

SCOOPY+ S


Mono/stereo portable audio codec for real time transmission over ISDN / POTS / MOBILE / IP



User Manual

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1. Presentation – Getting started

The SCOOPY+ S codec allows the bidirectional transmission of one or two audio signals with bit rate reduction, over various transmission media: ISDN lines, PSTN telephone lines, IP protocol networks, mobile networks...

The standard version of the codec includes an Ethernet interface for IP transmission. The unit can be complemented with many options providing additional network interfaces, coding algorithms, etc.


One outstanding feature of AETA codecs in ISDN mode is the 5A System[®]: on receiving an incoming ISDN call, the unit can automatically detect the coding algorithm and parameters of the calling codec, and then adjust itself in a compatible configuration so that the connection succeeds regardless of the initial configuration and that of the remote unit.

In IP mode, the codec features the same ease of operation thanks to the use of the SIP and SDP protocols.

This chapter gives **basic instructions** for a quick start. It obviously does not provide all the information for full control. For comprehensive information one can refer to the rest of this manual:

- Chapter 2 describes all the **functions** and features of the SCOOPY+ S (but not necessarily with all the operating modes).
- Chapter 3 gives a **physical description** of the unit, shows its **setting up** and **operation principles**.
- Chapter 4 details **menus** and **operating modes**.
- Chapter 5 deals with using the **html server** embedded in SCOOPY+ S.
- Chapter 6 shows how to use the **remote access service** via Internet.
- Chapter 7 provides all the **technical characteristics** of the SCOOPY+ S.
- 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 ● in this table are available as options.





 *Note: this document is relevant for units with firmware version 1.10. Previous versions have differences in the features and the user interfaces.*

[®] 5AS = Aeta Audio Advanced Automatic Adjustment System


Characteristics	Optional
Operation modes Single wide band codec Single codec and coordination using a mobile voice link (with HD-4G option)	●
IP transmission interface Ethernet Interface, 10BaseT / 100BaseT; TCP/IP, UDP/IP, RTP protocols Audio transmission in unicast mode: SIP signalling protocol, SDP, RTP streaming Audio transmission in multicast mode: RTP streaming Net bit rate 16 to 256 kbit/s (depending on coding algorithm, linear coding excluded)	
ISDN transmission interface One S0 interface, single codec 64 or 128kbit/s SAS auto configuration on incoming calls	●
Transmission interface on PSTN telephone line One "2 wire" telephone interface "POTS codec" mode with integrated V34 modem and CELP coding, 12 to 24 kbit/s Telephone hybrid mode	●
Mobile network access Integrated 2G/3G/3G+/LTE network access module, double internal antenna and antenna socket Voice mode: standard telephone or "HD Voice" (7 kHz with AMR-WB) Packet data mode: IP protocol, SIP signalling, SDP, RTP streaming, net bit rate 16 to 256 kbit/s (depending on coding algorithm) External 3G/LTE module connection via USB socket (data mode only) SMS reception	●
Audio coding algorithms (audio modes) G711 (standard telephone) Mono GSM, AMR (mobile telephone) Mono AMR-WB / G722.2 (mobile "HD Voice") Mono G722 SRT, H221, H242 Mono CELP 7 kHz Mono MPEG Audio Layer II Mono, Stereo, Dual mono, Joint stereo MPEG AAC-LC , HE-AAC, HE-AAC v2 Mono, Stereo 4 sub-band ADPCM (low latency) Mono, Stereo OPUS (AoIP only) Mono, Stereo Linear L16, L24 Mono, Stereo TDAC (ISDN mode only) Mono	●
Audio interfaces 3 analog MIC/line inputs and two analog outputs with adjustable gain Level display for encoder inputs and decoder outputs 2 stereo headphone sockets for monitoring, balanced send/receive	
Auxiliary functions (available depending on transmission interface) Relay management : 2 inputs and 2 outputs Audio coordination channel with Mobile option (up to 7kHz) SMS reception via optional mobile access	●
Control and supervision Keyboard and LCD display on front panel Programmable set-up/dial memories Ethernet/IP remote control, embedded html server Control and status GPIO Remote access via Internet Secondary Ethernet interface for remote control	● ●
Recording	●

Table 1 – Main features of the SCOOPY+ S





1.1. Install and connect SCOOPY+ S

- Plug on a power source or insert 6 C-size batteries.
- Switch on SCOOPY+ S by holding down  for 3 seconds. (And hold down  for 3 seconds to switch off)
- Connect the necessary audio interfaces (*details: page 20*)
- Connect SCOOPY+ S on the transmission network (*details: page 38 and following*)
- Using the SCOOPY+ S menus: **Esc** key or  key to activate the main menu, use the joystick to select a sub-menu or a parameter, enter or validate with the  key. **Esc** jumps back up to the previous menu level, until getting back to the base screen (with level display). (*details: page 47*)






1.2. Audio settings

- The microphone(s) to be used must be active ("ON" LED); for instance press  for microphone 1 if not yet done.
- Set the microphone gain to the middle position, white dot up. Make a test with the microphone. If the level is too strong or too weak on the meters, adjust the gain via the menu **Setup / Audio / Inputs / Input x Settings / Gain**, so that you reach an adequate level (*see details further on page 62*).

1.3. Select and set up network to be used: wired networks

- Select network: **Setup** menu then **Network**, then select **Change Network**, . In the proposed choice, select the desired network (**ISDN, Ethernet, POTS**). Validate with .
- Select the audio coding: menu **Setup / Algorithm**, then **Other**, then . Browse the available choice with the arrows, and make a selection with . Restart the same procedure to change for another coding setup.
ⓘ The available choice depends on the transmission network! For more details on coding, see page 15.

1.3.1. Set up an ISDN link

- If needed, select the protocol with **Setup / Network / ISDN Parameter / Protocol**. *More details: see page 57.*
- Enter the remote number to dial, using the keypad, and press the  key.
- If more than one B channel is involved due to the coding algorithm used, you must enter a second number, then . If the last dialled number is adequate, just confirm by pressing  without typing a number again.
- Hang up with the  key to release the connection (you must confirm by pressing  again).




1.3.2. Set up a (wired) IP link






For a link over a public IP network via an access router with NAT¹, it is recommended to use a STUN server.

The address of a STUN server can be set in the SCOOPY+ S html pages (*see page 95*) or via the menu: **Setup / Network / AoIP Parameter / STUN Server**, enter the address of a STUN server (we propose our server stun.aeta-audio.com, look also the support pages on our web site www.aeta-audio.com). Enable or disable STUN with **Setup / Network / AoIP Parameter / STUN Mode (On or Off)**.

More details: see page 122.

- Check the Ethernet interface is active thanks to the LED on the Ethernet socket on the back, and check an IP address is allocated: menu **Tools / About / Local IP**.
- The default setting uses a DHCP server to get an IP address, which is suitable for most occasions. *In other situations, look for more details on page 42.*
- Using the keypad, enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used), then press the  key.
- Hang up with the  key (you must confirm by pressing  again).
- *When using a SIP server, some data must be entered beforehand using the **AoIP Parameter** menu; for more details, refer to page 54. Similarly, some preliminary settings may have to be done for the "Direct RTP" mode (without the SIP protocol).*





1.3.3. Set up a POTS link

- For connecting the line, you must plug the POTS line in ANALOG socket.
- If needed, adjust the POTS line parameters using **Setup / Network / POTS Parameter**. *Details on these settings: see page 58.*
- Enter the remote number to dial, using the keypad, and press the  key.
- Hang up with the  key to release the connection (you must confirm by pressing  again).

¹ Network Address Translation, which is performed in most cases by the access router.




1.4. Select and set up network to be used: mobile networks

To set links over a mobile network, you must have a SIM card with a subscription suitable for the use. Specifically, for an IP mode transmission the subscription must include access to data transmission, and RTP audio streams must be allowed.

- While the unit is switched off, insert the SIM card into the drawer on the rear side of the SCOOPY+ S.
- Switch on the SCOOPY+ S (depress  a few seconds).
- Select the network: **Setup** menu then **Network**, then select **Change Network**, . In the proposed choice, select the desired network (**Mobile (Int.)**). Validate with .
- Enter the menu **Setup / Network / Mobile Parameters / PIN**. Enter the PIN code for the SIM card using the keypad then .

1.4.1. Set up a link in voice mode

This mode allows communicating with any telephone terminal through the regular telephone service. It also allows to benefit from the 7 kHz wide band service known as "HD Voice" whenever the remote terminal is compatible and the network supports the service.

- In the menu **Setup / Network / Mobile Parameters**, select **Mode / Cellphone**
- Afterwards, go to the menu **Setup / Network / Mobile Parameters**. In the sub-menu **Preferred Techno**, if needed you can select a priority among the 2G, 3G and 4G networks. The normal choice is **Auto**.
- The **Network Select** menu enables you to select among the available operators, if your mobile subscription entitles you to do so.
- Come back to the main menu. Enter the remote number to dial, using the keypad, and press the  key.
- Hang up with the  key to release the connection (you must confirm by pressing  again).

1.4.2. Set up a (mobile) IP link



For a link over a public IP network via an access router with NAT¹, it is recommended to use a STUN server.

The address of a STUN server can be set in the SCOOPY+ S html pages (see page 95) or via the menu: **Setup / Network / AoIP Parameters / STUN Server**, enter the address of a STUN server (we propose our server stun.aeta-audio.com, look also the support pages on our web site www.aeta-audio.com). Enable or disable STUN with **Setup / Network / AoIP Parameter / STUN Mode (On or Off)**.

More details: see page 122.

- In the menu **Setup / Network / Mobile Parameters**, select **Mode / IP Mode**
- Come back to the menu **Setup / Network / Mobile Parameters**. In the sub-menu **Preferred Techno**, if needed you can force a priority for 3G or 4G networks. The standard setting is **Auto**.
- Still in the **Setup / Network / Mobile Parameters** menu, enter the sub-menu **Access Point / APN**: enter the operator's APN code, using the keypad, then press **OK**.
- Come back to the previous level (**Setup / Network / Mobile Parameters**) by pressing **Esc**. Go to the **Data Active** line and press **OK** to activate the mobile data (the box is then checked)².
- Come back to the home screen using the **Esc** key.
- Select the audio coding: menu **Setup / Algorithm**, then **Other**, then **OK**. Browse the available choice with the arrows, and make a selection with **OK**. Restart the same procedure to change for another coding setup.
- Using the keypad, enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used), then press the **Call** key.
- Hang up with the **End Call** key (you must confirm by pressing **End Call** again).
- *When using a SIP server, some data must be entered beforehand using the **AoIP Parameters** menu; for more details, refer to page 54. Similarly, some preliminary settings may have to be done for the "Direct RTP" mode (without the SIP protocol).*

i Note: the sequence is much simpler for further calls as long as you keep the same SIM card, because the network related settings are kept memorized even if the unit is switched off; so you don't need to make these settings again (IP Data mode, APN, data activation...). It is even possible to memorize the PIN code: check the box in **Setup / Network / Mobile Parameters / PIN Save**.

¹ Network Address Translation, which is performed in most cases by the access router.

² Like on a smartphone, this data activation stays memorized, and you need not do this again each time the unit starts. Conversely, make sure to disable it if later you don't want any data traffic.

2. Functions

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

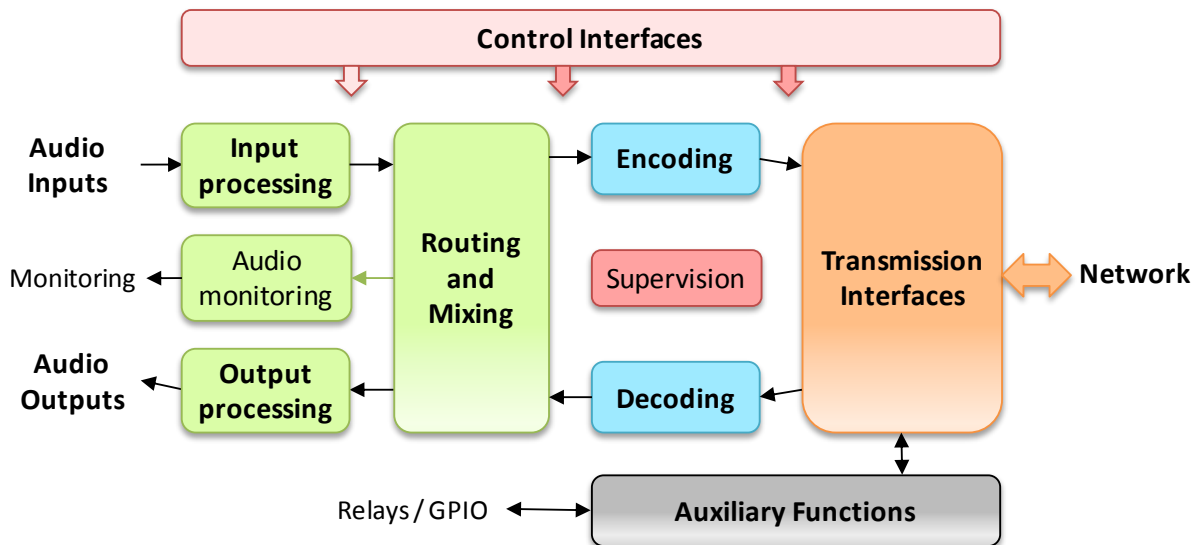


Figure 1 – Functional diagram of the equipment

The audio signals to be transmitted are mixed and digitized, then the encoding function reduces the bit rate, using a selectable algorithm; the resulting bit flow is sent to one of the available transmission interfaces: Ethernet interface, ISDN interface, PSTN interface, mobile network...

The transmission interface functional block also extracts compressed data coming from the network and sends them to a decoding block that reproduces uncompressed audio data. Last, the audio signals are fed to the outputs.

Monitoring the audio interfaces is possible thanks to headphones and level meters for the inputs/outputs.

In addition to the main task of transmitting an audio programme, the SCOOPY+ S can also transmit auxiliary information, usually by embedding them inside the transmitted audio streams.

Supervision and controlling the unit is performed using various remote control interfaces, and of course by means of the displays and controls on the front panel.

2.1. Transmission interfaces

The SCOOPY+ S features in all versions an Ethernet interface for IP protocol networks.

An optional ISDN interface can be added, as well as optional PSTN/POTS and mobile network access interfaces.

2.1.1. Ethernet/IP interface

The IP interface is a 10BaseT/100BaseT Ethernet interface allowing transmission of the audio programmes in a wide range of possible bit rates. The audio stream is always transported under the RTP/UDP protocol.

AoIP unicast mode

The most classical transmission mode is unicast: audio connection with one remote device, generally bidirectional. This mode can be used on all types of networks links, LAN or WAN, including links via Internet. The SCOOPY+ S implements the SIP protocol, which allows it to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Links can be set up in two ways:

- "Peer to peer" connection between two compatible units
- Use of a SIP proxy server to set up the link, or a SIP PBX

Details about the SIP protocols can be found in the annex (see 8.2, Overview of the SIP protocol).

It is also possible to operate in a "Direct RTP" mode, without using the SIP protocol. 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.

Available audio coding algorithms

The audio coding algorithm can be selected depending on the required quality and the available network bandwidth. The following algorithms are currently available:

Codec	Bit rate (coding)	Bit rate (total) ¹	Audio bandwidth	Typical use, main features
G711	64 kbit/s	86 kbit/s	3 kHz	Voice, telephony Compatible with IP phones
CELP	24 kbit/s	28.5 kbit/s	7 kHz	Suitable for high quality speech; Low network bandwidth consumption
G722	64 kbit/s	86 kbit/s	7 kHz	High quality speech. Compatible with some IP phones.
MPEG Layer II	64 to 256 kbit/s	73 to 275 kbit/s	Up to 20 kHz	Highest quality, suitable for speech and music
MPEG AAC-LC	16 to 256 kbit/s	30 to 277 kbit/s	Up to 20 kHz	Low bit rate, suitable for speech and music
MPEG HE-AAC and HE-AAC v2	16 to 128 kbit/s	23 to 139 kbit/s	Up to 20 kHz	Very low bit rate, suitable for speech and music
4SB ADPCM	128 or 256 kbit/s	173 or 301 kbit/s	15 kHz	Low latency, suitable for speech and music
OPUS	12 to 256 kbit/s	28 to 272 kbit/s	Up to 20 kHz	Low bit rate and low latency, suitable for speech and music
Linear L16/L20/L24	512 to 2304 kbit/s	592 to 2384 kbit/s	Up to 20 kHz	Best quality, very low latency

Table 2 – Overview of algorithms available in IP mode

IP multicast mode

The multicast mode allows an encoder device to transmit an audio programme to several decoders by sending a single encoded stream to a multicast group address. The link is unidirectional by nature. This mode can be used on a local area network, and on larger private networks that can manage the multicast mode. On the other hand, Internet cannot support this routing mode.

In this mode, the SCOOPY+ S uses the RTP protocol to manage the audio stream, like in the unicast mode, but the SIP protocol is not applicable here; instead a proprietary signalling system is used. As the link is unidirectional, the unit has to be set either as a "sender" in order to encode and transmit the audio stream to the selected group address, or as a "receiver" to receive and decode such stream coming from a "sender" device.

The audio coding algorithm can be selected with just the same capability as for the unicast mode described above.

SIP and SDP protocols

The SIP protocol is a signalling protocol, used for IP connections, which allows the SCOOPY+ S to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Details about the SIP protocols can be found in the annex (refer to 8.2, Overview of the SIP protocol).

¹ Informative value; higher than the "net" encoded audio bit rate because of the protocol overhead

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.

i *Note: the SIP protocol does not mandatorily 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.

"Factory" SIP account

AETA has set up a public SIP server, dedicated to broadcast audio over IP applications. Our customers can subscribe accounts and register their codecs on this server.

In addition, each SCOOPY+ S is provided a "factory set" SIP account on this server. This account is permanently and definitively bound to the unit, and at any time it is possible to load its parameters and register the SCOOPY+ S on the server. However, you can also configure another account from any other SIP server.

Packet replication

SCOOPY+ S 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.

Remote control via IP

The Ethernet interface can also be used for configuring or remote controlling the unit.

SCOOPY+ S provides html pages which provide complete control over the unit using a web browser, via port 80 (default port for HTTP protocol), or port 443 (default port for HTTPS protocol). See in chapter 5 the detailed operating mode.

2.1.2. ISDN interface

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

The codec synchronises itself onto the ISDN network clock when a link is active.

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".

i *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 SCOOPY+ S. 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, stereo...), 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 programme. 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 active for ISDN connections.

J52

The ITU-T J52 recommendation was defined in order to allow the interoperability of multimedia terminals over the ISDN¹, using common coding standards. It includes the following features:

- Support of ITU-T recommended coding algorithms: G711, G722, MPEG Layer II
- Framing as per ITU-T H221 recommendation, ensuring byte synchronisation and interchannel synchronisation when more than one 64 kbit/s B channel is required for the desired bit rate;
- Interoperation procedures according to ITU-T H242 recommendation;
- In the case of MPEG encoding, optional protection against transmission errors (Reed-Solomon error correction codes).

Details about MPEG and J52 can be found in the annexes (refer to 8.1, Additional information on the algorithms and protocols used).

¹ J52 is only relevant for ISDN connections

It must be noted that, thanks to the interoperation protocol, J52 codecs, when setting up a link, can negotiate automatically and agree on a configuration that is compatible with the capability of both units (regarding bit rate, channel mode, etc.). In this way, when the units differ in their capability (or make), the resulting configuration may be different from expected beforehand, but in most cases the link will work and audio will be transmitted.

As another useful consequence, this also gives users more tolerance to mistakes when configuring the units on the two sides of the transmission links, as the codecs will adapt automatically even with differences in the initial settings of the two units.

2.1.3. POTS interface

The interface is a "two wire" analog telephone access, with characteristics adjustable depending on the country. Dialling normally uses DTMF, but for older switching equipment it is possible to use pulse dialling.

SCOOPY+ S includes a V34 modem which transmits via this line a bidirectional audio flow, encoded at a nominal 24 kbit/s bit rate. Depending on the line quality and the quality of the link with the remote codec, this bit rate is automatically negotiated and dynamically adjusted from 12 to 24 kbit/s.

A "protected" mode can be activated, which increases the resilience to transmission errors, at the cost of a higher latency (encoder to decoder delay). *You must make sure to set this parameter the same way on both devices / both ends of the link.*

2.1.4. Mobile network access

Units equipped with the "HD-4G" option include an integrated module for access to mobile networks, and a holder for a SIM card.

Depending on the version the accessible networks are 2G (GSM, EDGE), 3G (UMTS), 3G+ (HSDPA, HSUPA, HPA...), and 4G/LTE.

The SCOOPY+ S includes an integrated double multiband antenna. This antenna "diversity" feature is useful when the radio reception is poor: the signal strength and stability is improved thanks to the second antenna. This can specially increase the radio coverage area in rural areas.

An external multiband antenna (to be selected for compliance with the mobile network characteristics) can be connected to the SCOOPY+ S as well. It will in such case replace one of the internal antennas. Such arrangement takes a greater advantage of diversity, as the two antennas involved are set at a larger distance from one another.

Lastly, SCOOPY+ S can display the received SMS messages.

Mobile voice mode – HD Voice

The integrated module allows using 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.

No special subscription, other than to the regular telephone service, is needed, but for most operators only the 3G/3G+ base stations support the service.

i *This sometimes makes people believe that HD Voice is related to the mobile IP service, but this is definitely not the case.*

More and more mobile phones now support this service, especially (but not only, and not all) smartphones. All AETA codecs in "Wireless" version support HD Voice, namely:

- Scoop 4+ in "wireless" version
- Scoopy+ HD (*except old units, in doubt consult us*)
- ScoopFone HD, ScoopFone HD-R.
- Scoop 5 and Scoopy+ with the HD-4G option
- ScoopFone 4G and ScoopFone 4G-R
- Scoop5 S with the HD-4G option
- ScoopTeam with the HD-4G option

Mobile IP mode


The other 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 2.1.1, 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 (3G/3G+/LTE...), 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.
- The multicast mode is not available on mobile networks
- Setting a link implies first activating the data connection, before actually initiating an audio stream transmission link

Using an external USB device

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:*

- This capability is optional.
- The USB module must be from the list of devices supported by AETA. *As this list is evolving, please check our web site for up to date information.*
- The "HD Voice" mode is not available in this way.
- USB devices do not feature antenna diversity.

2.1.5. Managing calls

The audio transmission implies a link/session setup phase.

One of the transmission interfaces is selected as the *default interface* on the SCOOPY+ S.

A call towards a remote unit, initiated by the user of the SCOOPY+ S, is implicitly sent through this default interface.

On the other hand, an incoming call on any interface (regardless of the default interface) can be processed and the link established, under following conditions:

- The "called" interface must be connected and active. As an example, for mobile IP the data connection must be active.
- The codec must not be already busy with another connection.

If the call comes on an interface other than the default interface, the codec 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).

2.2. Audio encoding and decoding

SCOOPY+ S features a wide range of coding algorithms. Their availability depends on the transmission network used. Besides, the MPEG family algorithms offer a great configuration flexibility.

The table below synthetically describes the capabilities with the various transmission media:

Codec	Audio channels	Frequency (kHz)				Net bit rate (kbit/s)	Wired			Mobile	
		16	24	32	48		PSTN	ISDN	Ethernet	3G/3G+ LTE	Voice
G711	Mono					64					
GSM, AMR	Mono					-					
AMR-WB	Mono					-					
CELP	Mono					12 => 21,6					
CELP	Mono					24					
G722	Mono					64					
G722-H221	Mono					64					
G722-H242	Mono					64					
TDAC	Mono					64					
4SB ADPCM	Mono					128					
4SB ADPCM	Stereo					256					
MPEG L2	M / S	[1]				64					
MPEG L2	M / S	[1]				128					
MPEG L2	M / S					192					
MPEG L2	Stereo					256					
AAC-LC	M / S					16 => 56					
AAC-LC	M / S					64					
AAC-LC	M / S					96					
AAC-LC	M / S					128					
AAC-LC	M / S					192					
AAC-LC	Stereo					256					
HE-AAC	M / S					16 => 56					
HE-AAC	M / S					64					
HE-AAC	Stereo					96					
HE-AAC	Stereo					128					
HE-AAC v2	Stereo					16 => 56					
HE-AAC v2	Stereo					64					
AAC-LC LOAS	M / S					64					
HE-AAC LOAS	M / S					64					
HE-AAC v2 LOAS	M / S					64					
OPUS	M / S					12 => 256					
Linear L16	Mono					512 / 768					
Linear L16	Stereo					1024 / 1536					
Linear L20	Mono					640 / 960					
Linear L20	Stereo					1280 / 1920					
Linear L24	Mono					768 / 1152					
Linear L24	Stereo					1536 / 2304					

Possible audio bandwidth:

 3 kHz

 7 kHz

 15 kHz

 20 kHz

[1] : only for ISDN

IP networks

Table 3 – Available coding depending on network

The following chapters bring some precisions about the important features of the various algorithms and protocols available.

2.2.1. 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.

G711 is available only for IP or ISDN transmission

2.2.2. Mobile telephony coding: GSM , AMR

Application: telephony, mobile coordination. Moderate latency.

These algorithms are exclusively used for speech transmission over mobile telephone networks, with a 300 to 3400 Hz audio bandwidth. Gateways perform, whenever needed, transcoding in order to interface with fixed PSTN, ISDN and IP networks.

2.2.3. Mobile HD Voice coding: AMR-WB

Application: commentaries, mobile coordination. Moderate latency.

The AMR-WB coding (standardised as G722.2 by the ITU-T) is used between compatible mobile terminals, when the mobile network supports the so called "HD Voice" service, and provides speech transmission with a 50-7000 Hz bandwidth.

SCOOPY+ S automatically implements this algorithm in mobile voice mode every time it is possible, and automatically falls back to standard voice coding if not (when network does not support, or the remote terminal is not compatible).

i *Unfortunately it is not possible to see directly whether AMR-WB is active or not at a given moment. You have to rely on your listening skills! However, it has to be active if the conditions are met: a) support from the network on both sides of the link, b) both terminals compatible, c) service continuity from end to end¹*

2.2.4. CELP coding

Application: commentaries, coordination. Low capacity transmission channels

This algorithm operates in mono for a net nominal bit rate of 24 kbit/s, and provides 7 kHz bandwidth and a quality close to G722 for a much lower bit rate.

CELP is available for IP or PSTN networks. For PSTN (POTS), the bit rate may be reduced to adapt for the line quality, among the following values: 12 kbit/s, 14.4 kbit/s, 16.8 kbit/s, 19.2 kbit/s, 21.6 kbit/s, 24 kbit/s. The bandwidth is reduced accordingly.

Still for POTS lines, in addition to the normal mode two "protected modes" are available, bringing an increased resistance to transmission problems, however with an increased latency as well.

¹ At the time of writing, this requires both units to be on the same network: same operator, same country

2.2.5. 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.

It is available, ISDN, IP networks (wired or mobile).

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.

H221 synchronisation is highly recommended when possible, as it features higher reliability and faster recovery time, while degradation (because of the bit rate used for framing) is minimal.

H242 protocol, the most flexible mode, is recommended by the ITU-T, and is included in J52. However, the mode with H221 synchronisation but without H242 protocol can be useful for compatibility with old generation codecs which did not use this protocol.

No specific synchronisation is required for the IP mode.

2.2.6. TDAC coding

Application: commentaries, mono music via ISDN with only one B channel.

The SCOOPY+ S can also include the TDAC algorithm. TDAC is for Time Domain Aliasing Cancellation; this is a transform coding based on an MDCT (*Modified Discrete Cosine Transform*), encoding a 15 kHz bandwidth mono signal at a 64 kbit/s bit rate.

TDAC is available as an option, only for the ISDN mode.

2.2.7. 4SB ADPCM coding

Application: commentaries, mono or stereo music. Low latency.

4SB ADPCM operates either in mono at a 128 kbit/s rate, or in stereo at 256 kbit/s, for a 15 kHz bandwidth. It features a very low latency which makes it very interesting for live duplex transmission. It also has the advantage to be very little sensitive to tandem coding.

4SB ADPCM is available for ISDN (Mono only); wired or mobile IP. However it is not recommended for mobile networks, because it requires a high bit rate and it is highly sensitive to packet losses, which can be frequent on such networks.

2.2.8. MPEG Audio Layer 2 coding

Application: mono or stereo music, high quality.

As shown on Table 3, this coding algorithm features a maximum flexibility, with many variations for bit rate, mono or stereo channel mode, sampling rates...

The two channel modes exist in three variations:

- Stereo: coding of each channel stays independent
- Dual mono: coding is similar to the previous case, but this choice applies to channel with no acoustic relationship, e.g. two languages for commentaries
- Joint stereo: applies to stereo programme, but here the encoder exploits the interchannel correlation for improved coding. To be used only for a stereo programme

The 16 and 24 kHz sampling rates feature a moderate bandwidth (respectively 7 kHz and 10 kHz) and are rather useful for commentaries.

i *Latency is rather high with these sampling rates*

MPEG L2 is available for ISDN, wired or mobile IP networks.

ISDN mode specific aspects: J52

For ISDN transmission, MPEG L2 is proposed with two variations:

- "MPEG L2 J52" variation, using the ITU-T J52 protocol for the link negotiation and inverse multiplexing (B channel aggregation for connections requiring more than 64 kbit/s). Moreover, optional protection against transmission errors (FEC) is also available. For more details refer to 2.1.2, ISDN interface (page 11).
- MPEG L2" variation, without the J52 protocol, using a system for inverse multiplexing two B channels which is proprietary but compatible with several codecs on the market.

