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Scoop 6

Stereo audio codec for real time audio transmission

User Manual

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1. Presentation – Getting started

The Scoop 6 codec allows the bi-directional transmission of one or two audio signals with bit rate reduction, over various transmission media: IP protocol networks, mobile networks, AES 67 networks or even WIFI networks by using an external USB key.

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.

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

For IP transmission with SIP, the unit can be operated in a “double codec” mode. It is then equivalent to two independent mono codecs, both running over the IP interface.

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 Scoop 6 (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 Scoop 6
- Chapter 6 provides all the **technical characteristics** of the Scoop 6
- 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 1.00.*

Characteristics		Optional
Transmission interfaces		
Ethernet/IP transmission interface		
Ethernet Interfaces, 10BaseT/100BaseT & 1000BaseT; TCP/IP, UDP/IP, RTP protocols		
Audio transmission (AoIP) in unicast mode: SIP signalling protocol, SDP, RTP streaming		
Double AoIP/SIP codec operation mode		
Audio transmission in multicast mode: RTP streaming		
Net bit rate 12 to 256 kbit/s (depending on coding algorithm, linear coding excluded)		
Mobile network access		●
Integrated 2G/3G/3G+/LTE/5G network access module, 2 antenna sockets		
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 12 to 256 kbit/s (depending on coding algorithm)		
External 3G/LTE module connection via USB socket (data mode only)		
Audio coding algorithms	(audio modes)	
G711	Mono	
GSM, AMR (mobile telephone)	Mono	
AMR-WB / G722.2 (mobile "HD Voice")	Mono	
G722	Mono	
MPEG Audio Layer II	Mono, Stereo, Dual mono, Joint stereo	
MPEG AAC-LC , HE-AAC, HE-AAC v2, AAC-LD, AAC-ELD	Mono, Stereo	
OPUS	Mono, Stereo	
Linear L16, L20, L24	Mono, Stereo	
Audio interfaces		
Two analog inputs and two analog outputs with adjustable gain		
Digital input and output, AES/EBU format		
Level display for encoder inputs and decoder outputs		
Stereo headphone socket for monitoring, switchable send/receive		
AES 67 DANTE or RAVENNA		●
Auxiliary functions (available depending on transmission interface)		
Relay transmission: 2 isolated inputs and 2 isolated outputs, 6 non isolated inputs and outputs		
SMS reception via the optional mobile access		
Control and supervision		
Rotary encoder and graphic display on front panel		
Programmable set-up/dial memories		
Ethernet/IP remote control, Embedded html server		
Isolated control and status loops		
Secondary Ethernet interface for remote control		

Table 1 – Main features of the Scoop6

1.1. Install and connect Scoop 6

- Plug on a power source. To start it, press the key (bottom left corner on the front panel) for at least 3 seconds.
- Connect the necessary audio interfaces. (*Analog or digital via XLR)*
- Connect Scoop 6 on the transmission network. *(Ethernet, mobile,…)*
- Using the Scoop 6 menus: Esc key to activate the carrousel menu, select with the rotary encoder and press OK key (pressing the rotary encoder) to menu access. Use the rotary encoder to select a sub-menu or a parameter, enter or validate with the OK key. Esc to jump back up to the previous menu level, until getting back to the home screen (with level display).

1.2. Audio settings

- With factory settings, the active inputs are the analog ones, and the clipping level is set at +16 dBu for both inputs and outputs.
- Enter the menus using Esc, then and finally select **Audio**. Perform necessary settings. *(Analog, digital, AES67, line in/out*)

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

For a link over a public IP network via an access router with NAT[1], it is recommended to use a STUN server.
The address of a STUN server can be set in the Scoop 6 html pages using the tab: **Network** / **AoIP Parameters** / **STUN**, 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.
More details: see chapter 5.

- To choose the network: in the main menu, select **IP** / **IP interface** then the desired network, validate with OK.
- Check that the Ethernet interface is active (indicator on the display and/or LED on the Ethernet port on the rear panel), and that the IP address has been assigned: **Ethernet** / **IP Address** menu. Return to the main menu.
- The default setting uses a DHCP server to obtain an IP address, and is adequate in most cases. In other cases, see details on 3.4.1.
- Enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used) in the HTML page via the **Connections** / **Connection State** menu, then click on **Dial**.
- Hanging up will also be done via the HTML page by clicking on **Release**.
- *When using a SIP server, some data must be entered beforehand using the* ***IP*** *menu; for more details, refer to 7.2.* Similarly, some preliminary settings may have to be done for the "Direct RTP" mode (without the SIP protocol) or the multicast mode.

[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, an antenna connection is required, and 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.

1.4.1. Set up a (mobile) IP link

For a link over a public IP network via an access router with NAT[1], it is recommended to use a STUN server.
The address of a STUN server can be set in the Scoop 6 html pages using the tab: **Network** / **AoIP Parameters** / **STUN**, 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.
More details: see chapter 5.

- While the unit is switched off, insert the SIM card into the drawer on the front side of the Scoop 6.
- Switch on the Scoop 6 (depress ⏻ a few seconds).
- Select the network: **IP** menu then **IP Interface**, then select **Mobile**, validate with OK.
- Activate mobile data in the **Mobile** / **Mobile data** (**On**) menu, confirm with OK .
- Configure the PIN code of the SIM card beforehand via the HTML page using an Ethernet network. **Network** / **Mobile parameters** Menu.
- Check that the Mobile interface is active (indicator on the display).
- Enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used) in the HTML page via the **Connections** / **Connection status** Menu, then click on **Dial**.
- Hanging up will also be done via the HTML page by clicking on **Release**.

*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.

1.4.2. 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 carousel menu, key Esc . Select HD.
• Then, return to the main menu then select **Mobile** / **Mobile Network**. If necessary you can set a priority between 2G, 3G, 4G and 5G networks. The normal choice is **Auto**.
• Enter the remote number to call (numeric IP address, or SIP URI if a SIP server is used) in the HTML page via the **Connections** / **Connection status** Menu, then click on **Dial**.
It is also possible to select the call history via the carousel menu, key Esc. Select / **Call List**.
• Hanging up will also be done via the HTML page by clicking on **Release**.

2. Functions

Reminder: this chapter details the functions and capability of the Scoop 6, but not necessarily describes the detailed operating modes, developed in other chapters.

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

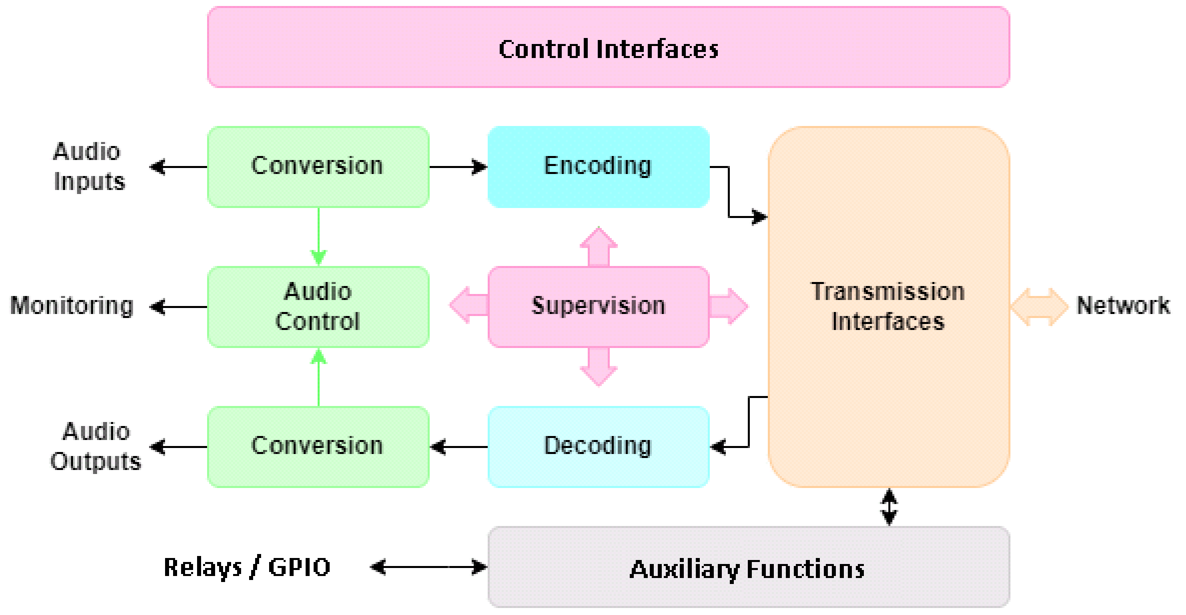


Figure 1 – Functional diagram of the equipment

The audio signals to be transmitted are converted (when needed) to digital format, 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, 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 output to both digital and analogue outputs.

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

In addition to the main task of transmitting an audio programme, the Scoop 6 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 Scoop 6 features in all versions an Ethernet interface for IP protocol networks. A Mobile interface can also be added with an option.

2.1.1. Ethernet/IP interfaces

The IP interfaces is a 10BaseT/100BaseT and a 1000BaseT 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.

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
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 AAC-LD	16 to 256 kbit/s	30 to 277 kbit/s	Up to 20 kHz	Low bit rate, suitable for speech and music
MPEG AAC-ELD	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
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

Chapter 2.2 "Audio coding and decoding" describes in more detail the characteristics of these various codings.

