read only) Current IP address of the primary Ethernet interface. Inside brackets: DHCP status, current link mode.
Ethernet (aux) 169.254.2.1 [DHCP, 10BaseT-HD]		(read only) Current IP address of the auxiliary Ethernet interface. *On this example, the interface is not connected, and a link-local address is set automatically.*
Mobile Current network is 4G/LTE [F SFR]		*Tap to enter the sub-menu (expanded below)* Information about the mobile network the unit is registered on.
	Radio Access Technology 4G/LTE	(read only) Radio access technology of the network.
	Operator Orange	(read only) Name of the mobile network operator.
	Signal Strength -75 dBm	(read only) Strength of the radio signal. This is also shown more roughly by the 5-bar radio strength indicator on the status panel.
	Data Service Connection Established	(read only) Status of the mobile data connection.
Mobile (ext 1) Current network is 4G/LTE [F SFR]		*Tap to enter the sub-menu (same as above)* Information about the network accessed via a USB device.
Wi-Fi (ext 1) WiFi_Net5 [192.168.20.68, -62 dBm]		(read only) Summary information about the Wi-Fi connection, visible if a USB device is plugged.
Remote Access Service Connection Established		(read only) State of the connection to the remote access server.
About		*Section header* Information about the embedded software
	Serial number 01165	(read only) Serial number of the unit.
	Firmware Version 1.03.02	(read only) Global firmware version. *Along with the unit serial number of the unit, this is essential information to provide whenever you contact AETA for support or maintenance issues.*
	AoIP Version 2.00.0035	(read only) Version of the AoIP firmware.
	Software Build 2017-05-30	(read only) Firmware date of release.

Table 36 – Status sub-menu description (2 of 2)

A specific sub-menu provides detailed measurements on a running AoIP link, or on the previous completed AoIP link. The table below describes the data displayed in this sub-menu.

"Tx" data are measurements on the transmission from the ScoopTeam to the remote unit, "Rx" data are measurements on the stream received from the remote unit.

ⓘ *Measurements for the "Tx" direction rely on the support of the RTCP protocol by the remote device. Because of this, these measurements may not be available in some situations.*

Item / Label	Type / Choices / Range / Description
Tx Duration 00:06:04	(read only) Duration (or time elapsed since the beginning) of the connection, in hours:minutes:seconds
Tx Packets Total: 15177, Lost: 64 (0.42%)	(read only) Statistics on packets lost during the connection or since its beginning.
Tx Jitter Average: 6 ms, Maximum: 17 ms	(read only) Statistics on the jitter on the link.
Rx Duration 00:06:05	(read only) Duration of the connection, in hours:minutes:seconds
Rx Packets Total: 15130, Lost: 78 (0.52%) Discarded: 0, Duplicated: 0, Reordered: 0	(read only) Statistics on packets lost during the connection, with additional data.
Rx Jitter Average: 4 ms, Maximum: 22 ms	(read only) Statistics on the jitter on the link.

Table 37 – "Detailed Statistics" sub-menu description

4.8. Access control

The unit provides selective access control. It features three access levels:

1. **Administrator**: this levels grants full access to all functions.

2. **Expert**: this levels grants access to most of the functions, except a few very critical ones which do not directly relate to routine operation (e.g. firmware update).

3. **Operator**: this level is for the normal operation; advanced or critical settings or actions are excluded, both for safety and for simplicity of use.

The features described in this document are accessible from the "Operator" level unless it is explicitly specified otherwise.

ⓘ *As a general rule, features not accessible at a given time are hidden from the pages or menus, or colored grey to warn that the corresponding action is not allowed from this level.*

Each access level has got its specific password. After a full factory reset, all three passwords are reset to <blank>. Then the passwords can be set or modified, only from the Administrator level. This is done from the **Tools > Passwords** menu.

When the unit starts up, the user is prompted to select the access level ("Operator" is selected by default). If the corresponding password is not blank, the user is prompted for the password before accessing the user interface on screen.

ⓘ *There is an "Auto Operator Login" option that can be activated (see 4.7.6 Tools sub-menu, Miscellaneous section), which allows to enter directly the "Operator" level at start-up, provided that the Operator password is blank.*

Afterwards, using the **Tools > Access Level** menu you can switch to another access level. You are prompted to enter the corresponding password (if not blank).

ⓘ *Lost the Administrator password? The only way to restore full access is to perform a backup reset, which will clear all the settings, including the passwords. The unit will be accessible again. The procedure is described in 3.5.2, "Backup reset".*

4.9. Operating mode – Embedded html interface

The embedded html server in ScoopTeam provides a means to control and monitor the unit. It needs the ScoopTeam to be connected to an IP network and to be reachable from a computer, or another device with an html browser: tablet, smartphone…

This control mode is usable regardless of the OS of the control unit, and the embedded pages are compatible with all common browsers. No software installation is needed on the control position.

4.9.1. Accessing the html pages

There are two methods for accessing the html pages of the ScoopTeam: LAN access, or remote access.

LAN access

In the most common case, the two devices are connected on the same local network (LAN). But more generally this applies for controlling the ScoopTeam by remote, provided that the control device can reach it (TCP/IP port 80, HTTP protocol, or HTTPS protocol via TCP/IP port 443).

Either of the two Ethernet interfaces (normal or "auxiliary" interface) can be used. The Wi-Fi interface can be used as well if available.

Once the ScoopTeam is connected on the network, the first step is to get its IP address, from the menu: **Status > Ethernet**[1]. Then, on the control device, launch the html browser and enter the IP address of the ScoopTeam in the "address" or "URL" field. This gives access to the html server integrated in ScoopTeam.

ⓘ *Alternative: you can get an HTTPS secure connection by typing **https://<IP address>**. You will have to override or bypass the security warning regarding the site's certificate. The procedure depends on the browser.*

Remote access

ⓘ *The "remote access" option must be installed on the ScoopTeam for using this method.*

This method allows to control the unit from remote across the Internet. It can be used with any available IP interface, including a mobile network interface, as long as it provides Internet access.

The detailed procedure for such remote access can be found further in chapter 6.3.3, "How to use the remote access service".

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[1] Or other IP interface: Ethernet (aux), Wi-Fi…

4.9.2. Principles of operation

Browsing the pages

Once you access the html server integrated in ScoopTeam, the page that is displayed is similar to the following picture:
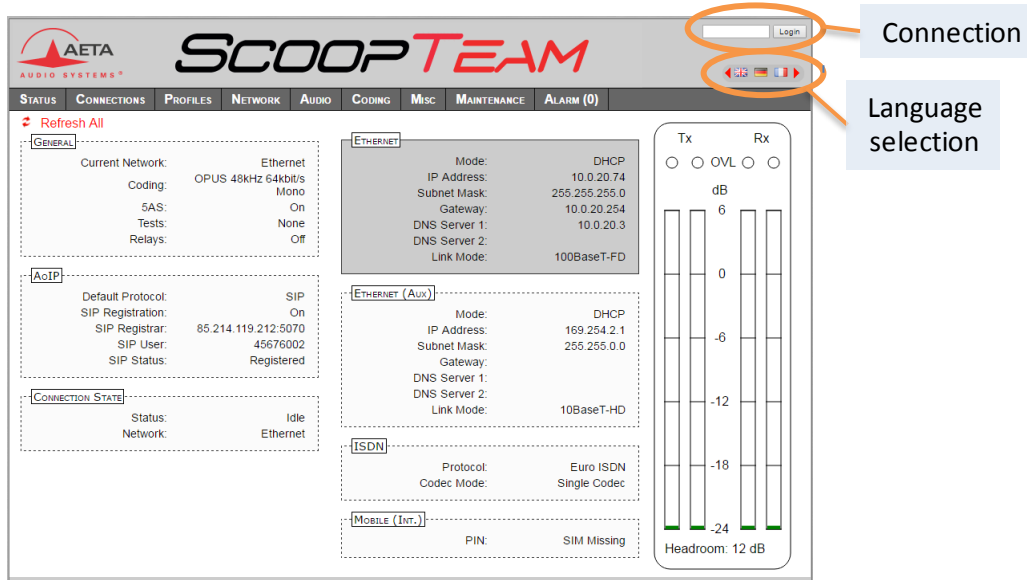
If needed, select another language by clicking the suitable flag (*this choice is only for the html interface*).

The home page displayed above is the "STATUS" page, which provides an overview of the unit status, but allows no action on it. This is the only "free access" page, with no limitation or access control.

To access the other pages, you must "log in", and get for the control device an <u>exclusive</u> access. Any connection request from another device[1] will remove this access.

To log in, enter the password and click the connection button. The initial [html access] password is blank: click directly on the button. To set a password and enable protection, go to the "Maintenance" page (cf. further).

ⓘ *This html access password is only valid for access to the html interface! You can't use it for any user level on the unit's control surface.*
Do not confuse this password with the "Codec password" possibly used for remote access.

To release control, click on "Logout" (also in the connection area). You are also logged out automatically after a long period of time with no action on the pages.

---

[1] This can also be a connection from another IP interface than the one through which you are connected.

Principles of operation with html pages

The picture below shows a typical page.
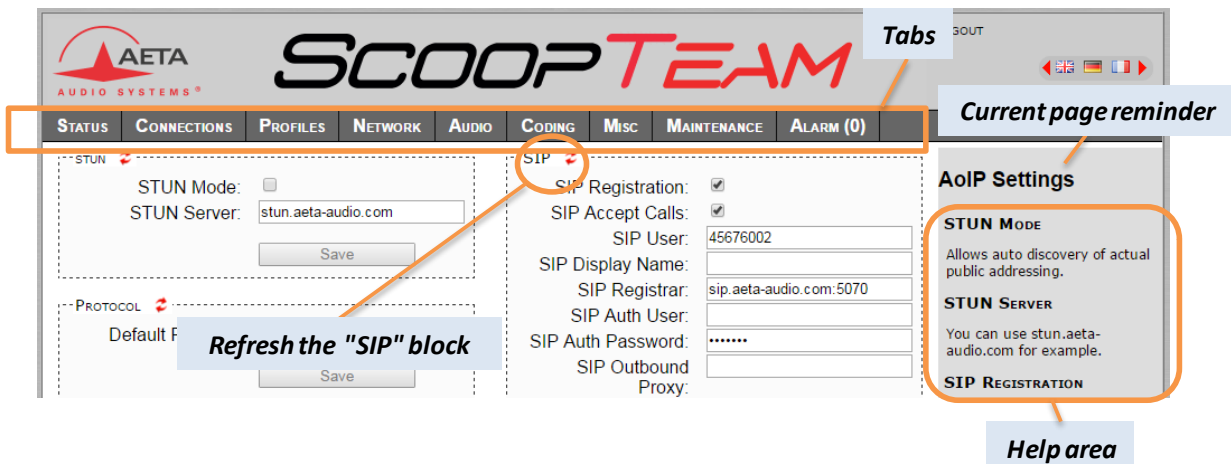
On top you find a bar of tabs corresponding to the various categories of functions and parameters for the ScoopTeam. Clicking a tab you access either a page, or a drop-down list for selecting a secondary page. *These tabs and pages are detailed in the following chapters.*

Under the bar can be found the information and adjustable parameters, with various selection or entry modes for these parameters, grouped in blocks (each surrounded with a frame). On the right side, a text area provides additional help and hints.

As a general rule, the displayed parameters are read when accessing the page, and are not refreshed automatically. To force a refresh, click the ⟲ icon: the data in the area or frame are read and refreshed.

ⓘ *Exception: some data on some pages is however periodically and automatically refreshed. This makes a modest bit rate, but you should exit the html pages if you want no traffic at all on the path between the ScoopTeam and the control device.*
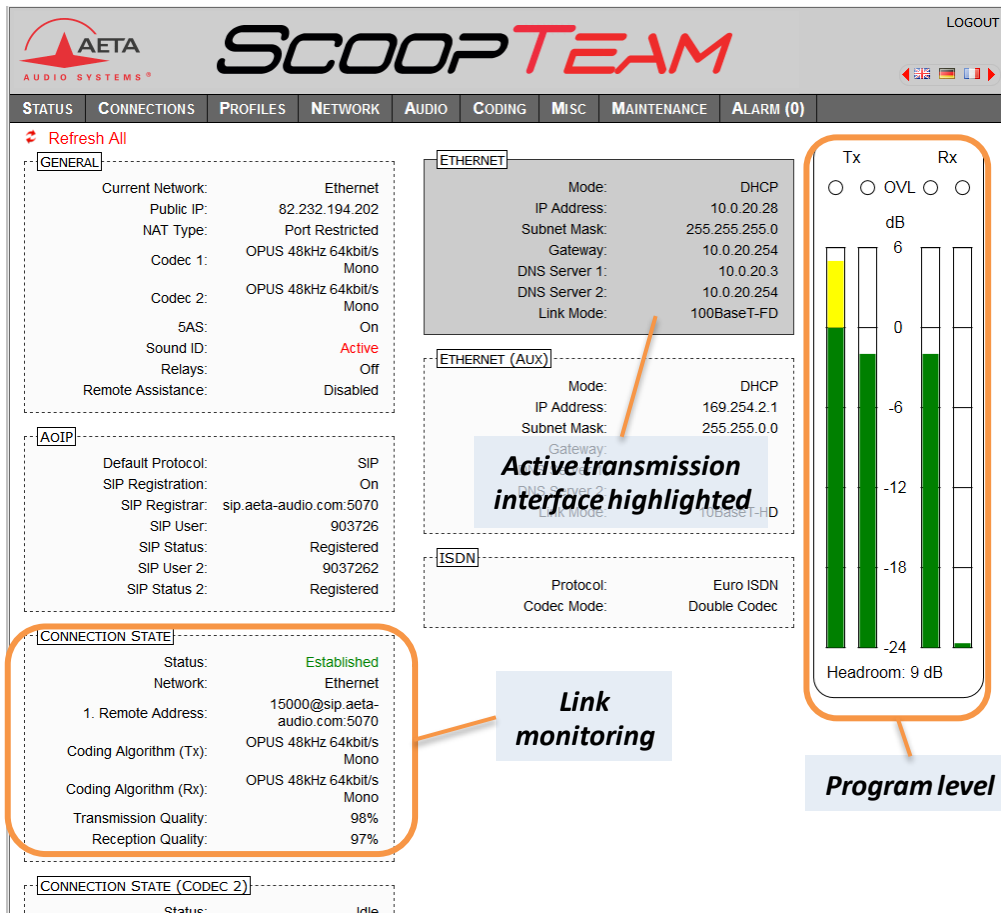
Two tabs have a specific behavior:

- "STATUS" is accessible without a login and some data are updated automatically.

- "ALARMS" is also updated automatically, and switches to red when an alarm triggers, showing the number of issues detected. You can then check for details by clicking the tab.

Access to the tabs (other than "STATUS") requires logging in beforehand. If you click a tab without being logged in, the login dialog box opens to allow you to enter the connection password. If it is blank, just click the "LOGIN" button.

4.9.3. "Status" tab

This tab leads to the "Status" page, which is also the home page. It provides an overview of the essential settings and parameters, sufficient for basic monitoring.