2.2.9. MPEG AAC algorithms

Application: mono or stereo music, low capacity transmission channels.

These algorithms feature a very high compression ratio, for a given audio quality, compared to Layer 2. They can operate at a sampling rate of 32 or 48 kHz, and several bit rates: 16, 20, 24, 32, 40, 48, 56, 64, 96, 128, 192, 256 kbit/s. Three coding variations are available:

- MPEG **AAC-LC** ("Low Complexity"): lower compression than other variations, but lower latency.
- MPEG **HE-AAC** ("High Efficiency" AAC): higher compression, and the bit rate is limited to 128 kbit/s for this variation.
- MPEG **HE-AAC v2** ("High Efficiency" AAC version 2): compared to the above, this coding further enhances the performance for a stereo program (not available for mono). The bit rate is limited to 128 kbit/s for this variation.

AAC codecs are available for ISDN, and wired or mobile IP networks.

2.2.10. OPUS coding

Application mono or stereo 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 SCOOPY+ S, the configuration stays very simple: the user just imposes the audio channel format (mono/stereo) and 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 SCOOPY+ S, 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.

You can adjust the bit rate during a link using the "CONNECTION PARAMETERS" frame of the "**CONNECTIONS**" web page, or using the joystick on the front panel.

Note: the quality indicators on the front panel or the "Status" page of the web interface provide information in real time about the transmission quality (for the transmit direction, this depends on the capability of the remote codec).

2.2.11. Linear coding

Application mono or stereo music, transmission media with very high capacity and reliability.

The linear coding is in fact the absence of coding/compression, the audio data are transmitted without information loss and with no bit rate reduction. Obviously this implies a very high bit rate; therefore linear coding is only usable over an IP network, and with a very high bandwidth and a very good quality of service, such as a local area network or a high reliability private WAN.

The coding features three variations L16, L20 and L24, corresponding respectively to a sample resolution of 16 bits, 20 bits, or 24 bits for the transmitted audio. With each variation it is possible to select a sampling frequency of 32 or 48 kHz, and a mono or stereo channel mode.

The linear coding is only available for IP networks.

i *It is not recommended on mobile IP networks due to the required bit rate and the poor quality of service expected from a mobile network.*

2.3. Audio interfaces and functions

The following diagram shows the main audio processing and mixing functions:

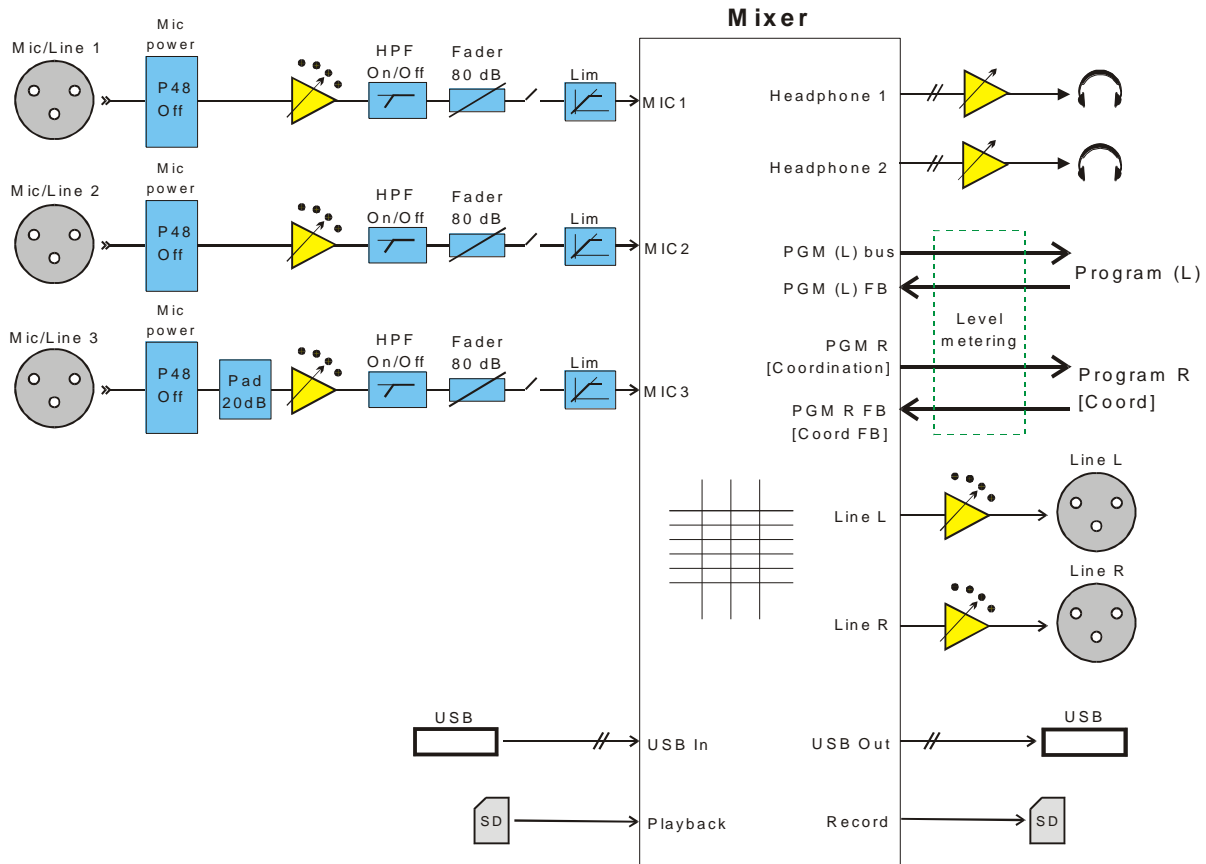


Figure 2 – Functional diagram of the audio sub-system

The SCOOPY+ S features two mixing buses, used for the left and right channels when the selected coding is in stereo mode.

With the "HD-4G Mobile" option, a mobile voice link can be used for coordination when the coding is on mono mode. In this case, the right "Program" mixing bus is used for coordination (as shown inside square brackets in the diagram above). *The coordination cannot be used along with stereo audio coding for the program channel.*

2.3.1. Mic/Line inputs

The SCOOPY+ S includes three microphone inputs, each with following features:

- Balanced input, 3-pin female XLR (located on the left and right sides)
- 48V phantom power (switchable)
- Input sensitivity selector (0 / 16 / 32 / 48 dB gain)
- 20 dB attenuator for high level line signals (pad on input 3 only)
- Switchable high-pass filter (50Hz, 18dB/o)
- Mixer fader, using a front panel potentiometer
- Switchable limiter (-8dBFS threshold).
- Mute switch located on the front panel

All settings, except the fader potentiometer, are controllable by remote.

In mono operation, each input can be routed to the main program and/or the possible coordination line. In stereo mode, the input can be routed to left, right, or both.

Input 3 can also be configured as "Sound Guide": in this case it is not included in the program mix (neither the coordination mix), but instead it is routed to the left ear of the headphone outputs, mixed with the other sources selected for monitoring.

2.3.2. Line outputs

The SCOOPY+ S has two line level outputs with the following characteristics:

- Balanced outputs, 3-pin male XLR (located on the right panel)
- Maximum level adjustable from -11 to +22 dBu

The signals on the outputs can be selected among:

- "Send": Left/Right mixing bus, or in mono mode Program/Coordination
- "Receive": remote signals received through the link (Left/Right, or Program/Coordination in mono mode)
- "Mix": send/receive mix (the balance is the same as for headphone output 2)

2.3.3. Headphone outputs: monitoring

The SCOOPY+ S has two outputs for listening on headphones, with following features:

- 6.35 mm (1/4") jack (on left and right sides)
- Volume setting with dedicated knobs
- Ability to monitor a mix of the "Send" and "Receive" signals; on each headphone the send/receive balance is adjusted thanks to a potentiometer on the front panel, combined with the volume knob.

When operating in mono with a coordination channel (mobile voice), various monitoring modes are available for listening to the program and/or the coordination. This configuration is detailed further (2.3.6, "Mixing and routing", and Table 4 – Operation of the monitoring matrix).

2.3.4. USB interface (device)

This interface (mini-B socket on the rear panel) is a USB device port which provides the capability to set a link with a Windows or macOS computer. The SCOOPY+ S is seen by the computer as a stereo sound card.

In the "Play" direction (output) from the computer, the two audio channels received by the SCOOPY+ S on its USB interface are injected into the Program mixing bus (see below, 2.3.6 and "Figure 3 – Main mixing and network sends").

In the direction towards the computer, the Program mixing bus is fed to the channels of the USB interface (see below, 2.3.6 and Figure 3).

i *The USB audio interface cannot be used along with the record/playback function to/from an SD card (see further).*

2.3.5. Recording/Playing audio files

This feature is optional on the SCOOPY+ S.

In mono operation, the SCOOPY+ S can record onto an SD card the mono "programme" signal. When transmitting, the Program feedback is also recorded on a second track. The recordings have following characteristics:

- Linear 16 bits coding, regardless of the transmission coding format.
- 48 kHz sampling frequency (except if the recording is started while transmitting and encoding at 32 kHz).
- WAV file format: mono for an "offline" recording (without a network link at the time the recording is started), 2-channel ("stereo") if the recording is started when a transmission is already active. This format allows post-processing the files on all types of editing software.

i *Recording is not possible when operating in stereo mode.*

A recording stored on the SD card can be played back by the SCOOPY+ S; in such case (in mono operation) it becomes one more source mixed on the "Program" bus. The two audio signals played back are injected on the mixing bus.

On demand, the SCOOPY+ S can send a recording to an FTP server.

i *The record/playback feature cannot be used along with the USB audio interface (see above).*

A simple editing function is available: on a given recording, you can set a "begin" marker and an "end" marker. This defines a virtual "take" that can be played like a "real" recording, but playing only the part between the markers.

- Several takes can be defined from the same recording.
- It is possible to convert a virtual take into a real WAV file; this file will then add to the library of recordings.

2.3.6. Mixing and routing

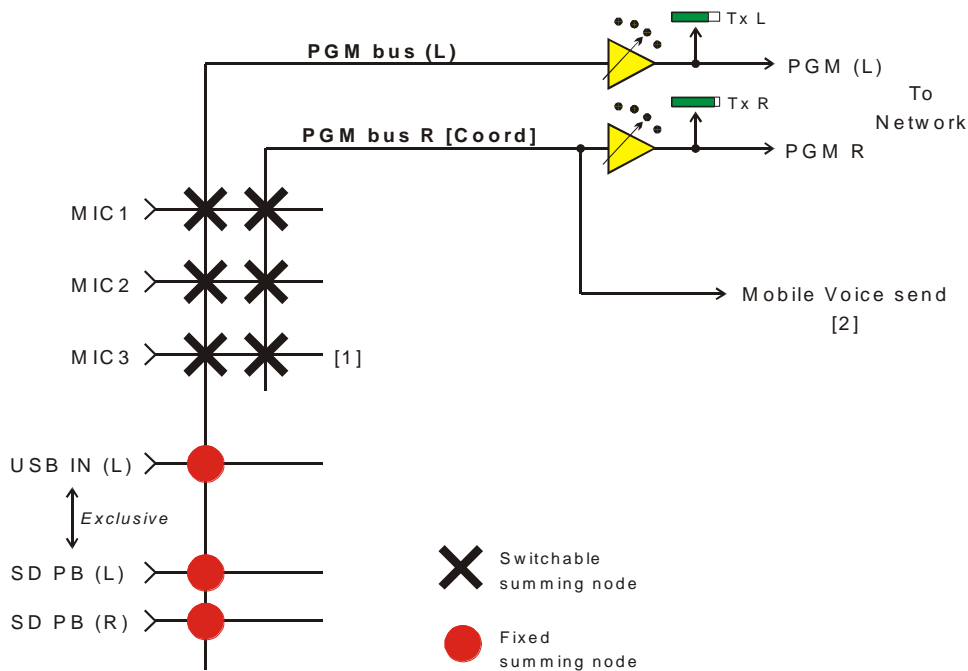
The audio sources are mixed for generating:

- The program and coordination send signals.
- The line outputs.
- The output to the USB device port or the signal recorded on the SD card.
- The headphone monitoring outputs.

The available sources are:

- The mic/line inputs (MIC1 to MIC3).
- The inputs from the USB device port or a file played from the SD card (SD PB).
- The Program and Coordination feedback signals received from the network link.

The SCOOPY+ S has two main mixing buses. The diagram below shows the available mixing features on each bus:



[1] Except if "Sound Guide" is activated on MIC3

[2] If "HD-4G Mobile" option is present

Figure 3 – Main mixing and network sends

The first mixing bus is used for the program send in the "standard" mono mode, and for the left program send in stereo mode. In stereo, the second mixing bus is assigned to the right program send. The "Tx L" and "Tx R" bargraphs on the display show the level of these send signals. A global gain adjustment (-12 to +12 dB) can be applied to these signals.

With the "HD-4G mobile" option, a mobile voice link can be used as a coordination channel when the program is mono. In such case, the second mixing bus is assigned to coordination, and the audio routing works in a "PGM & Coord" mode.

Coordination cannot be used when stereo coding is used for the main program.

Each microphone input can be routed to the left and/or right bus in stereo mode (left/right/center selection), otherwise, in the "PGM & Coord" mode, it can be routed to Program and/or Coordination.

i *Exception: microphone 3 can be switched as a "Sound Guide", in which case it is routed to the left ears of the headphones, but not routed to the send mixing buses.*

If a computer is connected to the USB device port, the audio signal received on this interface is injected on the Program mixing bus.

If a file is being played back from the SD card¹, the audio signal is injected on the Program mixing bus. If the file is stereo/2-channel, both channels are mixed on the bus.

i *The USB input and file playback are mutually exclusive: when connecting the USB interface, it is possible to confirm if you activate it, in which case the file playback is no more available.*

¹ Provided that the unit includes the "Recording" option

The diagram below shows the routing for the audio outputs:

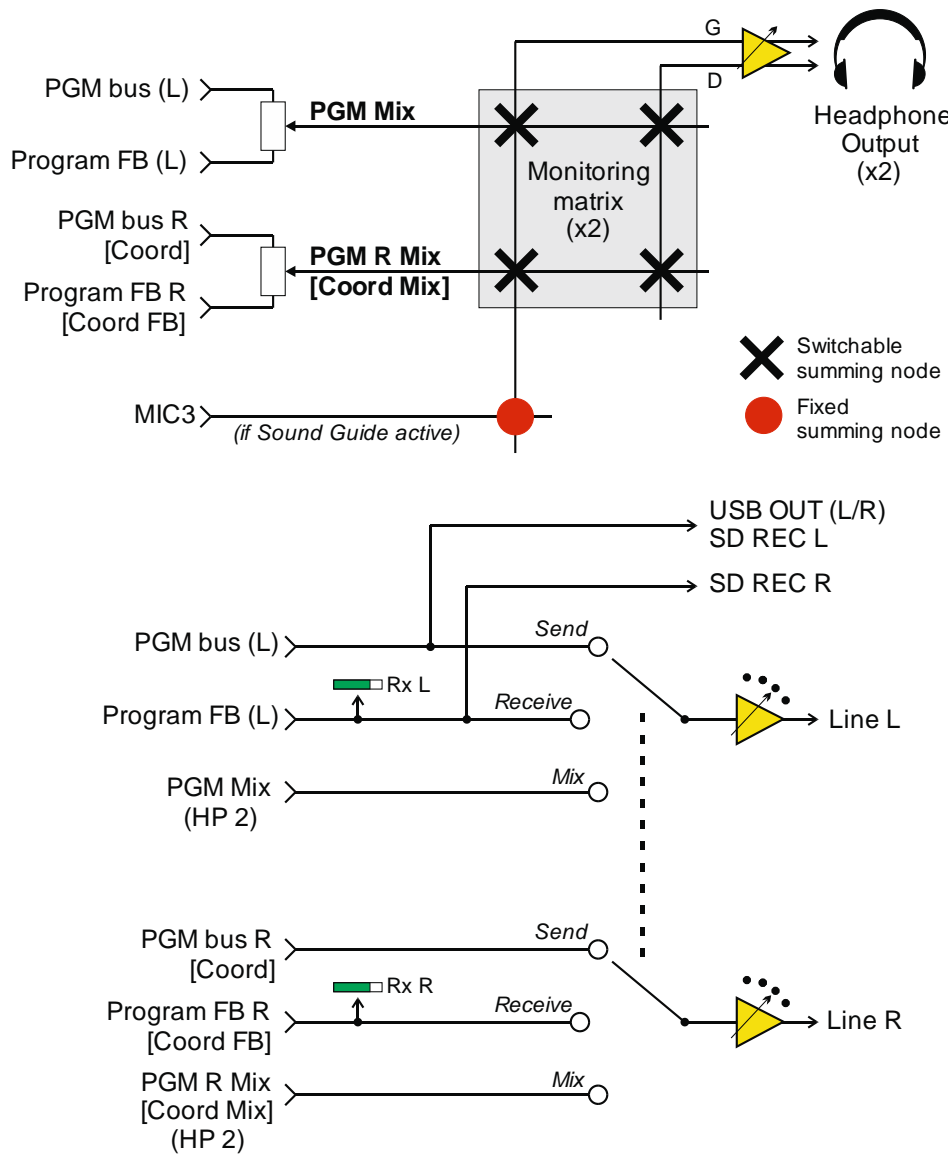


Figure 4 – Routing of the audio outputs

Each headphone output is fed a mix of send and receive signals, with an adjustment of the send/receive balance thanks to a potentiometer.

The line outputs can be fed the signals as sent to the network, or those received from the network, or a mix of both types, with the same balance as headphone 2.

A monitoring matrix routes the signals to the left and right ears depending on the operation mode (mono, stereo, Program&Coord...) and the configuration selected by the user for each headphone. This routing is described in the table below. Besides, the MIC3 signal is inserted in the left ear if it has been configured as "Sound Guide".

Operation Mode	Selection for the headphone output	Context	Headphone signals	
			Left	Right
"Standard" Mono	-		PGM	
Stereo	-		PGM L	PGM R
PGM & Coord.	Program		PGM	
	Coordination Line		Coord	
	PGM / Coord. Line		PGM	Coord
PGM & Coord. (auto)	Program		PGM	
	Coordination Line	Coordination line idle	PGM	
		Coordination line active	Coord	
	PGM / Coord. Line	Coordination line idle	PGM	
		Coordination line active	PGM	Coord
Main & Backup	Main		Main	
	Backup		Backup	
	Auto	No voice link (backup)	Main	
		Voice link active	Backup	

Table 4 – Operation of the monitoring matrix

In this table, the "selection for the headphone output" is the choice proposed in the menu, and which impacts the routing in the headphone. For better readability, the headphone signals are abbreviated "PGM" (for PGM Mix) and "Coord" (for Coord Mix). The various operation modes are determined by the "**Routing for Mono**" selection that can be found in the menu (and the "**Audio**" html page).

- **"Standard" Mono:** this is the default mode if using SCOOPY+ S in mono mode, and this is the only one if the unit is not equipped with the "HD-4G Mobile" option. No special configuration is necessary in this mode.
- **Stereo:** this mode is automatically active as soon as stereo encoding is selected. No special configuration is necessary for the headphone.
- **PGM & Coord:** used for adding a coordination link via a mobile voice call. It is possible to select one signal or the other, or one on each ear.
- **PGM & Coord (Auto):** a variation of the previous mode, in which the routing effectively changes (compared to the base "Program") only when a coordination link is active (for instance, when a mobile voice call is received for coordination).
- **Main & Backup:** in this special mode, the mobile voice link is used as a backup, normally used at the same time with the "Main" link. In the studio, as receiving from the two links, it is possible to switch over to "Backup" in case the "Main" link is lost or defective. Various automations are provided in this mode:
 1. The microphones are routed to both links in the send direction.
 2. On the headphones, it is possible to select the "Main" or "Backup" feedback.
 3. In the "Auto" case, the feedback listening is in priority on "Backup", with automatic foldback to "Main" when the mobile link is stopped. *Advantage of this option: although the mobile voice is of lower quality, it always has less latency for the feedback and then provides more fluency in duplex remotes.*

2.3.7. Audio level monitoring

Audio peak level indicators (bargraphs) are available on the graphical display:

- Left (and possibly right) level of the Program send (Tx L and Tx R bars).
- Level of feedback signals (Rx L and Rx R bars): left of the, and right level of Program feedback or Coordination feedback (depending on current operation).

The reference level of the bargraphs (default -12 dBFS) is adjustable in the "**Audio**" menu ("Headroom" setting).

2.4. Auxiliary transmission functions

The main function of the SCOOPY+ S is the transmission of one or two audio channels, but it also provides auxiliary functions for transmitting additional data, inside the same stream (or, more generally, the same session).

- Exchange of binary data with the remote codec during transmission: sending and receiving two Boolean states "INFO1" and "INFO2". On the receiving direction these states are shown on the front panel, and can close relay contacts (GPO), found on the "EXTENSION" connector of the SCOOPY+ S. The transmitted states can be set by GPI contacts on this same connector, or alternatively by keys on the front panel. A typical application is the transmission of an "On Air" signal. See operation details in 3.2.1 and 4.4.7. *This feature is not available through a mobile voice link.*
- Exchange of additional "INFOn" states, up to 4 signals. Using these is possible if complementing the SCOOPY+ S with a GPIO extension box, available as an accessory. This box provides 4 GPIO isolated inputs and outputs. See operation details in 4.4.7. *Only an IP network (Ethernet or mobile) can support such additional transmission.*
- Receiving SMS from the mobile network, available when the "HD-4G Mobile" option is installed and as soon as the unit is registered on a mobile network. The PIN code of the SIM card must be entered beforehand. When receiving an SMS (sent to the SIM card's subscriber number), the message is displayed directly on the front panel, and it can be read as well from a page of the embedded html server of the unit. This function is independent from the transmission link(s).
- "In-band" exchange of information ("RC data" in the table below): this function allows a remote unit to remote control the SCOOPY+ S via a running ISDN link. This control mode uses the dedicated AARC protocol. *This additional transmission exists only over ISDN and excludes INFO1/INFO2 exchanges).*

i *These various features are only compatible with AETA products, because they are not inside the scope of independent standards. Besides, for using them they must be activated on the remote unit as well.*

The availability of these functions depends mainly on the transmission network. The following table summarizes the capabilities.

Network	Transmission		SMS	RC data
	INFO1 / INFO2	INFO3 to INFO6		
ISDN				
IP				
POTS				
Mobile voice				

Table 5 – Auxiliary functions depending on the network used

Besides, over ISDN links auxiliary data transmission is only available for the following coding algorithms:

- G722-H242
- MPEG Layer 2 with J52
- 4SB ADPCM

2.5. Supervision and control interface

The control and supervision of the equipment (configuration, communication management, status monitoring), is carried out either "locally" thanks to the front panel interface, or using various remote control interfaces.

2.5.1. "Local" control

For local management, the front panel includes a large keypad, a graphic OLED, and various LED indicators for essential status information.

2.5.2. Embedded html server: "web pages"

SCOOPY+ S provides html pages that enable full control using a web browser, via port 80 (default port for the HTTP protocol), or port 443 (default port for the HTTPS protocol). See on chapter 5 the detailed operation mode for these pages.

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. With the "Remote Access" option, described further in 2.5.4, this control mode is also available across the Internet.

2.5.3. Remote control interfaces

Remote controlling the SCOOPY+ S is possible through various interfaces:

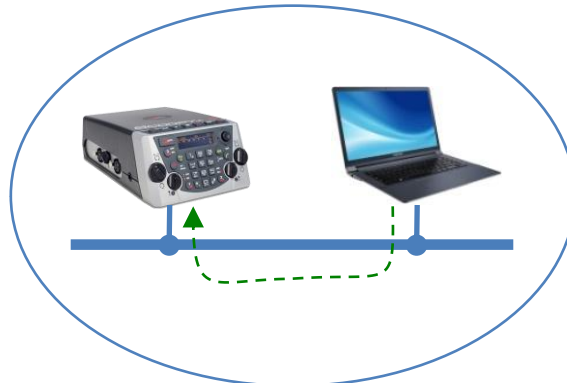
- The Ethernet/IP interface, used for audio over IP transmission (AoIP), can be used for accessing the html pages as well.
- It is possible to get a second Ethernet/IP interface by plugging a USB/Ethernet adapter¹ on the USB host socket.
- A data link via a mobile network can be used, but this requires the remote access feature to be used (see below, 2.5.4).

¹ Reference : Icy Box IB-AC509, or other device shown on our web site support pages.

2.5.4. Remote access

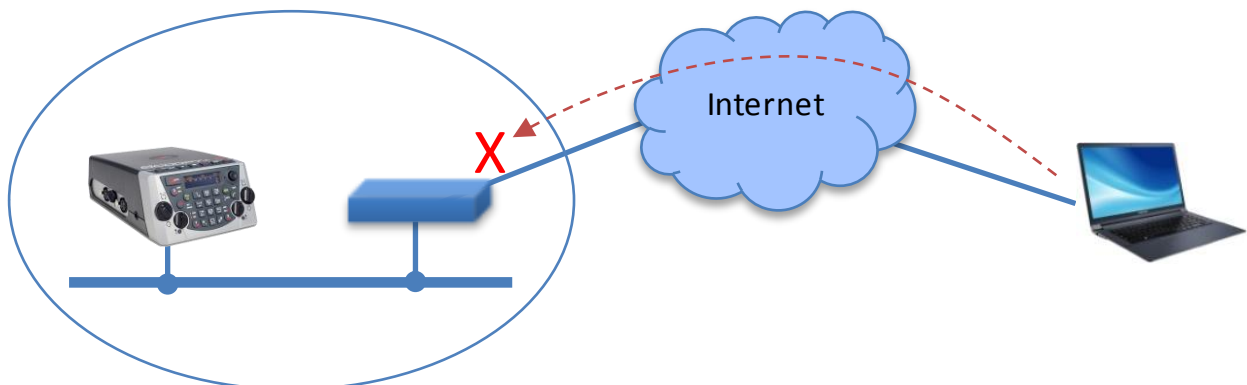
Background

A computer connected on the same LAN as the SCOOPY+ S can easily take control over the unit:



The computer can access the html pages 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):



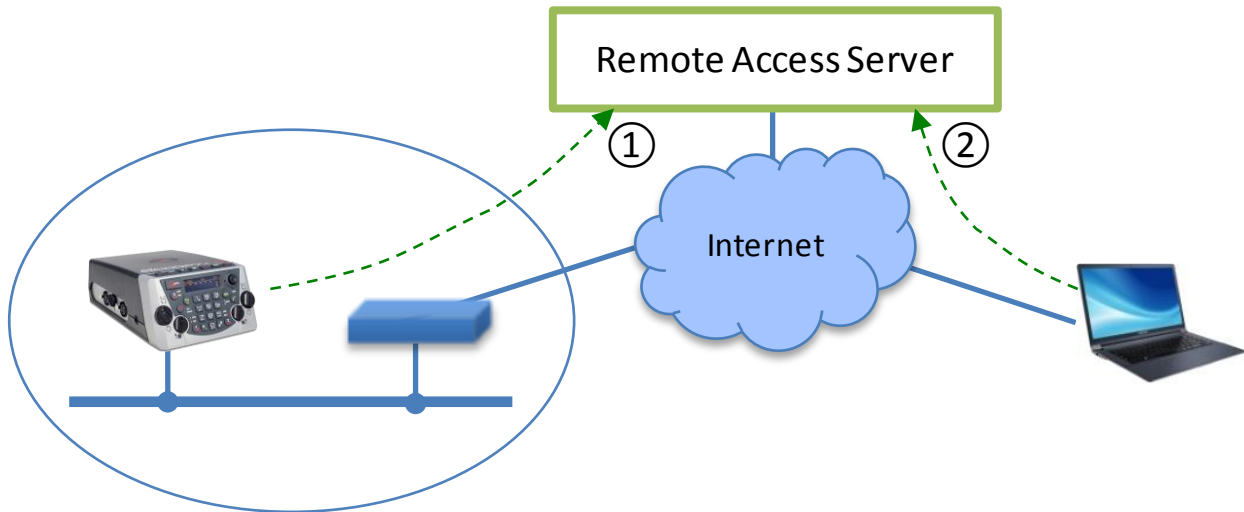
In such situation, it is not possible to get a connection as directly as over a LAN.

The AETA Remote Access service is designed for working around this obstacle and enabling to take control over a device even in such case¹.

¹ However, a very restrictive firewall can of course block the service

"Remote Access" service

The system relies on using a remote access server, as an intermediary between the SCOOPY+ S and the control device. A control session is performed in two phases:



1. The SCOOPY+ S connects to the server, and gets available for a possible remote access session.
2. A user who wants to remote control the SCOOPY+ S 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 SCOOPY+ S is connected to the network and when remote access is active on the unit. However this can be disabled by the user.

Two 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 SCOOPY+ S for using this service.

When the unit has more than one IP interface available, it is possible to use for remote access an interface different from the one used for AoIP transmission.

2.5.5. GPIO

SCOOPY+ S provides on its "EXTENSION" socket two isolated inputs (optocouplers) or "contacts", and two isolated "relay" outputs. Two categories of functions can be assigned to these GPIO:

- Use for exchanging "INFO" boolean data with a remote device (see above 2.4, "Auxiliary transmission functions"). A GPI can be set for driving such transmitted data, and a GPO can be set for showing the state of a received "INFO".
- Specific action triggering (for instance releasing a connection) for a GPI, or status indication for a GPO (for example "on line").

If a GPIO extension box is used (see 7.8, Accessories and related products), 4 GPI and 4 GPO are available in addition, complementing the available features.