[1] Informative value; higher than the “net” encoded audio bit rate because of the protocol overhead

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. Links can be set up in two ways:

- "Peer to peer" connection between two compatible units using a SIP protocol
- Use of a SIP proxy server to set up the link, or a SIP PBX
- Direct "peer to peer" connection without the SIP protocol, in a so-called "direct RTP" mode. This mode can be used to interoperate with devices without SIP support, or to bypass any network blockages specifically affecting the SIP protocol.

SIP and SDP protocols

The SIP protocol is a signalling protocol, used for IP connections, which allows the Scoop 6 to interoperate with IP phones and other SIP compatible audio codecs. Details about the SIP protocols can be found in the annex (refer to 8.3, Overview of the SIP protocol).

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

ⓘ *Note: the SIP protocol does 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.

Double SIP codec

This operation mode is available as an option.

In the unicast mode with SIP, the Scoop 6 can be used as two independent mono codecs. Each of these is then a mono SIP codec that can set an audio link in two ways:

- Direct "peer to peer" connection with a compatible remote unit.
- Use of a SIP proxy server to set up the link, or a SIP PBX. In such use case, each codec has got its own registration.

The available audio coding algorithms are a subset of those of the "standard" mode (single codec): see "

Table 4 – Available coding in double AoIP codec mode".

Double Streaming

Audio transmission over IP (AoIP) is sometimes subject to data loss which can cause noise or dropouts in the audio signal. The AETA "Double Streaming" system is an effective means of considerably reinforcing the robustness of an AoIP transmission to the SIP protocol, thanks to a redundant link which makes it possible to compensate for losses on the main link.
Chapter 8.1 "Using the "Double Streaming" system describes this mode in detail.
Dual Streaming requires the use of the SIP protocol and cannot be used when the Scoop 6 is operated in dual SIP codec mode.

Packet duplication

Scoop 6 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.

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 Scoop 6 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.

Factory SIP account

AETA has set up a public SIP server, dedicated to "broadcast" audio over IP applications. Our customers can obtain accounts in order to register their audio codecs on this server.
In addition, each Scoop 6 has a "factory" SIP account on this server. This account is permanently and definitively associated with the unit, and it is possible at any time to recall its settings to save the Scoop 6 on the server. However, you can also configure another account on any other SIP server.
A second "factory" SIP account is also installed in the device:
• This account can be used to implement "Double Streaming" (see 8.1, Using the "Double Streaming" system).
• If the "Double SIP codec" is used, this account can be assigned to the second SIP codec.

Remote control via IP

In addition, the Ethernet interface can be used for configuring or remote controlling the unit, with two control methods:

- Scoop 6 provides html pages which allow getting complete control over the unit using a web browser, via port 80 (default port for http protocol).
- The supervision software **Scoop Manager** from AETA can remote control the Scoop 6 via a TCP/IP connection on port 7001.
- TCP port 6000 can be used for “command line” control, suitable for other codec supervision software such as TeleScoop, Codec Live, MDC.Net, etc.

2.1.2. 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. An “HD-5G Mobile” option is also available and add a 5G Network to the product.

For the operation, at least one multiband antenna (to be selected for compliance with the mobile network characteristics) must be connected on Scoop 6. An antenna “diversity” feature gives an opportunity to improve the stability when the radio reception quality is poor, by connecting a second antenna.

Lastly, Scoop 6 can display the received SMS messages.

Mobile voice mode – HD Voice

The integrated module allows to use the mobile phone service, for communicating 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.

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

More and more mobile phones now support this service, especially (but not only, and not all) smartphones. All AETA codecs with “HD-4G” mobile network option 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
- ScoopFone 5G
- Scoop 6 with the HD-xG option
- ScoopTeam with de HD-xG option
- ScoopyFlex with the HD-xG option

Mobile IP mode

The other service available with mobile access is the data packet transmission mode, abbreviated as “PS” (for Packet Switched), 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 (2G/3G/3G+/4G/5G), 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.

2.1.3. Managing calls

The audio transmission implies a link/session setup phase.

One of the transmission interfaces is selected as the *default interface* on the Scoop 6.

A call towards a remote unit, initiated by the user of the Scoop 6, 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 conditions described in 4.5.5, "Call Reception"

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).

The detailed procedure for establishing a connection is described in 4.5, "Establishing a link".

2.2. Audio encoding and decoding

Scoop 6 features a wide range of coding algorithms. Their availability depends on the transmission network used, and on the single/double codec mode that is used. Besides, the MPEG family algorithms feature a large configuration flexibility.

2.2.1. Single codec

In this mode, the selected network interface (Ethernet/IP) is used to set a link with a single remote device. The table below synthetically describes the capabilities with the various transmission media:

Codec	Audio Channels	Frequency				Net bit rate (kbit/s)	Ethernet	Mobile IP	HD Voice
		16	24	32	48				
G711	Mono					64			
G SM, AMR	Mono					-			
AMR-WB	Mono					-			
G722	Mono					64			
MPEG L2	M/S					64			
MPEG L2	M/S					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			
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			
AAC-LD	M/S					16 => 56			
AAC-LD	M/S					64			
AAC-LD	M/S					96			
AAC-LD	M/S					128			
AAC-LD	M/S					192			
AAC-LD	Stereo					256			
AAC-ELD	M/S					16 => 56			
AAC-ELD	M/S					64			
AAC-ELD	M/S					96			
AAC-ELD	M/S					128			
AAC-ELD	M/S					192			
AAC-ELD	Stereo					256			
	3kHz		7kHZ		20kHz				

Table 3 – Available coding depending on network

2.2.2. Double AoIP codec

This mode is available for the transmission over an IP access (Ethernet or possibly a mobile network) and with the SIP protocol.

In this mode, two AoIP links can be set simultaneously, possibly with two separate remote devices. Either link is a mono link. The table below shows the capabilities for each link:

Codec	Frequency 8	 16	 48	Net bit rate (kbit/s)
G711				
G722				
MPEG L2				
MPEG L2				
AAC-LC				
AAC-LC				
AAC-LD				
AAC-LD				
AAC-ELD				
AAC-ELD				
HE-AAC				
HE-AAC				
OPUS				

Table 4 – Available coding in double AoIP codec mode

2.2.3. Details about the coding algorithms

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

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.

G711 is available only for IP (single or double codec).

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 IP networks.

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.

Scoop 6 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).

ⓘ *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 end*[1]

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 wired or mobile IP networks (single codec, or double AoIP codec).

No specific synchronisation is required for the IP mode.

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.

ⓘ *The latency is rather high with these sampling rates*

MPEG L2 is available for wired or mobile IP networks (single or double AoIP codec).

MPEG AAC algorithms

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

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

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 but lower latency.
- MPEG **AAC-LD** (“Low delay”) : lower compression but lower latency.
- MPEG **AAC-ELD** (“Low delay”) : upgraded version of AAC-LD.
- 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 as an option for wired or mobile IP networks (single or double AoIP codec).

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 (single AoIP codec).

ⓘ *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.*

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 Scoop 6, 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 Scoop 6, 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.
- Ability to change the bitrate during a communication by accessing from web page.

Opus coding is only available for wired or mobile IP networks (single or double AoIP codec).

2.3. Audio interfaces

2.3.1. Analog interfaces

The analogue inputs and outputs are balanced, and the input and output gains are adjustable.

The input to the encoder is selectable between the digital audio input and the stereo analogue input.

The sampling frequency of the analogue ⇔ digital converters is automatically set depending on the coding algorithm used for transmission.

2.3.2. Digital interfaces

The equipment also provides digital audio inputs/outputs in AES/EBU format.

The input to the encoder is selectable between the digital audio input and the stereo analogue input.

The digital audio interfaces are usually locked to the digital audio input ("genlock" mode), but alternatively they can be synchronised to the internal clock reference of the codec (mode called "Master").

Sampling rate conversion is automatically performed, whenever needed, depending on the coding algorithm used for transmission.

ⓘ *As an important consequence, the selection or value of the sampling frequency of the AES/EBU input/output is **completely independent** of the sampling frequency of the compression algorithm.*

One should also be aware that the various audio settings have <u>no relationship</u> with those for the other side of the link (the remote codec), whatever they are: selection of analog or digital source, sampling, rate of AES interfaces, etc. This configuration is only relevant for the local installation.

2.3.3. Audio monitoring

Audio programmes can be monitored of the audio input (before encoding) and at the audio output (after decoding the received signal).

First, the programme level is displayed on the graphic screen.

Besides, a test output on a stereo headphone jack allows monitoring either the encoder or decoder audio signals.

2.4. Auxiliary functions

The main function of the Scoop 6 is the transmission of one or two main audio programmes, but it also provides auxiliary functions for transmitting data or additional signals, inside the same stream (or, more generally, the same session).

These features are only compatible with AETA products, because they are not inside the scope of independent standards.

The availability of these functions depends on the coding algorithms, and on the transmission network. The following tables show these capabilities for the various networks.