This page is accessible without a log in, and without a password.

The link status is monitored dynamically; e.g. you can see call reception and releasing. It is also possible to monitor calls that an operator is managing using the front panel.

The page displays the audio level of the two transmitted signals (encoder(s)) and the two received signals (decoder(s)), shown as bargraphs with a 0 dB reference. The "Headroom" value shown at the bottom is the difference between this reference and the clipping level. *The "Headroom" setting is adjustable using the "Audio" tab.*
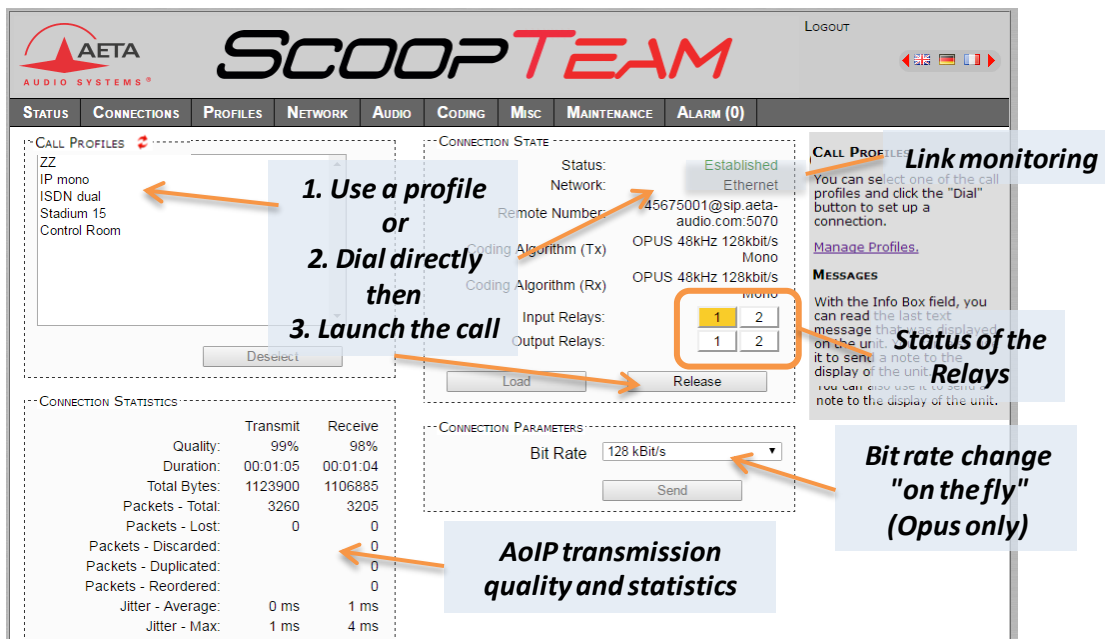
ⓘ *Note: the goal of these bargraphs is to provide an indication on the presence and level of the audio modulation. The measurement has a good precision for stationary signals (1 dB accuracy and resolution), but the update rate is rather slow. As a consequence these bargraphs are not appropriate for precisely monitoring a dynamic program, and they should not be considered an alternative to real program meters!*

One can find in the "Connection State" section global transmission and reception quality indicators (for IP links only). These are the same information that you can get from the front panel, as described above in 4.7.7, "Status sub-menu".

ⓘ *The "Transmission Quality" indicator is not always available; this depends on the capability of the remote codec.*

4.9.4. "CONNECTIONS" tab

This tab leads to the "CONNECTIONS" tab, allowing to remote control the links: monitoring, call set up and call releasing.

The "CONNECTION STATE" block allows monitoring and/or managing links.

- For direct dialing: the codec has to be set beforehand for the right interface (see "NETWORK" tab), and the suitable coding configuration ("CODING" tab). Enter the number (or numbers, or SIP URI…), then click "Dial". *You can also use one of the last called numbers: click the arrow on the right of the "Remote Number" field and pick a number in the dropdown list.*

- For using a call profile: select the profile in the list on the left, and click the "Dial" button. Alternatively, you can click "Load". The codec loads the profile settings but does not dial: this is useful for quickly setting the coding parameters.

During a connection, transmission statistics and quality indicators are available in the "CONNECTION STATISTICS" frame (for IP links only), and indicators show the status of the relays when the "Relay transmission" feature is active. *For auxiliary functions, see the "CODING" tab.*
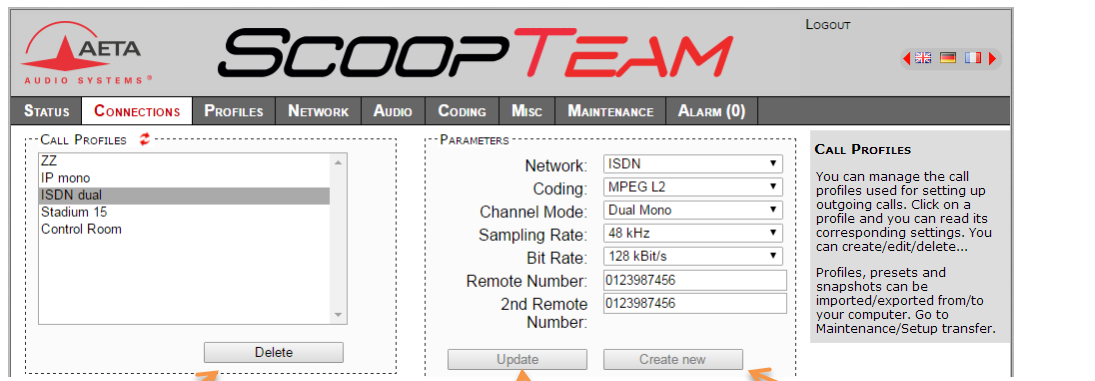
If the link makes use of the Opus coding algorithm, it is possible to change the transmission bit rate "on the fly", without interrupting the transmission and with no switching noise: select a bit rate and click "Send".

ⓘ *When ScoopTeam is in double codec mode, the "CONNECTION STATE" frame is duplicated, one frame for each codec, and also the "CONNECTION STATISTICS" frame. But selecting a call profile switches back to the normal display (for the single codec), because the profile data correspond to a single codec (call profiles cannot be used in double codec mode)*

4.9.5. "PROFILES" tab

This tab proposes three secondary pages, one for each profile type.

"CALL PROFILES" page
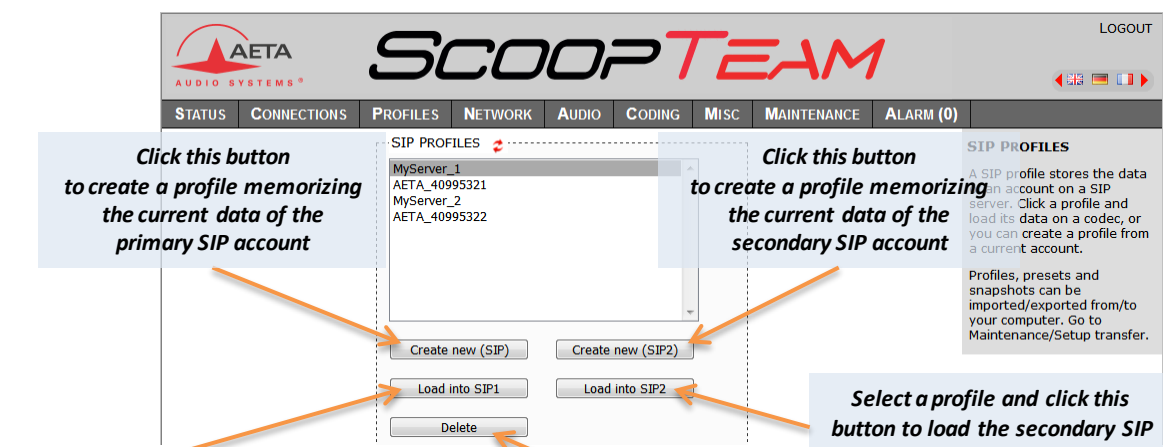
Select a profile
and click this button
to delete a profile

Select a profile, edit the parameters
and click this button
to update a profile

edit the parameters,
and click this button
to create a profile

Parameters that are not relevant in a given context are shaded and inactive.

"SIP PROFILES" page

Click this button
to create a profile memorizing
the current data of the
primary SIP account
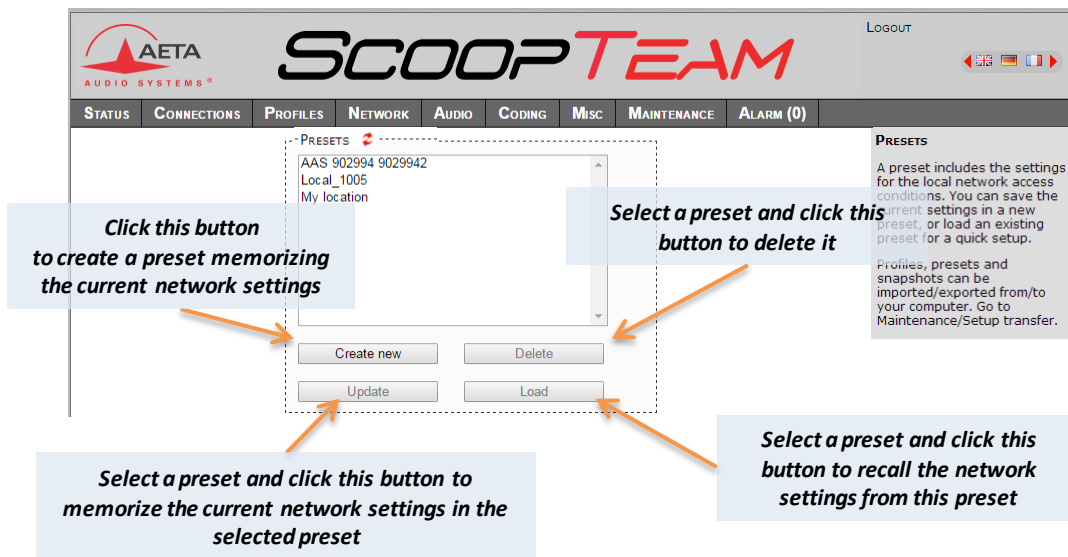
Click this button
to create a profile memorizing
the current data of the
secondary SIP account

Select a profile and click this
button to load the secondary SIP
account from this profile

Select a profile and click this
button to load the primary SIP
account from this profile

Select a profile and click
this button to delete it

SNAPSHOTS

Snapshots are in fact managed on the "AUDIO" page, and this "SNAPSHOTS" choice on the "PROFILES" tab just redirects to the "AUDIO" page. *See further the description of the audio page.*

4.9.6. "NETWORK" tab

This tab leads to several pages, which correspond, just like the sub-menu "**Network**" of the front panel interface (see 4.7.2 above), to the various network interfaces. For this reason, the number of pages actually proposed depends on the version and the options of the unit (with or without mobile access, etc.).

"CHANGE NETWORK" page: default interface selection

This page is used for selecting the default interface: this is the one that is implicitly used when dialing to set an outgoing call. Besides, codec algorithm settings (either via menus or the html pages) are applicable to this interface[1].
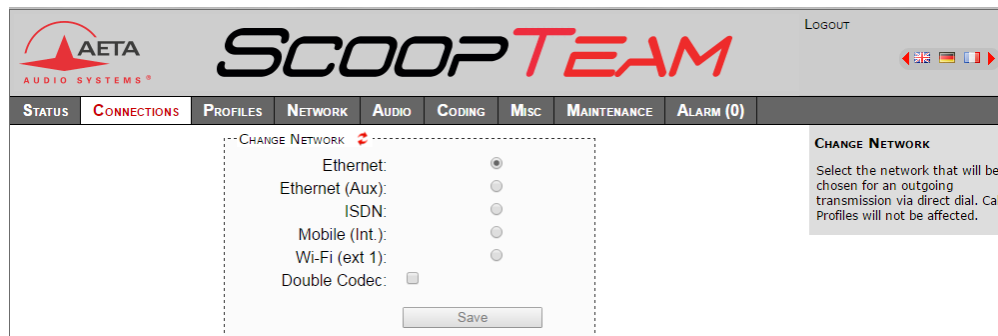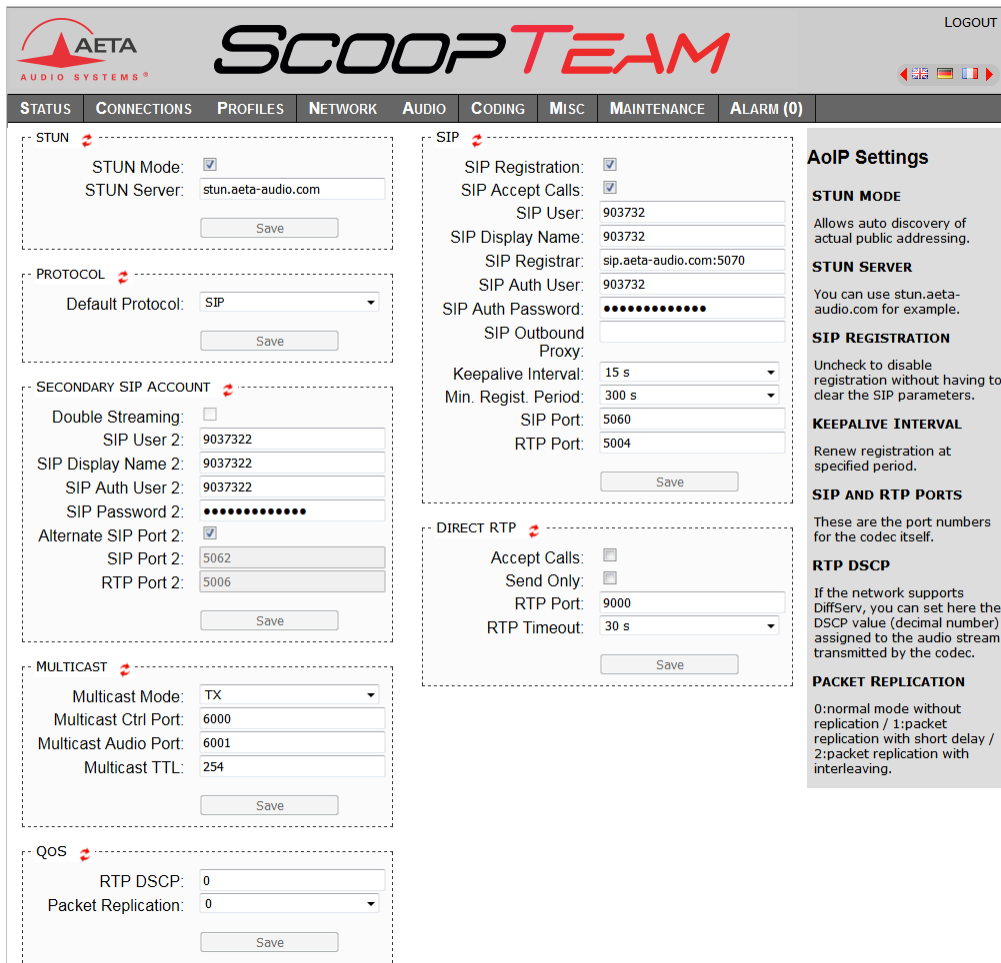
Select the desired interface and click "**Save**".