The following table shows the functions that can be assigned to the GPIO:

Interface	Available functions	Description
GPI 1	Disabled	
	Info 1	Sets the state of Info 1 sent to the remote device
	Redial	Redial the last number called on the Program line
	Release Call	Release the Program line
GPI 2	Disabled	
	Info 2	Sets the state of Info 2 sent to the remote device
	Redial	Redial the last number called on the Program line
	Release Call	Release the Program line
GPO 1	Disabled	
	Info 1	Show the state of Info 1 received from the remote device
	Line status	Relay closed when the Program link is established
	Sync status	Program link established and decoder synchronized
GPO 2	Disabled	
	Info 2	Show the state of Info 2 received from the remote device
	Line status	Relay closed when the Program link is established
	Sync status	Program link established and decoder synchronized

Table 6 – Functions assignable to the GPIO

The GPIO possibly added with the extension box are configurable with similar principles:

- Each GPI input can be assigned: Redial, Release, or send INFO.
- Each GPO can be assigned: Line status, Sync status, or received INFO.

2.5.6. Configuration and dialling memories

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

- "Call profiles", including the parameters for calling a given destination: dial numbers, coding algorithm, etc. A profile is similar to a phone directory item, but in addition it can also hold coding parameters. Recalling such profile directly configures the codec and/or triggers an outgoing call with the parameters that have been previously stored by the user into this 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.
- "Snapshots" which memorise all the settings for the audio interfaces.

These various profiles can be used locally and also through the web pages, and they can be imported/exported from/to a computer.

2.5.7. Test functions

For maintenance purposes, some test loops can be activated. The following drawing schematically shows these test loops:

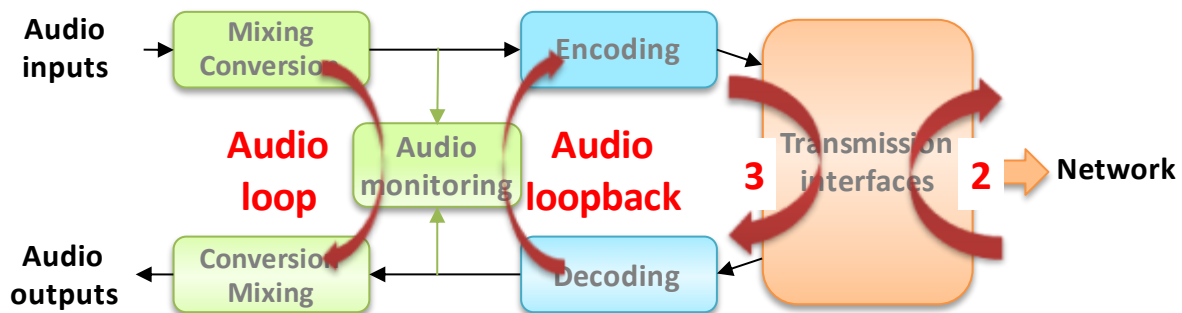


Figure 5 – Test loops

- "Audio loop": uncompressed audio data are looped from the input of the encoder to the input of the output conversion functional block. This loop redirects the audio input to the audio outputs;
- "Loop 3", or "Codec" loop: compressed audio data are looped just before the network interface;
Note: this test loop is not available for every network interface.
- "Loop 2", or "Network" loop: this loop sends the received data back to the network; for the remote codec, the effect is the same as a loop 3 when the transmission works correctly;
Note: this test loop is not available for every network interface.
- "Audio feedback" loop (audio output to audio input); this allows the codec to send back to the remote codec the signal it receives, after decoding and re-encoding.

i Note: due to the location of the audio test points, the level indicators are not influenced by the test loops.

3. Operation

i *This chapter physically describes the unit, shows how to install it and explains the main operation principles. The details about the menus or HTML pages can be found in chapters 4 and 5.*

3.1. General principles – Control means

The equipment control and supervision (configuration, status monitoring) is possible either in "Local" mode (front panel keypad and display, status indicators), 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 web browser, thanks to the **embedded HTML** server, is detailed in chapter 5: Operating mode – Embedded HTML pages.

The GPIO can be used for a simple management of the link setting up and releasing, as described above in 2.5.5, "GPIO".

3.2. Physical description of the equipment

The SCOOPY+ S codec is housed in an ABS chassis, equipped with metal panels supporting the various connectors on the sides and the back. It is powered by an external 12V DC source, and/or by a set of rechargeable batteries (charging circuits integrated in the unit).

3.2.1. Front panel


All the elements needed for local controls are located on the front panel:



Figure 6 – Front panel of the SCOOPY+ S

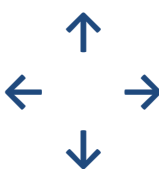

1. OLED screen
2. Joystick
3. Headphone 1: volume adjustment (center knob) and send/receive mix balance (outside ring). Ring turned leftwards (Rx): more feedback (or remote), rightwards (Tx): more for the send (or local) signals.
4. Gain adjustment for input 1
5. Keypad and LED indicators
6. Gain adjustment for input 2
7. Headphone 2: volume adjustment (center knob) and send/receive mix balance (outside ring). Ring turned leftwards (Rx): more feedback (or remote), rightwards (Tx): more for the send (or local) signals.

On/Off switch and standby

First, press the on/off key  for at least 3 seconds to switch on the unit. When it is running, keep this same key pressed for at least 3 seconds to switch the unit off.


Display and navigation keys

Besides the OLED screen you can find the keys for navigating in the menus:



Touche	Fonction
OK	Confirm a selection, enter a sub-menu or enter data. From the base screen: switch to the root menu.
Esc	Escape to upper menu level; From the base screen: switch to the root menu. <i>Long press: switch on/off</i>
Joystick 	Equivalent to arrow keys and a push-button (press down the joystick), used to move the cursor or browse through menu options.  is also used to go back one step up the menu structure, or delete the last character while entering data. Pushing down the joystick is equivalent to pressing the OK key.

More details can be found about the menus in chapter 4, which deals with operating modes.

Keys for audio handling

Keys	Function
One "Mute" key for each input 	Activates or cuts the input; the red LED "ON" shows that the input is on.

Call management and dialling keys

Key	Function
"Hang up" 	Release an established Program link.
"Unhook" 	Start a call for a Program link.
Dialling keys 0 .. 9 * . #↑	Keys for entering numbers or URI for the destination to call. These keys are also used to enter texts such as profile names, etc. The "#↑" key is used for switching between numbers, lower case letters or capital letters. An indicator on the screen recalls the current type of characters. Repeatedly press a key to get the characters other than the main figure. Keys 1 and ".*" also allow entering special additional characters (not all marked on the keypad).

LED indicators

The LEDs provide the following information:

Marking	Color	Function
DC/Charge	<i>Green/Red</i>	Off : unit off and no DC power connected; <i>Green</i> : DC source connected; <i>Red</i> : DC source connected, internal batteries are charging
DEC	<i>Green/Red</i>	Off when unit is idle; <i>Green</i> when the decoder is synchronised on "Program"; <i>Red</i> if interface is active/connected but not synchronised or in error
LIM (each input)	<i>Green/Red</i>	Off when limiter on concerned input is off; <i>Green</i> when the limiter is activated; <i>Red</i> if the limiter is active and currently limiting the level
ON (each input)	<i>Red</i>	On when the input is enabled
INFO 1	<i>Amber</i>	Shows the state of the received "INFO 1", when the auxiliary "relay transmission" is active
INFO 2	<i>Amber</i>	Shows the state of the received "INFO 2", when the auxiliary "relay transmission" is active

"Recording" keyboard

Just above the front panel you can find a small keypad, dedicated to the recording function (optional):



Figure 7 - Recording keyboard

Keys	Function
◀◀	Jump backwards by 10 s while playing back a recorded file.
▶	Playback a recorded file. While playing back: pause, then toggle Play/Pause on each press. While recording: pause, then toggle REC/Pause on each press.
■	Stop recording or playback (and jump back to the beginning of the file).
●	Record. 1 st press: "arm" or "pre-record". 2 nd press: start actual recording.
▶▶	Jump forward by 10 s while playing back a recorded file.
F1	During transmission : toggle the state of "INFO 1" transmitted to the remote device. <i>Caution: if GPI 1 is assigned to INFO 1, toggling with F1 is temporary and the state quickly comes back to that defined by GPI 1).</i> While playing a recorded file: set a <i>take begin</i> marker.
F2	During transmission : toggle the state of "INFO 2" transmitted to the remote device. <i>Caution: if GPI 2 is assigned to INFO 2, toggling with F2 is temporary and the state quickly comes back to that defined by GPI 2).</i> While playing a recorded file: set a <i>take end</i> marker.

3.2.2. Rear panel: transmission interfaces

This panel mainly includes the interfaces to the transmission networks.

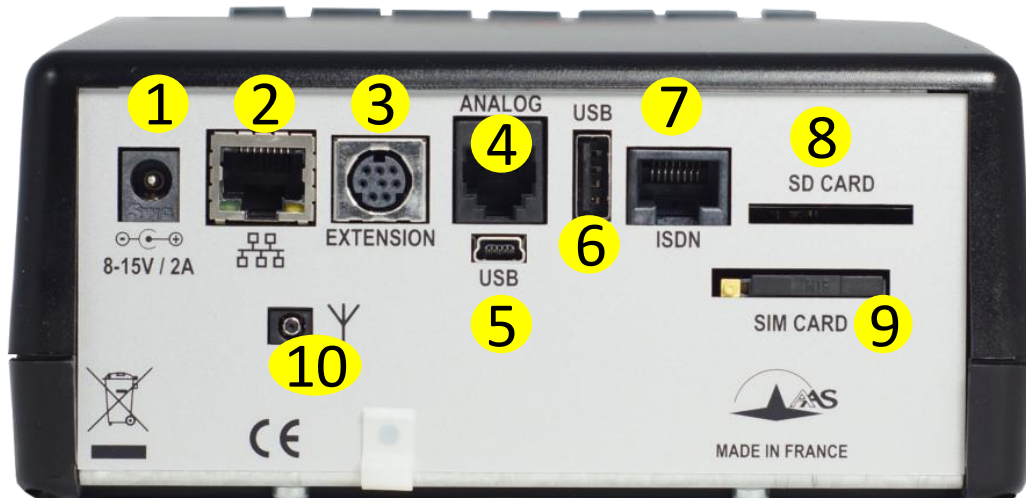


Figure 8 – Rear panel

The characteristics of the interfaces and their pinout are detailed in chapter 7, Technical characteristics. You can find on this panel the following items¹:

DC power ①

2.1 mm jack. When the DC power supply is plugged, the unit stays off, but the internal charger charges the batteries if they are installed in the battery compartment. Use the front panel on/off button to start the unit. *Details: see 7.4.1, DC power supply.*

Ethernet interface ②

This socket is a 100BaseT/10BaseT port, used for audio transmission over IP and/or for remote controlling the unit. This RJ45 socket is devised for a normal "straight" cable to an Ethernet hub or switch. The two integrated LEDs show the presence and activity of the network (green LED) and the interface mode: half-duplex (yellow LED off) or full-duplex (yellow LED on).

The configuration of the interface is described in 3.4, Initial setup of the Ethernet interface.

"Extension" socket ③

GPIO 1 and 2 interfaces: 2 optocoupler inputs, 2 relay outputs.

Wiring and characteristics: see 7.3.1.

POTS socket (marked "ANALOG") ④

For the units equipped with the PSTN/POTS option, this socket is used to connect the POTS line. You must plug here a cable from the POTS wall socket² and terminated with an RJ11 plug.

USB mini-B socket ⑤

This "device" socket allows the connection to a computer in order to use the SCOOPY+ S as a sound card.

¹ Some items may be missing, depending on the options installed in the unit.

² Whose type depends on the country and the building wiring

USB host socket ⑥

Host USB port, type A, for connecting peripheral devices (4G keys, etc.)

This "host" socket allows the connection of a peripheral device, e.g. a mobile access USB module or "key" in order to access mobile IP transmission.

Refer to chapter 2.1.4 about this function.

As another use, an Ethernet adapter can be plugged for providing a secondary Ethernet/IP remote control interface; *see chapter 2.5.3 for this topic.*

ISDN socket ⑦

This RJ45 socket allows the connection to the ISDN, for the product versions which include this capability. The wiring is for connecting an S0 (BRI) ISDN line using a standard RJ45 cable.

SD card slot ⑧

Slot for an SD card, used as storage for recording audio files.

Drawer for SIM card ⑨

In the units fitted with the "HD-4G mobile" network access option, this drawer houses the SIM card enabling the access to the network and services.

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

To remove the drawer and the card, push the small button at the left of the drawer (as seen from the back), using a pen or other sharp object.

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 down. Check that the drawer is well in its guides before pushing it completely in place.

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

Antenna socket ⑩

On the products fitted with the "HD-4G mobile" network access option, this HIROSE MS-151NB socket allows to connect an external antenna (a multiband antenna is included with the mobile network option).

The antennas must cover the band(s) used for the operator and network services. In doubt, refer to the operator for advice. The provided antenna covers the 1, 2, 3 and 8 bands (900 MHz, 1800 MHz, 1900 MHz and 2100 MHz). It is compatible with almost all the 2G and 3G/3G+ networks in Europe.

3.2.3. Left side: inputs 1 and 3



Figure 9 – Left panel

1. Input 3 socket: balanced input on 3-pin female XLR
2. Gain adjustment for input 3
3. Input 1 socket: balanced input on 3-pin female XLR
4. Headphone socket 1, 6.35 mm jack (1/4")

3.2.4. Right side: input 2 and line outputs



Figure 10 – Right panel

1. Headphone socket 2, 6.35 mm jack (1/4")
2. Input 2 socket: balanced input on 3-pin female XLR
3. Line output L: balanced output on 3-pin male XLR
4. Line output R: balanced output on 3-pin male XLR

3.2.5. Battery compartment

This compartment is located on the bottom side of the SCOOPY+ S. It houses 6 C-size batteries.

Use only NiMH batteries, such as those supplied as accessories by AETA (see further in 7.8).

An internal charger recharges the batteries when the external DC source is connected, regardless whether the SCOOPY+ S is running or on standby.

3.3. Putting in service

Recommendations for the operation:

- The unit is cooled by natural convection; you should not confine it too much for proper operation.
- With the "HD-4G mobile" option, the unit includes two antennas integrated beneath the top cover. Do not cover the SCOOPY+ S with metal objects or liquids prone to degrade the performance of the antennas. For the same reasons, do not confine it in a place where the reception of the mobile networks is typically bad: underground place, metal furniture, vehicle, etc. These are example cases where an external antenna can reveal useful.

For operating the codec, the minimum connections to set are (refer to the details in the rear panel description):

- Power source (Batteries and/or external DC)
- Audio inputs and outputs (XLR sockets)
- Network connections: depending on the network used, Ethernet interface, ISDN line, POTS.

The pin out of the connectors is indicated in chapter 7.1: Characteristics of the audio interfaces.

Before the first use, the equipment must be configured according to the desired operation mode: audio input/output format, local conditions (network interface parameters...). Then to set up links you must select the coding type and parameters.

For using the configuration menus, a password may have to be entered. After factory setting or after a complete configuration erasure, the password is "blank" (no password) and the access is unrestricted. Afterwards, if needed, a password can be programmed by the user and the restricted access mode can be activated.

For more details about the codec configuration, see chapter 4 - Detailed operating mode. The setup for the Ethernet interface is described in 3.4 below (Initial setup of the Ethernet interface).

3.4. Initial setup of the Ethernet interface

The SCOOPY+ S includes a 100BaseT / 10BaseT Ethernet interface, and the audio transmission can take place over an IP network through this interface. In addition, it is always possible to use the Ethernet interface to access the embedded html server.

An initial set up is needed for using one of these features of the Ethernet interface. For setting into operation, first connect the Ethernet interface to the network, using CAT5 wiring.

- Connection to 10BaseT or 100BaseT interfaces are both suitable, as the SCOOPY+ S automatically switches to the adequate 10 Mbit/s or 100 Mbit/s mode.
- "Straightforward" (MDI) patch cables should be used for a connection to a hub or a switch. Conversely, a "crossed" cable (MDI-X) might be needed for special configurations (e.g. a test connection to a computer).

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 menu to use is reached by [Setup / Network / Ethernet Parameters](#).

3.4.1. DHCP server available

This is the simplest case, because the server will allocate a suitable IP address and provide the unit the right settings. Select "**DHCP**" in the menu ([Setup / Network / Ethernet Parameters / Mode](#)). 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 "About" menu ([Tools / About](#)).

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.

3.4.2. "Static" IP configuration

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 [Setup / Network / Ethernet Parameters / Mode / Manual](#). Then, still in the menu [Setup / Network / Ethernet Parameters](#), you must enter the following parameters:

Parameter	Notes
Local IP	Must be unique on the network
Subnet Mask	A typical value is 255.255.255.0
Gateway	
DNS Server 1	Domain Name Server (main)
DNS Server 2	Domain Name Server (secondary)

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

i 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.

3.4.3. 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 "About" menu (**Tools / About / Local IP**).

You can then 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 SCOOPY+ S.

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

3.4.4. 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.

i *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 **Setup / Network / Ethernet Parameters / Link Mode** . The available choices are:

- **Auto-negotiation** (*standard setting*)
- **100BaseT, full-duplex**
- **100BaseT, half-duplex**
- **10BaseT, full-duplex**
- **10BaseT, half-duplex**

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

3.4.5. Configuration of the secondary Ethernet interface

In case an external USB/Ethernet adapter is used to provide an additional interface (see 2.5.3), an initial setup is also required for this interface. The procedure is similar to that for the main integrated Ethernet interface:

- First plug the adapter on the USB Interface.
- Connect the Ethernet interface to the network, using CAT5 wiring. *Normally the adapter's LED should blink depending on the network activity.* The additional interface is designated "**Ethernet 1**" in the unit's menus.
- Apply the same steps as described above (3.4.1 to 3.4.4), the settings are in this case accessed via the sub-menu **Config / Network / Ethernet 1 Parameters**.
- To check the IP address and status of the interface, go to the menu **Config / Network / Ethernet 1 Parameters / Network Config**.

3.5. Managing links

3.5.1. 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.

A 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 4.5 (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 SCOOPY+ S (local release).

3.5.2. 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. 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 especially applies in the following cases:

- Abnormal loss of the link because of a problem on the network;
- Unit switched off or power failure; in such case, after rebooting the codec will automatically redial and set-up the link back.

i *Note that, while "auto redial" is active, an established link can be definitively stopped only by releasing the line on the calling codec side. 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 the "Auto Redial" sub-menu ([Tools](#) / [Misc](#) / [Auto Redial](#)).

3.5.3. GPIO control

The normal operation is via menus and/or remote control. In addition it is possible to assign to the GPIO functions for partly managing the links thanks to these interfaces: redial last called destination, or release the line. These capabilities are described in 2.5.5, GPIO.

3.6. First level maintenance

3.6.1. Using the test loops

The test loops accessible from the "**Tools / Maintenance / Tests**" menu can help improve the analysis of a problem:

- In order to check if the audio part functions correctly, use the "Audio" loop and check if the audio is OK at the output.
- To check if the coding part functions correctly, activate "Loop 3"¹ and check if the alarm disappears (and the decoding indicators come back to normal), and if the audio is present at the output.
- "Loop 2"² sends back to the remote codec the compressed data received from the network (see the description of test loops in 2.5.7, Test functions). In this way, it is possible to test the integrity of the transmitted data and/or check that the remote codec works properly.
- The decoder out to encoder in loop ("Audio feedback" loop) can be used for overall functional check, and also for aligning the overall chain.

Note: not all test loops are available for all the network interfaces.

3.6.2. Clearing / Saving / Restoring settings

Clearing all settings

It may be necessary in some cases to come back to the initial "factory" settings.

For clearing all the configuration and coming back to the default parameters, you can go to the sub-menu **Tools / Misc / Reset** and select "Factory Reset" or "Reset Settings" (which does not delete stored profiles and snapshots neither audio recordings). You also can find this capability on the html interface.

The default passwords are blank.

i *If it is not possible to access the menu for performing a complete reset (for example the access is restricted and the password is lost), you can use the html interface. If this one is locked as well, or not accessible, you can then follow the instructions for a "Backup reset", described further (3.6.3).*

Saving and restoring the configuration

It is possible to export/save all the settings of the unit in a file, and conversely to import all the configuration from a file previously stored in this way. Transfers are performed using the html interface; see the operation in 5.10.6, "**SETUP TRANSFER**" page.

Partial exports can be done as well, selecting the elements to include: call profiles, presets, snapshots, settings...

¹ Note: this loop is not available for all transmission configurations

² This loop is not available for all transmission configurations


3.6.3. Backup reset

This procedure should be applied to recover control over the unit if it is in a blocked status, in which it is impossible to access either the front panel interface or the html page.

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

Go through the following steps:

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

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

3.6.4. Firmware updating


When necessary, updating the firmware can be done via the html interface of the SCOOPY+ S. The procedure is described in 5.10.3, ""SYSTEM UPDATE" page".

4. Detailed operating mode – User interface

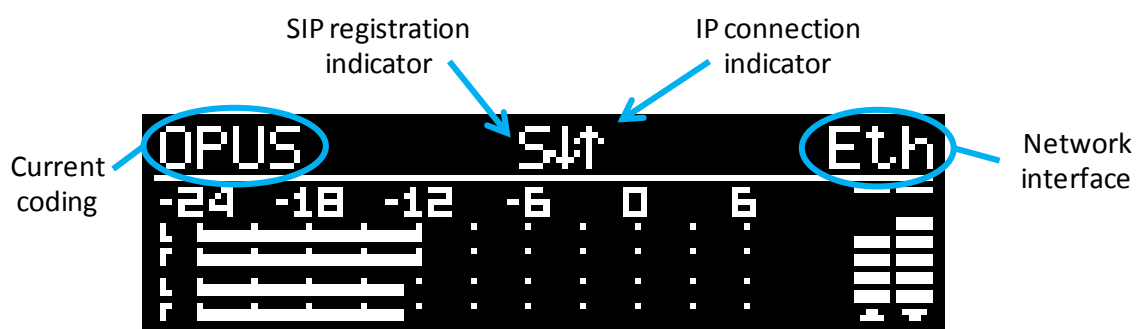
In local mode, the unit is operated thanks to a keypad and display on the front panel. The display is an OLED graphic display.

Operating from the keyboard can be protected by a password (numeric sequence, that can be programmed or erased by the user. In such case, the operation is in a "restricted" mode, without direct access to settings or dialling. Only the profiles menu is accessible, for loading a profile or call a call profile (*see later in 4.8, Restricted operation mode for more about the restricted mode*).

4.1. Equipment start-up

Press the  key for at least 3 seconds to start up the unit.


During start-up, the unit displays temporary messages. This initialisation lasts around 30 seconds. Then the main menu is displayed.



On this "base screen", transmission and reception audio levels are permanently monitored for the two audio channels.

At this time, the unit can be operated and receive calls. It is also possible, using the menus, to change settings and/or set up calls.


4.2. Principles for the navigation

From the base screen, if pressing the  key the display switches to the highest level menu, called "root menu"¹.



Pressing  again brings back the base screen.

The LED indicators and the main keys surrounding the display are described above in 3.2.1, Front panel.

¹ You can also press the  key

The unit proposes a tree structure of menus, and the joystick is used to navigate through the menus. The **OK** key is used to confirm choices or enter values for parameters, while the **Esc** key allows to come back up to the upper menu level. Repeatedly pressing this key brings you up to the root menu, and the base screen.

From the root menu, you can enter one of the three main menus by using the joystick and pressing **OK** to enter the highlighted menu.

i If the unit has been set in "restricted access" mode, for recovering full access to all menus type directly the password (numeric sequence), and the device unlocks as soon as the last figure of the password is pressed. See further in 4.8 how to use and manage this restricted access.

In these main menus, the screen is organized in the following way:



The sub-menus have the following aspect:



At the end of a "branch", lists of choices appear in this way:

Reminder of parameter to set



Current value for the parameter

In all cases, the principles are as follows:

- Use the joystick to browse sub-menus or choices.
- Enter the highlighted sub-menu by pressing **OK** or pushing the joystick to the right.
- Move to upper level with the **Esc** key or pushing the joystick to the left.
- Validate a choice or a value using **OK**, or pushing down the joystick.
- Give up a selection or action by pressing **Esc** (repeatedly as needed)

i After some time without pressing any key (about 30 seconds), the display comes back to the base screen.


4.3. Dialing and text input keypad

The keypad is used for entering numbers and/or texts. It is used in a similar way to a mobile phone:

- The keypad works in "numeric" only mode, or in "alphanumeric" mode (where both numbers and letters can be entered).
- In numeric mode, only the numbers are used, and the "*" key (as a separation between ISDN address and sub-address, or as a dot in a numeric IP address).
- The "alphanumeric" mode allows text entry. A letter can be entered by pressing repeatedly a key; for example, a "B" is entered by hitting the "2" key two times (sequence A, B).
- Use the #↑ key to switch between "numeric", "alphanumeric (lower case)" and "alphanumeric (capital)" modes.
- The "0" and "1" keys provide access to various special characters which are not all printed on the keypad: "@", ":", "-", "_" ...

4.4. Description of the menus

The root menu shows the following three main menus:

- **Setup** configuration of the codec
-  (directory): access to configuration memories or "profiles"
- **Tools** Maintenance and miscellaneous functions, access to status information




The "**Setup**" menu leads to the following sub-menus:

- **Network** Selection and configuration of the network interface and parameters
- **Algorithm** Selection and configuration of the coding algorithm
- **Audio** Configuration of the audio interface and parameters
- **Coordination** Establish a coordination link (mobile voice)

The  menu for profiles leads to the following sub-menus:

- **Call Profiles** Selection and edition of call profiles
- **Presets** Selection and edition of the presets
- **Snapshots** Selection and edition of snapshots (memories for audio parameters)
- **Recordings** Manage audio recorded files
- **Takes** Manage takes (virtual extraction of parts of recorded files)

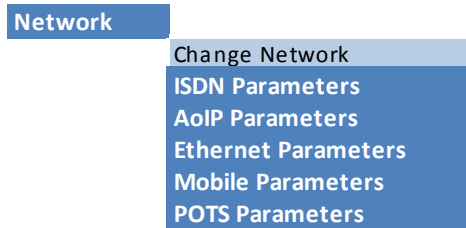
 *This menu is the only branch of the menu tree that is accessible even in restricted mode.*

The "**Tools**" menu is itself divided in sub-menus:

- **About** Display information on addressing and firmware version
- **Misc** Miscellaneous functions
- **Maintenance** Maintenance and test functions

The following diagrams show the various sub-menus and the available parameters.

4.4.1. Network sub-menu



The **Change Network** sub-menu shows a selection among the available network interfaces. This is how you select the "default" interface for the Program link:

- This is the interface that is used for an outgoing call. This menu must be used before setting a call (except if the default interface is already the desired one).
- In the case of an IP interface (Ethernet or mobile), this is the interface that will receive incoming AoIP calls.

When entering this menu, the current interface is highlighted in the list.

The other sub-menus provide access to the settings for each existing interface. This is why some interfaces may not be visible in the sub-menu depending on the unit version, if it does not include such interfaces.

- ❗ *If you have plugged a 3G or LTE "mobile key" on the USB interface, you can also see this module in the list, designated as "Mobile 1 (Ext.)", and you can select it just like another network interface. If more than one mobile device is available (example: presence of an integrated module and also an "external" device connected on the USB socket), then each is identified in a unique way.*
- ❗ *In a similar way, if an Ethernet adapter is plugged on the USB Interface, you will see in the list a new interface, named "Ethernet 1". The corresponding sub-menu has the same structure and settings as the sub-menu "**Ethernet Parameters**".*

Ethernet Parameters

This sub-menu groups all settings for the Ethernet interface, and it is useful especially when setting up on a local area network.

Ethernet Parameters	
Mode	<ul style="list-style-type: none"> DHCP Manual PPPoE BGAN
Network Quality	<ul style="list-style-type: none"> Very high High Middle Low Bad Very Bad
BGAN Detection	<input checked="" type="checkbox"/>
Network Config	
Local IP	Local IP
Subnet Mask	Subnet Mask
Gateway	Gateway
DNS Server 1	DNS Server 1
DNS Server 2	DNS Server 2
Local IP	<Entry>
Subnet Mask	<Entry>
Gateway	<Entry>
DNS Server 1	<Entry>
DNS Server 2	<Entry>
PPPoE_user	<Entry>
PPPoE_password	<Entry>
PPPoE_service	<Entry>
PPPoE Connection	<ul style="list-style-type: none"> Connection [Disconnection]
Link Mode	<ul style="list-style-type: none"> Auto negotiation 100BaseT-FD 100BaseT-HD 10BaseT-FD 10BaseT-HD

Notes:

- **Mode:** the selection is operative immediately after selecting an addressing mode. *This selection strongly impacts other items of this menu!*
- **Network Quality:** 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 "**Low**", the codec sets a large size for the reception buffer, in order to stand a larger 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).
- **Network Config:** only appears when DHCP is active, in which case this sub-menu displays the addressing parameters that the Scoopy+ S got from the server:

```
Network Config
Local IP
10.0.20.39
```

Using the joystick, you can browse the various settings: local IP, sub-network mask, etc. Exit this display with the `Esc` key.

- If DHCP is not used, several menus allow you to enter the needed settings, such as **Local IP**, **Gateway**, etc. These are not shown if DHCP is active.
- **Link Mode** is related to the physical Ethernet interface; the default setting "**Auto-negotiation**" is suitable for almost all situations.