ⓘ *No auxiliary function is available for mobile voice transmission, except SMS reception.*

Codec	Loop	
	Relay	GPIO
G711		
G722		
MPEG L2		
AAC-LC		
AAC-LD		
AAC-ELD		
HE-AAC		
HE-AAC v2		
OPUS		
L16, L20, L24		

Table 5 – Auxiliary functions: over IP networks (wired or mobile)

ⓘ *The auxiliary functions are not available in “Direct RTP” mode.*

2.4.1. Transmission of isolated relays

When this function is activated, the codec transmits to the remote unit the status of two isolated current loops. The remote unit then opens or closes relay contacts according to the transmitted status. Conversely, as the function is bi-directional, the codec activates its two relays (“dry” isolated contacts) depending on the status of the two current loops on the remote unit.

For transmission over IP, this feature is always available regardless of the coding algorithm (except linear coding L16, L20, L24).

A typical application is the transmission of “on air” signals; a contact closure can be used for instance to light up an indicator or switch on other equipment.

2.4.2. Transmission of GPIO

For IP transmission, Scoop 6 also allows, in the same way as the two isolated relays, to transmit 6 additional binary signals, routed to “GPIO” interfaces, which are not isolated. This feature is available regardless of the coding algorithm (except linear coding L16, L20, L24).

2.4.3. SMS reception

When the Scoop 6 includes the “HD-xG” option for mobile access, it can display the SMS that are sent to the phone number of the SIM card installed in the unit. On receiving a SMS, the message is displayed directly on the screen, and it is also visible on a page of the embedded html server.

This feature is independent of the network interface in current use; it is permanently active as soon as the unit is registered on a mobile network. The PIN code of the SIM card must have been entered beforehand.

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 a keyboard, a graphic display, LED indicators, or using various remote control interfaces.

2.5.1. "Local" control

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

2.5.2. Embedded html server: "web pages"

Scoop 6 provides html pages that enable full control using a web browser, via port 80 (default port for the HTTP protocol). See on chapter 5 the detailed operation mode.

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.

2.5.3. Supervision using Scoop Manager

AETA's Scoop Manager software can supervise the Scoop 6 via a TCP/IP connection on port 7001. Scoop Manager is especially efficient for managing the traffic on the various networks for a pool of codecs, providing an overview of their status on a single screen.

2.5.4. Remote access

Context

A computer, for example, can easily take control of a Scoop 6 when it is connected to the same local network as the codec. The computer can then access the embedded html pages of the codec.

When the computer is at a remote site, connected via the Internet, it is most often necessary to pass through a NAT router(s) and/or firewall(s). In this type of situation, it is not possible to obtain a connection as easily as through a local network.

The AETA *"Remote Access*" service is designed to circumvent this difficulty and allow you to take control of a device even in this case.[1]

Service "Remote Access"

The system is based on the use of a remote access server, intermediate between the codec and the control device. An order session takes place in two phases:

1. The Scoop 6 connects to the server, and makes itself available for a possible remote control session.
2. A user who wants to control the unit connects to the server, and the latter will establish a virtual link between the control device and the codec, identical in its operation to a direct connection via a local network.

Both connections are secure and encrypted. Phase 1 above is done when the Scoop 6 is connected to the network and remote access is active on the device. This can, however, be disabled by the user.

Two types of service are offered:

1. *Remote Assistance* : this service allows AETA's support service to access the codec for possible help or investigation. *This cannot be done without activation on your part.*
2. *Remote Access* : allows remote access to the html pages of the codec. The remote control means can be any device with a web browser, without an application to install. *A "Remote Access" option must be installed on the SCOOP 6 to use this service.*

When the codec has more than one IP interface, it is possible to use an interface for remote access that is separate from that used for AoIP transmission.

2.5.5. Alarm contacts

Besides configuring the equipment operating mode, this module supervises its status (detection of alarm conditions). On detecting operation or transmission faults, the equipment switches on indicators and relay contacts. Two alarm classes are defined:

- “Internal” alarm; corresponds to a major fault internal to the equipment;
- “External” alarm; corresponds to a fault whose origin is deemed external to the equipment (for example, transmission fault);

[1] However, a very restrictive firewall can obviously block the service.

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. Such profile is similar to a phonebook entry, but in addition coding parameters can be stored as well. Recalling a profile directly configures the codec and/or sets an outgoing call with the parameters previously recorded by the user in the profile.
- “Presets” which memorise the network access characteristics. Recalling a preset is a quick way to recover the configuration needed for connecting on a given line/network.
- “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.

3. Operation

This chapter physically describes the device, shows how to install it and the operation principles. Details on the menus or the html pages are provided 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 display, status indicators, rotary encoder), or "Remote control" mode, thanks to an asynchronous serial port or an Ethernet 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.

Using the Scoop Manager software (installed on a Windows PC computer), it is possible to manage calls on a pool of Scoop 6 codecs. Please consult us for more information about the features of the Scoop Manager software.

The Scoop 6 can also be remote controlled by third-party codec management software and systems, such as Codec Live, MDC.Net, etc.

For controlling connections, it is also possible to use the "Loop control" function. When this special connection mode is selected, one can trigger a call by activating an input current loop (optically isolated), and release the line by de-activating this loop. In such case, an outgoing connection is established or released only by this way, and no more from the front panel or the remote control interfaces (however, all other parameters are still controlled from these interfaces as in the normal mode).

If "loop control" is not activated, it is always possible to use the loop to release a running connection (a pulse on the loop will release the line).

The loop control interfaces are described in 3.5.3 and 6.1.8

Besides, whatever the connection mode (normal or loop control), a "dry loop" is closed when a connection is active.

3.2. Physical description of the equipment

The Scoop 6 codec is housed in a half 19 inches chassis of 1U height (44 mm or 1.75”). It includes a universal mains power supply. There is an option for powering from a 12V DC source (which can be used in parallel with the mains input, with priority to the latter).

3.2.1. Front panel

All the elements needed for local control are located on the front panel (see picture below).

On the right-hand side, one can find several LED indicators, an audio jack port and a USB port. On the left-hand side are located the display, an escape/on-off key and a rotary encoder. Lastly, at the extreme left side one can find a sim port.

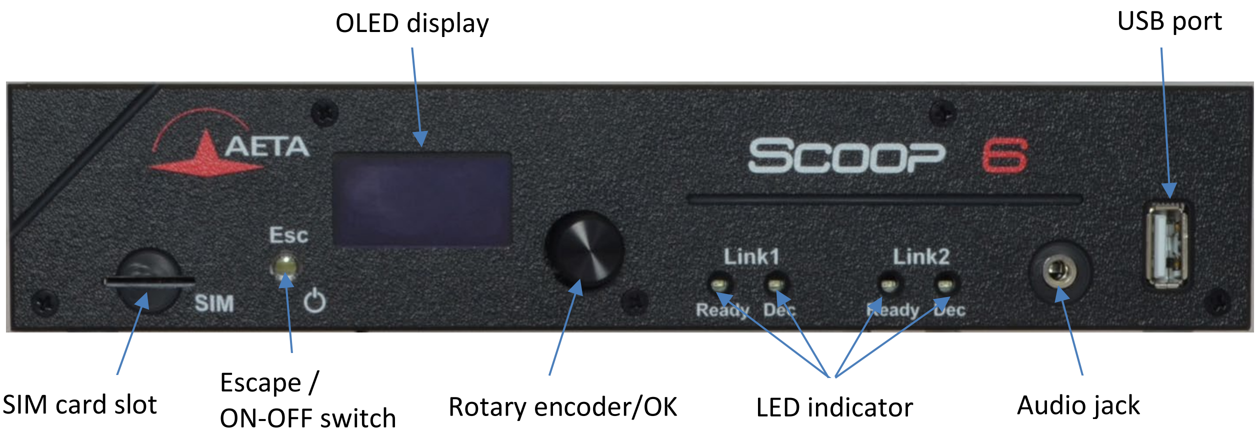


Figure 2 – Scoop 6 front panel

On/Off switch and standby

First, completely on the left is located the ⏻ on/off key which also serve at the standby indicator. When the unit is powered but in standby (Red LED on), keep the key pressed for at least 3 seconds to switch on the unit (White LED on). When it is operating, keep the key pressed for at least 3 seconds to switch it off.

Status LED indicators

The LEDs have the following meaning:

Marking	*Color*	Function
Dec	***Green/Red***	This LED is off when idle, green when a link is established, red in case of a sync loss of the decoder.
Ready	***Green/Red***	This LED goes red at startup time or in case of an alarm, and goes green when the unit is ready for operation. The LED also blinks red if SIP registration is active but fails.
⏻	***White/Red***	This LED goes red when alimented and goes white when it is started.

Display and navigation keys

Besides the OLED screen, one can find the keys for navigating through the menus:

Key	Function
Esc	When press hold 3 sec : On/Off; From the main menu press once to go to the upper menu level; From the home screen: switch to the carousel menu;
Rotary encoder	Encoder used to adjust settings or to browse through menu options; From the home screen : press once to change the headset volume; From any menu : press to select

More details on the navigation and sub-menus can be found in chapter 4, dealing with detailed operating modes.

Audio monitoring

The audio signals can be monitored with a low impedance headphone connected on the front panel (3.5 mm stereo jack). The headphone volume is adjustable thanks to the rotary encoder.

USB sockets

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 2.1.2 about this function.

Sim card socket

In products equipped with the mobile network access option, this socket is designed to accommodate the SIM card allowing access to the network and services.