On this page you also select the desired codec mode: "single" or "double codec". Once selected, this choice applies to any network, even if you switch to another network interface. *Refer to 4.4.1, "Basic options" regarding this setting.*

---

[1] Remind that the algorithm settings are memorized separately for each type of network interface (ISDN or IP). When switching the default interface, the codec settings for this interface are recalled.

"AoIP Settings" page
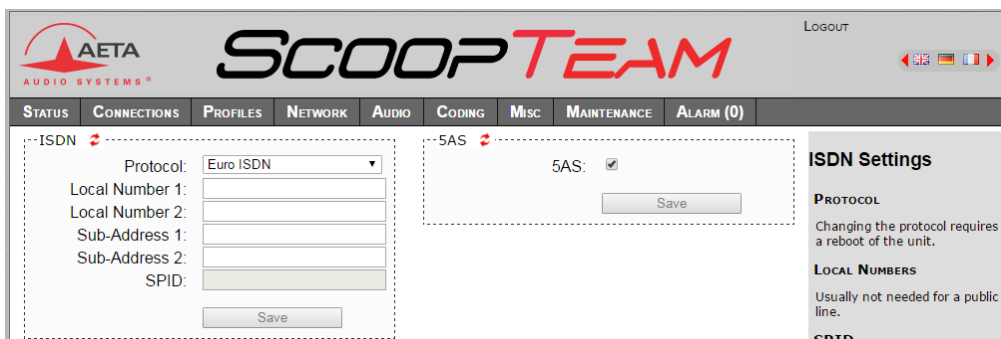
*More details on these parameters in 4.7.2, "*Network > AoIP settings* sub-menu".*

Remember to **save the changes**, clicking the "Save" button!

"ISDN Settings" page

This page is only visible for units equipped with ISDN interface(s). *See more details on the parameters in 4.7.2,* "Network > ISDN settings*".*
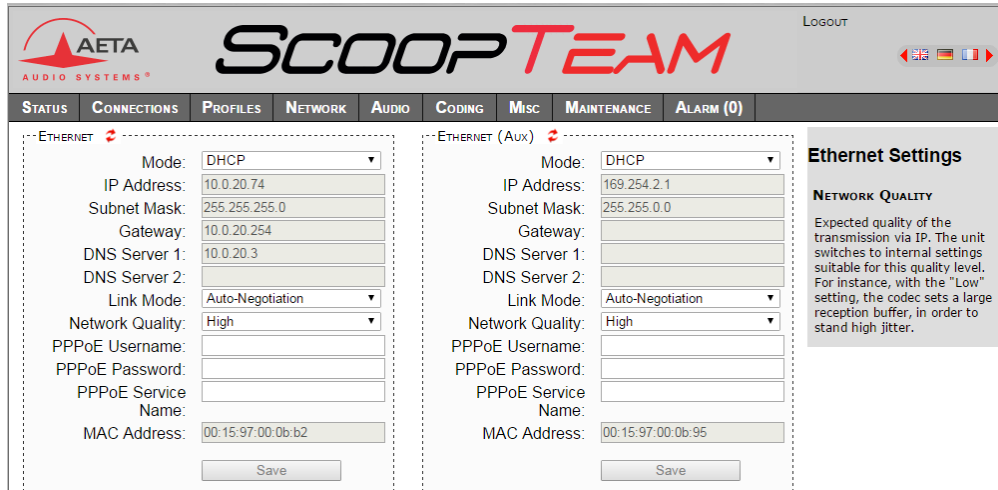
Remember to **save the changes**, clicking the "**Save**" button!

"Ethernet Settings" page

This page provides access to the settings for the Ethernet interfaces.

More details on these parameters in 4.7.2, "Network > LAN1 settings / LAN2 settings".

ⓘ *Note: this page also allows you to read the MAC address of the Ethernet interface.*

Remember to **save the changes**, clicking the "**Save**" button!
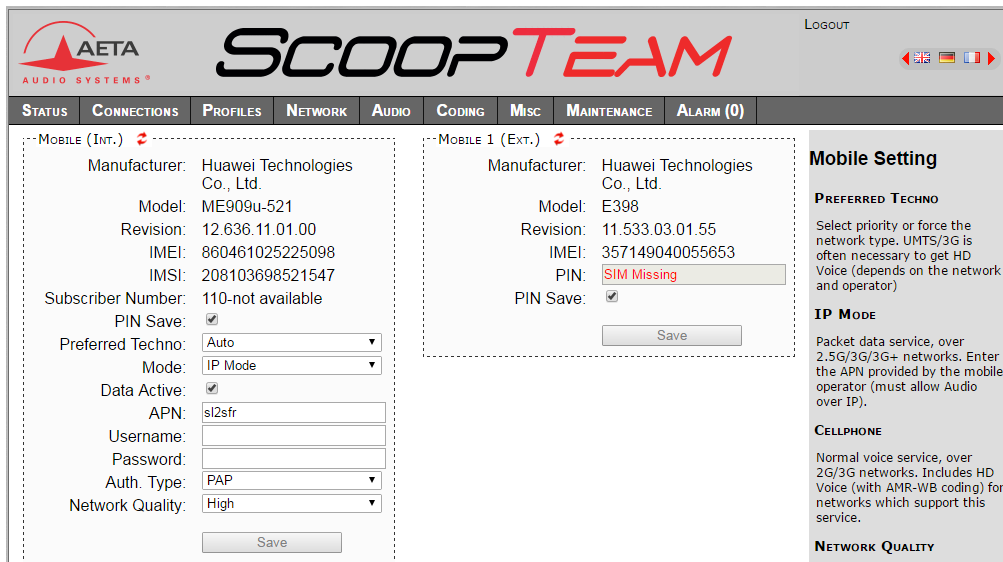
ⓘ *Settings regarding the Ethernet interface and IP addressing are especially critical, because this is the interface you are controlling ScoopTeam through. You can lose control because of a mistake in the settings; it might even be the normal expected consequence for a configuration change. So be very careful with the settings on this page!*

In case control is lost after an error, you must regain control and correct the settings using the front panel of the ScoopTeam.

"Mobile Settings" page

This page is only visible for units with the capacity to access mobile networks:

- Either the unit is equipped with the "HD-4G" option. In this case, the page includes a "Mobile (Int.)" section, as on the screen capture below (left side).

- And/or the "Mobile via USB" option is included, and a suitable USB device is inserted in the unit. In such case, the page includes a "Mobile 1 (Ext.)" section, like on the screen capture below (right side).

See more details on the parameters in *4.7.2,* "Network > Mobile settings (Internal)". You cannot access all settings unless the PIN code has been entered (otherwise the view is partial only, as can be seen above for the external USB device).

Remember to **save the changes**, clicking the "**Save**" button!

"Wɪ-Fɪ Sᴇᴛᴛɪɴɢs" page

This page is only visible for units equipped with the "Wi-Fi interface" option, and with the provided USB device inserted in one of the USB sockets.
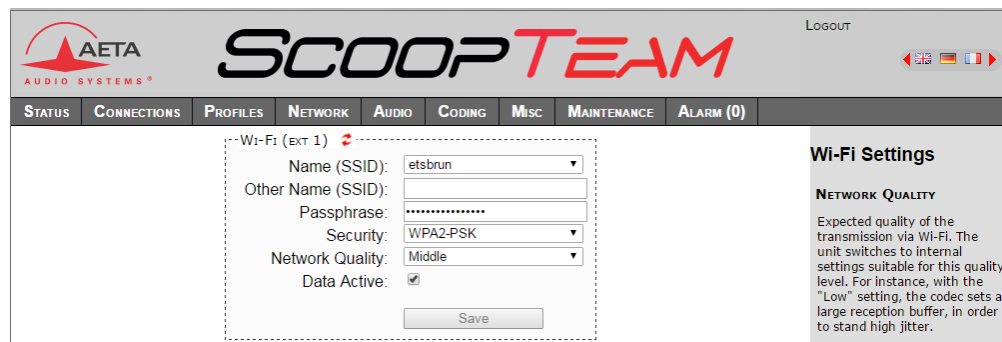
For connecting to a Wi-Fi network, the standard procedure is as follows:

- Click "Name (SSID)": a drop-down list shows the Wi-Fi networks detected by the device. Select the desired one.

- Enter for "Passphrase" the password for the network.

- Check the "Active" checkbox, and click the "**Save**" button.

- With this simple procedure, the "Security" setting is irrelevant, because the right setting is selected automatically by the ScoopTeam.

For a full "manual" setting, first select "-" from the drop-down list, and then enter/select the appropriate values for the parameters: "Other Name (SSID)", "Passphrase", "Security". Click the "Save" button to complete the configuration.
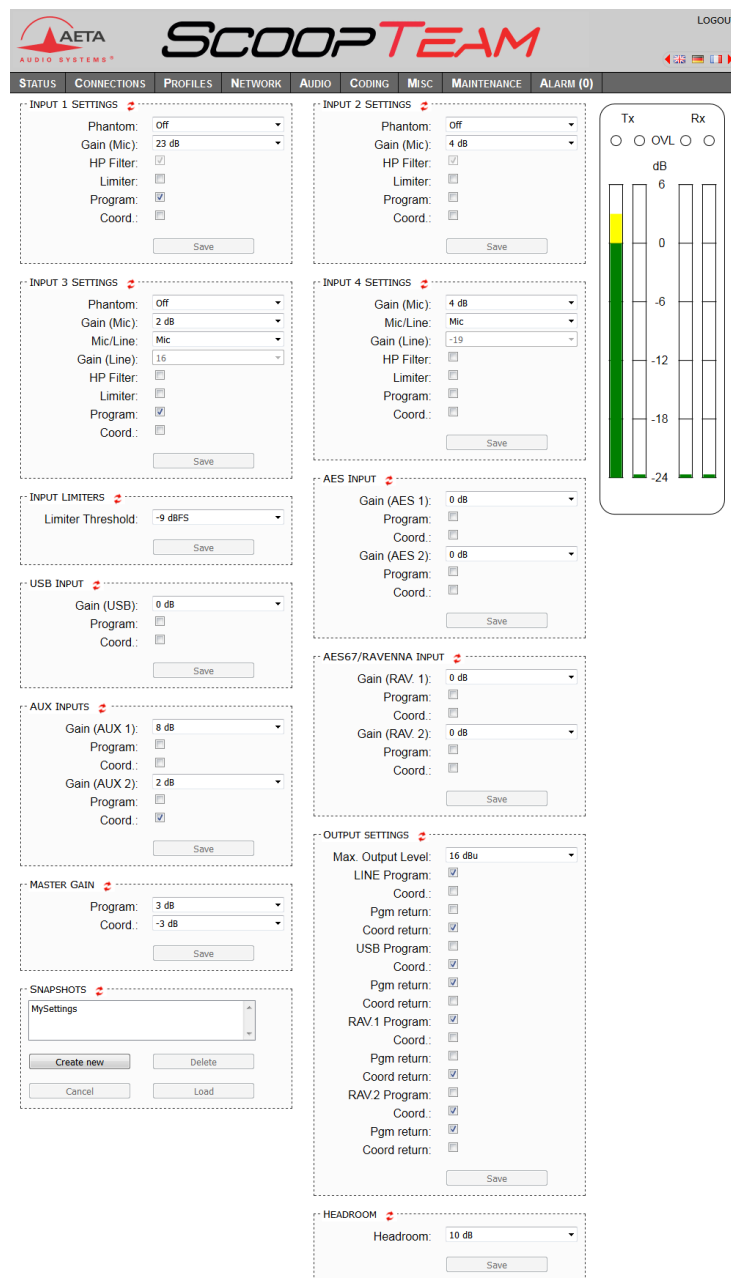
For more details on the parameters, refer to *4.7.2, "*Network > Wi-Fi settings (External)*"*.

ⓘ *The status of the Wi-Fi connection and the radio signal level are displayed on the "Sᴛᴀᴛᴜs" page.*

4.9.7. "AUDIO" tab

The audio mixing and processing (excluding the monitoring mixes) can be controlled via this page:

- Settings for the four microphone and mic/line inputs, as found on the "**Input Settings**" page of the control surface (see 4.3.4).

- Gains for these inputs in the mix and Master gains, as found on the "**Mixing**" page (see 4.3.3).

- Setting for the limiter threshold, applicable for these inputs, as found in the "**Audio**" sub-menu (see 4.7.4, "Audio sub-menu").

- Settings for the other inputs (USB, AES, AUX, AES67/RAVENNA), as found on the "**Aux Settings**" page of the control surface (see 4.3.5).

- Settings for the outputs, as found on the "**Output Settings**" page (see 4.3.6): routing for the outputs, and gain for the line outputs.

After changes on the audio parameters, make sure to **save the changes**, clicking the "**Save**" button.

This page includes the management of *snapshots*, memorizing the parameters controlled in this page.

To **create** a snapshot based on the current settings, click the "**Create new**" button, and enter a name for the snapshot.

The other buttons can be used after selecting an existing snapshot. Click a snapshot: its parameters are recalled in the fields of the page, but yet no change is done (the "**Save**" buttons warn of this). Then you can:

- Click the "**Cancel**" button to simply come back to the previous situation (*also possible using the*

  ⟲ *icons, but one frame at a time*).

- Click the "**Load**" button and directly apply the snapshot (*also possible using the "Save" buttons, but one frame at a time*).

- Click "**Delete**" to remove the selected snapshot from the list of snapshots.

Like the "STATUS" page, this page shows the audio level transmitted/received to/from the network, shown as bargraphs with a 0 dB reference. The "Headroom", which is the difference between this reference and the clipping level, can be adjusted on this page.

ⓘ *This display is <u>relative</u> to a "0 dBr" reference which can be set here. The "Headroom" parameter is the available headroom for a signal displayed as 0 dBr, i.e. the difference between the reference and the 0 dBFS maximum level (clipping level). For instance, a headroom of 10 dB means that the reference is set at -10 dBFS.*

4.9.8. "CODING" tab

The "CODING" tab groups the selection of the algorithm and its parameters, as well as the auxiliary functions. See details on all these parameters in 4.7.3, "Coding sub-menu".

The available coding algorithms depend on the current network interface. The coding parameters depend on the algorithm and the network interface. This is also true for the auxiliary functions.

The settings are saved separately for each type of network interface. This means that when you change the current network interface, the coding configuration that is related to it is recalled. Example:

- Current interface = ISDN, selected configuration MPEG L2 mono, 24 kHz and 64 kbit/s.

- Change the interface and select the Ethernet interface, then configure for Opus mono at 64 kbit/s.

- Switch again and come back to the ISDN interface: the MPEG L2 setting is recalled.

- Switch the network interface and select an Ethernet interface (or mobile IP): the Opus configuration is restored.