AoIP Parameters

This sub-menu groups the settings related to the Audio over IP (AoIP) transmission. It is relevant for the Ethernet interface but also for using a mobile access network, be it with the integrated module or an external module via USB.

AoIP Parameters	
Default protocol	<i>SIP</i> <i>Direct RTP</i> <i>Multicast</i>
STUN Mode	<input checked="" type="checkbox"/>
STUN Server	<Entry>
SIP Registration	<input checked="" type="checkbox"/>
SIP Accept Calls	<input checked="" type="checkbox"/>
SIP User	<Entry>
SIP Display Name	<Entry>
SIP Registrar	<Entry>
SIP Auth User	<Entry>
SIP Auth Password	<Entry>
SIP Outbound Proxy	<Entry>
Keepalive Interval	5 s ... 180 s
Min. Regist. Period	30 s ... 3600 s
SIP Port	<Entry>
RTP Port (SIP)	<Entry>
RTP Port (Direct)	<Entry>
RTP DSCP	<Entry>
Packet Replication	0-None 1-Simple Mode 2-Interleaved
Direct RTP Accept	<input checked="" type="checkbox"/>
Direct RTP Send only	<input checked="" type="checkbox"/>
Direct RTP Timeout	1 s ... 30 s
Multicast Mode	<i>Sender</i> <i>Receiver</i>
Multicast Ctrl Port	<Entry>
Multicast Audio Port	<Entry>
Multicast TTL	<Entry>

Changes done via this menu come in effect when you go back to the upper menu; some validation time is needed (a message is shortly displayed on the screen).

Notes:

- **Default Protocol:** this is the protocol used by default for setting an outgoing call. You can find here the three possible AoIP session types: *SIP (unicast)*, *Direct RTP (unicast)*, or *Multicast*. SIP is the standard choice, "Direct RTP" is a mode without signaling, used for instance for compatibility with devices that do not support SIP.
- **STUN Mode:** enable or disable STUN. In this way you can disable STUN without erasing the STUN server address.
- **STUN Server:** enter the address of the STUN server, either in numeric form or with a symbolic name. This kind of server allows Scoopy+ S 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**.
- **SIP Registration:** enable or disable the registration on a SIP *registrar* server. The following menus **SIP xxx** are used to enter the parameters of the account (or the accounts) on this server, when using such service. These parameters should be provided by the administrator of the SIP server. With this **SIP Registration** parameter, you can disable the registration and hence stop using a SIP server, without having to erase account parameters; these stay ready for later use.
- **SIP Accept calls:** when this box is checked, the codec accepts incoming SIP calls (regardless whether the codec is registered or not on a SIP server). If it is not checked, incoming SIP calls are rejected, with a "busy" indication.
- **SIP User:** 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, the unit does not register on the SIP server.*
- **SIP Display 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.*
- **SIP Registrar:** URL of the SIP server, in case one is used. You can also enter a numeric IP address.
- **SIP Auth User:** enter here the authentication ID as provided with the account. *If this field is left empty, the actual parameter is the same as the **SIP User** field, which is appropriate in most cases.*
- **SIP Auth Password:** password provided with the SIP account.
- **SIP Outbound Proxy:** 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:** 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.
- **Min. Regist. Period :** sets the interval between two re-registrations on the SIP server. Actually the server always imposes a maximum 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.
- **SIP port:** the default port for SIP signaling is 5060 (UDP), you can enter a different value here.
*Note: this port number is valid for the SCOOPY+ S itself, not for the server. For the server side, if a non-standard port is used, e.g. 5070, the "**SIP Registrar**" field must specify the port after a colon like in "mySIPserver:5070".*

- **RTP Port (SIP):** the default port for RTP/SIP audio streams is 5004 (UDP), but you can enter a custom value here if desired.
- **RTP Port (Direct):** port allocated to the transmission in Direct RTP mode; *should not be the same as the RTP port for SIP.*
- **RTP DSCP:** 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:** (see also 2.1.1 and page 10 regarding this feature) you can select here the setting for packet replication:
 - 0** => standard mode without replication (default setting)
 - 1** => packet replication without interleaving
 - 2** => packet replication with interleaving: the second packet is delayed; robustness is improved but latency is increased.
- **Direct RTP Accept:** when this box is checked, the codec accepts incoming RTP streams. If it is not checked, incoming streams on the Direct RTP port are ignored.
- **Direct RTP Send only:** if this is 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 "RTP Timeout" parameter (see further) is irrelevant. *This parameter should not be checked at the same time as "Direct RTP Accept".*
- **Direct RTP Timeout** (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 Accept** is unchanged).
- **Multicast Mode:** if the selected mode is Multicast, the transmission is unidirectional. Select with this menu if the codec must be *sender* of a multicast stream, or rather *receiver* of such stream.
- **Multicast Control Port:** this port number (UDP) is used for sending control packets accompanying the multicast stream. Default value is 6000.
- **Multicast Audio Port:** this port number (UDP) is used for transporting the multicast audio stream. Default value is 6001.
Note: this setting is not related to the RTP Port settings valid for RTP/SIP or Direct RTP.
- **Multicast TTL:** TTL value applicable to the RTP multicast audio stream. Default: 254.

ISDN Parameters

This sub-menu is not available on versions with no active ISDN interface.

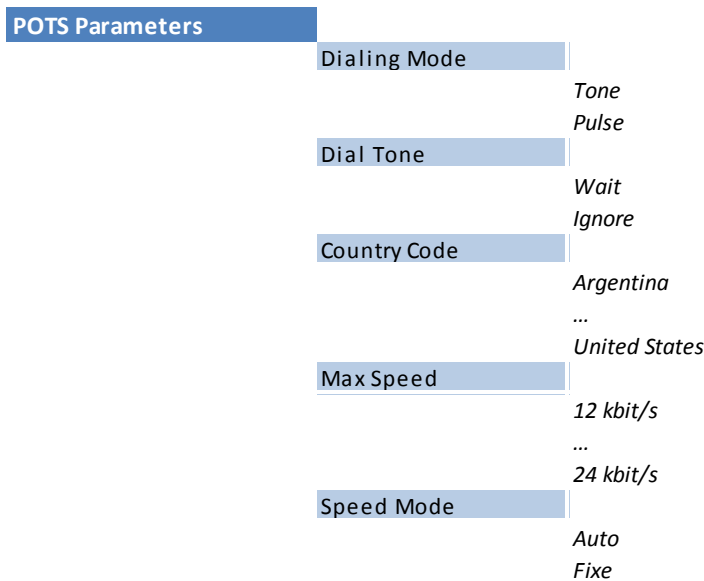
ISDN Parameters	
Protocol	Euro ISDN NI-1 NTT
Local Number 1	<Entry>
Local Number 2	<Entry>
Sub-Address 1	<Entry>
Sub-Address 2	<Entry>
SPID	<Entry>
Codec Mode	
5AS	<input checked="" type="checkbox"/>

Notes:

- **Protocol:** 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 (*see more details on page 11, Network protocol*).
- **Local Number** and **Sub-address:** these settings are irrelevant for the NI-1 protocol. *Their meaning and use is detailed further, page 71, in 4.5.2, "ISDN interface"*.
- **SPID:** this setting only applies to the NI-1 protocol, and is not presented for the other protocols. See in 4.5.2, "ISDN interface", more about this setting.
- **5AS:** by default the 5AS system for ISDN (cf. page 11, 5A System[®]) is active. If required, it can be disabled using this menu entry.

POTS Parameters

This sub-menu is only visible on units with the POTS option.



Notes:

- **Dialing Mode**: tones are used by default (DTMF), but for specific cases (old PBXs, etc.) it is possible to select pulse dialling.
- **Dial Tone**: choose "**Wait**" so that the codec waits for the dial tone before dialing the numbers. The dial tone may be specific with some particular equipment or networks, and may not be detected by the SCOOPY+ S; in such case, disable this function ("**Ignore**"), and the codec will dial the numbers just after picking up the line. *This latter choice also works correctly with most modern PBXs.*
- **Country Code**: this choice will set the internal modem for the characteristics applicable in the selected country. The full list is not shown above as it is quite large! For a country not found in the list, usually it is possible to select a country with similar standards (such information may be found from local specialists).
- **Max Speed**: in nominal use, the POTS codec runs at 24000 bit/s, but it can automatically adapt according to the line quality, and change the bit rate if required at the beginning and during the link. With this setting, you can constrain the bit rate to a lower value. Reason for this: on a line featuring a variable quality¹ you can avoid undesirable changes during the communication by "limiting" the modem to a speed that you know to be stable. The speed may nonetheless go down further if the quality decreases too much.
- **Speed Mode**: for a line/transmission with variable quality, and/or to avoid repeated changes of the bit rate, with this setting you can force a fixed bit rate. In such case, the "Max Speed" above setting becomes a fixed value. Be aware you must be sure that transmission conditions will not require a lower speed; otherwise you incur a synchronization loss in such eventuality.

¹ This can happen for example when the line suffers a strong crosstalk from neighboring lines.

Mobile Parameters

This sub-menu is only visible on units with a "wireless" option.

If more than one mobile access module is available (example: integrated module and "mobile key" on the USB socket), then one such sub-menu is presented for each module.

Mobile Parameters	
PIN	<Entry>
PIN Save	<input checked="" type="checkbox"/>
Mode	IP Mode Cellphone
Network Quality	Very high High Middle Low Bad Very Bad
Data Active	<input checked="" type="checkbox"/>
Preferred Techno	Auto Only 2G Only 3G Only 2G, 3G Only LTE
Network Operator	Auto Operator X
Access Point	APN <Entry> Username <Entry> Password <Entry>
Auth. Type	PAP CHAP
Network Status	Network Mobile network type Signal strength
Hardware	Manufacturer Model Revision IMEI IMSI Subscriber Number

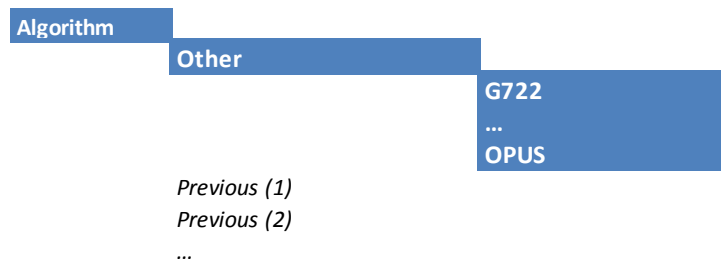
Notes:

- PIN:** 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.
The usual safety rules apply to this setting, so be careful not to lock the SIM card with repeated mistakes!

- **PIN Save:** if this option is checked, 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.
- **Mode:** select the desired transmission mode: "**IP Mode**" for transmission of data packets (IP protocol), or "**Cellphone**" for the telephone mode, which can also be wide band "HD Voice" when all conditions are met (*see above in 2.1.4, Mobile voice mode – HD Voice*).
Note: the telephone mode is not available for a "mobile key" device on the USB interface, because such devices only support IP data transmission.
- **Network Quality:** select here the expected quality of the transmission via the mobile interface in IP mode. This is the same type of setting as for the Ethernet interface (see above in 52, Ethernet Parameters), but a different value can be set here for the transmission via a mobile network (typically a lower quality can be expected).
- **Data Active:** activate or disable the mobile data service. This is necessary for using the mobile IP mode (but useless for the mobile voice mode). 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. For instance it can receive an incoming call. It can also initiate an outgoing call.
Note: once this parameter is "checked", it is memorized: the data connection is carried out automatically by the SCOOPY+ S 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, mobile data are useless.
- **Preferred Techno:** select the network technology. "**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). It is also possible to force a specific technology xG ("**Only xG**" selection), or exclude the LTE mode ("**Only 2G, 3G**" choice).
Note: if you force the LTE mode it is not possible to perform a voice/HD voice telephone link.
- **Network Operator:** normally the unit automatically selects a suitable network. If you want to force a specific operator (which can happen 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!*
- **Access Point:** this sub-menu allows to enter the necessary parameters for data connections (IP mode): APN, user name and password. The essential setting is the **APN** (Access Point Name), mandatory for accessing the mobile IP network. SCOOPY+ S automatically sets the APN for many operators, but you may have to enter this yourself, or change the default setting. The other parameters can usually stay empty; otherwise they must be provided by the network operator, along with the SIM card and the APN.
- **Auth. Type:** type of authentication to be used for the data service connection. The default setting PAP fits most use cases.
- **Network Status:** information on the mobile network (for the current connection): network/operator identification, technology (GSM, EDGE, UMTS, HSDPA, HSPA, LTE, etc.), radio signal strength. *Note: bars on the home screen also show the mobile reception level.*
- **Hardware:** information on the mobile access module (type, firmware, IMEI) and about the SIM card (IMSI, subscriber number).

4.4.2. Algorithm sub-menu

This sub-menu allows to select the desired configuration for an outgoing call, using the current transmission interface.



It is possible to quickly recall the last used configurations ("Previous" above), scrolling the list and selecting one of these recent settings.

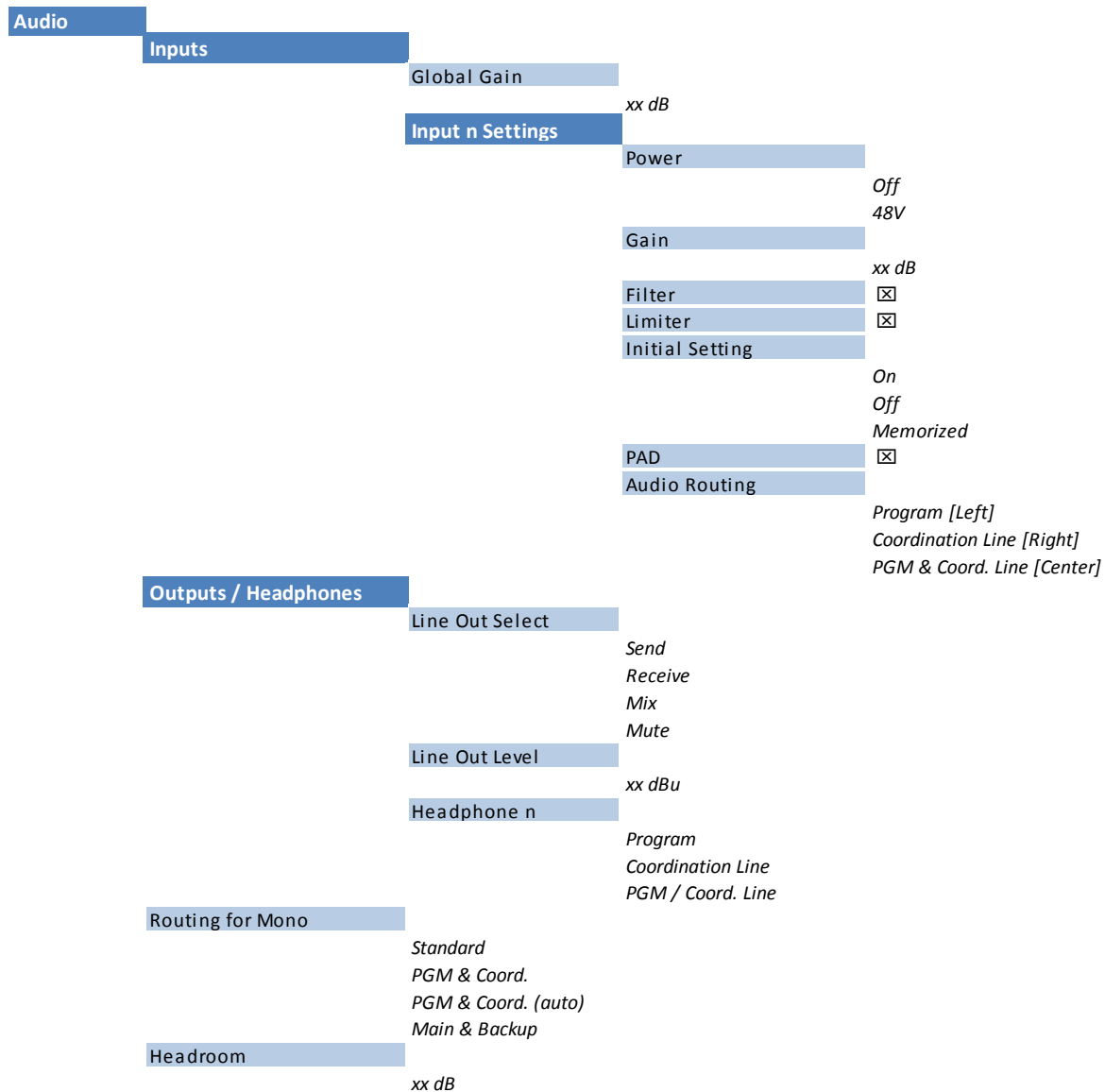
i This list can be cleared if needed: see menu "[Tools / Misc](#)".

Otherwise, selecting "**Other**" you get the list of the coding algorithms available for the current network interface and the installed options. Choices that are not compatible with the network interface or not installed are not displayed.

After selecting the coding algorithm, for some of these you must select additional parameters, such as: channel mode (for stereo-capable coding), sampling frequency (for MPEG L2, AAC, linear coding), bit rate (for MPEG L2, AAC, Opus), protection mode (only for MPEG J52 and CELP).

It is possible to quickly recall the last used configurations ("Previous" above), scrolling the list and selecting one of these recent settings.


4.4.3. Audio sub-menu



Notes (for more details on the audio processing, refer to 2.3, Audio interfaces and functions):


- **Global gain:** gain applied to the Program mix before transmission.
- **Input n Settings:** settings for Mic/Line input number n
- **Power:** activation of phantom power on the concerned input.
- **Gain:** coarse gain setting for the microphone preamplifier, selection 0 / 16 / 32 / 48 dB.
- **Filter:** activation of the high-pass filter (50 Hz, 18 dB/octave).
- **Limiter:** activation of the analog limiter (fixed threshold -6 dBFS).
- **Initial setting:** select the setting of the input "mute" at power on.
- **PAD:** 20 dB attenuation for standing a very high input level; only available on input 3.
- **Audio routing:** routing of the input to the mixing buses. The presented choice depends on the operation mode (mono/stereo, with coordination, etc.). For example, the choice in the above picture is found for a configuration with coordination.
For input 3, a "Sound Guide" option is also available.
- **Line Out select:** selection of the signals sent to the line outputs.

- **Line Out Level:** gain setting for the outputs. The value is the maximum level (absolute level expressed in dBu) found at the output when the digital signal (send or receive) is at its maximum (0 dBFS).
- **Headphone n:** selection of the monitoring mode on the headphone. The presented choice depends on the operation mode (mono/stereo, with coordination, etc.), and especially on the type of mono routing that is currently selected (next setting in this menu). *See the detailed description in 2.3.6, Mixing and routing.*
- **Routing for mono:** this setting is not found in stereo mode (when stereo coding is selected). It determines the audio operating mode. *See the detailed description in 2.3.6, Mixing and routing.*
- **Headroom:** this setting defines the reference level for the program level display. This display is relative 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. For instance, a headroom of 12 dB means that the 0 reference is set at -12 dBFS.


 On the display, the "OVL" clipping indicator shows that the absolute maximum level is over -3 dBFS.

4.4.4. Coordination sub-menu

This sub-menu enables to set a call and establish a coordination link (mobile voice call). The procedure is the following:

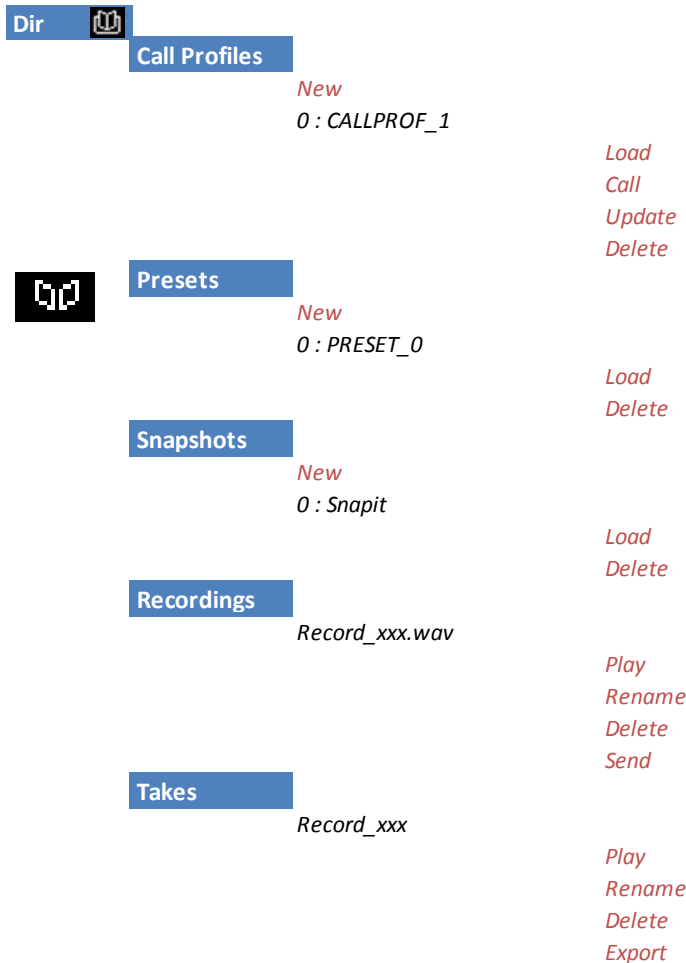
- Enter the sub-menu: an invitation to dial is displayed.
- Enter the telephone number to call, then press the "Call" key .

Later, for releasing the link, the procedure is similar:

- Enter the sub-menu: a message asks for confirmation.
- For effectively releasing the line, press the "Hang up" key .

4.4.5. Directory menu

This is the only menu available in restricted access mode.






The screenshot shows a menu structure with the following items and actions:

- Dir** (with a folder icon)
- Call Profiles**
 - New
 - 0 : CALLPROF_1
 - Load
 - Call
 - Update
 - Delete
- Presets** (with a phone icon)
 - New
 - 0 : PRESET_0
 - Load
 - Delete
- Snapshots**
 - New
 - 0 : Snapit
 - Load
 - Delete
- Recordings**
 - Record_xxx.wav
 - Play
 - Rename
 - Delete
 - Send
- Takes**
 - Record_xxx
 - Play
 - Rename
 - Delete
 - Export

"Profiles" are configuration memories that can be recorded and later recalled at will (see their description in 2.5.6 above). For each profile category, a sub-menu provides the list of profiles already recorded in the SCOOPY+ S memory, each with its index and name.








On top of the list, the "New" choice allows you to create a new profile:

- Enter a name for this new profile, then . For a "preset" or a "snapshot", the current settings (respectively network settings or audio settings) are recorded into the new profile with the name you just entered.
- For a preset, first you can elect to include or not the coding parameters: select "Current" (include the current codec settings) or "None" (do not include an algorithm with this profile).
- Then you can enter the dial number (or IP address, or SIP URI), then . You can also leave this field empty; in this case the profile is used for memorizing only coding parameters, rather than a remote number.
- Lastly, for ISDN and coding at 128 kbit/s, you must enter an additional number.

To use the profiles, just scroll the list, select one of the recorded profiles, and press : a choice is presented:

- **Load** this profile, recalling the memorized parameters;
- **Delete** the profile (not available when in restricted access);
- **Call** the remote destination in the profile (choice available only for a call profile): in this case the codec recalls the memorized parameters, and then sets the call to the memorized number(s);
- **Update** the profile (for a call profile only): you can edit an existing profile, going through the same steps as for creating a new call profile. *This choice is not available in restricted access mode.*

When the SCOOPY+ S includes the "Recording" option, the menu includes items related to this feature:

- **Recordings:** this sub-menu shows the list of recordings available in the SD card. Scroll the list and reach the desired recording, then: either you press the playback key  and the file starts playing, or you press  and the following options are proposed (scroll with the joystick).
For a quick access to this sub-menu from the base screen, press directly the  key.
 - **Play** : press  and the playback starts.
 - **Rename** : change the file name.
 - **Delete** : remove this file from the SD card
 - **Send** : send this file to an FTP server (the access to this server is configured via the menu **Tools / Misc / TFP Parameters**), via the main IP interface.
- **Takes:** this sub-menu shows the list of virtual "Takes" you have created (see details in 4.7.3). Scroll the list and reach the desired take, then: either you press the playback key  and the playback starts, or you press  and the following options are proposed (scroll with the joystick).
 - **Play** : press  and the playback starts.
 - **Rename** : change the name of the take.
 - **Delete** : delete this take.
 - **Export:** generate a "real" WAV format file including only the part of the original file that is selected in the take.

 Look for the detailed operation of the recording function in 2.3.5, *Recording/Playing audio files.*

4.4.6. **About** sub-menu

About
Local IP
Public IP
NAT Type
Version
AoIP Version
SW Build
MAC Address
Ethernet Setup
SIP User
SIP Registrar
SIP Status
Serial Number
Remote Access Status
Battery Level

This sub-menu provides a list of data about the identification and status of the unit. Scroll the list using the arrows and access all the information:

- **Local IP:** IP address of the active IP interface (Ethernet or mobile depending on the current one).
- **Public IP:** public IP address, when SCOOPY+ S accesses the Internet through a NAT router. This address is possibly detected thanks to a STUN server (configured via the "AoIP parameters" menu, see in 4.4.1).
- **NAT Type:** in addition to the above public IP, thanks to STUN Scoopy+ S can detect the type of NAT.
- **Version:** SCOOPY+ S firmware version global identification. Along with the unit serial number (shown on the label stuck on the bottom of the unit, and also further in this "**About**" menu), this is part of the essential information to provide whenever you contact AETA for support or maintenance issues.
- **AoIP Version:** version number for the audio via IP module.
- **SW Build:** date of generation of the firmware.
- **MAC Address:** of the main (integrated) Ethernet interface.
- **Ethernet Setup:** current setting of the main Ethernet interface (10 or 100 Mbit/s, half/full duplex...).
- **SIP Registrar:** address or domain name of the SIP server, if one is configured.
- **SIP Status:** shows whether the codec is registered on the SIP server. This information is also visible on the base screen ("S" indicator).
- **Serial Number:** shows the serial number of the unit.
- **Remote Access Status:** if you have activated remote access (or remote assistance), you can check here if the connection to the service is successful.
- **Battery level:** voltage of the batteries inside the device.