ⓘ The SIM card must be inserted when the device is switched off (or in standby).

ⓘ If you need to use a card in µSIM or nanoSIM format, which is smaller, you can use a µSIM/SIM adapter (available on request from AETA): first place the µ/nanoSIM in this adapter, then the whole is used like an ordinary SIM card.

3.2.1. Rear panel

All connections are done on the rear panel of the codec. The characteristics of the interfaces and layout of the sockets are detailed in chapter 6.1, Characteristics of interfaces.

Figure 3 – Rear panel

The following elements are available (*numbers such as **[12]** refer to the following* Figure 3 – Rear panel :

Mains power socket [1]

This is an IEC type power socket. The unit starts up as soon as power is applied.

See details in 6.1.9 and 6.4 (Power supply).

DC 12V power socket [2]

This 2.1mm jack is optional. *See details in 6.1.10, DC power supply (option).*

Audio inputs/outputs

- Analog inputs **[3]** /outputs **[4]**: at the input, plug the audio cables into the female XLR sockets. At the output, plug the audio cables into the male XLR sockets. In mono mode, only "A" channel is used. In double codec mode, the codec 1 uses input A and output A, while the codec 2 uses input B and output B.
- Digital inputs **[5]** /outputs **[6]**: a digital input (mono or stereo) in AES/EBU format (or SPDIF) can be connected on the female XLR socket, and a digital output in AES/EBU format is available on a male XLR socket.

Ethernet interfaces [7]

The first one is a 10/100/1000BaseT socket and the second socket (Upper) is a 10BaseT/100BaseT port. They are 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 (green LED) and activity (yellow LED) of the network.

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

Antenna sockets [8]

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

At least one antenna must be plugged on the main socket, the one that is shown on the picture above (the outmost socket, left side of the unit).

A second antenna is optional, but it allows to improve the reception quality in less favourable areas; it must be activated (configuration menus) if one is connected.

The antennas must cover the band(s) used for the operator and network services. In doubt, refer to the operator. The provided antennas cover bands 1, 2, 3, 7, 8, 20 (800 MHz, 900 MHz, 1800 MHz, 1900 MHz, 2100 MHz, 2600 MHz). They are compatible with almost all the 2G/3G/4G/5G networks in Europe.

USB socket [9]

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 2.1.2 about this function.

“AUX” socket [10]

This 8-pin RJ45 socket includes the interfaces for the relay transmission function (described in 2.4.1).

It also includes loop interfaces for the loop control function (cf. **Erreur ! Source du renvoi introuvable.**), as well as a +5 V power supply that can be used to provide current for the loop and relay interfaces.

“Digital I/O” socket [11]

This 8-pin RJ45 socket is the interface for the GPIO transmission function (as described in 2.4.2).

Its wiring is described in: “Digital I/O” interface*.*

“AES67” socket [12]

These interfaces are 1000BaseT ports. The unit can get two audio inputs from this AES67 interface, and transmits two audio outputs via this interface. Two options are available: RAVENNA and DANTE

3.3. Installation and set up

3.3.1. Mounting and connections

Natural convection cools the equipment. Avoid obstructing the openings on the flanges.

To operate the codec, the minimum necessary connections to set up are (see details in the rear panel description):

- Power supply (mains and/or DC);
- Audio inputs and outputs (XLR sockets);
- Network interface: depending on the networks used, Ethernet interface or antenna(s) for mobile network access[1];

The pin out of the connectors is indicated in chapter 6.1: Characteristics of interfaces.

3.3.2. Initial set up

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 more details about the codec configuration, see chapter 4 and 5. The setup for the Ethernet interface is described in 3.4 below (Initial setup of the Ethernet interface).

[1] In such case, a SIM card should be set as well in order to enable the mobile services.

3.4. Initial setup of the Ethernet interface

The Scoop 6 includes a 100BaseT / 10BaseT and a 1000BaseT 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 or for remote control the unit via a TCP/IP connection (TCP port 6000).

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 Scoop 6 automatically switches to the adequate 10 Mbit/s tor 1000Mbit/s mode.
- "Straightforward" patch cables should be used for a connection to a hub or a switch. Conversely, a "crossed" cable might be needed for special configurations (e.g. a test connection to a PC).

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 / ETHERNET**.

3.4.1. DHCP server available

This is the simplest case, because the server will allocate a suitable IP address and give the unit the right settings. Select "**DHCP**" in the menu (**Setup / ETHERNET / Ethernet**). 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 (**Setup / 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 / ETHERNET / Ethernet / Manual.** Then in the HTML menu, 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.

ⓘ *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 "**IP Address**" menu (**Setup / About**).

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 Scoop 6.

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

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.

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.
- "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 Scoop 6 (local release).

3.5.2. Auto-redial feature

The IP modes are "dial up modes", where a link can be set up and released at will. 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 applies in the following situations:

- If the initial call fails for any reason (e.g. called party is busy); the codec then redials and retries to establish the link.
- The codec can also redial if the link is already established and the link is lost, for any reason else than "local release" (e.g. the remote unit mistakenly dropped the line).
- After a power failure, after rebooting the codec will automatically redial and set-up the link back.

ⓘ *Note that, while "auto redial" is active, an established link can be definitively stopped only by releasing the line on the 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 (**Setup / Tools / Auto Redial**). In double codec mode, the function can be activated separately for each codec.

Advanced setting should be made via the HTML page (see chapter 5).

3.5.3. Loop control

In normal operation, outgoing calls are sent or released using the menus and/or the remote control interface. When the loop control function is enable, outgoing calls are controlled by activating or not optically isolated input loops. One loop is available for each codec when in double codec mode. When the input loop is activated (i.e. current is flowing), the corresponding codec establishes a link by calling the last number (or IP address, or SIP URI in the IP mode) previously dialled by the unit. When the loop is de-activated, the codec releases the line and stays idle as long as the loop is not active (except if receiving an incoming call).

ⓘ *In normal operation, it is nevertheless possible to release a running connection by briefly activating ("pulse") the control loop.*

The "auto-redial" feature is implicitly active when loop control is active: the codec tries to keep the link, and automatically recalls the remote unit if the line drops, as long as the input loop is active. The "time before redial" parameter described in the above is also applicable to the loop control mode. On the other hand, the "redial attempts" parameter is not applicable here, because the unit will always try to recover the link, until the loop is left inactive.

ⓘ *Note that, as an important consequence, when using loop control, the termination of a link must always be done on the calling party side by de-activating the input loop. Whenever the line is released by the receiving party, the calling unit will redial and re-establish the link.*

When loop control is active, the input loops are the only means of setting up an outgoing call; setting a call from the menu is not allowed. Hanging up with the keypad is rejected.

ⓘ *Remind that the first step is to set-up the link once in normal mode, and later activate the loop control mode; afterwards the input loop is used to trigger a redial to the previous number.*

3.6. First level maintenance

3.6.1. Analysis of malfunctions

The following table indicates the detected alarm conditions and their classification:

Alarm condition	Internal	External	Minor[1]
Power or fuse fault	X		
Overload on an audio input			X
Fault on AES/EBU audio input		X	
Decoder synchronization error		X	

Table 6 - List and classification of alarm conditions

Excluding the case when an internal failure disables the management micro-controller, messages are displayed to indicate the anomaly, or the fault can be searched using the HTML page.

3.6.2. 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 Scoop 6 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 Scoop 6).
- 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 Scoop 6.
- After the initialization phase, Scoop 6 will go back to its “factory” settings.
- Remove the USB stick (no matter with or without power on), before restarting the Scoop 6, otherwise all your possible new settings will be deleted again.

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

[1] Minor alarms are readable on the display, but do not trigger alarms (contacts and LEDs)

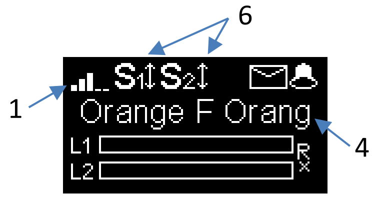
4. Detailed operating mode – User interface

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

4.1. Starting up

When the device is powered and in standby (red light on), press and hold the Esc button for at least 3 seconds to start the device.
During startup, the device displays temporary messages. This initialization phase lasts approximately 30 seconds.
The basic screen is then displayed:

Ex : Basic Start Screen

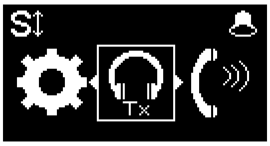


With Mobile option

1. **Network level indicator**: 5-bar display.
2. **SMS reception indicator**: stays on until all the new SMS are read.
3. **Buzzer indicator**
4. **Mobile network operator name**, current network.
5. **Audio level**: From -30 dBFS to 0 dBFS.
6. **Data connection active**
7. **SIP registration active**: on when the unit is registered on a SIP server.

4.2. Navigation principles

From the basic screen, by pressing the Esc key, the display switches to the circular menu. Also from the basic screen, pressing the rotary encoder OK will adjust the volume of the headphones.



Circular Menu

Headphone Volume

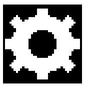
Pressing again Esc returns to the basic display.

The device offers a tree structure of menus, and the rotary encoder is used to navigate through the menus. The key OK is used to confirm certain choices or enter values, while the key Esc is used to return to the higher menu level. Pressing this key repeatedly takes you back to the circular menu and the basic screen. From the circular menu, you can enter one of the main menus by using the rotary encoder and then pressing OK to enter the framed menu.