Remember to **save the changes**, clicking the "**Save**" button!
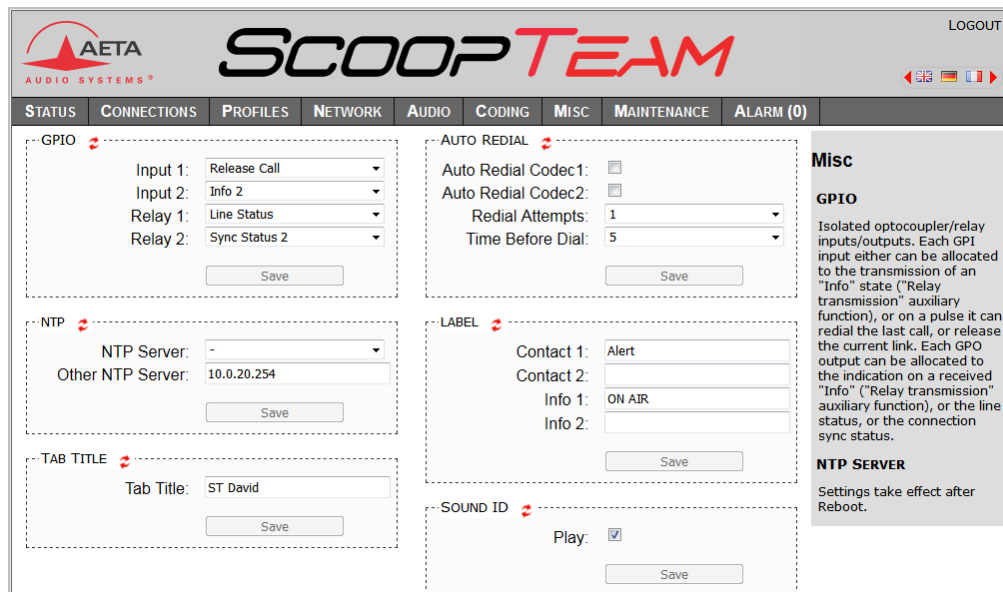
ⓘ *When the ScoopTeam is set as a double codec, an additional "CODEC 2" block is available. The range of allowed coding configurations is less wide than in single codec mode (see 2.2.2, "Double codec mode").*

4.9.9. "MISC" tab

This **MISC** tab includes several settings that can be found in the **Tools** sub-menu of the front panel user interface:

- Configuration of the GPIO (also found in the **Tools > Miscellaneous** menu).
- Parameters for the "auto redialing" function (also found in the **Tools > Miscellaneous** menu).
- Settings for using an NTP server (also found in the **Tools > Miscellaneous** menu).
- Labels for the "Contact" and "Info" boxes of the user interface (also found in the **Tools > Labels** menu).
- Activation of the Sound ID.

Please refer to 4.7.6, "Tools sub-menu", where you can find additional details on these settings.

Remember to **save changes done**, clicking the "Save" button!

In addition to the above, the page includes a setting that is specific to the html interface: you can define in the "Tab Title" block a label that is applied to the tab in the browser. This can help identify the unit if tabs are opened for more than one ScoopTeam.

ⓘ *Note about the **NTP server**: this kind of server provides a time reference (from the Internet or a server on the local network), useful and recommended for the absolute time stamping of the events in the log (see further 4.9.10, ""EVENT LOG" page"). If such a server is available and accessible via the IP interface, enter its address in the "NTP Server" field. A few public servers are also proposed in the dropdown list[1].*
A reboot is needed for this setting to become effective!

---

[1] Note: to be able to use these servers 1) access to the Internet from the codec must not be restricted, 2) a DNS must be set so that the symbolic names can be resolved.

4.9.10. "MAINTENANCE" tab

This tab leads to a selection between several pages dedicated to maintenance aspects.

"LOGIN DATA" page

This page is for setting the password for logging in the html pages. In a classical way, to set up a password you must first enter the current one, and then enter the new one and confirm it.

Remember to **save the change**, clicking the "**Save**" button!

ⓘ *Reminder: this password has no relationship with that for unlocking the possibly restricted access on the front panel interface.*

If you lose the password: using the front panel interface of the ScoopTeam, perform a complete reset of the settings, or a complete "factory reset". The passwords (front panel and html access) are then reset as well as the other parameters in the unit.

"SYSTEM UPDATE" page

This page should be used to upload a system update file into the unit and update the unit's firmware.

For an update, the file must be provided to you by AETA (or possibly your distributor/dealer). The procedure is rather simple:

- From this page, click "**Browse**…", find and select the update file.
- Click the "**Update**" button.
- The process begins, and a warning message is also displayed on the ScoopTeam display.
- Wait for the completion of the update. Normally the unit should reboot by itself at the end of the process.

This procedure requires maximum care, because such update always implies some risk. Please note these additional recommendations:

- Beforehand backup/export the recorded profiles and snapshots, if you would like to use them again afterwards.
- Make sure the link between the control computer and the ScoopTeam is stable (no undesirable interruption during the process).
- Make sure the ScoopTeam is not powered off and its Ethernet interface is not disconnected during the process, do not use its keys during the process.
- Re-import your profiles and snapshots after the update. *But be aware that, although we always try to keep a maximum upward compatibility, AETA cannot guarantee the re-usability and consistency of these memories after a firmware update.*

"RESET" page

This page provides two functions: clearing the settings and/or memories/profiles recorded in the unit, and rebooting the unit.

You can reset the unit completely or only clear a category of data; the available options are:

- "**Reset Snapshots**": all snapshots are cleared from the memory.
- "**Reset SIP Profiles**": SIP profiles are cleared from the memory.
- "**Reset Presets+CallProf**": call profiles and presets are cleared from the memory.
- "**Reset Settings**": all parameters are brought back to their default setting, but all memories are kept: call profiles, presets and snapshots.
  Warning: at the time of writing, this function does not reset 100% of the unit settings.
- "**Factory SIP Accounts**": reload the SIP account data integrated in the unit.
- "**Factory Reset**": all parameters are brought back to their default setting, and all memories (call profiles, presets, snapshots) are deleted. *Afterwards, the unit automatically reboots to ensure the parameters are properly reset.*
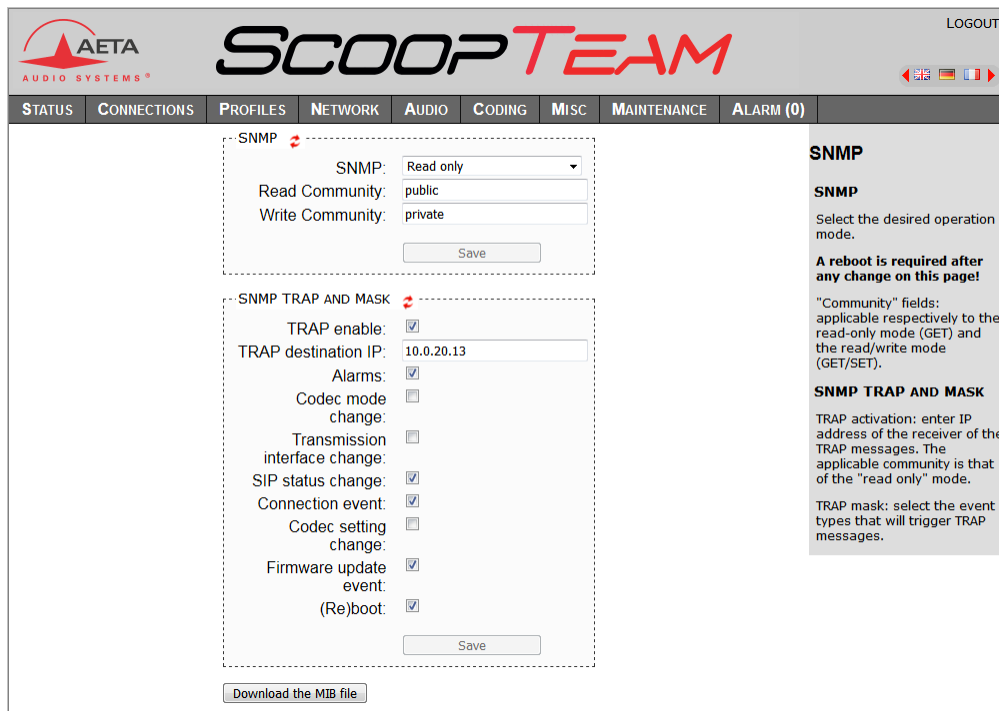
The selected action is performed after you click the "**Reset**" button.

The page also allows you to reboot the unit by remote, clicking the "**Reboot**" button. This is equivalent for ScoopTeam to an off/on power cycle. Of course, you must wait for the return to the normal operative state before regaining control via the html pages.

"SNMP" page

This page allows you to set the SNMP operation mode and configure traps.

The first section "SNMP" deals with the general settings:

- **SNMP**: the SNMP agent can be disabled, or "Read only" (SET messages rejected), or "Read/Write" (full access GET/SET from a managing agent).

- **Read Community**: applicable to the read-only mode. *A reboot is required after changing this setting.*

- **Write Community**: applicable to the read-write mode. *A reboot is required after changing this setting.*

The "SNMP TRAP AND MASK" section configures the TRAPs possibly generated by the ScoopTeam:

- **TRAP enable**: check to activate the TRAP function.

- **TRAP destination IP**: address of a management server to send the TRAP messages to. *A reboot is required after changing this setting.*

- The following checkboxes ("Alarms", etc., up to "(Re)boot") allow to select the type of events that should trigger the generation of TRAP messages.

Lastly, you can get the MIB description (for using in a network management system) by clicking the "**Download the MIB file**" button.

"Event log" page

This page displays a history if the system events, which can be useful for operational records, or for investigating operation issues, etc.

Events are logged by the codec in its internal memory; they are recorded in plain text (unformatted ASCII), and the web page displays the 200 last lines of this log (with a scroll bar).

You can select the detail level for this history, depending on the target application:

- "Coarse": only essential events are logged, such as reboots, link set up, alarms… This type of history can be useful as a link record, and gives a very synthetic overview.

- "Normal": gives more details, and this is the default setting, suitable for most operation needs.

- "Debug": absolutely all events are logged; this makes a very "verbose" and technical history, reserved for testing and fixing issues.

ScoopTeam can also send event messages to a SYSLOG server is such server is available on the network: enter the IP address of this server and click the "**Apply**" button. From then on, ScoopTeam sends the designated server <u>all</u> events, regardless of the detail level selected as described above. The two "histories" run in parallel: on one hand messages to the SYSLOG server, on the other hand the "filtered" events (depending on the detail level selected) recorded in the ScoopTeam log file.

Events are time-stamped (date, hour, minute, second) from the internal clock of the unit. This clock is not backed (no battery inside the unit), but the unit can synchronize at boot time using the NTP protocol. The address of such server must be set in the "Misc" page (see 4.9.9 above).

ⓘ *Important notice: time stamps are universal time (UTC), hence they take no account of geographical location, neither any DST (daylight saving time). Remember to add the appropriate time difference when analyzing the logs.*

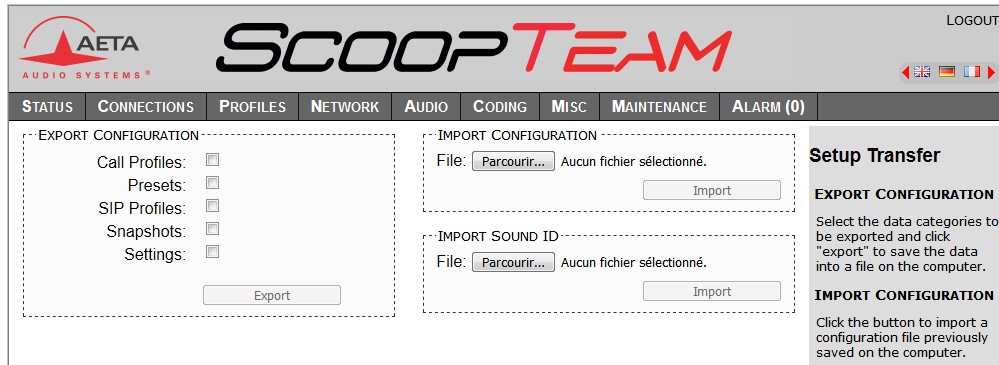Under the log window you can find some control buttons:

- "**Refresh**": click this button to update the display and show the most recent events. This button is relevant if the display has been frozen (with "Hold"), in order to refresh the event display on demand.

- "**Hold**": normally the event window scrolls at any time while new events occur. Click this button to stop this and freeze the display. The button becomes "**Continue**". *The events are still logged, only the page update is interrupted.*

- "**Continue**": resumes the automatic updating of the event display.

- "**Save Log File as…**": allows you to download on the control computer the history file. This takes place as a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

ⓘ *If you want to submit to AETA information about a possible issue or bug, you should set the log level at "Debug", save a log file right after the relevant event or scenario, and send to AETA the log file.*

"Setup Transfer" page

This page provides a means to backup all the settings of the unit into a file, and reciprocally to restore a complete configuration from a file previously saved in this way.

Partial exports are also possible, selecting the items you wish to include: call profiles, presets, SIP profiles, snapshots, settings…

The left part of the page is dedicated to "exporting" the device configuration: the configuration is then saved in a file, downloaded in the control computer. Before doing the export you can select which type of data is to be included in this backup/export: check the desired categories (multiple choice allowed), and click "**Export**". The rest is carried out like a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

The right side is for "importing" the device configuration: click "**Browse…**" to select the file to be imported, and launch the process by clicking "**Import**". The file must be a file previously exported from a ScoopTeam (same unit, or another with a compatible hardware configuration). The settings or memories included in the file are respectively applied to the codec, or recorded in its memory of profiles/snapshots.

ⓘ *Warning: the export files can be imported into a device with the <u>same firmware version</u>, but there is no guarantee of "portability" of the configuration files from one firmware version to another. In other terms, the outcome is not guaranteed when importing into a unit files which were exported from a unit with a different firmware version. We recommend to carefully check the settings in such case, after importing the file.*

You can also import the audio file to be used as the sound ID: click "**Browse…**" to select the file, and click "**Import**". The file must be a WAVE file (PCM16 mono, 48 kHz) and have a maximum duration of 10 seconds. After importing a new sound ID, you should reboot or restart the unit.

ⓘ *Include a silence period in the audio sample if you want to have a pause before the sample is repeated.*

You find on this page the settings related to the remote assistance and/or remote access. These settings are the same found on the unit's local menu (**Tools > Remote Access**).
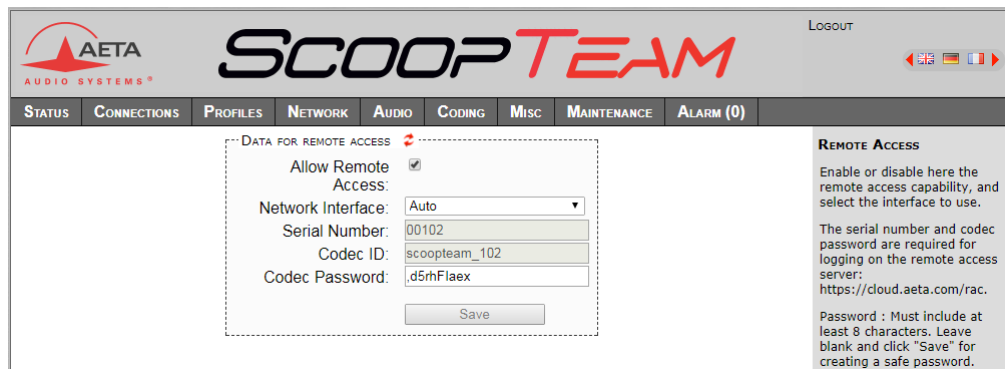
There are some differences depending on the installed options:

- "Allow Remote Access" becomes "Allow Remote Assistance" if no "remote access" option is installed.