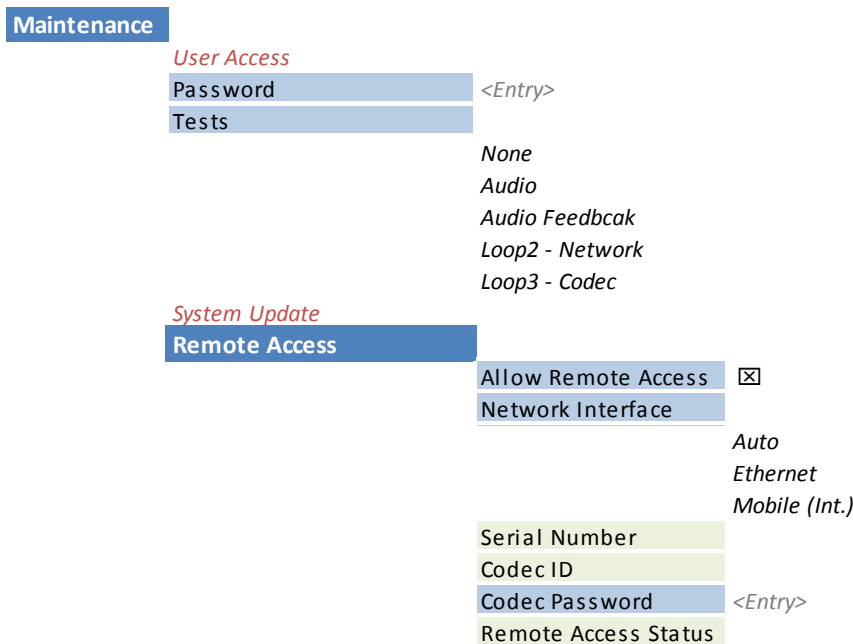
4.4.7. Misc sub-menu

Misc	Reset	<ul style="list-style-type: none"> <i>Reset Call History</i> <i>Reset Algo. History</i> <i>Reset Snapshots</i> <i>Reset Presets+CallPro</i> <i>Reset Recordings</i> <i>Reset Settings</i> <i>Factory SIP account</i> <i>Factory Reset</i>
	Aux. Functions	<ul style="list-style-type: none"> Data Channel <input checked="" type="checkbox"/> Baud Rate Relays <input checked="" type="checkbox"/> SMS <input checked="" type="checkbox"/>
	GPIO	<ul style="list-style-type: none"> <li style="background-color: #4a7ebb; color: white; padding: 2px;">Inputs Assignment <li style="padding-left: 20px;">Input 1 <li style="padding-left: 40px;"><i>Disabled</i> <li style="padding-left: 40px;"><i>Info 1</i> <li style="padding-left: 40px;"><i>Redial</i> <li style="padding-left: 40px;"><i>Release Call</i> <li style="padding-left: 20px;">Input 2 <li style="background-color: #4a7ebb; color: white; padding: 2px;">Relays Assignment <li style="padding-left: 20px;">Relay 1 <li style="padding-left: 20px;">Relay 2 <li style="padding-left: 40px;"><i>Info 2</i> <li style="padding-left: 40px;"><i>Line Status</i> <li style="padding-left: 40px;"><i>Sync Status</i> <li style="padding-left: 40px;"><i>Disabled</i>
	GPIO Extension	<ul style="list-style-type: none"> <li style="background-color: #4a7ebb; color: white; padding: 2px;">Inputs Assignment <li style="padding-left: 20px;">Input 3 <li style="padding-left: 20px;">... <li style="padding-left: 20px;">Input 6 <li style="padding-left: 40px;"><i>Info 6</i> <li style="padding-left: 40px;"><i>Redial</i> <li style="padding-left: 40px;"><i>Release Call</i> <li style="padding-left: 40px;"><i>Disabled</i> <li style="background-color: #4a7ebb; color: white; padding: 2px;">Relays Assignment <li style="padding-left: 20px;">Relay 3 <li style="padding-left: 40px;"><i>Info 3</i> <li style="padding-left: 40px;"><i>Line Status</i> <li style="padding-left: 40px;"><i>Sync Status</i> <li style="padding-left: 40px;"><i>Disabled</i> <li style="padding-left: 20px;">... <li style="padding-left: 20px;">Relay 6
	Brightness	<ul style="list-style-type: none"> <i>Low</i> <i>Middle</i> <i>High</i>
	Display Dimming	<input checked="" type="checkbox"/>
	Language	<ul style="list-style-type: none"> <i>Francais</i> <i>English</i> <i>Deutsch</i> <i>Italiano</i>
	SAS	<input checked="" type="checkbox"/>
	FTP Parameters	<ul style="list-style-type: none"> FTP-Username <input type="text" value="<Entry>"/> FTP-Password <input type="text" value="<Entry>"/> FTP-Address <input type="text" value="<Entry>"/>
	Auto Redial	<ul style="list-style-type: none"> Auto Redial Codec1 <input checked="" type="checkbox"/> Redial Attempts <li style="padding-left: 20px;">1 <li style="padding-left: 20px;">... <li style="padding-left: 20px;">20 <li style="padding-left: 20px;"><i>Infinite</i> <li style="background-color: #4a7ebb; color: white; padding: 2px;">Time Before Redial <li style="padding-left: 20px;">1 s <li style="padding-left: 20px;">... <li style="padding-left: 20px;">30 s

Notes:

- **Reset:** you can select to delete only the snapshots or the profiles, or conversely to reset all parameters but keep these memories ("**Reset Settings**" choice). It is also possible to clear the history of calls or the history of selected algorithms, or the audio recordings. For a complete erasure with the restoration of all default settings, select "**Factory Reset**". *Caution: this cannot be cancelled!* Lastly, "**Factory SIP account**" allows to reload (into the AoIP settings) the "factory" registration account on the AETA SIP Server.
- **Aux. Functions:** this sub-menu allows to enable the auxiliary functions (details on these functions can be found in 2.4, Auxiliary transmission functions), and select the baud rate for the data channel. Besides, it also allows to enable or disable the displaying of the received SMS on the screen (active by default).
- **GPI or GPIO Extension / Inputs Assignment:** for each GPI input you can assign either its corresponding "INFO" transmission function (default choice, see 2.4, Auxiliary transmission functions), otherwise use it to trigger a "Redial" or "Release" action (by a pulse on the GPI).
- **GPI or GPIO Extension / Relays Assignment:** for each GPO output you can assign either its corresponding "INFO" reception function (default choice, see 2.4, Auxiliary transmission functions), otherwise use it for indicating the status of the line or the decoder.
- **Brightness:** select the desired front panel display brightness.
- **Display Dimming:** to preserve the OLED, this feature automatically decreases the brightness after about 5 minutes of inactivity on the keypad, and if no link is active. The display switches further to screensaver mode after about 30 minutes.
- **Language:** select the preferred language and the menus are immediately updated.
- **FTP Parameters:** you can send recorded audio files to an FTP server, and the connection to this server can be configured in this sub-menu. The common parameters can be found: address or domain name of the server, identifier (login) and connection password.
- **Auto Redial:** this sub-menu deals with the auto redial feature, whose operation is described in 3.5.2 above.
- **Redial Attempts:** you can adjust from 1 to 20 the number of attempts to reset the link in case of a drop, or even "**Infinite**" for redialing indefinitely.
- **Time before dial:** (in seconds) adjust between 1 and 30 seconds, after a link loss, the waiting time before redialing. It is often worth leaving such a pause, so that the cause of the link loss is possibly cured, or the remote unit gets back to its normal state, etc.

4.4.8. Maintenance sub-menu



This sub-menu allows managing the following functions:

Restricted user access

The restricted access mode is detailed specifically in 4.8, Restricted operation mode.

For setting up the restricted access, first you must set a password (that will be needed to later unlock the device), and for this use the "**Password**" sub-menu.

To activate the restricted mode, select the "**User Access**" sub-menu, and confirm by selecting "**Yes**". Then you must enter the password to lock the unit (this in order to check that you know this password!). The unit immediately enters the restricted mode and will stay in this mode (even after a power cycle or reboot) until it is unlocked using the password.

i *If you lose the password: it is still possible to apply a complete factory reset; this will also clear the password and restore full access. The various methods are recalled in chapter 4.9 further, Clearing all settings.*

Test loops

The test loops are described in 2.5.7, Test functions.

Remote access

When the SCOOPY+ S does not include the "Remote Access" option, this sub-menu is entitled "Remote assistance" and it allows to make the unit accessible for remote assistance by AETA; the "Codec Password" parameter does not appear in the menu in this case.

The principles of the remote access are described in 2.5.4, Remote access. The procedure for using it is described in chapter 6: How to use the remote access service.

4.5. Setting up a link

For dealing with links with the SCOOPY+ S, generally you should go through the following steps:

1. Select the **basic options** that will essentially determine the operation: audio operation (mono/stereo, with/without coordination), transmission network type. This is described below (4.5.1, Basic options).
2. Prepare or check the settings for the **network interface(s)** you will use. See below (4.5.2, Preparing the network interface(s)).
3. Select the suitable **coding configuration** for the link. See below (4.5.3, Setting the coding configuration).
4. Select or dial the destination address/number and **launch a call** to the desired destination. This is described below (4.5.4, Launching calls).
5. Alternatively, the unit can also **receive calls** from remote units. See below (4.5.5, Receiving calls).

4.5.1. 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? Depending on the equipment of the SCOOPY+ S and the available resources, the type of network is ISDN, Ethernet, mobile IP, mobile voice or POTS. In addition, there may be alternate interfaces: for instance, the mobile access may be using the internal module (HD-4G option), or an external USB device.
- Mono or stereo for mixing and transmitting? Of course, this choice is linked to the previous one.
- Will you use coordination thanks to a mobile voice link? This is possible if the "HD-4G mobile" option is installed, and in mono operation.

The table below summarizes the possible combinations available:

Case	Usage of HD-4G option	Audio mode	Transmission				
			ISDN	Ethernet	Mobile IP	POTS	Mobile voice
1	-	Mono	PGM				
2	-	Stereo	PGM				
3		Mono	PGM				Coord
4		Mono	Main				Backup

Table 7 – Basic operating modes

- **Case 1:** only one link is set for **mono** operation; any available network interface can be used.
- **Case 2:** only one link is set for **stereo** operation. **This case is automatically selected as soon as stereo encoding is selected.** It is not supported by all network types.
- **Case 3:** a mobile voice link (using the integrated mobile access module) is used for a **coordination** channel. This case requires the presence of an internal mobile network access module (HD-4G mobile option). It is selected when the choice for "Routing for Mono" is "Program & Coord."
- **Case 4:** a variation of the previous case; this time the mobile voice link is used as a backup, to which the remote control room can switch in case the normal program link is lost. This case is selected when the choice for "Routing for Mono" is "Main & Backup".

4.5.2. Preparing the network interface(s)

This part describes how to prepare the network interfaces for the subsequent connections, in order to be ready to handle 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.

ISDN interface

Use an RJ45 cable to connect the ISDN line to the ISDN socket on the back of the unit (see 3.2.2, ISDN socket ⑦).

The network interface must be configured depending on the local ISDN line that is used, thanks to the "ISDN Parameter" sub-menu ([Setup / Network / ISDN Parameters](#)).

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

- 5AS: on
- Protocol: Euro ISDN
- Local numbers and sub-addresses: leave blank
- SPID: only relevant for North America

Protocol

The default setting is "Euro ISDN", also known as ETSI protocol. Change this setting if another protocol is needed in your location.

Local address

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 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) are often 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.

i *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.*

Sub-address SA

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 SCOOPY+ S on the same line.

Whenever a sub-address is set, the unit will only accept incoming calls specifically directed to this sub-address.

i *Most often, the best setting is to leave this blank.*

SPID

This "Service Profile Identifier" is only relevant for the NI-1 protocol (used by some operators in North America). 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 must be provided by the network operator.

AoIP settings

These settings apply to any connection using the IP protocol, hence all Ethernet or mobile IP interfaces are concerned. They can be found in the sub-menu **Setup / Network / AoIP Parameters**, and details about these settings are described in 4.4.1, "AoIP Parameters" section.

The most common and convenient connection method is by using the SIP protocol: select "SIP" as first parameter (**Default protocol**).

If a SIP server is used, you should register the SCOOPY+ S on this server. Use the sub-menu to enter the SIP account data and activate the SIP registration. *Check that the codec is properly registered thanks to the indicator on the base screen ("S" sign at the middle of the top line), otherwise via the sub-menu **Tools / About, SIP Status** information.*

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 **AoIP Parameters** sub-menu.

*Check that the public address discovery has succeeded: **Tools / About** sub-menu: **Public IP** information (its mere presence shows the discovery has succeeded). The type of NAT detected by the Scoopy+ S is also shown in this sub-menu (**NAT Type**).*

If you wish to use the "Direct RTP" mode instead of SIP for outgoing calls, select "RTP Direct" for the first parameter (**Default protocol**). This mode involves in fact no signalling protocol such as SIP, but its operation is somewhat similar: a codec "calls" another, in fact initiates a link by directly sending a stream to the other end. The "receiver" of the call should be configured for accepting this incoming "call" and sending its own stream back to the initiator of the link.

Ethernet interface

Use an RJ45 cable to connect the SCOOPY+ S to a router or switch. A couple of LEDs on the Ethernet socket (back panel, see 3.2.2) indicate the status and activity of the Ethernet link.

It may be necessary to configure the network interface depending on the local installation, especially when it does not include a DHCP server. The Ethernet interface settings are reachable via the sub-menu **Setup / Network / Ethernet Parameters**.

The various parameters are described in 4.4.1, "Ethernet Parameters" section.

In most cases, you can use the following settings:

- **Mode:** "DHCP"
- **Link Mode:** "Auto-Negotiation"
- **Network Quality:** "Middle"

Mobile network (integrated module)

This part applies to units equipped with the "HD-4G mobile" option, which integrate a mobile network access module and antennas.

Prerequisites

A mobile network can be used as interface for AoIP connections. It can be used in voice mode as well (regular telephone links), especially for the coordination channel (case 3 or 4 mentioned above in 4.5.1).

You must have a subscription and a SIM card granting access to mobile transmission. *In The subscription must include access to the service(s) you intend to use: mobile data (3G/4G) for setting AoIP connections, and/or telephone service for coordination or backup (case 3 or 4 above) or even as program link.*

Mobile networks often set obstacles to AoIP towards a mobile terminal: NAT routers, often symmetric (which prevents benefitting from a STUN server), etc. Above all, it is rare to be assigned a public IP address, and is never possible to get a public *and fixed* IP address.

As a consequence it is very difficult to receive a call without using a SIP server. On the other hand, if the SCOOPY+ S is registered on a SIP server, once the data connection is active the codec is ready to receive an incoming call.

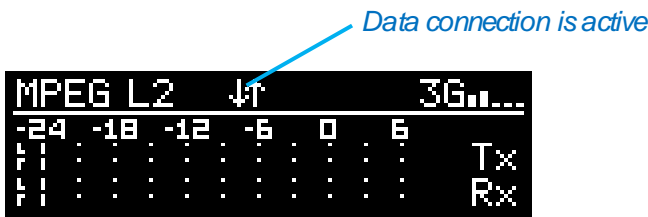
Preliminary settings

First insert the SIM card in the SCOOPY+ S (the unit must be powered off for this). See how to proceed in 3.2.2, "Drawer for SIM card ⑨" section. Power up the codec, enter the sub-menu **Setup / Network / Mobile Parameters / PIN**. Enter the PIN code of the SIM card using the keypad and press **OK**. *This is useless, of course, if the card's PIN code is blank or disabled. Besides, if you check **PIN Save**, you will not need to enter the PIN again, even after rebooting the unit.*

The network access must be configured using the "Mobile Parameters" menu (**Setup / Network / Mobile Parameters**), see details in page 59, Mobile Parameters. The following settings are suitable in most cases:

- Leave "**Auto**" for the **Preferred Techno** setting. *Conversely, avoid "Only 2G", as the performance in such case is too poor for AoIP.*
- Leave "**Auto**" for the **Network Operator** setting, unless you have a specific reason to force this setting.
- Select the data mode: **Mode / IP Mode**.
- Activate the mobile data by checking "**Data Active**". The other parameters are usually adjusted automatically.
- If however the **Access Point** parameters are not set, or their preset values are not suitable, you can enter new values: enter **Access Point / APN**, enter the appropriate access point name using the keypad and **OK**.
- You can define a **Network Quality**: in doubt, select "**Middle**". The best is to perform a test on a significant amount of time, and change the setting if the resulting stability is not good enough.

Once these settings are done, you can check the mobile network registration status in the sub-menu **Setup / Network / Mobile Parameters / Network Status**: operator, current radio technology, radio signal strength. In addition, icons on the base screen provide information on the mobile network: radio strength bars, double arrow showing that the data connection is active¹.



i It may take a while from editing the settings to actually getting the successful registration on the mobile network.

Using mobile voice

If your goal is to use the mobile network only for voice (or HD voice) links, it is not necessary to activate IP data as above: select for **Mode** the "Cellphone" option. The settings related to the data service are irrelevant in such case.

Mobile network (USB device)

The other method for accessing a mobile network is to use a 3G/4G USB device, for instance for following use cases:

- Accessing a mobile network from a unit that is not equipped with the integrated "HD-4G mobile" option (however a "USB mobile" option is required).
- Using a second SIM card for an alternate network, with the ability to switch quickly from one network to the other (switching from the integrated module to the external USB key, instead of having to switch the SIM cards).

You must have a subscription and a SIM card granting access to mobile data transmission. *Contrarily to the integrated module, you will not be able to use this alternative for setting mobile telephone links, neither for a coordination channel.*

First insert the SIM card into the USB key (refer to the device's specific documentation if needed) and plug the USB module on the USB host socket on the back. You may do this while the SCOOPY+ S is on. After a few seconds, the module is detected and included in the list of network sub-menus, under the title "**Mobile 1 Parameters**".

Enter this sub-menu to access the settings; these are almost identical to those of the integrated module, described in the previous pages.

PSTN interface (POTS)

The network interface must be configured depending on the PSTN line that is used, thanks to the "POTS Parameters" sub-menu (**Setup / Network / POTS Parameters**)

i You should preferably disable services such as call waiting tones. Tone signals injected on the line during the communication have a high risk of disturbing the codec operation and even dropping the link. It is not recommended to use a line if it is not possible to disable such tones.

¹ If the mobile interface is currently set as the default network.

4.5.3. Setting the coding configuration

This configuration is accessible through the sub-menu **Setup / Algorithm**.


When entering the sub-menu, a list of the most recently used configurations is displayed: you can scroll this list and select a configuration (enter OK) for loading it directly. For setting another configuration, select "Other". You then have to select the coding algorithm and adjust its various parameters:

- The list of algorithms depends on the current default network.
- The possible choices for the parameters depend on the algorithm; it is not possible to build an invalid combination.
- Especially, the number of parameters depends on the coding. Some algorithms just need no additional parameter (example: G711 A-Law).
- For a POTS line, the alternative is CELP or "Phone" (*the latter is not really a coding algorithm, as this is telephone hybrid operation*). In the case of CELP, pay attention to the "FEC" setting: it must be set identically on both ends of the link. The P1 and P2 settings are provided for compatibility with older models of codecs.
- There is no choice at all if the current network is mobile and the "Cellphone" mode is selected (integrated module).
- Selecting a stereo mode has a big influence on the audio routing, and in effect precludes the use of a coordination channel and/or the recording function.

i *Warning: if the goal is to set up an AoIP link in the Direct RTP mode, make sure to set up the coding exactly the same as the remote codec; otherwise the link might fail to set up.*

4.5.4. Launching calls (outgoing)

Once the preliminary settings are done, the procedure for setting a call is rather straightforward: dial a destination "number" and launch the call.


First go back to the base screen: if necessary press the  key several times.


Preliminary notice about call "numbers"



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 POTS or mobile voice call. A leading "+" sign is used if the international prefix is included. If a sub-address is necessary (ISDN), append a "*" sign and the sub-address number.
Example "+33141361268*12"
- **Numeric 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 Port**" in the **AoIP Parameters**). For "Direct RTP", it is the RTP port, if different from the default ("**RTP Port (Direct)**" in the **AoIP Parameters**).
- **SIP URI (Uniform Resource Identifier):** this is an identifier with a general format *username@domain_name*, using alphanumeric characters.
Example: Alice@myserver.com
Actually *username* most often only includes numbers, for easier dialing on compact 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 "S" indicator on the base screen.*



From the base screen, enter the number to call. Then press the "Call" key  for launching the call to the destination.


 *Whenever an ISDN connection requires two numbers (128 kbit/s coding with two different numbers for the remote device), the codec asks for a second number. If the first number is suitable as well, just press the  key instead of entering a new number.*

An error message is displayed if the link fails to establish, or if the remote end releases the link.



To clear it up and come back to the base screen, press .

Releasing/Hanging up

To stop a link and release the line, press the "hang up" key . Confirmation is required; press the  key again for confirming (*this is a security to avoid an undesired release*).

-  *The SCOOPY+ S 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 4.4.1 , "AoIP Parameters" section.*



Quick redialing


Once a number (or two for ISDN links with two B channels) has been called, it is easy to redial without having to type it again: press the  key, then scroll through the "history" (last dialled numbers) using the joystick. Press again the  key when the desired number is displayed. This is especially efficient for quickly redialling the last number called.

If you wish you can clear this history of calls: see the menu **Tools / Misc / Reset / Reset Call History**.

Using the directory


A quick way to dial is to use the "call profiles" in the directory.

Go to the "Directory" menu  / **Call Profiles**. Scroll with the joystick to reach the desired profile, then press  to launch the call.


-  *With this method you can at the same time: select the network interface, select a coding setup, and of course recall the number(s) of the destination. Indeed the flexible profile management allows you to include or not each of these elements in a call profile.*

Dialing or releasing the coordination link

A call on the coordination channel is handled on a different way: you must use the sub-menu **Setup / Coordination**.

- Enter the sub-menu: you are asked to dial the remote number.
- Enter the phone number to call¹, then press the "Call" key .

For later releasing the line, the procedure is similar:

- Enter the sub-menu: a message asks for confirming you release the line.
- For effectively releasing the line, press the "Hang up" key .

¹ If needed, use the "+" sign before the country code for an international call.

4.5.5. Receiving calls

Actually there is no additional setting to do for receiving calls. Incoming calls are automatically accepted by the SCOOPY+ S, 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. *However, if J52 is used by both parties, the link will succeed even without 5AS active.*

i *If an ISDN line is connected but this line is not the current network interface, the SCOOPY+ S will however accept incoming ISDN calls, provided that it is not already busy with another connection. Once the connection is released, the Scoopy+ S 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 or the POTS line, the unit can still register on a SIP server via the 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, the SCOOPY+ S "follows" the calling unit preference.

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 SCOOPY+ S 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: 4.4.1, "AoIP Parameters" section.*

i *For being able to receive a Direct RTP call, the **Direct RTP – Accept** must be active. As a shortcut for switching this parameter on/off, dial "0" (a pop-up message briefly shows the new setting).*

POTS incoming calls

When the call comes in, the codec "unhooks" and the link is established after a negotiation of the bit rate between the two involved units (this phase may last several seconds).


"Network quality" setting

Depending on the quality of service provided on the network, especially its jitter performance, it is possible to change the stability/latency compromise used by the SCOOPY+ S. For this purpose, a setting is available among the settings of each IP interface (for instance, for the Ethernet interface: sub-menu **Setup / Network / Ethernet Parameters / Network Quality**). Six options 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 (Quality of Service), 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, as can be found 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 very 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 setting is adjustable separately for every available IP network interface.*

4.6. Management of the configuration profiles

The configuration profiles ease configuration changes and the setting up of routine links. There are three 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 sub-menu **Setup / Network**. Recalling a *preset* is a quick way to restore the configuration needed for a given line or network.
- "**Snapshots**" memorize the settings for the **audio** interfaces. These are settings found in the **Setup / Audio** sub-menu.

These various memories are usable locally, but also through the embedded html pages, and also using these pages they can be imported/exported from/to a computer.

Each **call profile** includes the following elements:

- *Name* assigned to the profile
- *Network* to use with this profile
- *Number(s)*: ISDN call number(s), POTS or mobile call number, or IP URI, 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 "**Directory**" menu (refer to details in 4.4.5, **Directory** 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 previously recorded: the codec then configures itself with the network and coding configuration memorized in the profile.
- *Call* directly a profile: the codec then configures itself with the network and coding configuration memorized in the profile (if any), and sets a call to the number(s) included in the profile.
- *Delete* a profile previously stored.
- *Update* a profile previously stored, either for renaming it or change its content.

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 audio settings (for a snapshot), assigning it a name.
- *Load* a preset or snapshot: the codec then restores the settings included in the profile.
- *Delete* a preset or snapshot.

4.7. Recording function

To use this (optional) function, you must insert in the suitable slot (on the rear side) an SD card, FAT32 formatted (only file system supported).

i The SD card must be inserted when the unit is off (no "hot plug" capability).








Recordings are handled thanks to the "transport" keypad, located on top of the front panel:



4.7.1. Recording

i Reminder: recording is not possible when the current coding is stereo.

Recording can be started regardless if a link is active or not:

- Push the record key  : the recording is "**armed**" but not yet effective. The audio signal is recorded in a buffer memory. The display shows this with a  icon. It also shows (on top right) the remaining recording time available on the SD card.
- At the desired moment, press again the  key: the **recording** actually starts, which is shown by a  icon. The recorded file also includes the audio that was recorded in the buffer memory before this moment, up to a maximum of 30 seconds. *In this way, having armed the recorder beforehand, you won't miss the beginning of a report even if you started recording a little late.*
- You can **pause** the recording by pressing the play/pause key , then resume the recording by pressing it once again.
- To stop recording, press the Stop key . A pop-up message asks for confirmation: press the key again to confirm (or cancel and keep recording, by pressing ).

While recording, counters are displayed on top of the screen:



The recorded time is shown on top left, with 1 s resolution. On top right you can see the remaining recording time available on the SD card.

The audio file is named *record_xxx.wav*, where "xxx" is a sequence number, automatically incremented. You can rename the file afterwards. Once the recording is completed, it is accessible via the sub-menu [Directory / Recordings](#).

The file is one-channel (mono) if the recording is performed while idle, without an active transmission. The "Program Send" signal is recorded.


If you start recording while a link is active, the recording is two-channel ("stereo"¹): Program Send in channel 1 ("left") and Program Feedback/Return in channel 2 ("right").

i *Conversely, if you start a link while already recording, the file will stay one-channel, as it was when the recording was started.*

The recording goes on even if the link is interrupted while recording.

4.7.2. Playing back

To start playing back a recording, go to the list of recordings, using the sub-menu **Directory / Recordings**.





i *Shortcut: press the Playback key  .*

The cursor is located directly on the last recording you have accessed (last file recorded or played back). If needed, scroll the list and select another file.

- Press the Playback key  to start playing back. The display is similar to the one above:



Here, the left counter shows the current position in the file, while the right counter shows the remaining time *in this file*.

- You can pause the playback by pressing the play/pause key , and resume playing by pressing it again.
- You can also skip backward and forward (10 s), using the  and  keys.
- To stop the playback, press the Stop key  .

You can play a file at any time, even when a link is active. The playback signal is mixed into the Program Send.

i *However, beware of the following restriction: the recorded files cannot be played during the transmission when the coding sampling rate is 24 kHz.*

¹ Actually the channels are not left and right signals.




4.7.3. Simple editing

From a recorded file, it is possible to create virtual "takes". A take is made by setting a start point or *take begin* marker, and an output point or *take end* marker. This set (recording + begin marker + end marker) can be saved and named. It can be replayed just like a "real" recording, but the playback starts directly from the take begin location, and it will stop by itself at the take end location. The take thus defines a part extracted from the recording.

- Take creation: start playing back the recording, and during playback press **F1** to set the begin marker, **F2** to set the end marker. The display looks like the one below:



You can see the location of the markers.

- You can delete a marker by pressing again the corresponding F1 or F2 key, for instance F2 for erasing the end marker. Then you can set the marker again by pressing the key again, etc.
- You can move backward / forward and playback to check the result.
- When you reach the desired setting for the markers, press the Stop key  : you are asked if you wish to save the take. To cancel press  or validate "No". If you confirm saving ("Yes" then ), you can enter a name for this new take, or simply accept the name automatically assigned (which is that of the original recording).
- From then you can select this take from the take list, and play it like a normal "real" recording. The difference is in the markers displayed.

A "take" is virtual in the sense that only the position of the markers is stored, no audio file is actually created.

You can create as many takes as you want from a single recording. You can also re-edit a take, i.e. start playing the take and redefine the markers.

You can access the created takes by going to the sub-menu **Directory / Takes**. You will notice that regular recordings are also listed here. Then the functions are the same as for the recording directory:

- ▶ to play the take, etc.


4.7.4. Management of recordings





It is possible to extract the SD card from SCOOPY + S, insert it into a computer and recover the various recorded files for editing and further use. The wav format makes this easier for all types of available tools.

You also have a number of embedded management functions in SCOOPY + S. You can access them by going to the sub-menu **Directory / Recordings** or **Directory / Takes**. Scroll through the list using the joystick to select a recording or a take.

You can first obtain information about the file (or take) by repeatedly pushing the joystick to the right:

- Duration
- File size (not for a take as this is irrelevant)
- Name of the original recording (only for a take)
- Type: mono or stereo
- Sampling frequency

To manage the files or takes, press  (or push the joystick): a secondary menu appears (here also you can scroll to see all the proposed options):

- **Play:** press  to start playback.
- **Rename:**  to enter a new name.
- **Delete:** to delete, without confirmation.
- **Send** (only for a file):  to send the file to an FTP server. The connection data on the server are configured in the **Tools / Misc / FTP Parameters** menu (see 4.4.7, **Misc** sub-menu), or in the html pages, see below. The transfer will take place through the same IP interface as the possible AoIP links; it can be a mobile interface. *In the latter case, make sure mobile data is enabled.* The connection to the server is done automatically when the file is sent.
- **Export:** this function, only proposed for a virtual take, creates an audio file, in wav format, containing only the part of the recording included in the take. Press  to enter a name for the new file. The generated file joins the other "real" recordings. This function makes it possible to have on the SD card a file usable by post-processing tools. On the other hand it is also the first step for transferring the take to an FTP server.

4.8. Restricted operation mode

4.8.1. Principles

SCOOPY+ S can be locked in a restricted user access mode, which limits user access to the essential controls and features. For routine operation, this mode allows to avoid risks such as random settings, with unpredictable effects, handling or adjustment mistakes, accidental erasure of useful memories, etc. When SCOOPY+ S is in restricted access, it is still quite operative, especially if suitable memories (call profiles, presets, snapshots) have been prepared beforehand, because the following functions are still accessible:

- Receiving calls
- Releasing links
- Dialing and redialing (recalling previous numbers) for outgoing calls
- Loading and calling call profiles
- Loading presets or snapshots

On the other hand, it is not possible to change settings via the menus, as only the "Profiles" menu is accessible. It is also impossible to delete or edit the recorded profiles.

A password is needed for switching to the restricted mode (locking) or back to full access (unlocking). This password can be programmed (of course, not when in restricted access...).


i *Warning: this access protection is completely independent from the one that can be set for the "web" interface (embedded html pages). The passwords have no relationship, and are set separately. One of the interfaces (front panel or html) can be locked while the other is not.*

4.8.2. Locking the front panel

Managing the restricted mode is done with the sub-menu **Tools / Maintenance**. See in 4.4.8 (**Maintenance** sub-menu) the items of this sub-menu.


First you must program the desired password, because the password is blank when the unit comes from the factory or after a complete "factory reset". The password is an arbitrary sequence of numbers.

For switching to the restricted mode, select **User Access** and confirm (**Yes**). You must then enter the password to actually lock the unit (this makes sure you really know the password before locking). The display warns you about the switch to restricted access.

Press  and come back to the base screen. From then on, on every attempt to navigate in the non authorized menus, the following message is displayed:



4.8.3. Unlocking the front panel

From the root menu, press a numeric key. When prompted with a message, hit  and type (without waiting) the password sequence; SCOOPY+ S unlocks:



The full access is restored. Press  to come back to the home screen.

4.8.4. Lost the password?

You have forgotten or lost the password ? The only way to restore full access is to clear all the settings, which will also clear the passwords. The unit will be accessible again.

Chapter 4.9 below shows the procedure for such erasure.

i *In order not to lose all the memorized settings, you can perform a complete reset of the settings without deleting the profiles. These can be useful for you to come back more easily to the desired configuration.*

4.9. Clearing all settings

In some cases (especially if the password is lost), it can be necessary to come back to the "factory" configuration.

To clear up all the configuration and come back to the factory settings, normally you should go to the sub-menu **Tools / Misc / General Reset** and select **Factory Reset**, or **Reset Settings** (the latter does not delete the memorized profiles and snapshots).