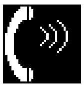
4.3. Menus presentation

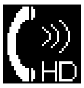
4.3.1. Circular Menu

The circular menu is the first menu that can be accessed by pressing the Esc key. This menu can be considered as the main menu of the machine. This allows access to different sub-menus which will be described below:


Allows access to the machine parameters (Set-up)

Allows access to change headphones source (Tx/Rx)


Used to dial the last number called (Redial) or to access the profile directory



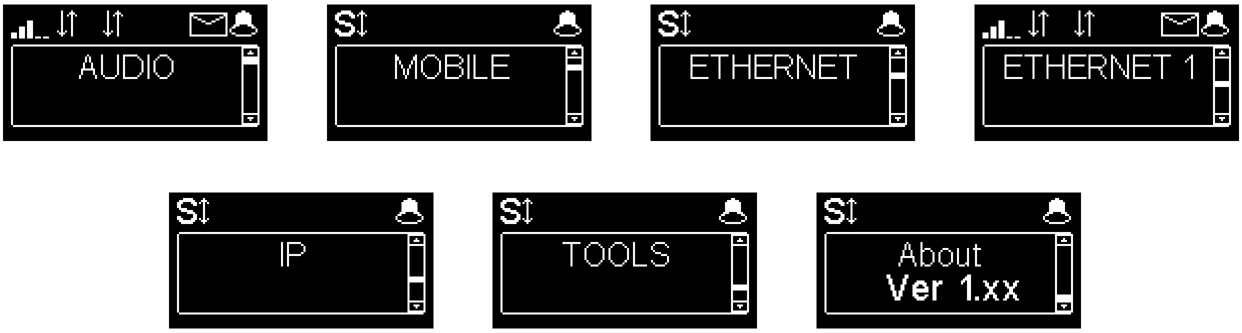
Allows you to call saved phone numbers in GSM mode (HD-Voice)

It is possible to activate the double codec mode using the HTML page, which will have the effect of adding a second call menu numbered 2.

4.3.2. Set Up Menu

The settings menu represented by a toothed wheel provides access to various sub-menus which themselves provide access to other menus. The different menus will all be described below.

First level of menu:



Audio Menu : menu for configuring the audio interfaces

Mobile Menu : configuration of the internal mobile access module

Ethernet Menu : configuration of the Ethernet interface (Ethernet 1 is used for the second ethernet port)

IP Menu : configuration of the IP transmission mode

Tools Menu : user interface configuration, reset settings

About Menu : shows information such as:

- Firmware version number
- SIM card phone number
- SIP accounts and server address
- Ethernet interface IP address
- Reference of the internal radio module

Audio Menu

Audio Format

It is possible to change the audio format of the device for an analog, digital or AES67 format.

Line input level adjustment

Maximum input level can be adjusted from +8 dBu to +20 dBu.

Line output level adjustment

The maximum output level can be adjusted from +8 dBu to +20 dBu.

Mobile Menu

Set the mobile Radio Access Technology

You can force the mobile on a specific radio technology: 2G, 3G, 4G, 5G or just let the mobile module select the best available network by using the Auto mode.

Enable mobile data transmission

You can disable mobile data transmission via the mobile network. This does not affect the IP data transmission over the Ethernet interface.

Set the mobile operator APN

The Scoop6 Features a preset APN for each operator. You can enter another one if this preset is not suitable, or if the operator is not included in the internal list of presets.

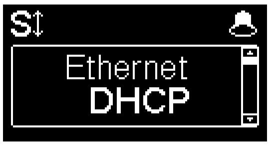
Line quality setting

This setting adapts the size of the reception jitter buffer depending on the expected transmission quality.

Alternative line quality setting

If the "Auto Jitter Buffer" system is activated (see IP menu below), this setting becomes "AJB Policy": selection of the policy for this system.

Ethernet Menu



Set Ethernet IP address allocation

Select DHCP on a network equipped with a DHCP server, otherwise select Manual to set a static IP address.

IP Address

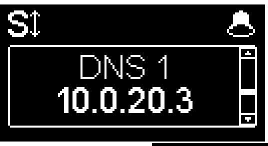
Display the current static IP address.

Network mask

Display the current network mask.

Ethernet gateway

Display the gateway address.



DNS 1

Display the domain name server.

Display the Ethernet MAC address

Press OK to see the complete address.

Line quality setting

This setting adapts the size of the reception jitter buffer depending on the expected transmission quality.

Alternative line quality setting

If the "Auto Jitter Buffer" system is activated (see IP menu below), this setting becomes "AJB Policy": selection of the policy for this system.

IP Menu

IP interface selection

It is possible to select mobile, Ethernet or Wi-Fi. In Auto mode, connecting the Ethernet cable switches the choice of the interface and vice versa in mobile.

Set the default IP transmission bite rate (OPUS)

This menu allows you to set the desired transmission rate for the OPUS algorithm, from 12 kbit/s to 192 kbit/s.

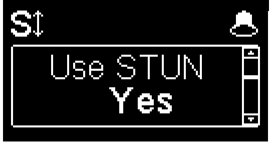
Activate the automatic IP bite rate system (OPUS)

This menu allows you to activate the automatic flow system, and to choose the variant of the system. The "**Stable**" variant reacts less quickly than the "**Standard**" variant, to limit audio disturbances, and finally the "**Dynamic**" variant seeks faster adaptation to changes in transmission quality.

« Auto Jitter Buffer » system activation

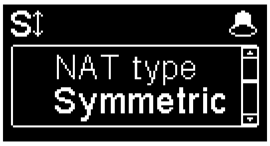
This menu is used to activate the automatic adjustment system of the size of the receiving buffer. Latency is variable with this system.

STUN activation



STUN is generally recommended with the use of a SIP server, but in some cases this function is not suitable and needs to be disabled.

NAT type



When the STUN is used the Scoop 6 can detect the type of NAT applied by the router through which it accesses the Internet. *(Disable STUN if NAT = Symmetric*)

Public IP address

When the STUN is used the Scoop 6 can detect the public address, with which it accesses the Internet.

Select default protocol

IP connections normally use SIP. "Direct RTP" makes calls without SIP protocol and "Multi Cast" transmits to several machines.

Set the SIP name account

It is possible to customize the name displayed on the remote device.

Display SIP account number

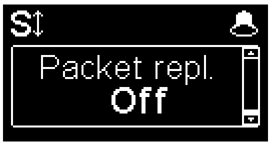
Display the SIP account number of the device. SIP account switching can only be done via HTML pages.

Server SIP port

The default value is 5060, in some cases it is better to use 5070. To use another port value, you must go through the HTML pages.

RTP port for the audio system

The port value can only be changed via HTML pages.



Activate packet duplication (double stream)

When this mode is enabled, a dual-stream link is established, securing the audio link.

Tools Menu

PIN number saving

If you select "On", the PIN code is saved in Scoop 6 and it will not be necessary to re-enter it each time you boot for the same SIM.

Automatic redial

Active the automatic redial. If enable, when the Scoop 6 calls a remote device, it automatically calls it back in the event of an untimely interruption.

Brightness setting

Three choices are available: low, medium, high brightness.

Remote access setting

Remote access can be enabled directly from the device.

Reset setting

- Load the factory SIP account into memory.
- Reset settings, including SIP account.

About Menu

The About menu allows you to display different data related to the Scoop 6 such as the IP address (Ethernet 1/2, mobile), the SIP account number, the port number, the phone number of the SIM card and the model of the modem.