- The "Codec Password" is not found if no "remote access" option is installed. One must know this password for getting remote access to the html pages, so it protects the unit against undesired access. No such password exists if no "remote access" option is installed.

ⓘ *Be aware, if you are currently on this page using remote access, that any change here can make you lose the connection. In the best case (password change), you will have to re-connect on the remote access server, using the new password.*

You can find the procedure for using remote access in 6.3.3, "How to use the remote access service".

4.9.11. "ALARM" tab

The "ALARM" tab switches to red as soon as at least one anomaly is detected. In addition the number of abnormal conditions is shown, like in the example below with one alarm condition detected:
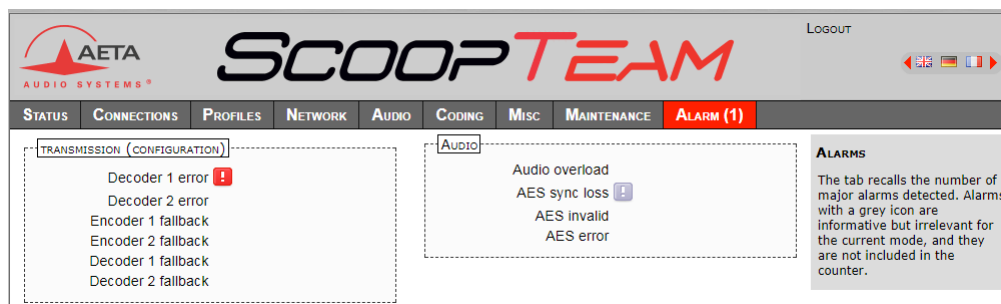
Clicking the tab, you reach this "Alarm" page showing all the alarm conditions. All possible issues are listed, but only those actually detected are highlighted with the 🟥 icon.

The issues are grouped in two areas:

- Alarms related to "TRANSMISSION": essentially issues related to the decoder synchronization. In addition, for ISDN links, there may be encoder or decoder fallback cases ("fallback" is the situation when the encoder or decoder configuration actually set on the link differs from the one which was initially programmed or expected).

- Alarms related to the AUDIO interfaces: audio overload, AES input errors.

As can be seen on the example above, minor or irrelevant alarm conditions do not contribute to the alarm count displayed in the tab label.

4.10. Operating mode – "My ScoopTeam" remote control applications

My ScoopTeam is a Windows application with a graphical user interface that allows to remote control the ScoopTeam over an IP connection. My ScoopTeam comes in two variations which differ in the type of link between the ScoopTeam and the computer:

- **My ScoopTeam LE** (LAN Edition) can connect to a ScoopTeam over a LAN, or more generally whenever there is a direct network path to the ScoopTeam under control. You must purchase a license from AETA.

- **My ScoopTeam RE** (Remote Edition) makes use of the "remote access +" service and provides the capability to control a ScoopTeam across the Internet. The application is free to install but the controlled units must include the "remote access +" option.

The two variations both provide a graphical user interface that is very similar to the display of the ScoopTeam. Once control is taken over a ScoopTeam, both applications have got the same operation mode.

However, the way they are activated and/or enabled is quite different.

4.10.1. Installing and activating the application

Either version of **My ScoopTeam** can be installed on a computer with following characteristics:

- Microsoft Windows Vista or more recent operating system;
- 2 GB RAM minimum;
- 1 GB available hard disk space;

Administrative rights are required for installing the application. You need to get the installation file for the application; it can be downloaded from AETA's web site.

Once you have installed the software on the computer, read the instructions that are installed along with the application.

Activation is required first of all for **My ScoopTeam LE**. You must use the product key provided by AETA when you purchase the application. Open the instruction document and follow the described procedure in order to activate the software.

4.10.2. Taking control over a unit

The two applications differ also on the way they select a unit and take control.

- **My ScoopTeam LE** accesses a unit by its IP address on the LAN. You can also browse the LAN and detect the ScoopTeam units found on the network.

- **My ScoopTeam RE** connects to a unit by using its serial number as identifier, and its codec password for authentication. You can build a list of known devices and easily switch from one to another.

You will find the detailed operation modes for this in the instructions that come along with the application.

Once you open a control session over a unit, you get a graphical user interface that is very similar to the display of the ScoopTeam:

Click this icon and you can enter a message that will pop up on the ScoopTeam's display. The message is cleared when the user touches the display.
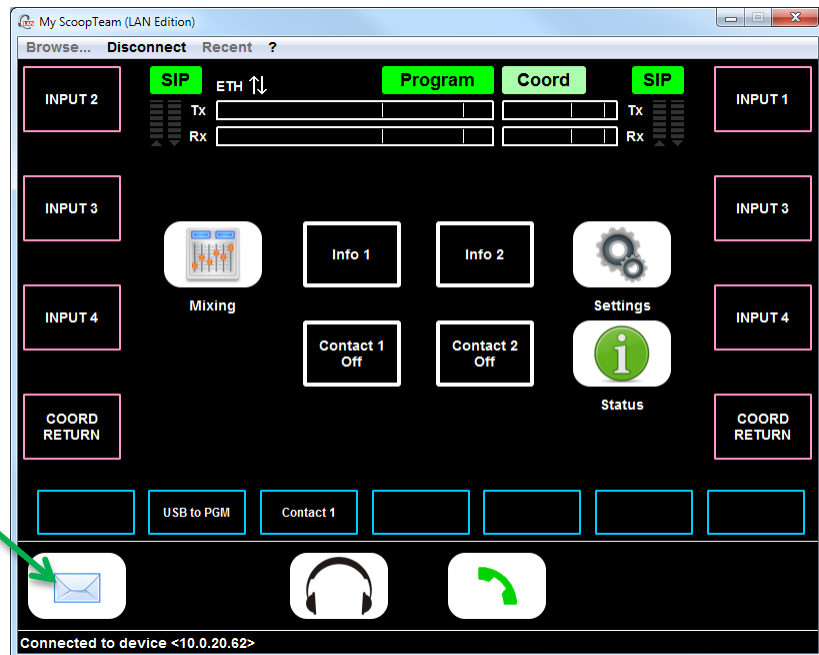
4.10.3. Using the interface

The operation should be straightforward for someone already familiar with the operation of the ScoopTeam. For example, the **user interface structure** is just the same as on the unit itself. However some actions are done in a slightly different way.

- **Buttons** such as the ⬜ key are replaced by virtual buttons on the screen; clicking such button is the same as pushing the physical key.
- The other buttons on the screen are just the same as the virtual buttons on the unit's display; clicking them is the same as touching the virtual buttons.
- In most cases, the mouse wheel is used instead of the rotary encoder of the ScoopTeam. For example, for setting a gain on the "Mixing" page, click the gain to select it, and rotate the mouse wheel to adjust the gain.
  As an alternative (for instance if you haven't got a mouse but a touchpad), you can use the "+" and "-" keys.
- Virtual knobs on the monitoring pages: click the gain to select the knob (it turns blue). Then rotate the mouse wheel to adjust the gain, or right-click to change the routing (L+R, L, R, "=="...).
- Controlling a "physical" knob (assignable knobs beside the unit's display):
  Click the corresponding button to select the control, scroll through the sources using the mouse wheel or right-click to change the routing (L+R, L, R, "=="...). Click OK when over.
  Right-click the button and a virtual knob pops up: click it and rotate the wheel to adjust the gain of the control. Click OK when over.
  Note: doing such adjustment by remote results in a mismatch between the gain and the physical knob's position. Any movement of the physical knob will override the gain.

ⓘ *You will notice that a few controls or parameters are purposely not replicated on the **My ScoopTeam** interface. This avoids mistakes or "dead end" situations; for instance it is not possible to switch off the unit from **My ScoopTeam**, and the corresponding menu is replaced by a "Reboot" item.*

5. Technical characteristics

5.1. Characteristics of audio interfaces

ⓘ *Note: this chapter does not apply to the "Premium" version and its optional AES67/Ravenna interface; these feature significant differences in their audio interfaces. Please refer to the user manual in its edition G for adequate information on these interfaces.*

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated.

5.1.1. Microphone inputs 1/2

The inputs are electronically balanced, using 3-pin female XLR sockets, and include a fixed high-pass filter.

Maximum input level:	+11 dBu	(with input limiter on)
Input impedance:	> 2 kΩ	
Frequency response:	500 Hz – 20 000 Hz ± 0.1 dB (high-pass cut-off 80 Hz)	
Gain range:	0 to 50 dB => input −18 dBu to −68 dBu for PGM = -12 dBFS	
THD + Noise:	< - 70 dB	(PGM = -3 dBFS)
EIN[1]:	< -122 dBu	(200 Ω input, maximum gain)
Phantom power voltage:	48 V	

5.1.2. Microphone/Line inputs 3/4

The inputs are electronically balanced, using 3-pin female XLR sockets.

Maximum input level (Mic position):	+11 dBu	
Maximum input level (Line position):	+22 dBu	
Input impedance (Mic position):	> 2 kΩ	
Input impedance (Line position):	> 5 kΩ	
Frequency response:	20 Hz – 20 000 Hz ± 0.1 dB	(80 Hz high-pass filter off)
Gain range (Mic position):	0 to +50 dB => input −18 dBu to −68 dBu for PGM = -12 dBFS	
Gain range (Line position):	-30 to +16 dB => output -48 dBFS to −2 dBFS for input = +4 dBu	
THD + Noise (Line position):	< - 70 dB	(Gain=0 dB, PGM = -3 dBFS)
EIN (Mic position):	< -122 dBu	(200 Ω input, maximum gain)
Phantom power voltage:	48 V	(Input 3 only)

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[1] Equivalent Input Noise

5.1.3. Aux audio inputs (Aux 1 / Aux 2)

These inputs are available in the "Premium+" version. Aux 2 can be switched to digital format (see below, 5.1.4)

These inputs are electronically balanced, using 3-pin female XLR sockets.

Maximum input level:	+22 dBu
Input impedance:	> 5 kΩ
Frequency response:	20 Hz – 20 000 Hz ± 0.2 dB
Gain range:	-30 to +16 dB => output -48 dBFS to –2 dBFS for input = +4 dBu
THD + Noise:	< - 70 dB (Gain=0 dB, PGM = -3 dBFS)

5.1.4. Digital audio input (Aux 2)

This interface is included in the "Premium+" version. It can be switched to analog line format (see above, 5.1.3).

This interface complies with recommendations:

- AES3-2009
- EBU Tech. 3250-E

The input includes sampling rate conversion and supports a sampling rate from 28 to 96 kHz.

Frequency response:	20 Hz – 20 000 Hz ± 0.05 dB
Gain range:	-30 to +16 dB => input –48 dBFS to –2 dBFS for PGM = -12 dBFS
THD + Noise:	< - 110 dB (Gain=0 dB, PGM = -3 dBFS)

5.1.5. Line audio outputs

The outputs are electronically balanced, using 3-pin male XLR sockets.

Maximum output level:	adjustable from +4 dBu to +22 dBu ± 0.5 dB
Nominal load impedance:	10 kΩ
Output impedance:	< 50 Ω
Frequency response:	20 Hz – 20 000 Hz ± 0.5 dB
THD + Noise:	< - 70 dB (Max level +16 dBu, PGM = -3 dBFS)

Line out 2 is only available on the "Premium+" version.

5.1.6. Headphone outputs

These outputs (6.35 mm jack) are suitable for low impedance headphones. High impedance headphones can be used as well; however, the maximum available power will be lower.

Maximum output level:	+18 dBu
Nominal load impedance:	32 Ω
Frequency response:	20 Hz – 20 000 Hz ± 3dB

5.1.7. AES67/RAVENNA interface

This interface is only available on the "Premium+" version.

Physical interface:	Ethernet 1000BaseT, RJ45 sockets
Input channels:	5
Output channels:	8
Sampling frequency:	44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, or 192 kHz
Gain range:	-30dB to +16dB

5.1.8. System level performance

Input to output delay: < 5 ms   (any analog input to any analog output)

From a RAVENNA input channel, the delay depends on the settings for the interface.

5.2. Characteristics of network interfaces

5.2.1. Ethernet Interfaces

These RJ45 socket have standard Ethernet pinout (for use of a normal "straight" cable to an Ethernet hub or switch). The interfaces normally operate in 100BaseT full-duplex mode, with auto negotiation, but it is possible to select other configurations for the main interface.

The installation and operation of these interfaces is detailed in 3.3, Initial setup of the Ethernet interface.

5.2.2. ISDN interface

This interface is available optionally.

The socket is an ISDN S0 interface (BRI), the supported protocols are ETSI (Euro RNIS), NI-1 / NI-2 or NTT.

5.2.3. Antenna sockets (mobile networks)

Available on units equipped with the "HD-4G" mobile network option. See also chapter 2.3.4: Mobile network access.

These sockets are used to connect one or two antennas (multiband antennas are included with the mobile network option).

The interfaces are male SMA (female contact), 50 Ω impedance.

The internal mobile access module can operate in following bands:

Frequency band (n°)	800 MHz (20)	850 MHz (5)	900 MHz (8)	1800 MHz (3)	2100 MHz (1)	2600 MHz (7)
2G GSM/EDGE			■	■		
3G/3G+ UMTS/HSDPA/HSUPA/HSPA/HSPA+		■	■		■	
4G/LTE	■	■	■	■	■	■

5.2.4. IP network protocols and ports

The ScoopTeam implements or complies with the following protocols (non exhaustive list):

- Physical and link layers: Ethernet, 100BaseT, 10BaseT
- Network/Transport layers (IPv4): TCP/IP, UDP/IP, RTP/IP
- Application: DHCP, DNS, HTTP, HTTPS, NTP, STUN, SYSLOG
- Audio transmission: SIP signaling, SDP, RTP, RTCP, RFC3550/3551, RFC3640
- Compliant with EBU recommendation Tech 3326 (interoperability of audio codecs for contribution)

The TCP ports used by the device are the following:

Type	Port		Designation	Dir	Notes
	Local	Remote			
TCP	22		SSH	⇄	Debug interface
	80		HTTP	⇄	Embedded html server
	443		HTTPS	⇄	Embedded html server
	7001		Control	⇄	Remote control; used by My ScoopTeam
		234	Control	⇄	Connection to cloud.aeta.com; used for the remote access feature

None of these ports is involved in the setting up of an audio connection.