The default password is blank. *In the factory setting, the menus are in English language. Other languages can be selected via **Tools / Misc / Language**.*

i *If it is not possible to access the menu to perform the factory reset (password lost), the erasure should be done using the embedded html pages: see chapter 5 below for the operating mode.*

4.10. Backing up and restoring the configuration

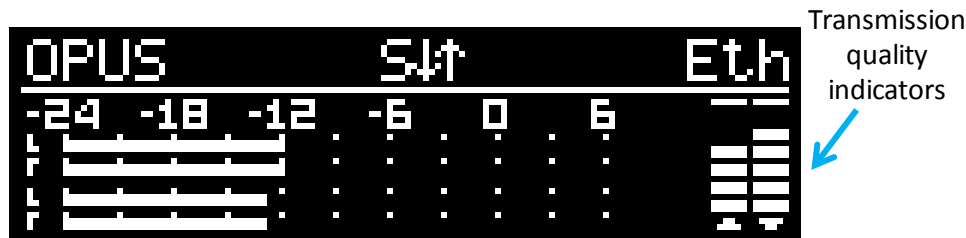
It is possible to save all the equipment settings in a file, and conversely to restore a complete configuration from a file previously recorded in such way. Such transfers can be done using the embedded html server; refer to 5.10.6 for the operating mode.

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

4.11. Specific display screens

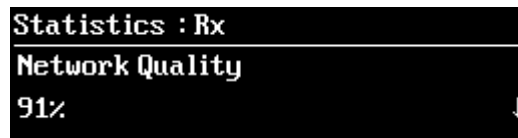
4.11.1. Display statistics about the IP transmission

During an AoIP connection, one can find, on the right side of the screen, 6-segment transmission quality indicators (full bar = 100%):



The left bar shows the quality estimated on the transmit direction, the right bar shows the quality in the receive direction.

It is possible to display more detailed information related to the transmission quality. To do that, once the link is established, and from the home screen, push the joystick to the left. A reception quality indication is displayed as follows:

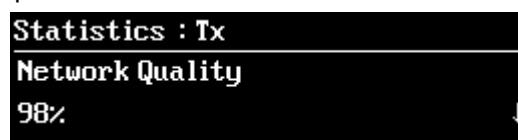


The value provides an assessment of the quality (100% corresponding to the best quality) that takes into account various relevant elements: packet loss rate, jitter, reception jitter buffer variations...


Moving the joystick vertically, you can display detailed information:

- "Duration": time elapsed since the beginning of the link (hh:mm:ss)
- "Total Bytes": total number of bytes received since the beginning of the link
- "Jitter (ms) – Avg": average jitter (in ms)
- "Jitter (ms) – Max": maximum jitter since the beginning of the link (in ms)
- "Packets – Lost": number of packets lost since the beginning of the link
- "Packets – Reordered": number of packets received out of the normal sequence
- "Packets – Total": total number of packets received since the beginning of the link

Pushing the joystick left, you can also access to similar data regarding the stream *transmitted* to the remote end. The title on the top line becomes "Statistics : Tx":

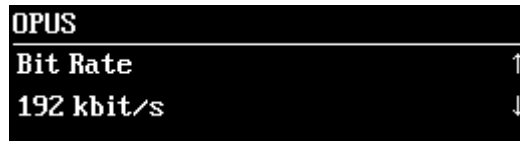


i Be aware that displaying data about the transmitted stream is not always possible, depending on the nature of the remote unit which receives this stream. This unit must support the RTCP protocol so that the relevant information is available for such display.

Press the  key to exit this display and come back to the home screen.

4.11.2. Case of the Opus algorithm: bit rate change "on the fly"


During a connection using the Opus coding, if you push the joystick to the left, you can change on the fly the Opus coding bit rate, without interrupting the transmission and with no switching noise:



Use the vertical arrow keys to bring the coding bit rate to the desired value.

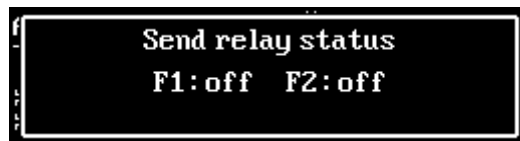
i *The modified setting is applied at least in the transmit direction, towards the remote codec. If this codec is another AETA codec, it will "follow" and change its transmitted stream as well, and the selected value will apply in both transmission directions.*

By pushing the joystick to the left again, you can access the detailed statistics as described above (4.11.1): statistics for the reception first, then (push the joystick left again) for transmission. Then you can come back to the bit rate setting by pushing the joystick left, and so on.

Pres the  key to go back to the home screen.

4.11.3. Display the state of the transmitted INFO 1 and INFO 2

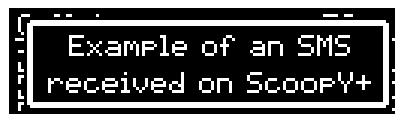
When the "relay transmission" function is on, and a connection is active, you can display the status of the *transmitted* info: from the home screen, push the joystick to the right. The status is shown as follows:




i *Reminder (see 3.2.1, "Recording" keyboard): if the INFOx are not controlled by GPI, you can switch their state using the F1 and F2 keys.*

4.11.4. Displaying received SMS

When the unit is equipped with the "HD-4G mobile" option and it is registered on a mobile network, SCOOPY+ S can display SMS received on the SIM card's subscriber number. The operation is very simple: on receiving the SMS the text is directly displayed on the screen:



Whenever the complete message cannot be displayed on the screen, use the joystick to scroll the message and read all the text. After the message is read, press  to come back to the normal display. Caution: the message is now cleared and it cannot be displayed again.

i *If you do not want to display the received SMS, you can disable this feature: go to the sub-menu **Tools / Misc / Aux. Functions** and disable the **SMS** parameter.*

5. Operating mode – Embedded HTML pages

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

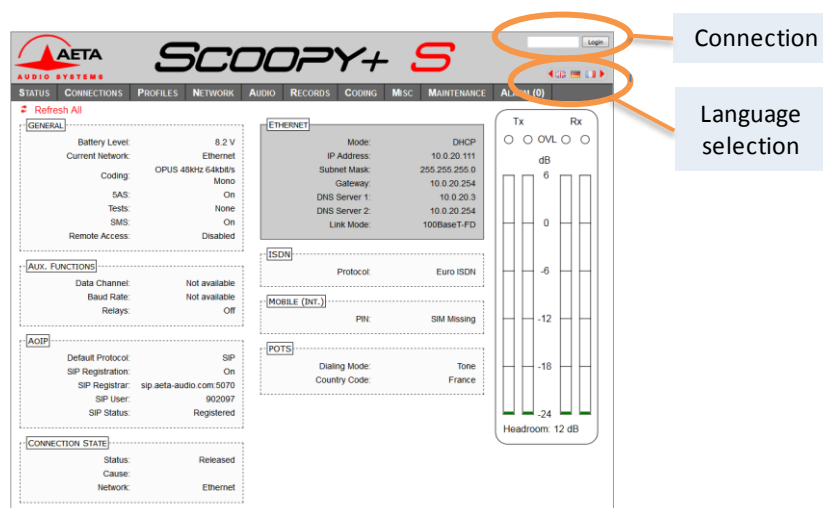
In the most common case, the two devices are connected on the same local network (LAN). But it is also possible to control the SCOOPY+ S by remote, provided that the control device can reach it (TCP/IP port 80, HTTP protocol, or port 443 for the HTTPS protocol).

If you also have a secondary Ethernet interface (using a USB/Ethernet adapter), you can use either of the two interfaces for this remote control. You can also use the remote access capability (see further in 6, How to use the remote access service).

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.

5.1. Accessing the SCOOPY+ S html pages

Once the SCOOPY+ S is connected on an IP network, the first step is to get its IP address, from the menu: **Tools / About / Local IP¹**. Then, on the control device, launch the html browser and enter the IP address of the SCOOPY+ S in the "address" or "URL" field. This gives access to the html server integrated in SCOOPY+ S. The page which is displayed is similar to the following picture:



If needed, select another language by clicking the suitable flag (*this choice is not linked to the language selected for the menus on the front panel 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 exclusive access. Any connection request from another device will remove this access.

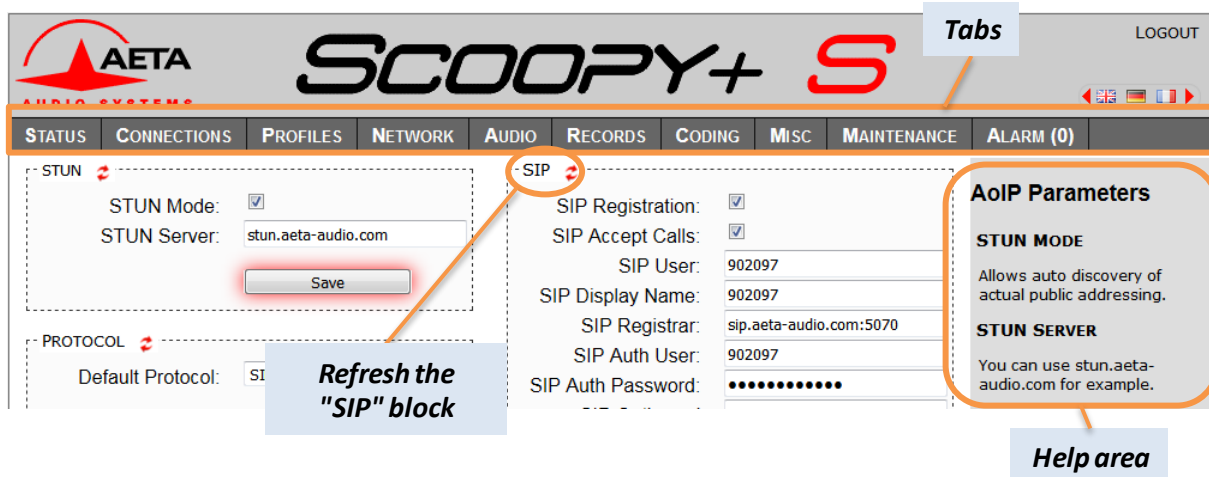
To log in, enter the password and click the connection button. The initial password is blank: click directly on the button. To set the password and enable protection, go to the "MAINTENANCE" page (cf. further).

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 For a possible secondary Ethernet interface, menu Config / Network / Ethernet 1 Parameters


5.2. 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 SCOOPY+ S. 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 automatically¹. To force a refresh, click the  icon: the data in the area or frame are read and refreshed.

i *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 SCOOPY+ S and the control device.*

Two tabs have a specific behavior:

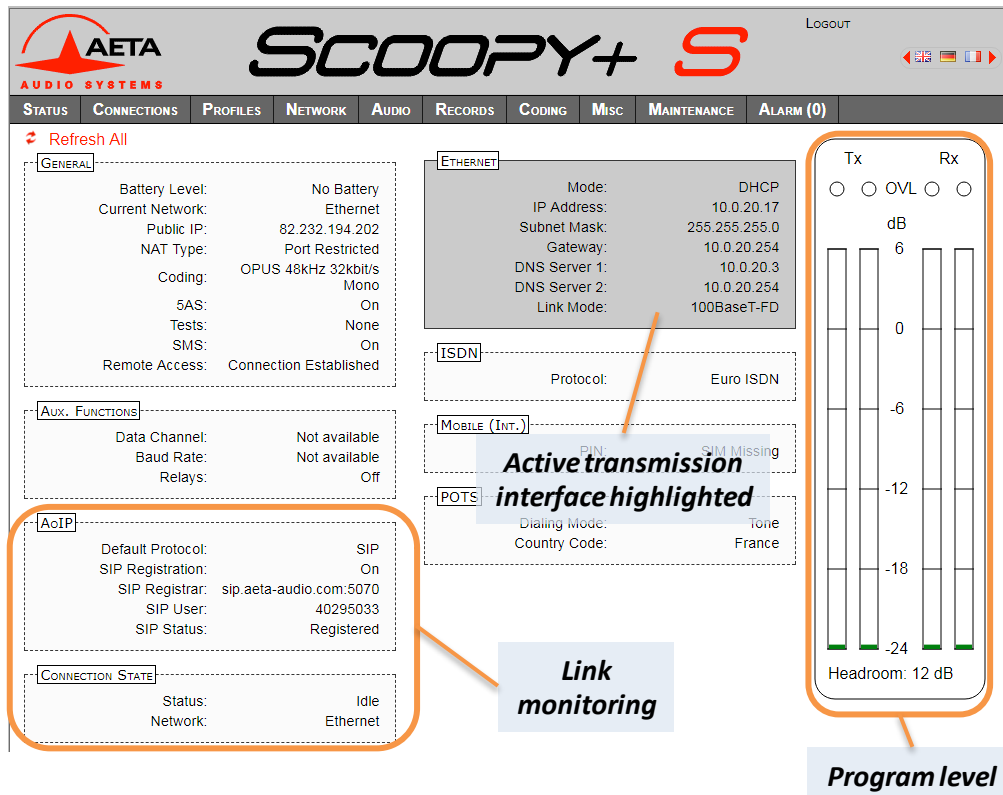
- "STATUS" is accessible without a login and some data are updated automatically.
- "ALARM" is also updated automatically, and goes red when an alarm is detected, while showing the number of errors detected. You can see details by clicking the tab.

Access to 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.

¹ This is on purpose, in order to avoid a permanent high rate of queries to the unit, which may be questionable in some situations.

5.3. "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.



The screenshot shows the SCOOPY+ S STATUS page with the following sections and parameters:

- GENERAL:** Battery Level: No Battery; Current Network: Ethernet; Public IP: 82.232.194.202; NAT Type: Port Restricted; Coding: OPUS 48kHz 32kbit/s Mono; 5AS: On; Tests: None; SMS: On; Remote Access: Connection Established.
- ETHERNET:** Mode: DHCP; IP Address: 10.0.20.17; Subnet Mask: 255.255.255.0; Gateway: 10.0.20.254; DNS Server 1: 10.0.20.3; DNS Server 2: 10.0.20.254; Link Mode: 100BaseT-FD.
- ISDN:** Protocol: Euro ISDN.
- MOBILE (Int.):** PIN: SIM Missing; SIM Missing: SIM Missing.
- POTS:** Dialing Mode: Tone; Country Code: France.
- AUX. FUNCTIONS:** Data Channel: Not available; Baud Rate: Not available; Relays: Off.
- AoIP:** Default Protocol: SIP; SIP Registration: On; SIP Registrar: sip.aeta-audio.com:5070; SIP User: 40295033; SIP Status: Registered.
- CONNECTION STATE:** Status: Idle; Network: Ethernet.

Annotations on the screenshot include:

- "Active transmission interface highlighted" pointing to the MOBILE (Int.) section.
- "Link monitoring" pointing to the AoIP section.
- "Program level" pointing to the audio bargraphs on the right.

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) and the two received signals (decoder), shown as bargraphs with a 0 dB reference. Refer to chapter 4.4.3 (Audio sub-menu) for more details on the setting for this reference. The "Headroom" setting is adjustable using the "AUDIO" tab.

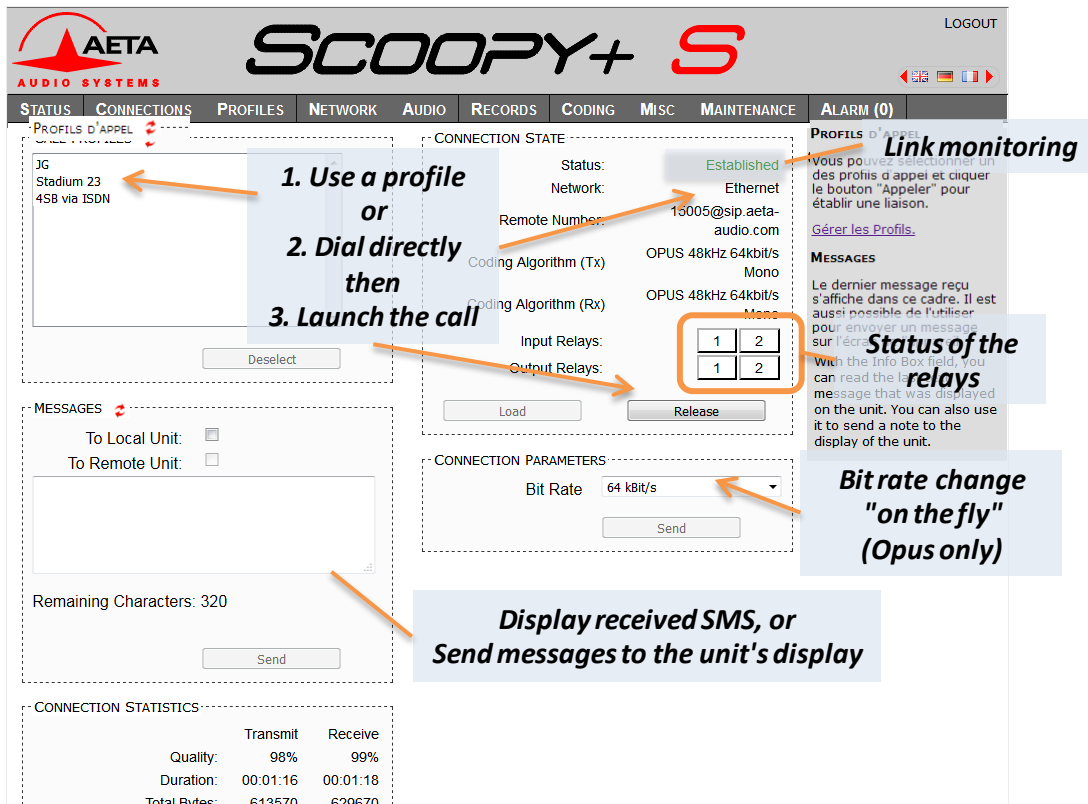
i Note: the goal of these bargraphs is to provide an indication on the presence and level of the audio modulation. The measurement has a good precision for stationary signals (1 dB accuracy and resolution), but the update rate is rather slow. As a consequence these bargraphs are not appropriate for precisely 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 4.11.1, "Display statistics about the IP transmission".

i The transmission quality indicator is not always available; this depends on the capability of the remote codec.

5.4. "CONNECTIONS" tab

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



The screenshot shows the SCOOPY+ S interface with the CONNECTIONS tab selected. The interface is divided into several sections:

- PROFILS D'APPEL:** A list of call profiles, including "JG Stadium 23 4SB via ISDN". An arrow points to this list with the text: "1. Use a profile or 2. Dial directly then 3. Launch the call".
- CONNECTION STATE:** Displays the current call status as "Established" over "Ethernet". It shows the "Remote Number" as "13005@sip.aeta-audio.com" and the "Coding Algorithm" as "OPUS 48kHz 64kbit/s Mono". Below this, there are "Input Relays" and "Output Relays" sections, each with a 2x2 grid of buttons labeled "1" and "2". An arrow points to these buttons with the text: "Status of the relays".
- CONNECTION PARAMETERS:** Shows the "Bit Rate" set to "64 kBit/s". An arrow points to this dropdown menu with the text: "Bit rate change 'on the fly' (Opus only)".
- MESSAGES:** A section for sending and receiving messages. It includes checkboxes for "To Local Unit" and "To Remote Unit", a text input field, and a "Send" button. An arrow points to this section with the text: "Display received SMS, or Send messages to the unit's display".
- CONNECTION STATISTICS:** A table showing transmission quality and duration.

	Transmit	Receive
Quality:	98%	99%
Duration:	00:01:16	00:01:18
Total Bytes:	613570	690670

The "Connection State" block allows to monitor and/or manage 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 nStauyumber (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 "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, detailed information on the transmission quality is displayed on this page (IP links only, "CONNECTION STATISTICS" frame), 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".

The SMS received by SCOOPY+ S are displayed in the "Messages" frame (in addition to being displayed on the unit's display). This block can be used also to send a text to the unit's screen from the control position.

5.5. "PROFILES" tab

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

5.5.1. "CALL PROFILES" page



CALL PROFILES

You can manage the call profiles used for setting up outgoing calls. Click on a profile and you can read its corresponding settings. You can create/edit/delete...

Profiles, presets and snapshots can be imported/exported from/to your computer. Go to Maintenance/Setup transfer.

PARAMETERS

Network: Ethernet
 Coding: OPUS
 Channel Mode: Mono
 Sampling Rate: 48 kHz
 Bit Rate: 64 kBit/s
 FEC: -
 Remote Number: 01421323594

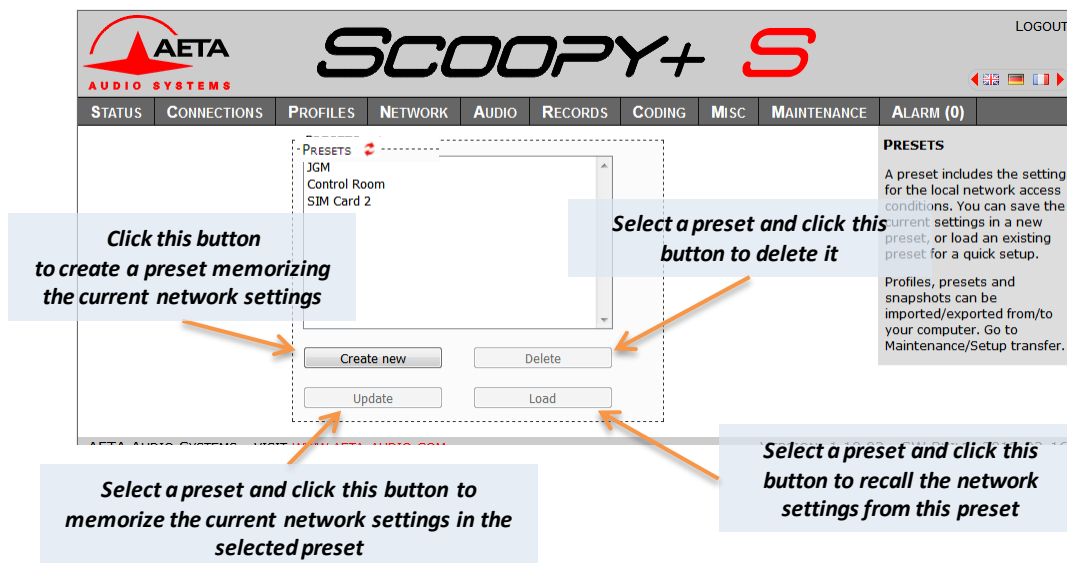
Buttons: Delete, Update, Create new

Callouts:

- 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.

5.5.2. "PRESETS" page



PRESETS

A preset includes the settings for the local network access conditions. You can save the current settings in a new preset, or load an existing preset for a quick setup.

Profiles, presets and snapshots can be imported/exported from/to your computer. Go to Maintenance/Setup transfer.

Buttons: Create new, Delete, Update, Load

Callouts:

- Click this button to create a preset memorizing the current network settings
- Select a preset and click this button to delete it
- Select a preset and click this button to memorize the current network settings in the selected preset
- Select a preset and click this button to recall the network settings from this preset

5.5.3. Snapshots

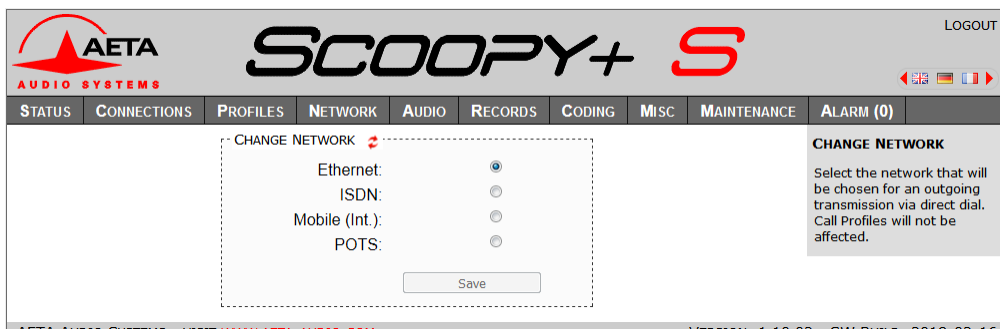
Snapshots are in fact managed on the "AUDIO" page, and this "SNAPSHOTS" choice on the "PROFILES" tab just redirects to the "AUDIO" page. See further the description of this page.

5.6. "NETWORK" tab

This tab leads to several pages, which correspond, just like the sub-menu "Network" of the front panel interface (see 4.4.1 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 ISDN, mobile access, etc.).

5.6.1. "CHANGE NETWORK" page: default interface selection

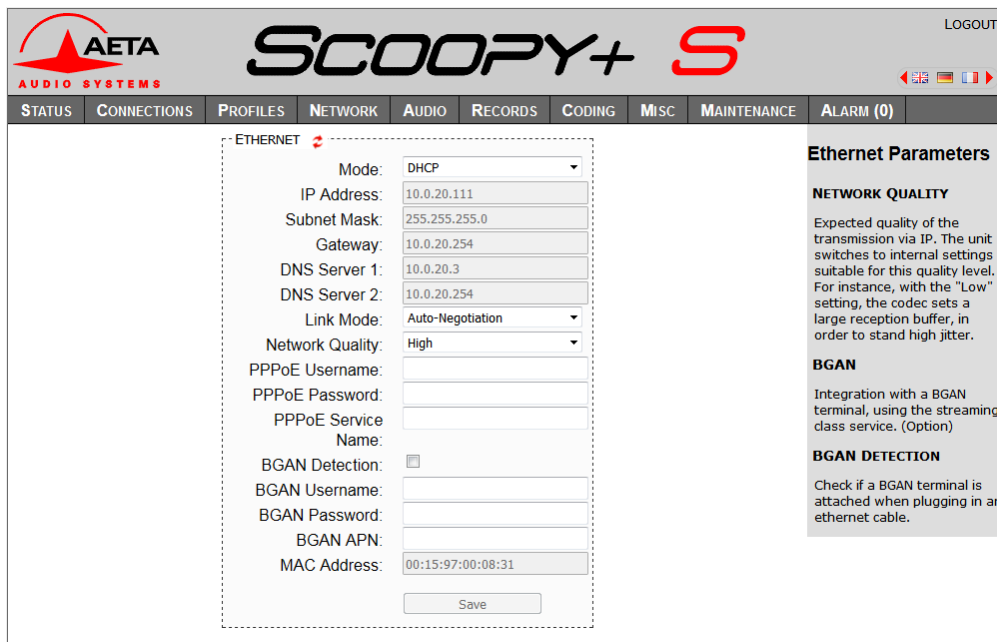
This page is used for selecting the default interface: this is the one that is implicitly used when dialing to set an outgoing call. Besides, codec algorithm settings (either via menus or the html pages) are applicable to this interface¹.



Select the desired interface and click "Save".

5.6.2. "ETHERNET PARAMETERS" page

More details on these parameters in 4.4.1, Ethernet Parameters.



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

i Settings regarding the Ethernet interface and IP addressing are especially critical, because this is just the interface you are controlling SCOOPY+ S 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!

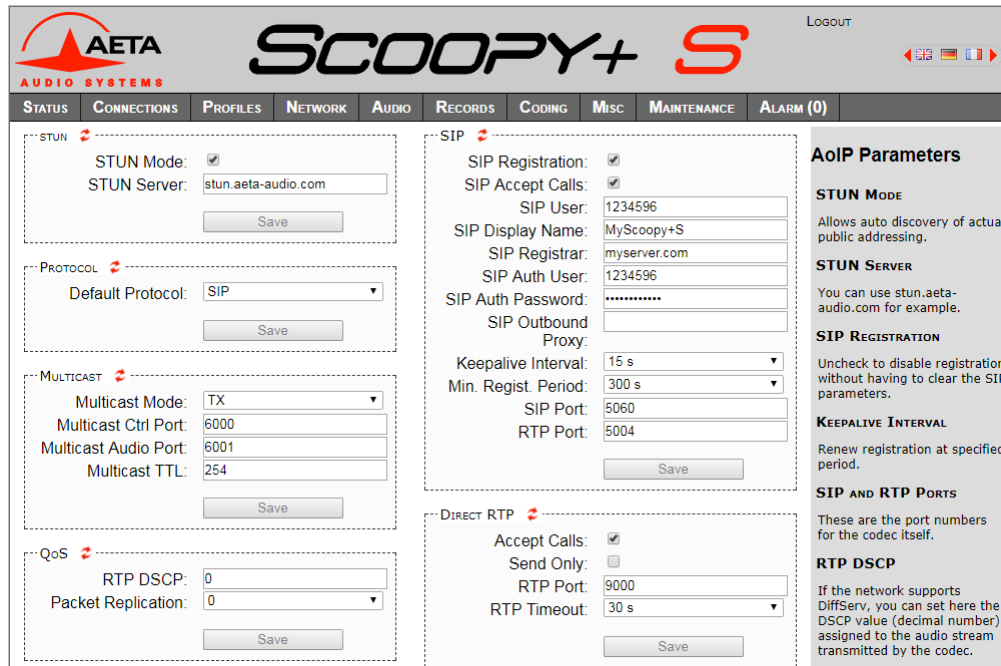
¹ Remind that the algorithm settings are memorized separately for each network interface type. When switching the default interface, the codec settings for this interface type are recalled.

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

Note: if a secondary Ethernet interface is available, it is also shown in this page, in an "ETHERNET 1" frame.

5.6.3. "AOIP PARAMETERS" page

More details on these parameters in 4.4.1, AoIP Parameter.