5. Operating mode – Embedded HTML pages

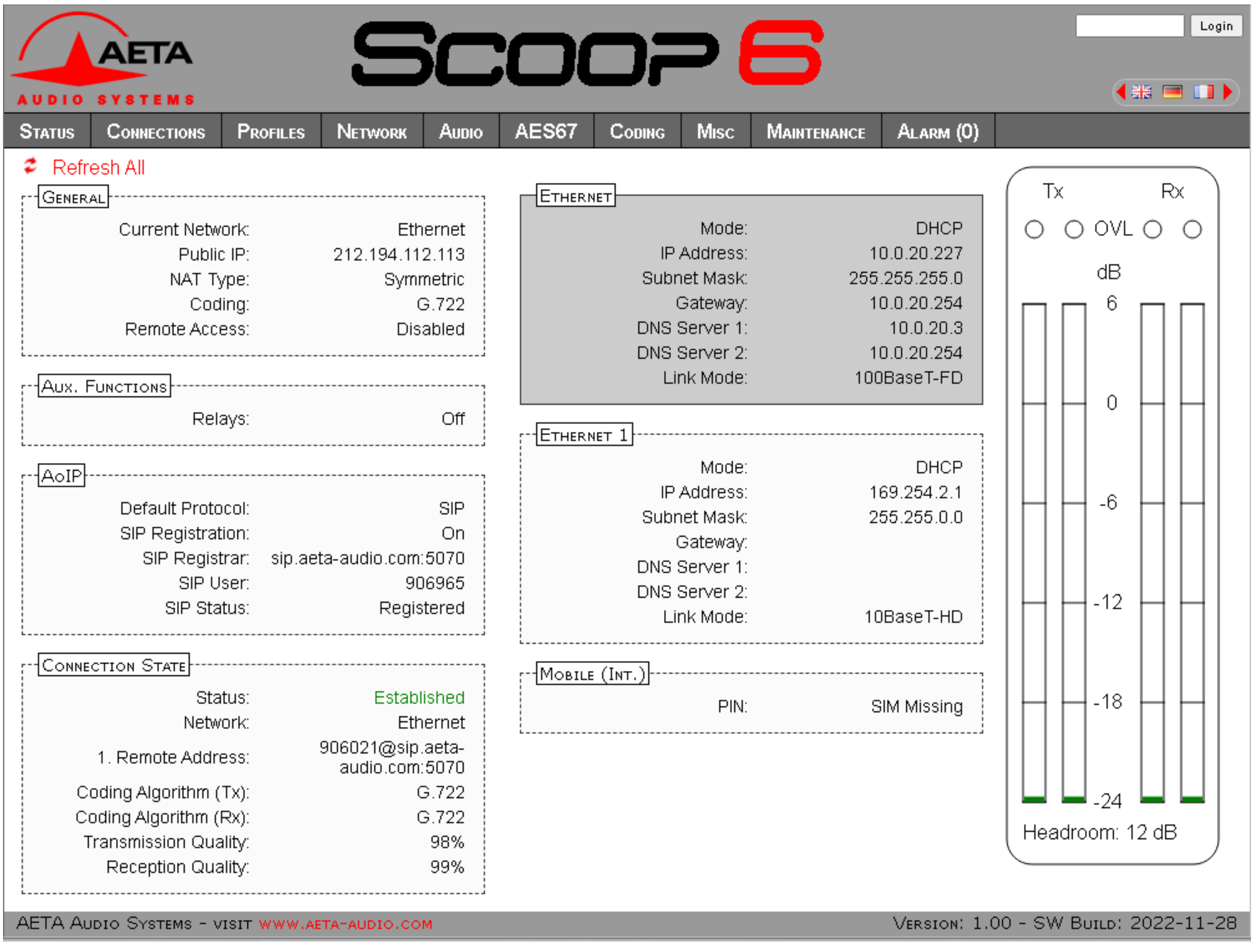
The embedded html server in Scoop 6 provides a comfortable and efficient means to control and monitor the unit. It just needs the Scoop 6 to be connected to an IP network and to be reachable from a computer, or another device with an html browser: tablet, smartphone…

In the most common case, the two devices are connected on the same local network (LAN). But it is also possible to control the Scoop 6 by remote, provided that the control device can reach it (TCP/IP port 80, HTTP protocol).

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 Scoop 6 html pages

Once the Scoop 6 is connected on an IP network, the first step is to get its IP address, from the menu: **Tools** / **About**. Then, on the control device, launch the html browser and enter the IP address of the Scoop 6 in the "address" or "URL" field. This gives access to the html server integrated in Scoop 6. 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.

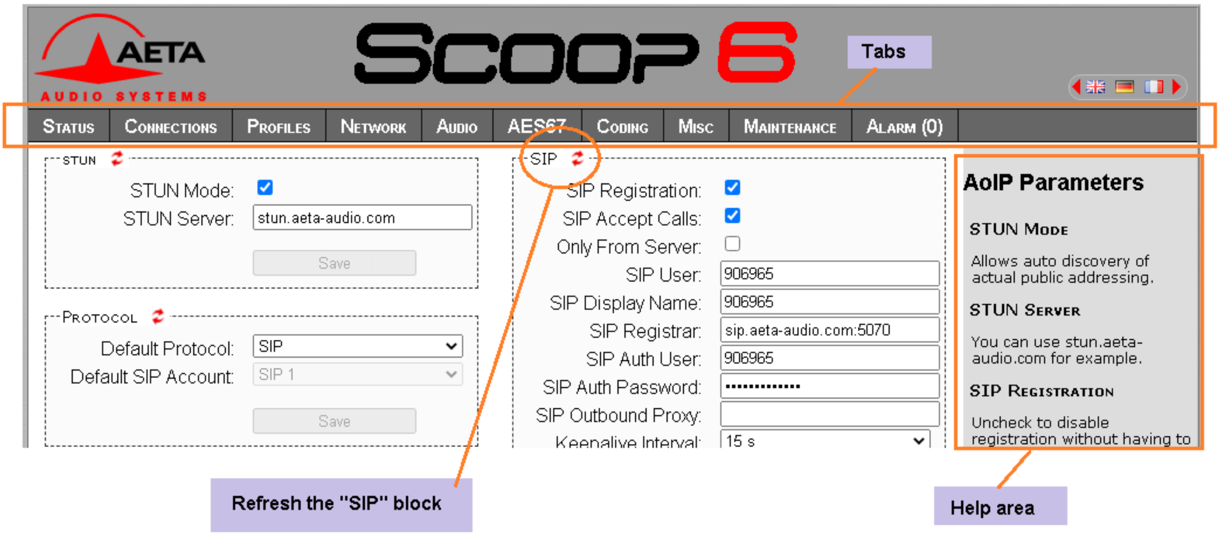
To access the other pages, you must "log in", and get for the control device an exclusive access. Any connection request from another device will remove this access.

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.

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 Scoop 6. Clicking a tab you access either a page, or a drop-down list for selecting a secondary page. *These tabs and pages are detailed in the following chapters.*

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

As a general rule, the displayed parameters are read when accessing the page, and are not refreshed automatically[1]. To force a refresh, click the icon: the data in the area or frame are read and refreshed.

ⓘ *Exception: some data on some pages is however periodically and automatically refreshed. This makes a modest bit rate, but you should exit the html pages if you want no traffic at all on the path between Scoop 6 and the control device.*

Two tabs have a specific behavior:

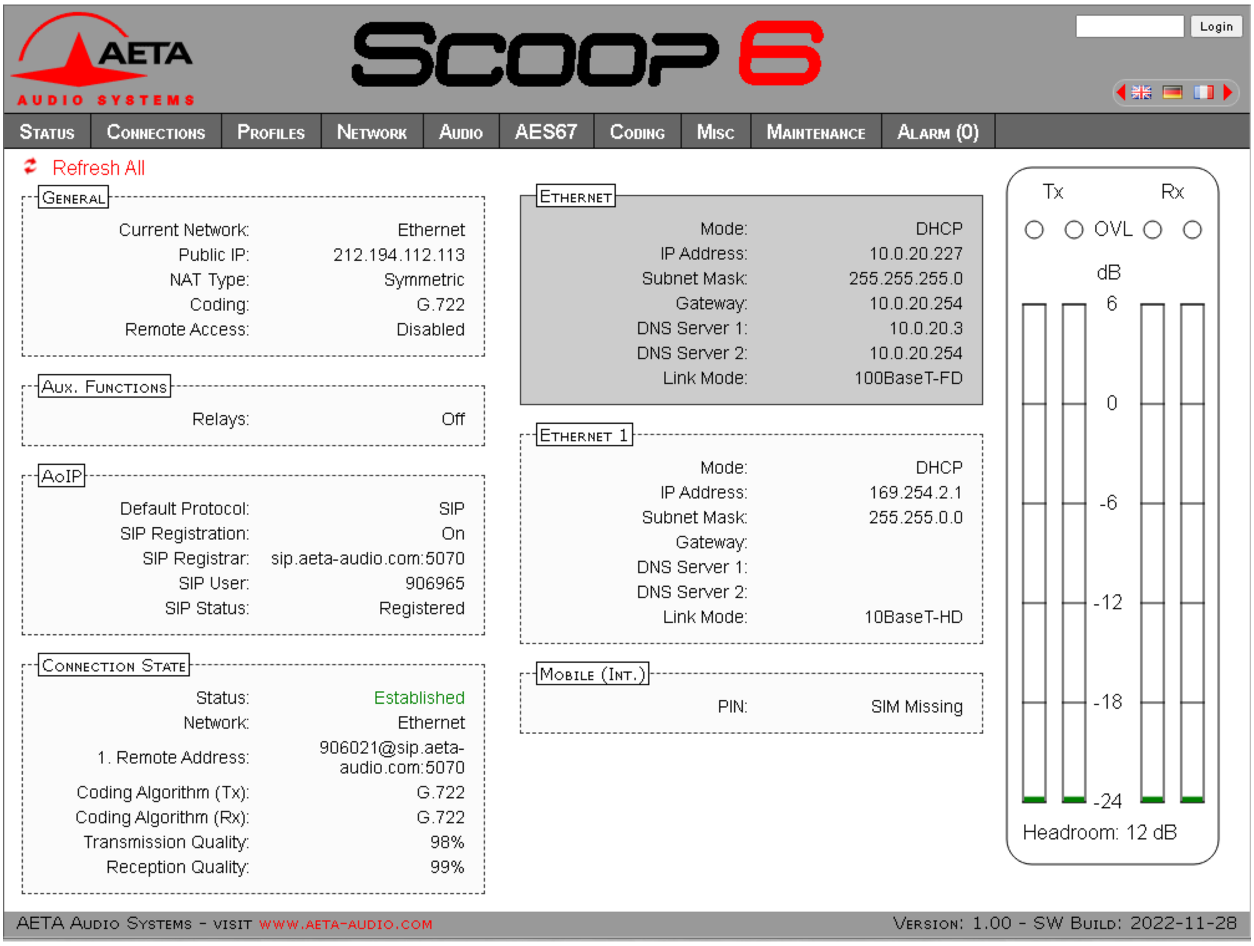
- "**STATUS**" is accessible without a login and some data are updated automatically.
- "**ALARMS**" is also updated automatically, and switches to red when an alarm triggers, showing the number of issues detected. You can then check for details by clicking the tab.