The UDP ports used by the device are the following:

Type	Port		Designation	Dir	Notes
	Local	Remote			
UDP		53	DNS	⇄	Domain name resolution
		123	NTP	⇄	For querying an NTP server
		514	SYSLOG	→	For sending messages to a SYSLOG server
	2382		AETA enumeration	⇄	Used by "AetaScan"[1]
		3478	STUN	⇄	For querying a STUN server
		3479	STUN	⇄	For querying a STUN server
	5004		RTP	⇄	For audio streaming (SIP)
	5005		RTCP	⇄	For audio streaming (SIP)
	5006		RTP	⇄	For audio streaming (double SIP codec or Double Streaming)
	5007		RTCP	⇄	For audio streaming (double SIP codec or Double Streaming)
	5060		SIP	⇄	SIP signalling
	5062		SIP	⇄	SIP signalling *(secondary SIP account, if "alternate SIP port" is selected)*
	6000		Multicast/description	⇄	Multicast stream description channel
	6001		Multicast/audio	⇄	Multicast stream transmission channel
	9000		RTP	⇄	For audio streaming (Direct RTP)
	9001		RTCP	⇄	For audio streaming (Direct RTP)

Ports shown in **bold** can be changed from the default values shown in the above table. *Note that the corresponding remote ports are not fixed; they only depend on the remote entity. For instance, the remote SIP port is by default 5060, but it may be another value.*

The grey background highlights those ports possibly involved in an audio connection using SIP.

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[1] AetaScan is a tool that scans the LAN to identify AETA codecs and e.g. detect their IP addresses.

5.3. Miscellaneous interfaces

5.3.1. Host USB interfaces

The two interfaces are available on A-type sockets.

"P1" interface type: USB 2

"P2" interface type: USB 1.1 (Full Speed)

5.3.2. GPIO interface

The relay transmission interface (refer to) is available on the 9 pin female sub-D "GPIO" socket. It includes two isolated current loop inputs and two relay outputs.

The following table shows the pinout of this socket:

Pin		Function	Direction
	1	+5V of internal power supply	Output
6		GPI 1 Input loop	Input
	2	GPI 1 and GPI 2 Common	Input
7		GPI 2 Input loop	Input
	3	Ground	
8		GPO 2 relay output loop (a)	Output
	4	GPO 2 relay output loop (b)	Output
9		GPO 1 relay output loop (a)	Output
	5	GPO 1 relay output loop (b)	Output

All loops are isolated and bi-directional (free polarity).

The characteristics of the input loops are:

Input loop control current:	4 mA	(max. 50 mA)
Resistance of input loop:	$\sim 1 \ k\Omega$	(current limiting series resistor)
Input loop isolation:	> 5000 VRMS	

A +5V to +12V source may be connected directly on an input loop, because the internal series resistor is dimensioned for this purpose. For a higher voltage source, it may be necessary to limit the input current.

The characteristics of the output loops are:

Maximum switching voltage: 350 V peak

Maximum switching current: 120 mA

Resistance of output loop: < 35 Ω

Output loop isolation: > 2500 V_{RMS}

The 5V power supply is available from the unit to power a low-consumption device (maximum 300 mA current consumption), e.g. to power the input loops, or LED indicators connected to the relay outputs. *Do not draw too high current from this line!*

5.4. Power supply

The unit is powered from an external DC source, or using rechargeable batteries.

5.4.1. External DC power supply

The ScoopTeam operates from an external 12V DC source. A suitable power supply is provided with the unit, and spares are available from AETA (see 5.8, "Accessories and related products").

The unit features a 4 pin male XLR socket for connecting the DC supply (non isolated), and it includes overvoltage and overcurrent protection (resettable fuse on the DC input).

The DC supply also charges the batteries inside the unit.

Characteristics	Min	Nominal	Max
Supply voltage	10 V	12 V	16 V
Supply current			2 A

ⓘ *The power consumption may vary in a wide range, depending on several factors: installed options, operating mode, possible presence of batteries, and input voltage.*

Connections on the XLR socket: Ground is connected on pin 1, and pin 4 must provide the +12 V voltage.

5.4.2. Rechargeable batteries

The ScoopTeam can operate from batteries (6 AA-size NiMH rechargeable batteries), and a charger is included that charges the batteries when the external DC source is connected.

Only fast charge NiMH batteries are allowed, such as those proposed by AETA as accessories (see further, 5.8).

The run time depends much on the network used and the transmission conditions. Here are two typical use cases:

Standard version, AoIP over Ethernet, 40% display brightness: > 1.5 hour
Premium+ version with mobile option, 4G transmission, 40% brightness: > 1 hour

5.5. Dimensions and weight

Overall size: 212 x 236 x 102 mm (Width x Depth x Height)
Weight:

- Standard version: < 2.0 kg.
- Premium version with all options: 2.1 kg (batteries not included).

5.6. Environmental characteristics

The equipment operates over a 0°C to 45°C ambient temperature range (32°F to 113°F), and a 5% to 90% humidity ratio range.

The ScoopTeam complies with "CE" directives regarding safety and EMC.

- Safety: compliance with EN60950
- EMI: radiated emissions complying with EN55103-1
- Susceptibility: compliance with EN55103-2

ScoopTeam with the "mobile" option complies with the RED directive.

5.7. Versions and options

ScoopTeam comes in three versions:

- "Standard" version (ordering code 80 00 260 01);
- "Premium" version (ordering code 80 00 260 02), with additional audio interfaces. *Refer to previous user manual (Edition G) for details.*
- "Premium+" version (ordering code 80 00 260 03), with additional audio interfaces (see details in 2.1, Audio interfaces and functions).

Options can be added to complement the capabilities of the ScoopTeam.

5.7.1. Network options

For adding more types of network interfaces, the following options are available:

Ordering code	Option	Description
80 00 261 11	ISDN option	Addition of an ISDN S0 interface
80 00 261 01	HD-4G Mobile option	Addition of the mobile network access (3G/4G + HD voice). *Only available for the "Premium" or "Premium+" versions.*
80 00 261 62	Wi-Fi interface	Capability to connect on a Wi-Fi network, using a USB adapter (included).

5.7.2. Other options

Ordering code	Option	Description
80 00 261 12	RAVENNA 2-channel option	RAVENNA interface for 2 audio input channels and 2 audio outputs channels. *Only available for the "Premium" version.*
80 00 261 15	AES67 8-channel option	AES67/RAVENNA interface for 5 input channels and 8 output channels. *Only available for the "Premium+" version.*
80 00 261 13	Remote Access	Capability to set up a remote control session via the AETA Remote Access Server and access the html pages of the ScoopTeam via the Internet.
80 00 261 14	Remote Access +	Capability to set up a remote control session over the Internet via the AETA Remote Access Server, and access the html pages of the ScoopTeam or control it from the "My ScoopTeam RE" application.

5.8. Accessories and related products
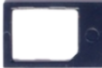
The ScoopTeam is delivered with an AC/DC adapter and a CAT5 Ethernet cable.

The "Mobile" option is delivered with two multiband antennas.

The following additional accessories or spare parts are also available from AETA:

Ordering code	Description	
80 00 262 02	Flight case *Including housing for power adapter and accessories.*	
80 00 262 04	Soft carrying bag	
80 00 262 05	Transparent rain cover	
80 00 262 01	Spare XLR4 power supply	
80 00 262 03	12 V car battery adapter	

(Continued)

Ordering code	Description	
80 00 224 01	Rechargeable Ni-MH battery set	
ZAAS00078	Multiband antenna, SMA connector, 1 m cable	
ZAAS00051	Multiband antenna, SMA connector, 3 m cable	
ZAAS00082	Hinged antenna, SMA connector	
80 00 262 10	USB adapter for 3G/4G mobile access in Europe, Middle-East, Africa	
ZAAS00056	µSIM adapter	
80 00 261 21	**My ScoopTeam (LAN Edition)**: Windows application for full remote control of the unit via a LAN, with a graphical user interface very similar to the unit's display.	
Free to download	**My ScoopTeam (Remote Edition)**: Windows application for full remote control of the unit via Internet, with a graphical user interface very similar to the unit's display. *"Remote Access +" option required on each controlled unit.*	

6. Annexes

6.1. Details on the coding algorithms

The following bring some precisions about the important features of the various algorithms and protocols available.

6.1.1. G711 coding

Application: telephony, coordination. Low latency.

G711 is the standard coding used for voice transmission on public telephone networks, and features 300 to 3400 Hz audio bandwidth. This algorithm is typically used for links over IP networks with IP telephones or VoIP gateways. Via ISDN, G711 is used for links with telephones or hybrid devices.

6.1.2. G722 coding

Application: commentaries, coordination. Low latency.

This mono coding algorithm at a 64 kbit/s bit rate is a reference for commentaries, and features a 50-7000 Hz bandwidth.

For ISDN, three synchronisation modes are possible:

- "Statistical recovery" byte synchronisation method (alias SRT);
- H221 synchronisation; in this case, 1.6 kbit/s from the compressed data are used for this;
- H221 synchronisation and H242 protocol. This is only available for the ISDN mode.

No specific synchronisation is required for the IP mode.

6.1.3. MPEG Audio Layer 2 coding

Application: mono or stereo music, high quality.

As shown on Table 2, this coding algorithm features a maximum flexibility, with many variations for bit rate, mono or two-channel mode, sampling rates...

The two channel modes come in three variations:

- Stereo: the two audio channels are bit-rate compressed independently.
- Dual mono: the encoding is just the same as the previous case, but this choice applies to channels without any acoustic relationship, such as for example commentaries in two languages.
- "Joint stereo": applies to a stereo program, but here the encoding process takes advantage of the inter-channel correlation. To be used only for a stereo program.

The 24 kHz sampling rate features a moderate bandwidth (10 kHz) and is rather useful for commentaries.

ⓘ *The latency is rather high with this sampling rate.*

For ISDN transmission and at 128 kbit/s, MPEG L2 uses a system for inverse multiplexing two B channels that is compatible with most codecs on the market.

6.1.4. MPEG AAC algorithms

Application: music, low capacity transmission channels.

These algorithms feature a very high compression ratio, for a given audio quality, compared to Layer 2. They operate at a sampling rate of 48 kHz, and several bit rates: 16, 20, 24, 32, 40, 48, 56, 64, 96, 128, 192, 256 kbit/s. Two coding variations are available:

- MPEG **AAC-LC** ("Low Complexity"): lower compression than HE-AAC, but lower latency.
- MPEG **HE-AAC** ("High Efficiency" AAC): higher compression, and the bit rate is limited to 128 kbit/s for this variation.

For ISDN, two synchronisation modes are possible:

- ADTS is used as the normal default system; the algorithm designation in such case is "AAC-LC" or "HE-AAC".
- LOAS can be used as an alternative for compatibility with some third party devices; the algorithm selection in such case is "AAC-LC LOAS" or "HE-AAC LOAS".

No such choice is required for the IP mode.

6.1.5. OPUS coding

Application music, low capacity transmission channels. Moderate latency.

Opus is an open source coding algorithm (under BSD license), described by RFC6716 from the IETF. Like MPEG coding algorithms, it is a wide band audio codec, but it features distinctive characteristics:

- Low latency; the algorithmic delay of Opus is very low compared to MPEG coding (Layer 2, AAC, HE-AAC...).
- Adaptability: Opus features many parameters influencing its performance, but it is possible to adapt them automatically depending on the external constraints and the signal characteristics. In the ScoopTeam, the configuration stays very simple: the user just imposes the desired bit rate, and the codec automatically adjusts the other parameters (for instance the audio bandwidth).
- Capacity to apply changes "on the fly": the coding allows changing its operation parameters on the fly, with no interruption and no artefacts when switching. In this way the bit rate can be changed during the link on the ScoopTeam, without any audio noise or dropout.
In addition, when the remote device is another AETA codec, it will automatically apply the same bit rate change to the stream it is transmitting.

Opus coding is only available for IP networks (wired or mobile).

6.2. Using the Double Streaming system

6.2.1. Context

Audio transmission over IP (AoIP) is sometimes subject to data losses for various reasons, as for instance:

- In case of network congestion, the overloaded routers happen to drop packets without forwarding them.
- On mobile radio links, radio signal loss or fading cause temporary interruptions, that can last for a rather long time.

Such problems are more common on connections via the Internet.

Packet losses cause disturbances, ranging from light artefacts to the complete muting of the signal for several seconds or minutes.

The AETA "Double Streaming" system is an efficient way of strongly increasing the robustness of an AoIP link, thanks to a redundant link that can compensate for losses on the main link.

6.2.2. Presentation of the "Double Streaming" system

Let us consider a classical AoIP link between two audio codecs:
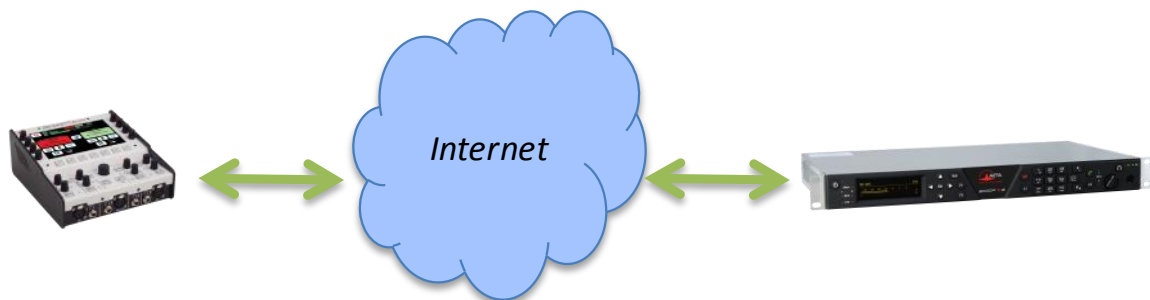
At each end the codec accesses the Internet through an interface which can be of various types:

- Ethernet connection
- Mobile access (integrated or USB device)
- Wi-Fi access

Packet losses can be experienced in various locations on the path between the two units.

General principle of operation

Double Streaming relies on setting up two SIP links, simultaneous but via two separate routes:
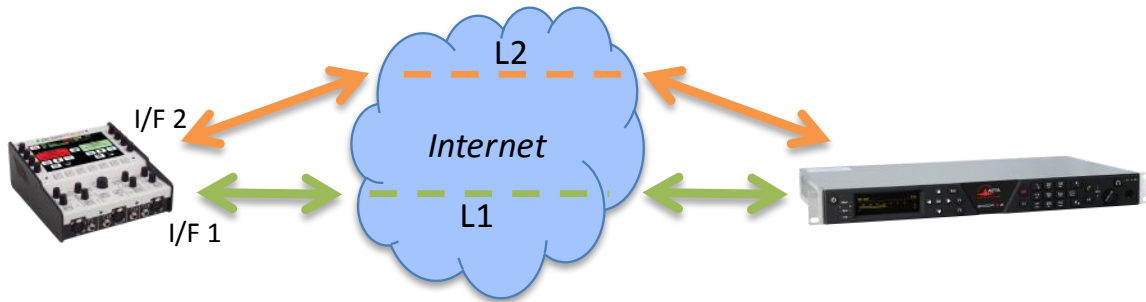
In addition to the "main" SIP link, a "secondary" SIP link is set up between the units. This link transports a copy of the audio stream of the main link.