The screenshot shows the 'AOIP PARAMETERS' page in the SCOOPY+ S web interface. The page is organized into several sections, each with a 'Save' button:

- STUN:** STUN Mode (checked), STUN Server (stun.aeta-audio.com).
- PROTOCOL:** Default Protocol (SIP).
- MULTICAST:** Multicast Mode (TX), Multicast Ctrl Port (6000), Multicast Audio Port (6001), Multicast TTL (254).
- QoS:** RTP DSCP (0), Packet Replication (0).
- SIP:** SIP Registration (checked), SIP Accept Calls (checked), SIP User (1234596), SIP Display Name (MyScoop+S), SIP Registrar (myserver.com), SIP Auth User (1234596), SIP Auth Password (masked), SIP Outbound Proxy (empty), Keaplive Interval (15 s), Min. Regist. Period (300 s), SIP Port (5060), RTP Port (5004).
- DIRECT RTP:** Accept Calls (checked), Send Only (unchecked), RTP Port (9000), RTP Timeout (30 s).

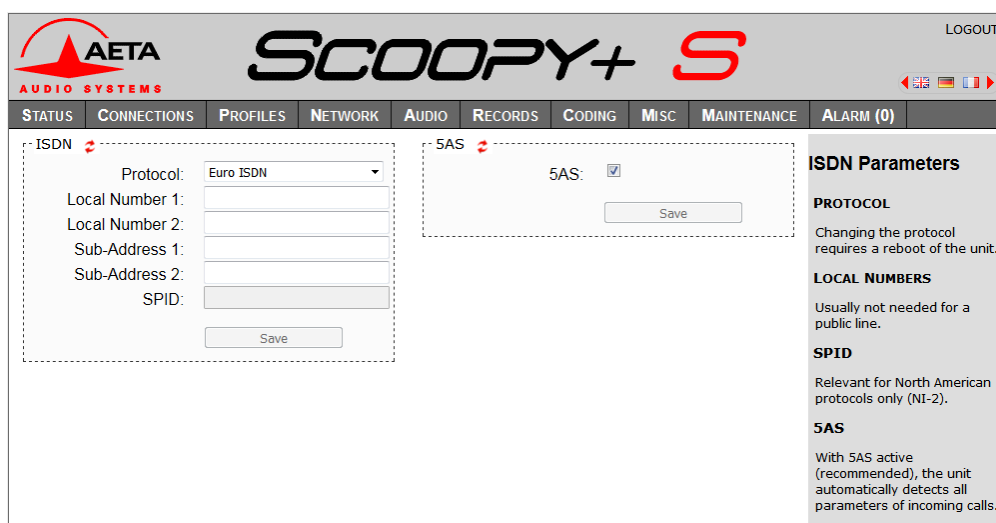
The right sidebar provides detailed information for each section:

- STUN MODE:** Allows auto discovery of actual public addressing.
- STUN SERVER:** You can use stun.aeta-audio.com for example.
- SIP REGISTRATION:** Uncheck to disable registration without having to clear the SIP parameters.
- KEAPLIVE INTERVAL:** Renew registration at specified period.
- SIP AND RTP PORTS:** These are the port numbers for the codec itself.
- RTP DSCP:** If the network supports DiffServ, you can set here the DSCP value (decimal number) assigned to the audio stream transmitted by the codec.

ⓘ Remember to save the changes, clicking the "Save" buttons!

5.6.4. "ISDN PARAMETERS" page

This page is only visible for units equipped with an ISDN interface. See more details on the parameters in 4.4.1, ISDN Parameters.



The screenshot shows the 'ISDN PARAMETERS' page in the SCOOPY+ S web interface. The page is organized into two main sections, each with a 'Save' button:

- ISDN:** Protocol (Euro ISDN), Local Number 1, Local Number 2, Sub-Address 1, Sub-Address 2, SPID.
- 5AS:** 5AS (checked).

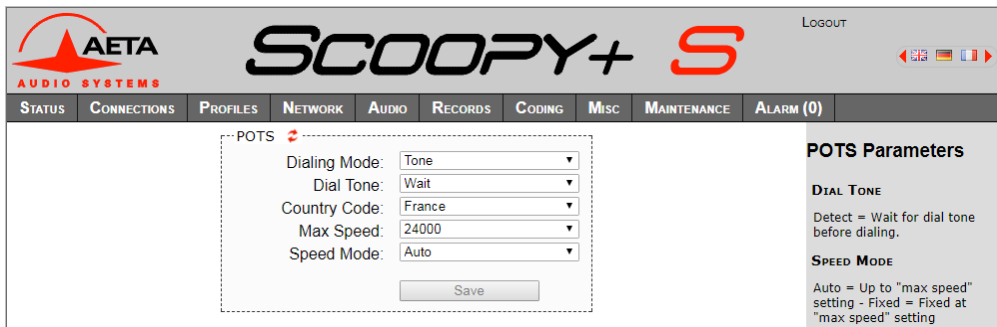
The right sidebar provides detailed information for each section:

- PROTOCOL:** Changing the protocol requires a reboot of the unit.
- LOCAL NUMBERS:** Usually not needed for a public line.
- SPID:** Relevant for North American protocols only (NI-2).
- 5AS:** With 5AS active (recommended), the unit automatically detects all parameters of incoming calls.

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

5.6.5. "POTS PARAMETERS" page

This page is only visible for units equipped with the POTS option. *See more details on the parameters in 4.4.1, POTS Parameters.*

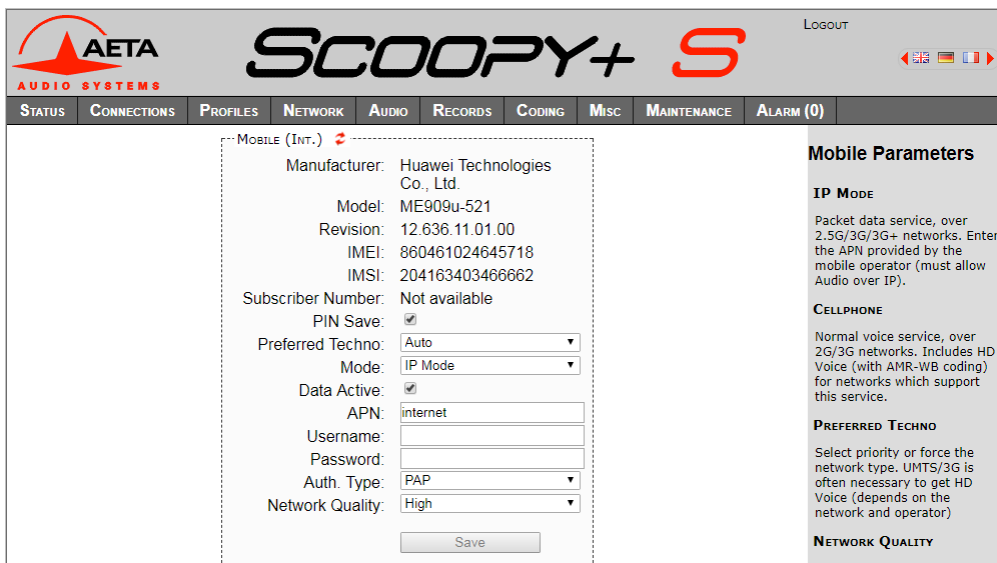


i Remember to **save the changes**, clicking the "Save" button!

5.6.6. "MOBILE PARAMETERS" page

This page is only visible for units equipped with the "HD-4G Mobile" option. See more details on the parameters in 4.4.1, Mobile Parameters. You cannot access all settings unless the PIN code has been entered (otherwise the view is partial only).

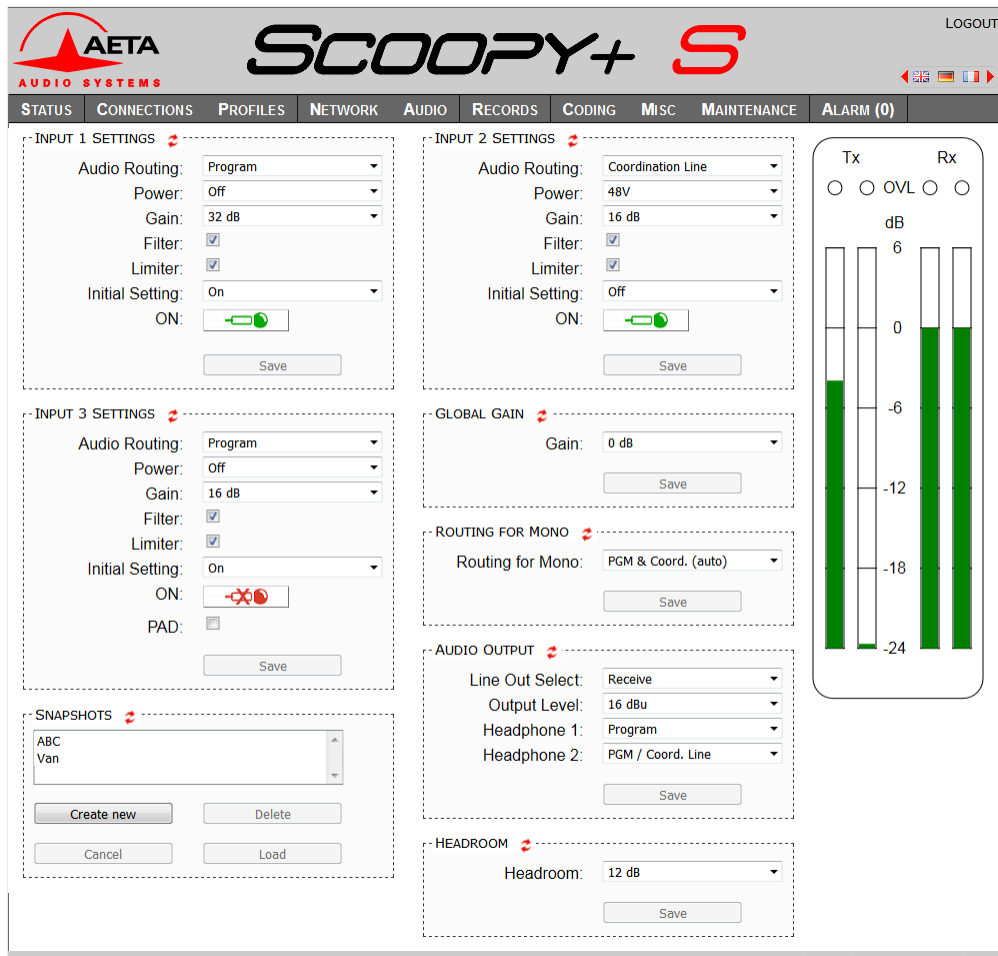
i If you have plugged a USB mobile access device, it is also assigned a specific parameter page.



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

5.7. "AUDIO" tab

See all details for these parameters in 4.4.3, [Audio](#) sub-menu. Like the "STATUS" page, this page shows the audio level transmitted/received to/from the network.



The screenshot displays the "SCOOPY+ S" interface with the "AUDIO" tab selected. The interface is divided into several sections:


- INPUT 1 SETTINGS:** Audio Routing: Program, Power: Off, Gain: 32 dB, Filter: [checked], Limiter: [checked], Initial Setting: On, ON: [button].
- INPUT 2 SETTINGS:** Audio Routing: Coordination Line, Power: 48V, Gain: 16 dB, Filter: [checked], Limiter: [checked], Initial Setting: Off, ON: [button].
- INPUT 3 SETTINGS:** Audio Routing: Program, Power: Off, Gain: 16 dB, Filter: [checked], Limiter: [checked], Initial Setting: On, ON: [button], PAD: [checkbox].
- GLOBAL GAIN:** Gain: 0 dB.
- ROUTING FOR MONO:** Routing for Mono: PGM & Coord. (auto).
- AUDIO OUTPUT:** Line Out Select: Receive, Output Level: 16 dBu, Headphone 1: Program, Headphone 2: PGM / Coord. Line.
- HEADROOM:** Headroom: 12 dB.
- SNAPSHOTS:** List containing "ABC" and "Van". Buttons: Create new, Delete, Cancel, Load.
- LEVEL METER:** Tx and Rx levels in dB, ranging from -24 to 6.

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.

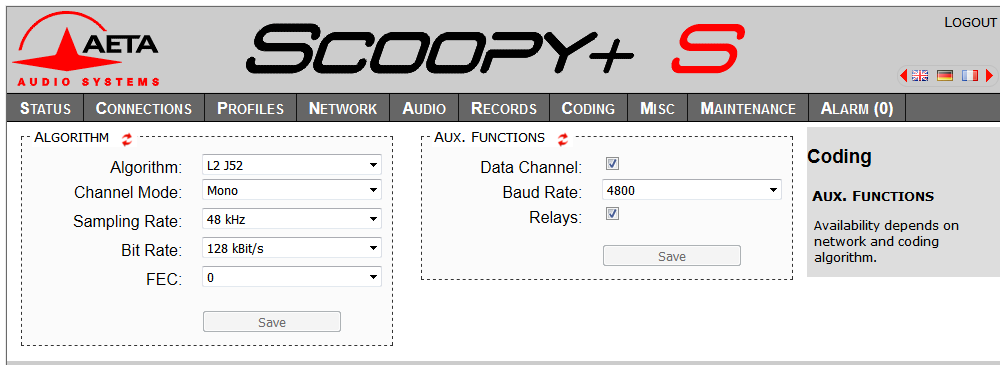
To **create** a snapshot based on the current settings, click the "Create new" button, and enter a name for the snapshot.

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.

5.8. "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 4.4.2, [Algorithm](#) sub-menu and 4.4.7, [Misc](#) 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.

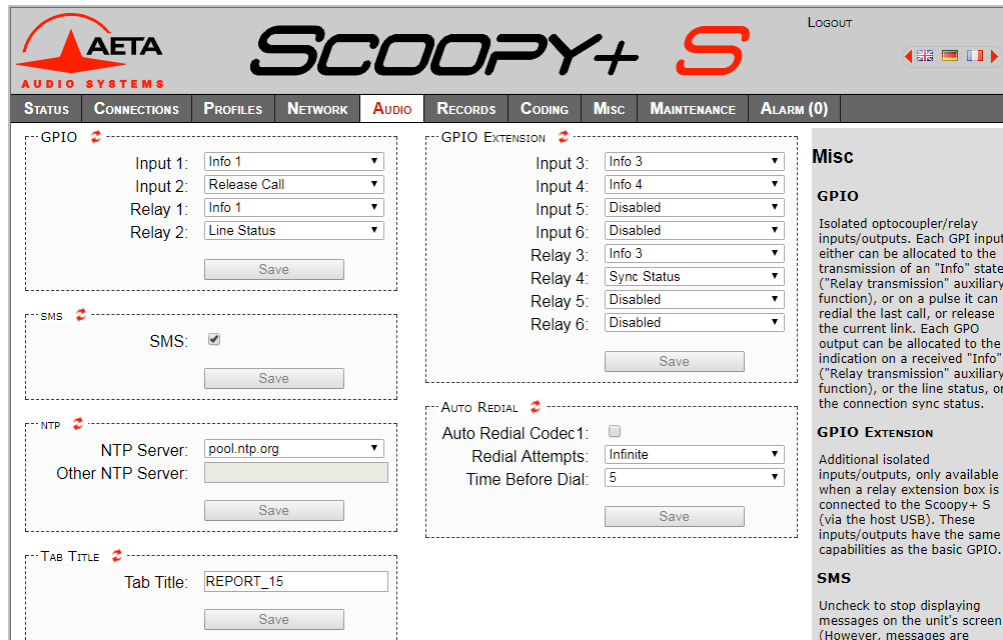
The settings are saved separately for each network interface type. This means that when you change the current network interface, the coding configuration that is related to it is recalled. Example:

- Current interface = ISDN, selected configuration 4SB ADPCM.
- 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 4SB ADPCM setting is recalled.
- Switch the network interface and select an IP interface: the Opus configuration is restored.

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

5.9. "Misc" tab

This **Misc** tab groups several settings that can be found in the "**Misc**" sub-menu of the front panel user interface (see 4.4.7, **Misc** sub-menu): auto redialing configuration and activation, GPIO configuration, SMS reception enabling/disabling...



The screenshot shows the 'Misc' tab configuration page in the SCOOPY+ S web interface. The page is divided into several sections, each with a 'Save' button:

- GPIO:** Input 1: Info 1, Input 2: Release Call, Relay 1: Info 1, Relay 2: Line Status.
- SMS:** SMS:
- NTP:** NTP Server: pool.ntp.org, Other NTP Server: (empty field).
- TAB TITLE:** Tab Title: REPORT_15.
- GPIO EXTENSION:** Input 3: Info 3, Input 4: Info 4, Input 5: Disabled, Input 6: Disabled, Relay 3: Info 3, Relay 4: Sync Status, Relay 5: Disabled, Relay 6: Disabled.
- AUTO REDIAL:** Auto Redial Codec 1: , Redial Attempts: Infinite, Time Before Dial: 5.

The sidebar on the right contains the following text:

Misc

GPIO

Isolated optocoupler/relay inputs/outputs. Each GPI input either can be allocated to the transmission of an "Info" state ("Relay transmission" auxiliary function), or on a pulse it can redial the last call, or release the current link. Each GPO output can be allocated to the indication on a received "Info" ("Relay transmission" auxiliary function), or the line status, or the connection sync status.

GPIO EXTENSION

Additional isolated inputs/outputs, only available when a relay extension box is connected to the Scoopy+ S (via the host USB). These inputs/outputs have the same capabilities as the basic GPIO.

SMS

Uncheck to stop displaying messages on the unit's screen. (However, messages are

Two items come in addition:

- Configuration of an 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 5.10.5, "**EVENT LOG**" page). If such a server is available and accessible via the Ethernet interface, enter its address in the "Other NTP Server" field. A few public servers are also proposed in the dropdown list¹.

i A reboot is needed for this setting to become effective!

- **TAB TITLE:** the text entered here is displayed on the tab of the html browser (the page should be refreshed in order to apply the setting). When several units are monitored/controlled from the browser, this will help to identify the units by assigning them distinctive names.

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

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 the symbolic names can be resolved.

5.10. "MAINTENANCE" tab

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

5.10.1. "LOGIN DATA" page

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

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

If you lose the password: using the front panel interface of SCOOPY+ S, 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.

You can also configure on this page the connection to an FTP server (used to upload recorded audio files). The common parameters can be found: address or domain name of the server, username and password.

Remember to **save changes**, clicking the relevant "Save" buttons!

5.10.2. "TESTS" page

This page provides the same choice as the **Maintenance / Tests** sub-menu (see in 4.4.8 and 2.5.7, "Test functions", the description of the various test loops available).

i *Do not forget to disable the test loops (setting "None") to come back to normal operation!*

5.10.3. "SYSTEM UPDATE" page

This page allows you to upload a file into the unit and update the firmware.

The firmware update file must be provided to you by AETA (usually available for downloading from the web site). The procedure is rather simple:

- **First read carefully the instructions and warnings** provided with the update file. In some cases, you might have to do some preparatory action.
- From this "SYSTEM UPDATE" 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 SCOOPY+ S front panel.
- Wait for the completion of the update. Normally the unit should reboot by itself at the end of the process.
- Usually the memorized settings are still valid after such update. However, for maximum safety you can perform a complete "Factory reset".

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 SCOOPY+ S is stable (no interruption should happen during the process).
- Make sure the SCOOPY+ S is not powered off and its Ethernet interface is not disconnected during the process, do not use its keypad during the process.
- If needed, re-import your profiles, presets and snapshots after the update. *But be aware that, although we try to keep a maximum upward compatibility, AETA cannot guarantee the re-usability and consistency of these memories after a firmware update.*

5.10.4. "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:

- **"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.*
- **"Reset Settings"**: all parameters are brought back to their default setting, but all memories are kept: call profiles, presets and snapshots.
- **"Reset Presets+CallProf"**: call profiles and presets are cleared from the memory.
- **"Reset Snapshots"**: all snapshots are cleared from the memory.
- **"Factory SIP Accounts"**¹: reloads the credentials for the "factory" SIP account of the unit, and registers the unit on the AETA SIP server.

The page also allows you to reboot the unit by remote, clicking the **"Reboot"** button. The device is actually reboot when you confirm by clicking "OK" in the dialog box which then shows up². This is equivalent for SCOOPY+ S 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.

¹ This feature may not be available on older units.

² If you mistakenly clicked the "Reboot" button, it is still possible to cancel this: close the page without clicking "OK" (do not hit the computer's Esc key neither).

5.10.5. "EVENT LOG" page

This page displays a history of 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.

SCOOPY+ S can also send event messages to a SYSLOG server if such server is available on the network: enter the address or name of this server and click the "**Apply**" button. From then on, SCOOPY+ S sends the designated server all 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 SCOOPY+ S 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 5.9 above).

i *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.*

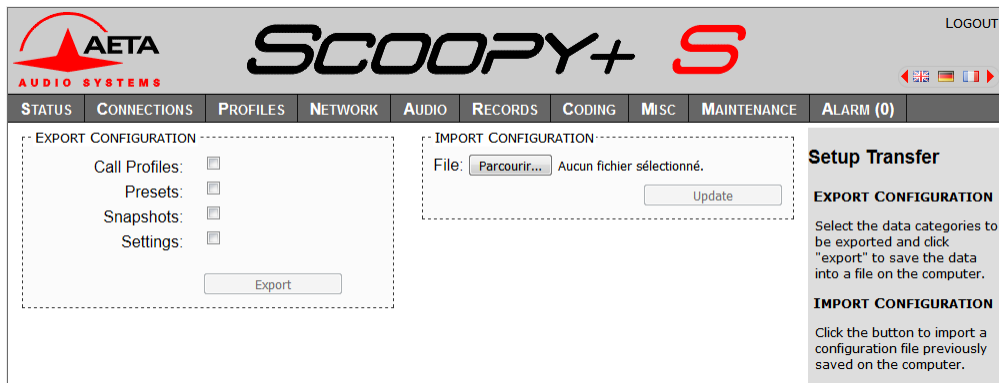
Under the log window you can find some control buttons:

- "**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.
- "**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.
- "**Save logfile 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...

5.10.6. "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, snapshots, settings...



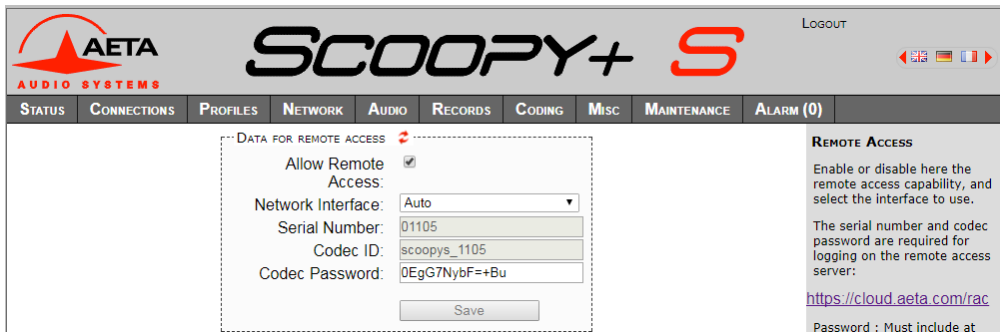
The left part of the page is dedicated to "exporting" the device configuration: the configuration is then saved in a file, downloaded in the control computer. Before doing the export you can select which type of data is to be included in this backup/export: check the desired categories (multiple choice 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 "**Update**". The file must be a file previously exported from a SCOOPY+ S (same unit, or another). The settings or memories included in the file are respectively applied to the codec, or recorded in its memory of profiles/snapshots.

i *Warning: the export files can be imported into a device with the same firmware version, 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.*

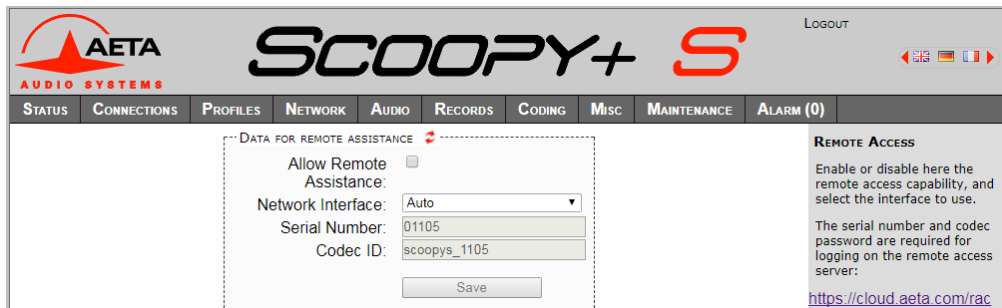
5.10.7. "REMOTE ACCESS" page

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 > Maintenance > 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:



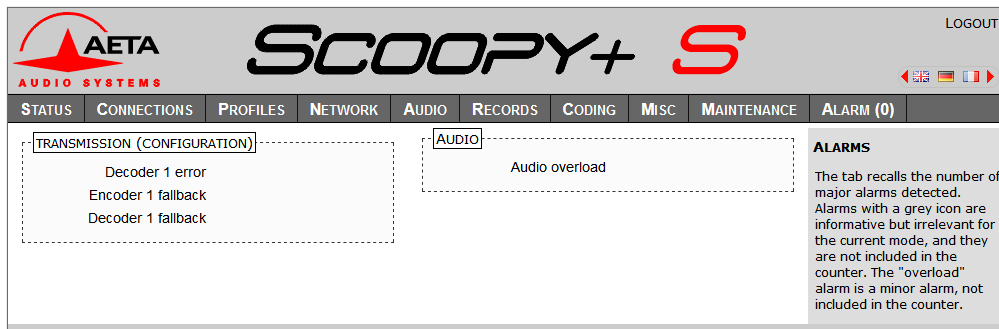
- The "Codec Password" is not found if no "remote access" option is installed. One must know this password for getting remote access to the html pages, so it protects the unit against undesired access. No such password exists if the "remote access" option is not installed.


i 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 detailed procedure for using remote access in chapter 6, "How to use the remote access service".

5.11. "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 in brackets.



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 clipping.

6. How to use the remote access service

6.1. 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 SCOOPY+ S:

- Up-to-date firmware (1.09 minimum). No additional option is required.
- Internet access, for instance via an Ethernet connection to a LAN. But other IP connections are suitable as well: mobile data access...

6.1.1. Step 1: connect the SCOOPY+ S to the server

Menu **Tools** > **Maintenance** > **Remote Assistance**: activate "**Remote Assistance**".

If a "Remote Access" option is installed, the menu is **Tools** > **Maintenance** > **Remote Access**, activate "**Allow Remote Access**".

If necessary, select the desired IP interface to be used for the remote access connection: "**Network Interface**" (with the "**Auto**" choice, it will be the same that is used for audio over IP transmission).

To check the successful connection: menu **Tools** > **About**, "**Remote Access Status**".

Take note of the MAC address of the main Ethernet interface: menu **Tools** > **About**.

Variation: you can also use the "**REMOTE ACCESS**" html page for this step:

Maintenance tab, **REMOTE ACCESS** page, check "**Allow Remote Assistance**".

If a "remote access" option is installed, **MAINTENANCE** tab, **REMOTE ACCESS** page, check "**Allow Remote Access**".

On the same page, you can select the IP interface that the codec will use for connecting to the service: "**Network Interface**". For using always the same interface as for AoIP transmission, select "**Auto**".

To check the successful connection: **STATUS** page, **GENERAL** section.

Take note of the MAC address of the main Ethernet interface: **NETWORK** tab, **ETHERNET PARAMETERS** page.

6.1.2. Step 2: open the remote session

Tell the AETA technician the MAC address of the codec; this is the unique 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.*

6.2. Using the remote html access

This service allows to access the html pages of the codec from remote.

What you need on the SCOOPY+ S:

- Up-to-date firmware (1.09 minimum).
- 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 ...

The control device can be any device with a web browser (no need to install any application): computer, tablet or even a smartphone...

6.2.1. Step 1: connect the SCOOPY+ S to the server

First you must connect the codec to the AETA server so that it is accessible. To protect the access to the unit, you can define a "codec password" that allows to get control over it.

Note: as a security measure, a password that is blank or too short is not accepted by the system (see more on this topic in 6.3, Security aspects).

Menu **Tools** > **Maintenance** > **Remote Access**: activate "**Remote Access**".

In the same menu, you can select the IP interface that the codec will use for connecting to the service: "**Network Interface**". For using always the same interface as for AoIP transmission, select "**Auto**".

To check the successful connection: menu **Tools** > **About**, "**Remote Access Status**".

If you want to change the password, go to the menu **Tools** > **Maintenance** > **Remote Access, Codec Password**. The following rules apply to this password:

- 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 SCOOPY+ S.

Variation: you can also use the "**REMOTE ACCESS**" html page for this step:

Maintenance tab, **REMOTE ACCESS** page, check "**Allow Remote Access**".

On the same page, you can select the IP interface that the codec will use for connecting to the service: "**Network Interface**". For using always the same interface as for AoIP transmission, select "**Auto**".

To check the successful connection: **STATUS** page, **GENERAL** section.

You can change the password, 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 SCOOPY+ S.