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

[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.



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. *The “Headroom” setting is adjustable using the “AUDIO” tab.*

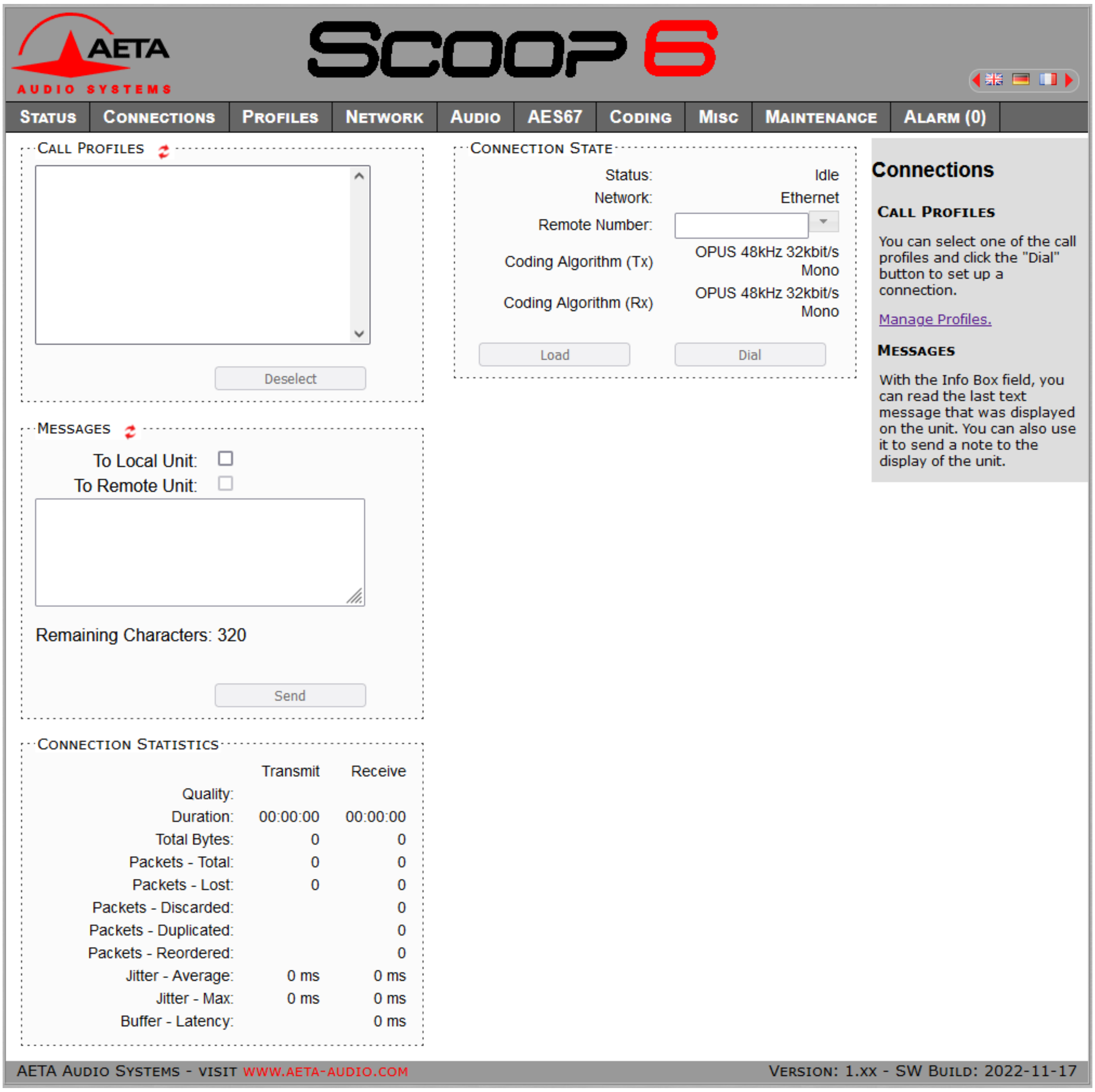
Note: the goal of these bargraphs is to provide an indication on the presence and level of the audio modulation. The measurement has a good precision for stationary signals (1 dB accuracy and resolution), but the update rate is rather slow. As a consequence these bargraphs are not appropriate for precisely monitoring a dynamic program, and they should not be considered an alternative to real program meters!

One can find in the “CONNECTION STATE” section global transmission and reception quality indicators (for IP links only).

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, call set up and call releasing.



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

- For direct dialing: the codec has to be set beforehand for the right interface (see "NETWORK" tab), and the suitable coding configuration ("CODING" tab). Enter the number (or numbers, or SIP URI…), then click "**Dial**". *You can also use one of the last called numbers: click the arrow on the right of the "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, the transmission quality indicators are available on this page (for IP links only), and indicators show the status of the relays when the "Relay transmission" feature is active. *For auxiliary functions, see the "CODING" tab.*

If the link makes use of the Opus coding algorithm, it is possible to change the transmission bit rate "on the fly", without interrupting the transmission and with no switching noise: select a bit rate and click "**Send**".

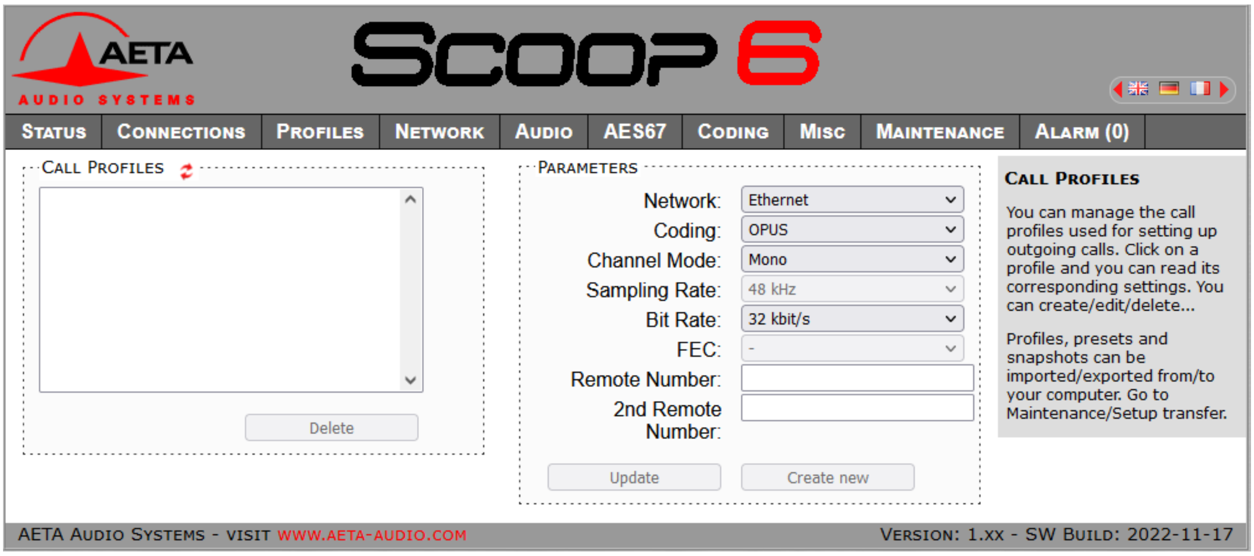
The SMS received by Scoop 6 are displayed in the "MESSAGES" frame (in addition to being displayed on the unit's display). This block can be used also to send a text to the front panel display from the control position.

ⓘ When the Scoop 6 is in dual codec mode, the "LINK STATUS" frame is split, one frame being dedicated to each codec. But selecting a call profile has the effect of returning to a normal display (simple codec), insofar as the profile data corresponds to a simple codec. (Call profiles cannot be used in dual codec mode).