On each codec, a specific transmission interface can be selected for each of the two AoIP connections. In this way two independent routes are created, whose possible losses are not correlated.

The codec which receives the two (normally) identical streams recombines them, discarding the duplicates:
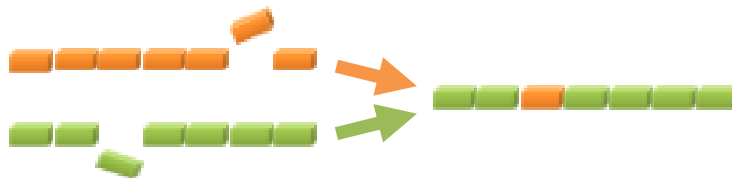
Whenever a packet is lost on the transmission, its copy from the other link compensates the loss, and a 100% lossless stream is delivered at the end of the process.

In an extreme case, if one of the two links is totally interrupted, the complete original stream is still available at the output. The few losses on the remaining link are no more compensated for, but they are much less noticeable than a complete loss.

The usage of SIP is a specific advantage of AETA's Double Streaming:

- The system works on a normal SIP architecture, and is compatible with any SIP server.
- The system is easily "portable", specifically it allows using mobile links.
- Like for a normal SIP link, an acceptable security level can be reached.
- Like for a normal SIP link, it is also possible to make links without a SIP server (with same limitations).

Double Streaming is available with all AETA codecs of the following product ranges:

ScoopTeam

Scoopy+ S

Scoop5 S and Scoop5 S-IP

µScoop

MultiScoop

Chapter 6.2.3 describes the operating instructions for using it.

Some examples of operation setups

Case 1: Redundancy on one end

In this first case, a redundant network access is used on one end of the link only, and the two redundant links are merged on a single network access on the other end.

The end with redundancy is protected against faults encountered on the main link, either at the local access level or on the path over the network towards the other end.

Conversely, the network access on the other end gets no additional protection.
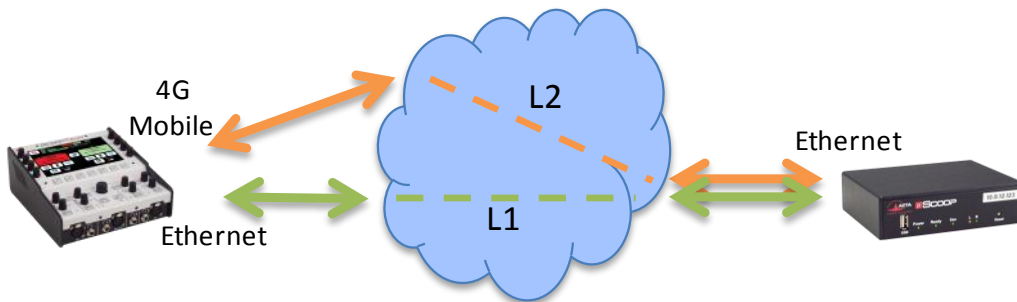
Example 1: Remote report, wired connection secured by a mobile link.

In the above example, in the field (left side on the picture) a 4G network connection is used for securing a main link via an Ethernet connection. On the control room side, the two streams are merged on a high reliability wired network that needs no redundancy.

Case 2: End-to-end redundancy

In this case, a redundant network access is used at each end.

A maximum protection is then available against faults experienced either at the network access level or inside the network.
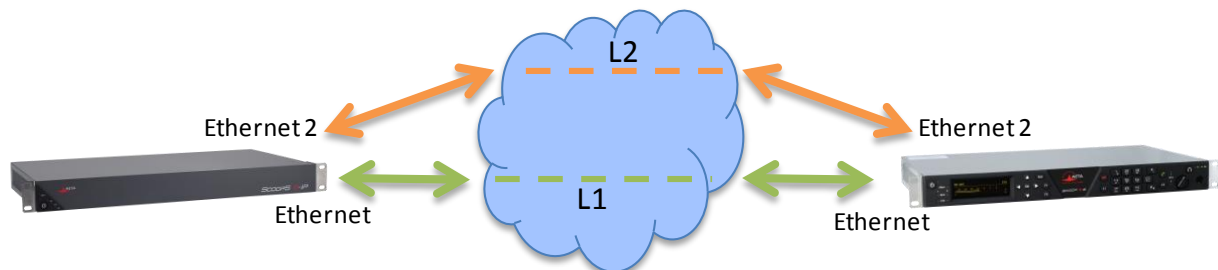
Example 2: Wired link, secured end-to-end by a secondary link

In the above example, the secondary link may for instance rely on cheap ADSL connections for providing high-grade availability at a moderate cost.

6.2.3. Setting up a link with Double Streaming

Pre-requisites

Each codec involved must have:

- Recent firmware, supporting the "Double Streaming" feature. *The feature must be enabled on both codecs.*

- Two network interfaces (depending on the unit: Ethernet, auxiliary Ethernet, integrated mobile, USB mobile device, Wi-Fi…). However it is possible on one end to use the same interface for both connections, like for instance in *Case 1* above.

- For links with a SIP server, one or two accounts on a SIP server (same server for both accounts). The first account is assigned to the main link, the second one is for the secondary link (it is optional if both links use the same interface, like on the non secured end in *Case 1* above). *It is possible not to involve a SIP server, but with same drawbacks as for regular links (i.e. without redundancy).*

Recent[1] AETA codecs are fitted with "factory" SIP accounts on the AETA SIP server, usable for Double Streaming. Other accounts and servers can be used as well.

Preparing the codecs

The following is to be done on each of the codecs involved in the link:

1. **Enable "Double Streaming"**

ScoopTeam, via menu:	**Network > AoIP settings**
Scoopy+ S and Scoop5 S, menu:	**Setup > Network > AoIP Parameters**
All products, html pages:	**NETWORK tab, AoIP PARAMETERS page**

   Warning: you must, where appropriate*, disable the "double codec" mode, as it is not compatible with Double Streaming.*

2. **Set up the network interfaces** that will be used (refer to the instructions of the user manual).

3. **Set up the two SIP accounts**, main and/or secondary (except if you do not intend to use a SIP server).

ScoopTeam, via menu:	**Network > AoIP settings > Account 1 / Account 2**
Scoopy+ S and Scoop5 S, menu:	**Setup > Network > AoIP Parameters**
All products, html pages:	**NETWORK tab, AoIP PARAMETERS page**

   For reloading the factory AETA accounts, if needed:

ScoopTeam, via menu:	***Tools > Troubleshooting > Reset,*** *"Reload the factory SIP Accounts" option*
Scoopy+ S and Scoop5 S, menu:	***Tools > Misc > Reset,*** *"Factory SIP Accounts" option*
All products, html pages:	***MAINTENANCE tab, RESET page,*** *"Factory SIP Accounts" option*

4. **Select the main interface** as for a single link (without redundancy):

ScoopTeam, via menu:	**Network > Network for Outgoing Calls**
Scoopy+ S and Scoop5 S, menu:	**Setup > Network > Change Network**
All products, html pages:	**NETWORK tab, CHANGE NETWORK page**

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[1] Scoop5 S, Scoopy+ S, µScoop, ScoopTeam and MultiScoop product ranges, manufactured since May 2017.

5. **Select the secondary interface**:

ScoopTeam, via menu:	**Network > Secondary IP Network for Double Streaming**
Scoopy+ S and Scoop5 S, menu:	**Setup > Network > Secondary Network**
All products, html pages:	**NETWORK** tab, **SECONDARY NETWORK** page

Note: it is possible to select the same interface as for the main link: end without redundancy in Case 1. For a µScoop this is always the case, because this product only features one single interface.

Establishing the link

Once the above preparation is performed, normally the situation should be as follows:

- The interfaces used (possibly just one) are connected and active.
- The codec registers on the server[1] with the main SIP account (SIP 1) via the main interface.
- The codec registers on the server with the secondary SIP account (SIP 2) via the secondary interface. *If it is the same as the main interface (Case 1, end without redundancy), this secondary registration is not required.*

Check that the SIP registration indicators are OK:

- On the ScoopTeam display, the secondary indicator is the one on top right.
- On Scoop5 S or Scoopy+ S display, you should see the "S_1" and/or "S_2" indicators.

The unit is now ready for receiving a call in Double Streaming mode.

For initiating a call (launching an outgoing call), the procedure is very similar to a regular call, using the two identifiers of the destination codec: SIP URI or IP addresses.

Call from a ScoopTeam:

Enter the two identifiers, separated by a comma: *dest_1,dest_2*   (example 902942,9029422)

If you enter just one identifier, it is used for both links (in such case only one interface is involved on the remote codec, as in Case 1 on the end without redundancy).

Launch the call as for a regular call.

Call from a Scoopy+ S or a Scoop5 S:

Enter the first identifier and press the call key . A prompt for entering a second identifier is displayed: enter it and press the call key again.

If you press the key without entering a second identifier, the first one is used for both links. In such case only one interface is involved on the remote codec, as in Case 1 on the end without redundancy.

Call from the embedded html CONNEXIONS page:

Enter the two identifiers in the two adequate fields: "Remote Number" and "2nd Remote Number". Click the "Dial" button.

If you leave the second number blank, it is automatically copied from the first one (in such case only one interface is involved on the remote codec, as in Case 1 on the end without redundancy).

What happens then: the codec calls destination 1 via the main interface, and makes a second call to destination 2 via the secondary interface. The link is established and secured.
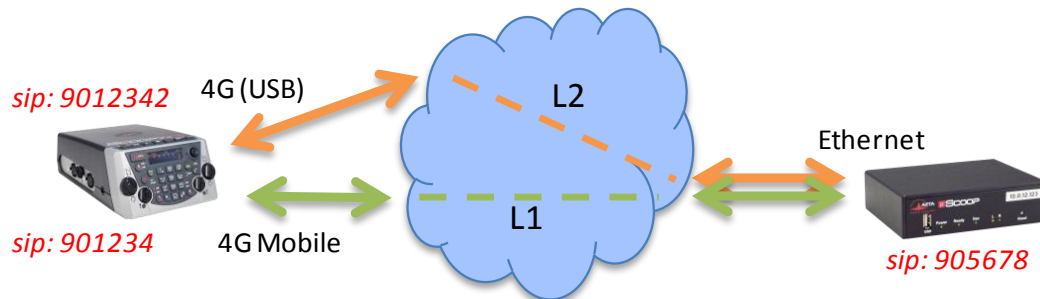
---

[1] Except if a SIP server is not used.

Releasing the link

The operation is just the same as for a regular non redundant link. The codec automatically releases the two redundant connections of the Double Streaming link.

6.2.4. Implementation examples

With single-ended redundancy

Outside remote, double mobile connection

This example involves:

- A Scoopy+ S on the outside location with 4G network access (integrated module), secured by a second mobile access (4G USB key with a SIM card from a different operator).
- A µScoop on the MCR side, with a high reliability wired Internet connection.

On the Scoopy+ S, two SIP accounts are used, for instance the integrated "factory" accounts.

In the MCR, a SIP account is used, for instance the integrated "factory" account. "Double Streaming" is enabled.

For a **"regular" call from the Scoopy+ S** without redundancy via the main 4G interface:

- Double Streaming disabled.
- Check the SIP 1 registration (integrated 4G interface).
- Call the SIP number of the µScoop: 905678 in this example.

For a **Double Streaming call from the Scoopy+ S**:

- Double Streaming enabled.
- Check the SIP 1 registration (integrated 4G interface) and the SIP 2 registration (4G mobile via USB).
- Call the SIP number of the µScoop: 905678 in this example.

For a **Double Streaming call from the µScoop**:

- Double Streaming enabled on both units.
- Check the SIP registration.
- Call the SIP numbers of the Scoopy+ S: in this example, 901234 and 9012342.

With end-to-end redundancy
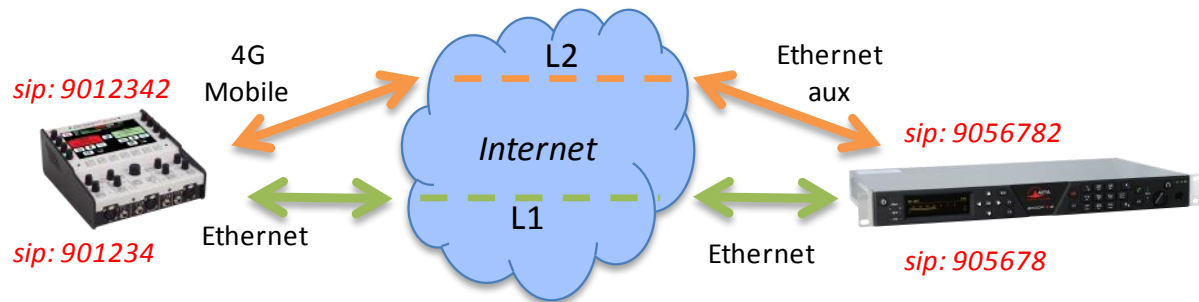
Outside remote, wired connection secured by a mobile link.

This example involves:

- A ScoopTeam on the outside location with 4G network access for securing a main link via Ethernet access.

- A Scoop5 S on the MCR side, with a main wired link, secured by a cheap ADSL connected through a second Ethernet interface (USB adapter).

On the ScoopTeam, two SIP accounts are used, for instance the integrated "factory" accounts.

In the MCR, two SIP accounts are needed, for instance the integrated "factory" account. "Double Streaming" is enabled.

For a **"regular" call from the ScoopTeam** without redundancy via the Ethernet interface:

- Double Streaming disabled.

- Check the SIP 1 registration (Ethernet interface).

- Call the SIP 1 number of the Scoop5 S: 905678 in this example.

For a **Double Streaming call from the ScoopTeam**:

- Double Streaming enabled.

- Check the SIP 1 registration (Ethernet interface) and the SIP 2 registration (4G mobile).

- Call the SIP numbers of the Scoop5 S: 905678,9056782 in this example.

For a **Double Streaming call from the Scoop5 S**:

- Double Streaming enabled on both units.

- Check the SIP 1 and SIP 2 registration.

- Call the SIP numbers of the ScoopTeam: in this example, 901234 and 9012342.

6.2.5. Additional information

Notes about the network settings

AoIP settings

Protocol: only the SIP protocol is supported with Double Streaming.

QoS: it is possible to use packet duplication, like for a regular link. This results in both "temporal" redundancy inside each link (packet duplication) and "spatial" redundancy between the two links (Double Streaming). Of course, the global cumulative bit rate is then quadrupled.

Parameters of the IP interfaces

The **Network Quality** setting, as it affects the size of the reception buffer, can have a significant impact on the performance of Double Streaming.

For correct operation of Double Streaming, this setting must be identical on both interfaces involved. If the settings are different, the codec will automatically apply the higher buffer size of the two for both links (corresponding to the poorer quality).

For better performance in terms of protection against losses, it is advisable to avoid the "High" and "Very High" quality settings, which are normally applicable to transmissions of... very good quality.