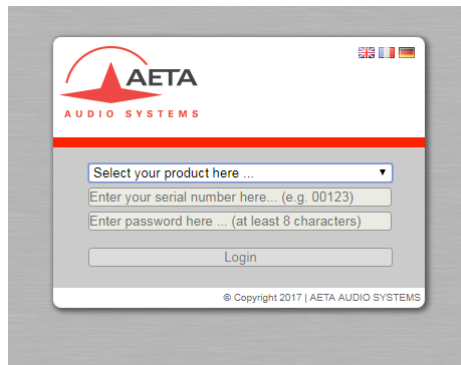
6.2.2. Step 2: open the remote session

The SCOOPY+ S 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:

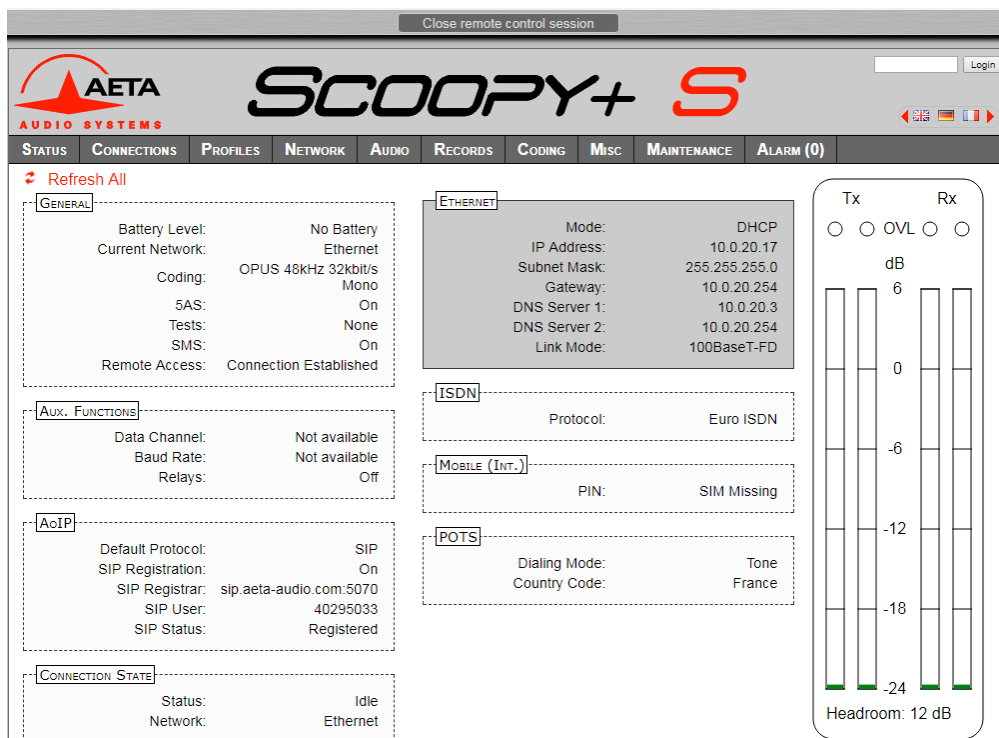
<https://cloud.aeta.com/rac>

You reach the AETA remote html access portal:



1. Select the product type (SCOOPY+ S) 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.

i *Warning: in this remote control mode, remember that any change on the "REMOTE ACCESS" page may cause you to lose control! In the best case you will need to reconnect to the remote access server, but the loss may be final if the codec can no longer reset its connection to the server. In such case a local intervention on the unit would be needed.*

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 SCOOPY+ S once the session is over. *This is recommended for avoiding useless data traffic.*

6.3. Security aspects

By its very principle, the remote access feature, as it makes possible remote control via Internet from any location, might imply security risks, for example:

- Undesired takeover of a device by an unauthorized person;
- Eavesdropping the transactions;
- Intrusion into a codec or a control device.

6.3.1. Protection measures

The AETA remote access system has several provisions to avoid these risks, including:

- The **connection between the codec and the server** is authenticated and encrypted (SSH).
- This connection requires a **voluntary activation** on the codec. As long as the feature is not activated, no one can take control of the codec via the Internet. The same is true for remote assistance and remote access by AETA staff. *For even more security, you may enable remote access only for the duration of an intervention.*
- The **connection between the control device and the server** is authenticated and encrypted (HTTPS and/or SSH).
- The **codec password** ensures that only authorized people can act on the codec, even when remote access is active. For this reason, an empty or short password is not accepted by the system. On the contrary, it is highly recommended to use a complex password. We recommend at least 13 characters, with a combination of uppercase and lowercase letters, numbers, and special characters among the following: _ - + * . : , ; # ? ! =.

6.3.2. Tips for firewalls

It is not necessary to create static routes or redirections from public ports to codecs (this is even not recommended), as it is the codec or the controller that initiates a connection to the server.

If a firewall prevents this connection, it may be necessary to enable packet exchanges with the AETA server: domain cloud.aeta.com (or, at the time of writing, the IP address 80.154.26.106).

7. Technical characteristics

7.1. Characteristics of the audio interfaces

7.1.1. Mic/line inputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated.

The inputs are balanced type, using 3-pin female XLR sockets.

Microphone powering:	Phantom 48V
Maximum input level:	+19 dBu (+39 dBu for channel 3 with Pad)
Input stage sensitivity adjustment:	+0 to +48 dB by 16dB steps
CMRR:	>80dB @ 1kHz
Nominal input impedance:	10 k Ω

7.1.2. Line outputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated. The outputs are balanced type, using 3-pin male XLR sockets.

Maximum output level:	adjustable from -11 to +22 dBu
Nominal load impedance:	600 Ω or 10 k Ω
Output impedance:	<50 Ω
Symmetry:	> 40 dB (Z _L = 150 Ω)

7.1.3. Headphone outputs

Theses outputs (6.35 mm jack on front panel) are for the connection of a 32 Ω headphone. It is also possible to plug a high impedance headphone; however, the maximum available power will be lower.

7.1.4. Audio performance

The audio performance in this part applies to the system without coding/decoding. The additional effect of the audio encoding and decoding on audio performance depends on the coding algorithm used and its parameters.

Except when differently stated, the following measurements are done at a +6 dBu input level and on the AD/DA path, with maximum input and output level set at +16 dBu.

Amplitude-frequency response

All measurements are done with a +6 dBu input signal, and a reference frequency of 1020 Hz. The measurements are done with a loopback before coding/decoding, so the possible effect of compression has no influence.

For a 48 kHz sampling rate:

Frequency range		Tolerance	
0 Hz	20 Hz	$-\infty$	0 dB
20 Hz	40 Hz	-0.2 dB	0.1 dB
40 Hz	20 000 Hz	-0.1 dB	0.1 dB

Group delay distortion

Taking the minimum group delay as reference, the group delay distortion on the AD/DA path is always less than 1 ms.

Idle channel noise

Background noise is measured with no audio modulation (idle channel), with maximum input and output level set at +16 dBu, through the whole encoder-decoder chain (wide band coding, with 48 or 32 kHz coding frequency).

Maximum noise level¹: - 56 dBm (quasi-peak detection, CCIR weighting)
(or - 62 dBq0ps)

This result in a signal to noise ratio (SNR) of more than 72 dB.

When the maximum input and output level is set at another level, both the signal and noise levels are shifted but the SNR remains in the same range.

Total distortion vs frequency and level

Total distortion relative to maximum level (or THD + N) is less than -78 dB over the whole audio bandwidth (20 – 20 000 Hz). This performance holds for audio signals from -80 dB to -1 dB relative to the maximum level (+16 dBu).

Crosstalk

Crosstalk is less than -80 dB over the whole bandwidth.

Gain and phase difference between channels

The gain difference between channels is less than ± 0.3 dB over the whole bandwidth, for any sampling frequency.

The phase difference between channels is less than ± 3 degrees over the whole bandwidth, for any sampling frequency.

¹ Worst case for all types of algorithms; MPEG algorithms performs better than the others

7.2. Characteristics of network interfaces

7.2.1. Ethernet Interface

This RJ45 socket has standard Ethernet pinout (for use of a normal "straight" cable to an Ethernet hub or switch). The interface normally operates in 100BaseT full-duplex mode, with auto negotiation, but it is possible to force other configurations.

The installation and operation of this function is detailed in 3.4, Initial setup of the Ethernet interface.

7.2.2. ISDN interfaces

This interface is optional.

The socket is an ISDN S0 interface (BRI), the supported protocols are ETSI (Euro RNIS), NI-1 or NTT.

7.2.3. POTS/PSTN interface

Interface available as an option.

The telephone interface is a 4-pin RJ11, only Tip and Ring signal are used:

Pin	Description
1	-
2	TIP
3	RING
4	-

7.2.4. Antenna socket (mobile networks)

Available on units equipped with the "HD-4G Mobile" option. See also chapter 2.1.4: Mobile network access.

This socket is used to connect an external antenna. This is a male HIROSE MS-151NB (female contact), 50 Ω impedance.

The internal mobile access module can operate in following bands:

Frequency band	800 MHz	850 MHz	900 MHz	1800 MHz	1900 MHz	2100 MHz	2600 MHz
(n°)	(20)	(5)	(8)	(3)	(2)	(1)	(7)
2G GSM/EDGE							
3G/3G+ UMTS/HSDPA/HSUPA/HSPA/HSPA+							
4G/LTE							

7.2.5. Network protocols and ports

The Scoopy+ S 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: HTTP, HTTPS, DHCP, STUN, NTP, SYSLOG
- Audio transmission: SIP signaling, SDP, RTP, RTCP, RFC3550/3551, RFC3640
- Compliant with EBU recommendation Tech 3326 (interoperability of audio codecs for contribution)

The ports used by the device are the following:

Type	Port	Designation	Dir	Notes
TCP	80	HTTP	↔	Embedded html server
	443	HTTPS	↔	Embedded html server
	7001	Control	↔	Remote control; used by Scoop Manager
UDP	123	NTP	↔	For querying an NTP server
	514	SYSLOG	→	For sending messages to a SYSLOG server
	2382	AETA enumeration	↔	Used by "AetaScan"
	3478	STUN	↔	For querying a STUN server
	5004	RTP	↔	For audio streaming (SIP)
	5005	RTCP	↔	For audio streaming (SIP)
	5060	SIP	↔	SIP signaling
	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 table. Besides, these are mandatory for setting up AoIP links.

7.3. Miscellaneous interfaces

7.3.1. GPIO ("EXTENSION" socket)

The "EXTENSION" socket includes the GPIO interfaces: 2 inputs (optocouplers) and 2 relays.

Pin	Name	In/Out	Function and characteristics
1	+5V Source	Output	Source provided by the SCOOPY+ S; can be used for powering devices connected to this interface (low power)
2	OPTO1	Input	Bidirectional optocoupler input 1 (with common on 4). May be fed directly with +5V, control current 6 mA
3	OPTO2	Input	Bidirectional optocoupler input 2 (with common on 4). May be fed directly with +5V, control current 6 mA
4	Common (opto)	Input	Common node for optocoupler inputs
5	GND		Electrical ground of the unit
6	Relay1	Output	Relay output 1 (with common on 8), max switched current 120 mA, max voltage 350 Vp
7	Relay2	Output	Relay output 2 (with common on 8), max switched current 120 mA, max voltage 350 Vp
8	Common (relays)	Output	Common node for the relays

7.4. Power supply

The SCOOPY+ S is powered by rechargeable batteries or an external DC power adapter. The latter also recharges the batteries when it is present.

- i** *The changeover from the external DC source to the batteries, or vice versa, is automatic and causes no break in the operation.*

7.4.1. DC power supply

The SCOOPY+ S accepts an external 8V to 15V DC source. A suitable AC/DC adapter is provided with the unit, and spare power adapters are available (see 7.8, Accessories and related products).

The unit includes overvoltage and overcurrent protection (resettable fuse).

Characteristics	Min	Nominal	Max	Notes
Supply voltage	8 V	12 V	15 V	
Supply current	0.5 A		2 A	[1]

[1]: The power consumption may vary in a wide range depending on the input voltage, the installed options, the possible presence of batteries, and the operating mode.

Connection: 2.1 mm jack, with ground on the outer conductor.

7.4.2. Batteries

The unit can be powered by 6 type "C" or LR14 NiMH cells.

Only rechargeable NiMH batteries can be used, such as those provided by AETA as accessories (see further, 7.8).

Make sure to observe the proper polarity when inserting the batteries.

The autonomy depends on the transmission network and the algorithm used. A battery indicator on the display shows the remaining charge level. You can expect from 3 to 4 hours battery life with batteries in good condition and fully charged.

- i** *It is strongly recommended to remove the batteries when you are not using the device for a long time.*
- i** *It is strongly discouraged to use batteries of poor quality, especially saline batteries. Such batteries may leak and damage the SCOOPY+ S.*

7.5. Dimensions and weight

Dimensions: (D x W x H) 234 x 155 x 80 mm

Weight: 1.14 kg without batteries, 1.5 kg with batteries

7.6. 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 SCOOPY+ S complies with "CE" directives regarding safety and EMC.

- Safety: compliance with EN60950
- EMI: radiated emissions complying with EN55103-1
- Susceptibility: compliance with EN55103-2

7.7. Options

The standard version of SCOOPY+ S includes an Ethernet interface for IP transmission. Several options can be added to complement the capabilities of the SCOOPY+ S.

7.7.1. Network options

For adding more types of network interfaces, the following options are available:

Ordering code	Option	Description
80 00 243 01	ISDN	Addition of an ISDN interface, for 64 or 128 kbit/s transmission (1 or 2 B channels)
80 00 241 01	HD-4G mobile	Integrated access to mobile networks
80 00 244 01	"POTS" codec	Addition of the POTS/PSTN interface
80 00 245 01	Inmarsat BGAN	Direct control over an Inmarsat BGAN terminal, automatic use of the <i>streaming class</i>
80 00 241 61	USB mobile	Access to mobile networks with a USB key (not included)

7.7.2. Other options

Ordering code	Option	Description
80 00 246 01	Recording	Record and Edition of audio file
80 00 246 11	Remote Access	Additional 12V DC power supply input
80 00 053 01	Grounded DC adapter	Recommended for some Sennheiser headsets such as HMD24, HMD25...

7.8. Accessories and related products

The SCOOPY+ S is delivered with an AC/DC adapter and a CAT5 Ethernet cable.

The POTS option includes an RJ11 cable.

The "HD-4G Mobile" option is delivered with an external multiband antenna.

An SD Card is included with the "Recording" option.

Additional accessories or spare parts are available:

Code	Description	
80 00 146 01	SCOOPY+ S bag	
ZAAS00080	External multiband antenna	
80 00 147 01	Rechargeable Ni-MH battery set (2500 mA.h)	
80 00 030 01	12V car battery adapter	
80 00 017 11	Spare AC/DC adapter	
15 00 073 31	Grounded AC/DC adapter	
ZAAS00056	μSIM adapter (for using a μSIM card instead of the standard SIM format)	
ZAAS00055	Relay extension box	

 Contact your distributor for more accessories.

8. Annexes

8.1. Additional information on the algorithms and protocols used

8.1.1. Auxiliary data in the MPEG frames

The auxiliary data are used for the following purposes:

- Reed-Solomon error detection and correction (J52 standard)
- Data channel
- Other auxiliary information: relay transmission.
The insertion of this auxiliary information is an extension (AETA proprietary format) to MPEG. However, the frame structure remains compliant.

8.1.2. Reed-Solomon encoding (ISDN only)

In order to cope with possible transmission errors in the ISDN network, Reed-Solomon error correction coding can be added, compliant with J52 recommendation. Four correction modes are available in the SCOOPY+ S:

- Mode 0: no error correction, Reed-Solomon coding disabled
- Mode 1: protection of only the control information and scale factors in the MPEG frame, low redundancy
- Mode 2: protection of the whole frame, moderate (2.5 %) redundancy
- Mode 3: protection of the whole frame, high (10 %) redundancy

Higher redundancy increases the protection against errors, but slightly degrades the audio quality, as redundancy takes up part of the bit rate that could be allocated to audio coding.

Most often, for a normal quality transmission link, mode 1 is sufficient and it consumes little bit rate from the compressed data, so it hardly impacts the audio quality.

8.1.3. H221 framing

H221 defines a framing structure that allows byte synchronisation recovery, and the transmission of control data along with the main data.

8.2. Overview of the SIP protocol

8.2.1. What is SIP?

SIP is for Session Initiation Protocol, a protocol specified by the IETF for establishing media transmission sessions. SIP is considered the communication protocol of the future by most vendors, and as such it has deep influence on the VoIP applications.

As a signalling protocol, SIP brings methods and techniques to solve the issues related to the establishing of an audio link. Almost as important, it is a recognised standard, implemented on many network devices and systems. Using SIP helps you build modular and really evolutive systems, not being tied to a single vendor.

The SIP protocol is an essential requirement in the EBU Tech 3326 recommendation (a.k.a. "N/ACIP" recommendation, from the name of the EBU workgroup who elaborated it).

8.2.2. Setting a link with SIP

Let us look at an example (diagram below): a reporter on the move with a SCOOPY+¹ wants to make a call to a SIP compliant codec located in the main station. The reporter may be at home, or at another location, not necessarily known in advance.

Once the SCOOPY+ is on and connected to the network, it will register itself ❶ to a SIP "registrar". This registrar can be located on the LAN of the radio house, but it may as well be elsewhere in the network. Then the registrar "knows" where the SCOOPY+ is, what its IP address is. On the radio house side, a similar process takes place ❷.

To call the codec of the radio house (e.g. a Scoop 5), the reporter just needs to know its SIP address, which can be like studio12cod@radiomcr.com (indeed very similar to an e-mail address). To call the unit, the reporter has to select the preferred audio coding mode on the SCOOPY+ (e.g. mono G722), then call the remote unit, simply using this SIP address (SIP URI).

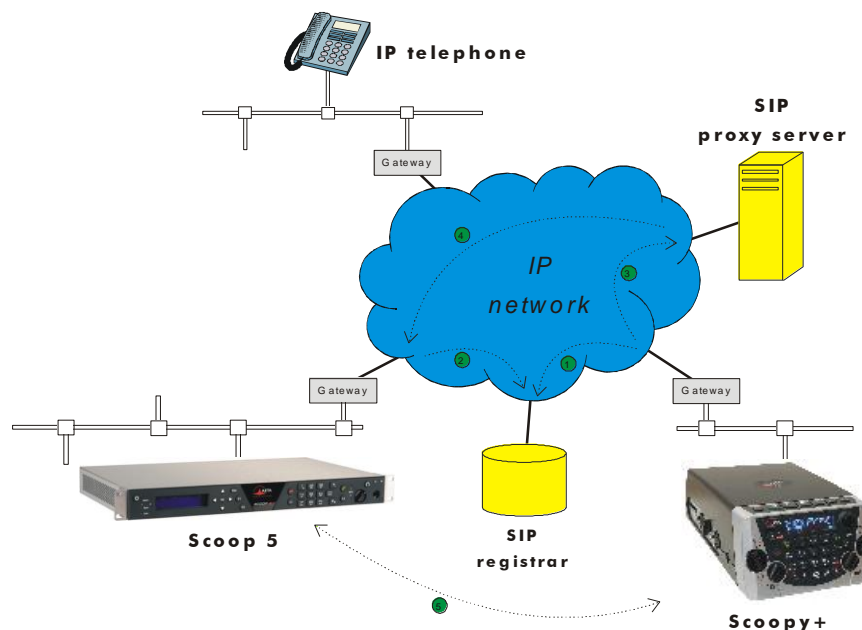


Figure 11 – Setting up an SIP session with server(s)

¹ Scoopy+ is a portable audio codec from AETA; the description here applies to any other SIP compliant codec.

What happens then on the network: the Scoopy+ sends the request ③ (INVITE in SIP protocol) to a proxy server (often the same device is also the registrar). To make things simple, this proxy then relays and routes the request ④ to its destination. Resolving the SIP URI to a physical network and address uses mechanisms similar to those used for resolving URLs. Several proxies may be involved in cascade to finally reach the desired destination, but this does not have to be known or dealt with by the end devices. The following is like the initiation of a phone call: the IP codec "rings"; if it accepts the call, this is notified to the Scoopy+.

At this stage, the proxy (s) provide the Scoopy+ and the IP codec with all the address data they need for the link, then the actual audio streams can be exchanged ⑤ between both units. As a very important feature, the end devices now can exchange data directly; the proxies do not have to be on the path, they are only involved in the setting (and later the ending!) of the session. The codecs will automatically exchange their coding capabilities, and agree on a coding mode, with no further user intervention.

Alternatively, the call can be done from the station to the reporter, in a way very similar to the above. In contrast with ISDN links, the operators at the station do not even need to know where the reporter is located! This is because the registrar deals with this issue.

Note that it is also possible to set a link with a SIP-compliant VoIP phone instead of another codec. This is one of the numerous advantages of using a standard.

8.2.3. Setting a link without a SIP server

Contrary to a commonplace but wrong idea, the SIP protocol does not impose using a SIP server. The SIP *protocol* can be used without a SIP *server*, i.e. it is possible to set "peer to peer" links without involving one or more SIP *registrars* or *proxies*. In such a case the process is more direct, and the codec contacts the remote destination with no intermediate entity. However, there are some drawbacks with such type of session:

- Without a *registrar*, the identification of the destination is its IP address; this is not as "stable" as a SIP URI, as it may change depending on the location, or time (with dynamic addressing).
- The two codecs must have open access to the Internet, or otherwise the link must be made possible by unlocking if necessary the appropriate routes and ports (conversely, when using a proxy, such allowances can be restricted to the server connection, for a better control on the access security).
- Most often the gateways perform address translation (NAT), which is an *a priori* issue for UDP protocols such as implemented for real time audio links. Proxies can help dealing with this issue, but without such servers it may be rather difficult to work around such obstacle. *The following chapter provides some recommendations about this.*

8.3. Some methods to deal with NAT routers and firewalls

Problems arise when the desired connection has to go through a NAT router, and/or a firewall, that blocks a direct IP communication.

This is a very common issue, especially if one needs to set up a transfer via the Internet. It is impossible here to describe in details the possible ways to deal with this problem, but the following just shortly discusses some typical solutions. Most probably, a network administrator should be consulted for support, and for granting adequate network authorisations and/or privileges.

The most classical issues are related to:

- The presence of a NAT router on the network path between the codecs.
- The presence of a firewall on this path.

It is always important to have available the information regarding the network organization and to be allowed to access the devices which need to be configured. Hence we highly recommend to involve the persons empowered for such tasks.

8.3.1. Links via a private network

No special problem should be met within a LAN. The operation is also possible with codecs from other manufacturers, provided that they comply with the Tech3326 EBU recommendation (also known as "N/ACIP" recommendation). However you should check for specific settings or preparation possibly needed on such devices.

A wide area network covers a wider geographic range, and the network topology most probably includes routers on the path between the codecs to be linked. However, usually there is not much difference with a local area network.

Note: using a VPN leads to just the same case; the operation is identical as far as the codecs are concerned.

8.3.2. Links through a public network (Internet)

If each of the units has got a "direct" access to Internet with a public address, we are in the same situation as the previous one, functionally speaking (private WAN). The addressing scheme is normally static, as DHCP can rarely be used on a public access. In fact, this situation is very seldom met in the field!

First, the Internet access is usually protected by a firewall which will, as a principle, block a priori the desired connection. In such case exceptions (to the firewall security rules) must be created, that will allow this connection; this has to be done by the person in charge of the network management.

Most often, on one access if not both, the codec accesses the Internet via a NAT router. This router shares Internet access, with one or a few public addresses, among the equipment on the LAN. On this LAN the devices get local private addresses, and the router carries out an IP address translation. Note that:

- As an example, a consumer ADSL modem-router is almost always a NAT router, sharing a single public IP address between the devices connected to the router.
- It is just the same on a 3G/3G+/4G mobile IP access; the terminals (phones or computers) access the Internet via NAT routing.
- NAT routing is often included in the firewall features; in fact NAT routing somewhat participates to the protection against direct attacks from the outside.

NAT routing is an obstacle to transmission with UDP, mainly for two reasons:

- It does not allow unsolicited data to come in from the outside network. In other words, data input is accepted on a port as an answer to a request from the local network, but an external agent cannot directly initiate the transmission of a packet.
- The terminal units on the LAN only know their private local address. On the other hand, agents implementing the SIP protocol have to communicate to each other the addresses and ports to be used for the media exchanges. Because of the NAT routing, agents do not get the real public addresses, which leads to failure of the session setup attempts.

We are now looking at various methods used to overcome these obstacles.

NAT and use of a STUN server

The STUN protocol is a method which is often successful¹ in helping the agents to discover their public address even when they are "hidden" behind a NAT router. Here is the operation principle:

- A STUN server is used, which is accessible over the Internet;
- The address of this server is programmed into the agent (i.e. the audio codec in our topic);
- The agent queries the server and discovers its public IP address and port number, as seen from outside of the NAT router and LAN;
- This addressing information is then used by the agent for negotiating and setting up a media session.

The STUN server address is programmable in the menu or the html pages of a SCOOPY+ S. Besides, there is also in the menu (keypad and display on the front of the unit) an enable/disable (on/off) selection, without having to clear the server address.

There are many public STUN servers available on the Internet; here are a few examples, valid at the time of writing:

Domain name	Numeric address
stun.aeta-audio.com	85.214.119.212
stun.sipgate.net	217.10.68.152
stun.counterpath.net	216.93.246.18

Examples of STUN servers

It is recommended to check that the server is operative. Moreover, numeric addresses may change, even if the domain name stays the same. A list of servers can also be found on the support page of our web site <http://www.aeta-audio.com>.

¹ Although not with so called "symmetric" NAT routers

Standard NAT router

Situation: codec A behind a NAT router with no specific programming (*a codec accessing Internet via a mobile network is almost always in such situation*).

Besides, we assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B cannot call codec A

Advantages	Drawbacks
Configuration is relatively simple	B cannot call A
No change is needed on the router	
Several codecs can be set behind the NAT router	
Method suitable for mobile network access ¹	

NAT router with DMZ

Situation: codec A behind a NAT router and placed in "DMZ".

We also assume that the other codec (B) is accessible with a public address.

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	Drawbacks
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	A is exposed to external attacks
	The DMZ may be already reserved for other network equipment
	Method not possible for a mobile network access

NAT router with port forwarding

Situation: codec A behind a NAT router and configuration of the router to forward to A the necessary ports.

We also assume that the other codec (B) is accessible with a public address.

Port forwarding to be set on the router²:

- UDP 5060 (= SIP port)
- UDP 5004 (RTP port) and 5005 (RTCP port)

¹ Except with symmetric NAT, which is often met for mobile network access

² If needed you can change these numbers on the SCOOPY+ S

Once codec A is configured for using a STUN server:

- codec A can initiate a connection to (call) codec B
- codec B can call codec A, using the public address of the NAT router

Advantages	Drawbacks
Each codec can set up a session	Need to configure the router
A is nearly equivalent to a codec with a direct public access	Only one codec can be set up in this way on a LAN
	Method not possible for a mobile network access

Use of a SIP server

In addition to the numerous features it brings along, using a SIP proxy server is a very powerful method to solve the issues related to NAT routers, because most SIP proxies are capable to detect the presence of NAT routers and/or deal appropriately with their traversal.

If a SIP server is available, and once the codecs are registered on this server:

- Any registered codec can call another registered¹ codec, regardless whether there is or not a NAT router on the path.
- The identifier (SIP URI) is stable and does not depend on the location of the called agent ("mobility" feature).

It is possible either to use a public server on the Internet, or to install a private server accessible via the Internet.

Advantages	Drawbacks
Each codec can initiate a session Each codec can receive calls	Installation may not be easy (private server)
Identification is simple and location/time-wise stable	Reliability of server questionable (public server)
Security: a private proxy can be linked with a firewall	
Also works with symmetric NAT routers	
Interoperation with telephony over IP	
Method suitable for mobile network access	

i For a fast implementation, you can use SIP accounts on AETA's SIP server sip.aeta-audio.com. This server is dedicated to a professional broadcast usage, and housed in a safe site, available 24/7. Contact AETA for subscribing SIP accounts.

¹ Depending on the access control policy, a server may accept "outgoing" calls to third party domains, or accept "incoming" calls from non registered agents.

8.3.3. Summary and reminder of essential rules

The table below sums up the situations where a link can be set up (not using a SIP proxy server) and reminds the needed specific settings:

	Codec A access	Possible calls	Codec B access	Notes
1	LAN	⇒ ⇐	LAN (same)	
2	Private WAN	⇒ ⇐	Private WAN	
3	Internet direct	⇒ ⇐	Internet direct	
4	NAT	⇒	Internet direct	STUN needed for A
5	NAT + DMZ	⇒ ⇐	Internet direct	STUN needed for A
6	NAT + port forwarding	⇒ ⇐	Internet direct	STUN needed for A UDP ports 5004, 5005, 5060
7	NAT	⇒	NAT + DMZ	STUN needed for A and B
8	NAT + DMZ	⇒ ⇐	NAT + DMZ	STUN needed for A and B
9	NAT + port forwarding	⇒ ⇐	NAT + DMZ	STUN needed for A and B UDP ports 5004, 5005, 5060
10	NAT	⇒	NAT + forwarding	STUN needed for A and B
11	NAT + DMZ	⇒ ⇐	NAT + forwarding	STUN needed for A and B
12	NAT + port forwarding	⇒ ⇐	NAT + forwarding	STUN needed for A and B UDP ports 5004, 5005, 5060

i *Basic rule: Codec behind a NAT router => use a STUN server.*

This allows the codec to set up *outgoing* calls. This is not sufficient to be accessible to connection requests *from* the outside.

i *Mobile network access without SIP server or VPN => use a STUN server*

i *NAT + DMZ or NAT + forwarding => incoming calls are possible.*

Incoming calls are not possible behind a NAT router without either such change or a SIP proxy.

i *SIP server => maximum versatility, at the expense of some initial effort (for installation)*

i *Reminder: the SIP protocol does not impose the use of a SIP server. Codecs can set up point-to-point links using this protocol in the above described conditions. When no SIP registrar is involved, the identifiers are simply the IP addresses of the codecs.*

8.4. Notice regarding open source code

The software of this product includes programs and libraries that are covered by the GNU General Public License (or "GPL"), available for example at following address: <http://www.gnu.org/licenses/gpl.txt> . Under this license, the source code for concerned elements is available on our Internet site ("Download" page); otherwise it can be obtained on request by e-mailing AETA Audio Systems (open_source@aeta-audio.com).

The Opus encoder and decoder are under BSD license.

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NOTES



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