In particular, if the latency is very different between the two means of transmission (this can be encountered with networks of very different technology), a low depth buffer cancels the benefit of the redundant stream.

Transmission quality monitoring

Indicators and transmission quality measurements apply to the primary link only. As a result, there may be an apparent inconsistency between these indications and the quality actually observed on the audio program:

- When the transmission is of poor quality on the main link but correct on the secondary link, the quality indications report a poor quality, while the overall audio result remains without defect, thanks to the redundancy.

- On the other hand, if only the secondary link is faulty, the indicators remain on an excellent quality level as long as the main link maintains a good quality.

Interoperability

If a codec configured for Double Streaming receives a regular SIP call from any SIP device: the call is processed normally on the interface corresponding to the identifier that has been called.

Conversely, if a codec configured for Double Streaming calls an unsupported unit, or a unit not configured for Double Streaming: the call fails with rejection.
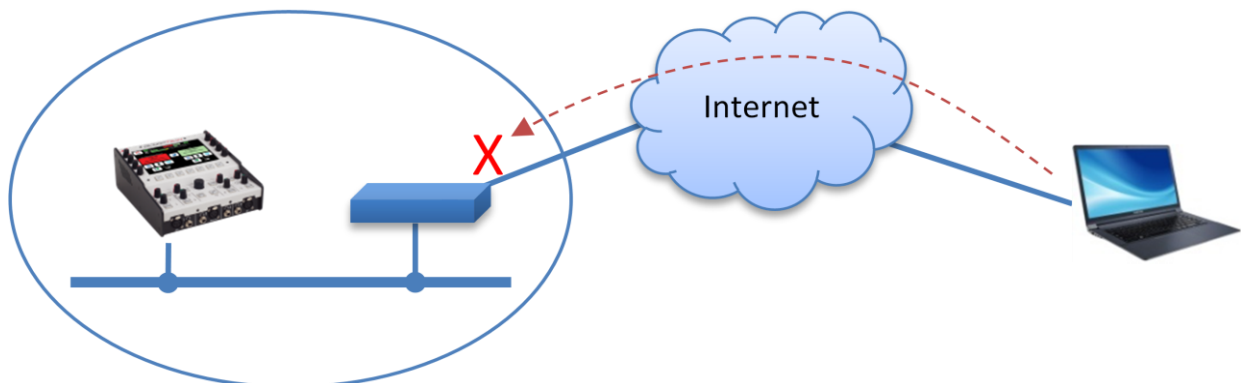
6.3. Remote access

6.3.1. Background

A computer connected on the same LAN as the ScoopTeam (via Ethernet or Wi-Fi) can easily take control over the unit:
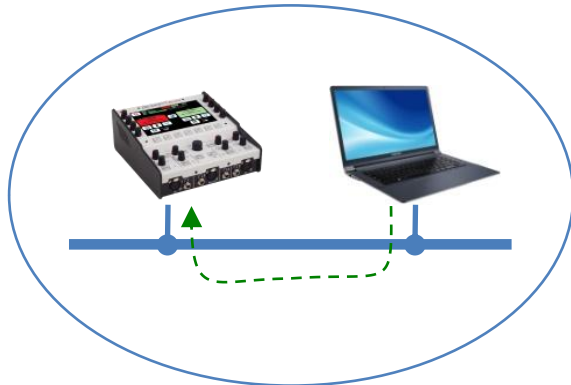
That is a configuration where the computer can access the html pages and/or use the "My ScoopTeam LE" application for controlling the unit.

When the computer is on a remote location, connected via the Internet, there is most often a need to go through NAT router(s) and/or firewall(s):


In such situation, it is not possible to get a connection as directly as over a LAN.

The AETA Remote Access service is designed for working around this obstacle and enabling to take control over a device even in such case[1].

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[1] However, a very restrictive firewall can of course block the service

6.3.2. "Remote Access" service

The system relies on using a remote access server, as an intermediary between the ScoopTeam and the control device. A control session is performed in two phases:
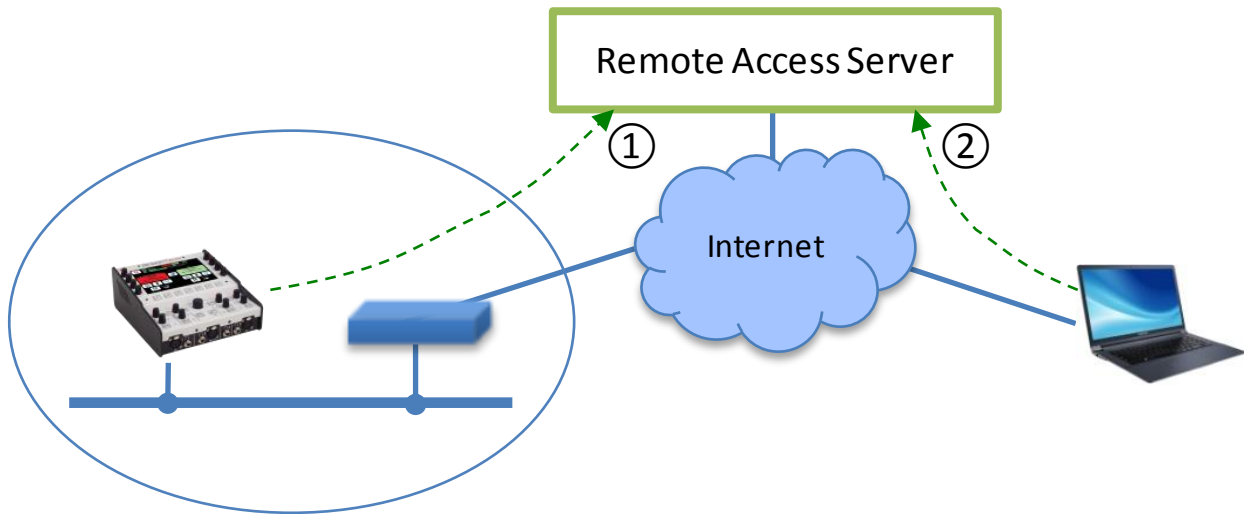
1. The ScoopTeam connects to the server, and gets available for a possible remote access session.

2. A user who wants to remote control the ScoopTeam sets a connection to the server, and the server sets a virtual link between the control device and the codec, identical in its operation to a direct link via a LAN.

Both connections to the server are secured and encrypted. The step 1 above is performed when the ScoopTeam is connected to the network and when remote access is active on the unit. However this can be disabled by the user. In addition it is possible to select which IP network interface is used for connecting the unit to the server.

Three types of service are proposed:

- **Remote assistance**: this service allows the AETA support staff to access the codec, for help or possible investigation. *Your explicit agreement is required for such operation.*

- **Remote html access**: provides access to the html pages of the codec from remote. The control device can be any device with a web browser, without the need to install any application. *A "Remote Access" option must be installed on the ScoopTeam for using this service.*

- **"Remote access +"**: this service, available for a ScoopTeam with the "Remote Access +" option, allows controlling the unit from a Windows PC equipped with the "My ScoopTeam RE" application.

6.3.3. How to use the remote access service

Using the remote assistance

This service allows the AETA support staff to access the codec, for help or possible investigation.

What you need on the ScoopTeam:

- Up-to-date firmware (1.03 minimum).
- Internet access, for instance via an Ethernet connection to a LAN. But other IP connections are suitable as well: mobile data access, Wi-Fi…

No specific option is required.

Step 1: connect the ScoopTeam to the server

Menu **Tools** > **Remote Assistance**: activate "**Remote Assistance**".

If a "Remote Access" option is installed, the menu is **Tools > Remote Access,** *"Remote Access".*

If necessary, select the desired IP interface to be used for the remote access connection: "**Interface for Control"** (with the "Auto" choice, it will be the same that is used for audio over IP transmission).

Check the successful connection: menu **Status,** "**Remote Assistance Service".**

Take note of the MAC address of the main Ethernet interface (LAN1): menu **Network > LAN1 settings** .

Step 2: open the remote session

Tell the AETA technician the MAC address of the codec; this is the identifier for the device.

AETA can then set a control session over the codec. The link is secure and encrypted.

You may disable remote assistance once the intervention is over. *This is recommended for avoiding useless data traffic.*

Using the remote html access

The remote html access allows to access the html pages of the codec from remote.

What you need on the ScoopTeam:

- Up-to-date firmware (1.03 minimum).
- The "Remote access" option (or the "Remote access +" option).
- Internet access, for instance via an Ethernet connection to a LAN. But other IP connections are suitable as well: mobile data access, Wi-Fi…

The control device can be any device with a web browser, without the need to install any application.

Step 1: connect the ScoopTeam to the server

Menu **Tools > Remote Access**: activate "**Remote Access**".

If necessary, select the desired IP interface to be used for the remote access connection: "**Interface for Control"** (with the "Auto" choice, it will be the same that is used for audio over IP transmission).

Check the successful connection: menu **Status,** "**Remote Access Service".**

If you want to change the password, you must be logged as an administrator (*if necessary, switch this via the menu* **Tools > Access Level**: ***Administrator***). In the menu **Tools > Remote Access** you can enter a password as you want, with following rules:

- The password must be **at least 8 characters** long. *For safety, it is recommended to use at least 13 characters, with a combination of uppercase and lowercase letters, numbers, and special characters among the following: _ - + * . : , ; # ? ! = .*
- If you enter a blank password, or one with less than 8 characters, a random password is automatically generated by the ScoopTeam.

You may also use the "REMOTE ACCESS" html page (from the "MAINTENANCE" tab) for activating this connection, and check the successful connection on the STATUS tab, GENERAL section.

Step 2: open the remote session

The ScoopTeam to be controlled must be connected to the server, and you must know the serial number and the "Codec password" of the unit.

Open a web browser, and go to the following URL:

https://cloud.aeta.com/rac

You reach the AETA remote html access interface:

1. Select the product type (ScoopTeam) from the drop-down list.
2. Enter the serial number (leading "0" don't care).
3. Enter the codec password.
4. Click the "Login" button.

After a while you get the home page of the codec, almost identical to that you would get with a direct connection over a LAN:
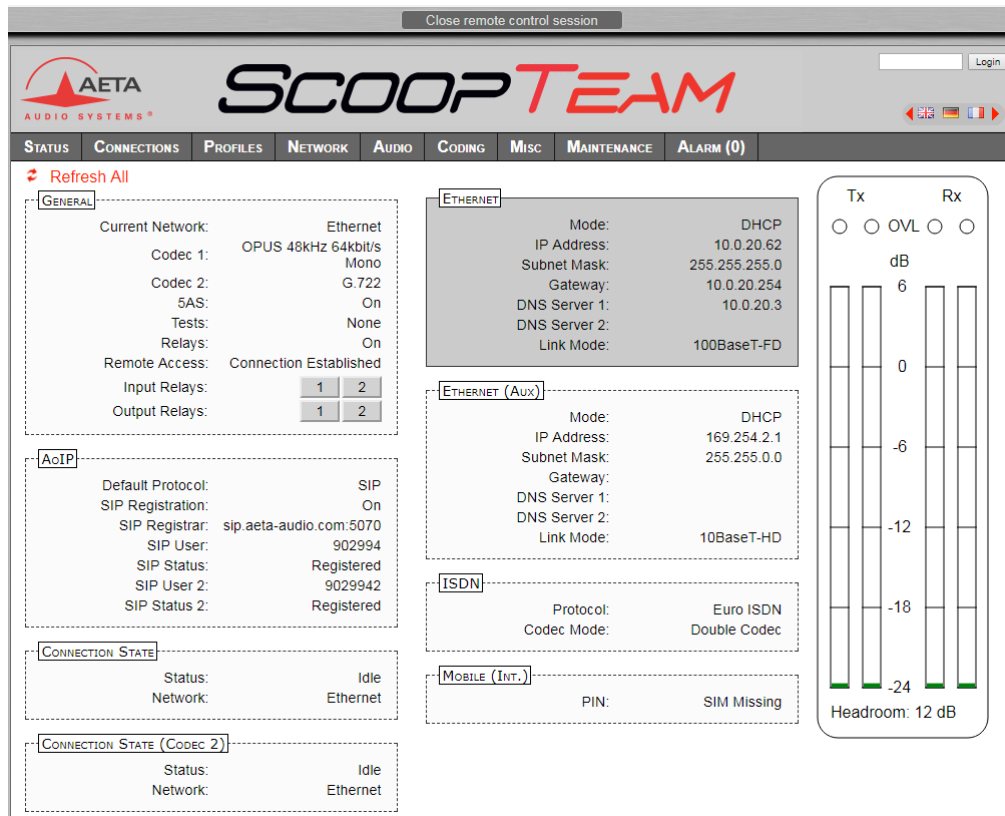
The difference is in the small bar at the top ("Close remote control session"). Click it when you wish to disconnect from the server.

Navigating the pages follow the same rules as a direct access through a LAN.

Warning: do not confuse the "password codec" that you used for this connection, and the password possibly configured to restrict the access to the html pages in a general way (i.e. regardless if the access is local or remote). These passwords are independent of each other.

The link is secured and encrypted.

You may disconnect the ScoopTeam once the session is over. *This is recommended for avoiding useless data traffic.*

Using "remote access +"

Using this service, you can control the ScoopTeam from a remote location, using the "My ScoopTeam RE" application.

What you need on the ScoopTeam:

- Up-to-date firmware (1.03 minimum).
- The "Remote Access +" option.
- Internet access, for example via its Ethernet connection to a LAN. However other IP connections are suitable as well: mobile data access, Wi-Fi…

The control device is a Windows PC, with the "My ScoopTeam RE" application installed.

You can download from AETA's web site the installation file for the application. Launch the installation file, and then open the instruction document that is installed along with the application. You can find there the instructions for using "My ScoopTeam RE".

Step 1: connect the ScoopTeam to the server

Menu **Tools > Remote Access**: activate "**Remote Access**".

If necessary, select the desired IP interface to be used for the remote access connection: "**Interface for Control**" (with the "Auto" choice, it will be the same that is used for audio over IP transmission).

Check the successful connection: menu **Status,** "**Remote Access Service".**

If you want to change the password, you must be logged as an administrator (*if necessary, switch this via the menu* **Tools > Access Level***:* ***Administrator***). In the menu **Tools > Remote Access** you can enter a password as you want, with following rules:

- The password must be **at least 8 characters** long. *For safety, it is recommended to use at least 13 characters, with a combination of uppercase and lowercase letters, numbers, and special characters among the following: _ - + * . : , ; # ? ! = .*
- If you enter a blank password, or one with less than 8 characters, a random password is automatically generated by the ScoopTeam.

*You may also use the "*REMOTE ACCESS*" html page (from the "*MAINTENANCE*" tab) for activating this connection, and check the successful connection on the* STATUS *tab,* GENERAL *section.*

Step 2: start the remote session with "My ScoopTeam RE"

Set up the virtual connection between the PC and the ScoopTeam; you need the serial number of the ScoopTeam and its "codec password". *You can find the details for the operation in the instruction document that is installed along with "My ScoopTeam RE".*

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ScoopTeam - User manual

AETA AUDIO SYSTEMS

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