5.5. "PROFILES" tab

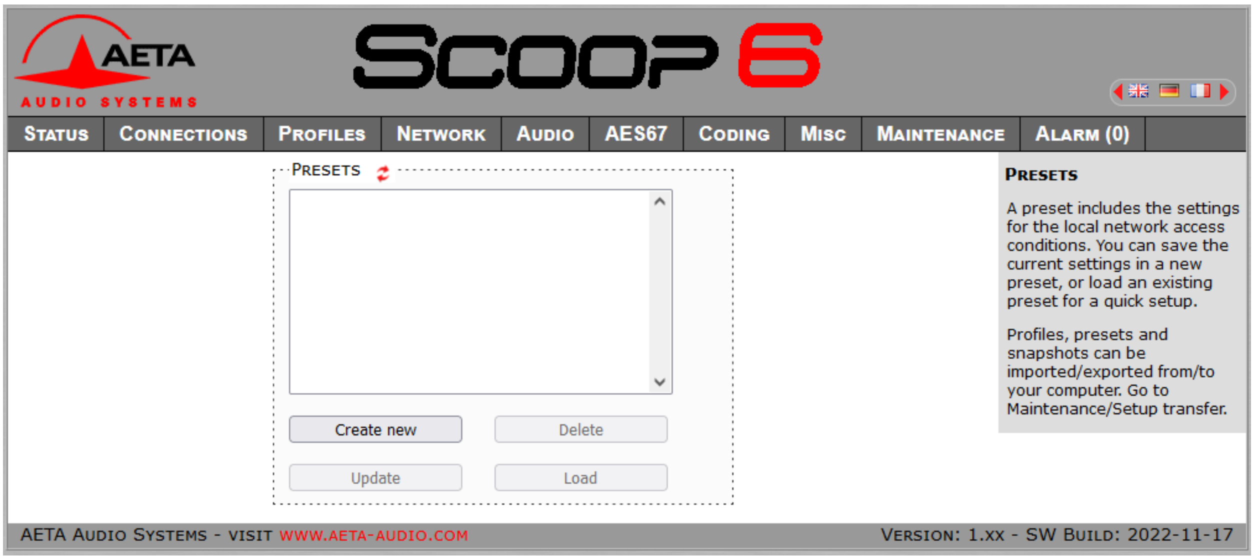
This tab proposes three secondary pages, one for each profile type.

5.5.1. "CALL PROFILES" page

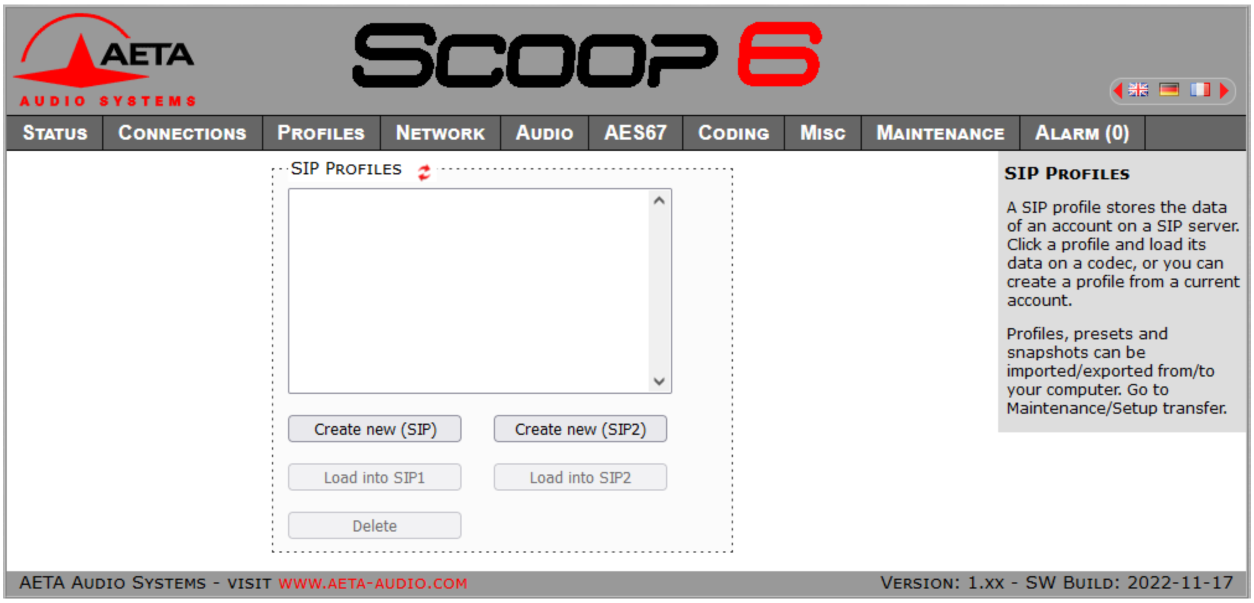


Parameters that are not relevant in a given context are shaded and inactive.

5.5.2. "PRESETS" page



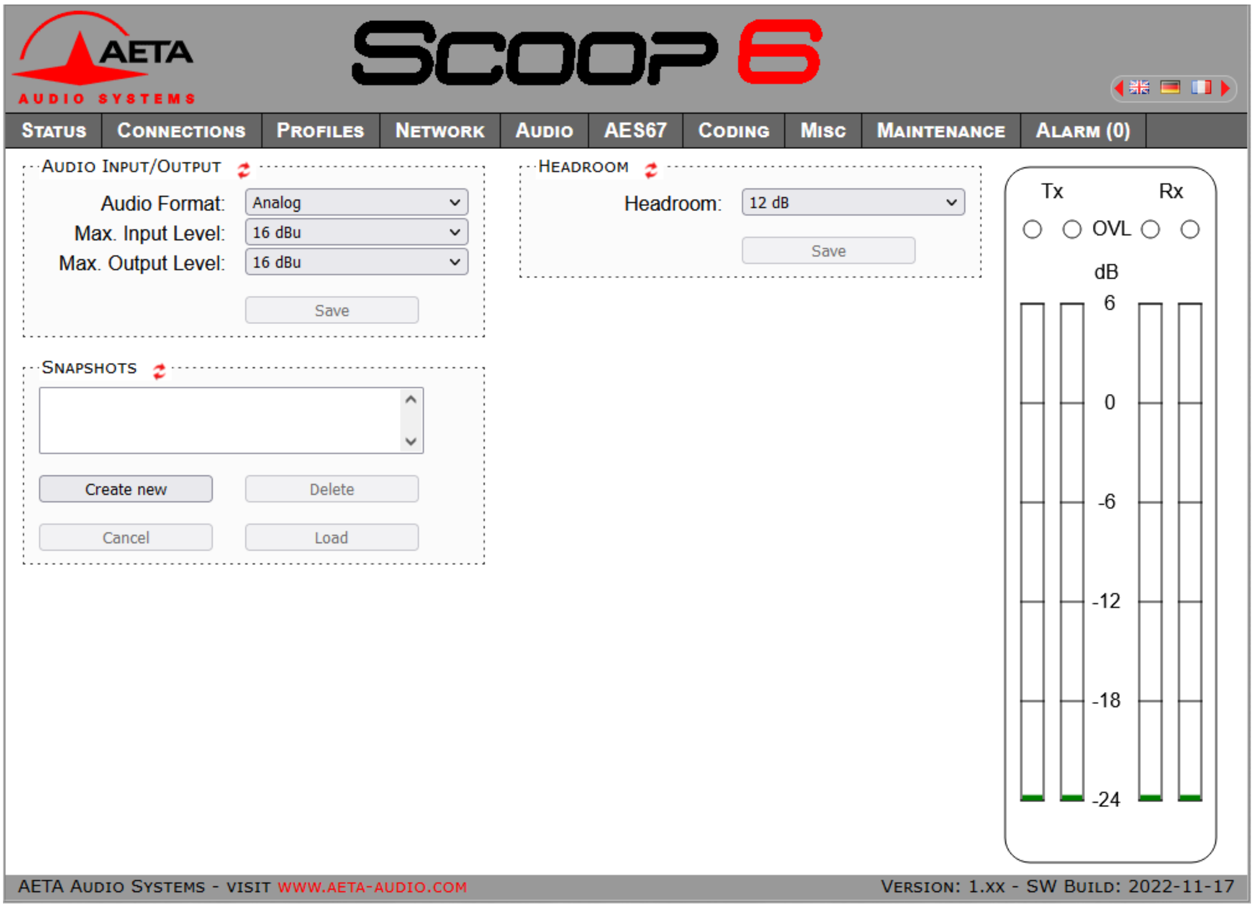
5.5.3. SIP profiles



This section is used in order to set up several SIP profiles.

5.5.4. Snapshots

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

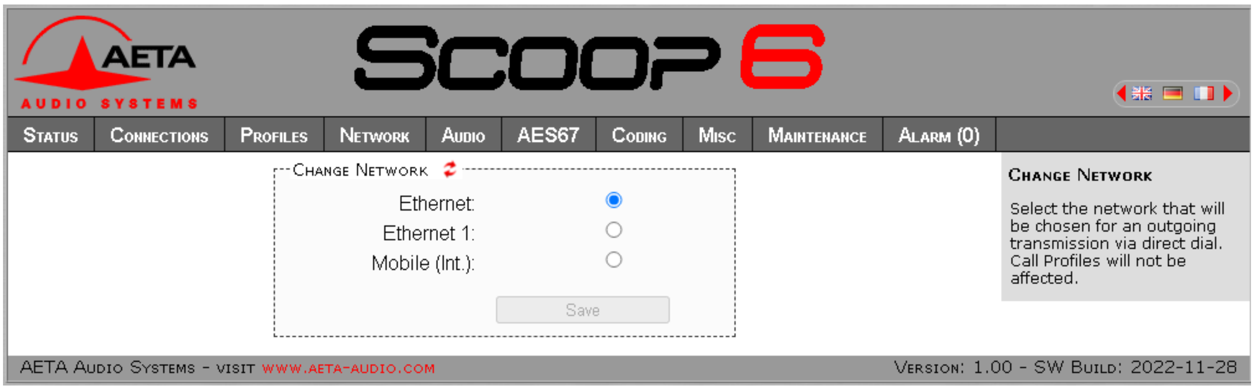


5.6. “Network” tab

This tab leads to several pages 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 mobile access, wifi ,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 interface[1].

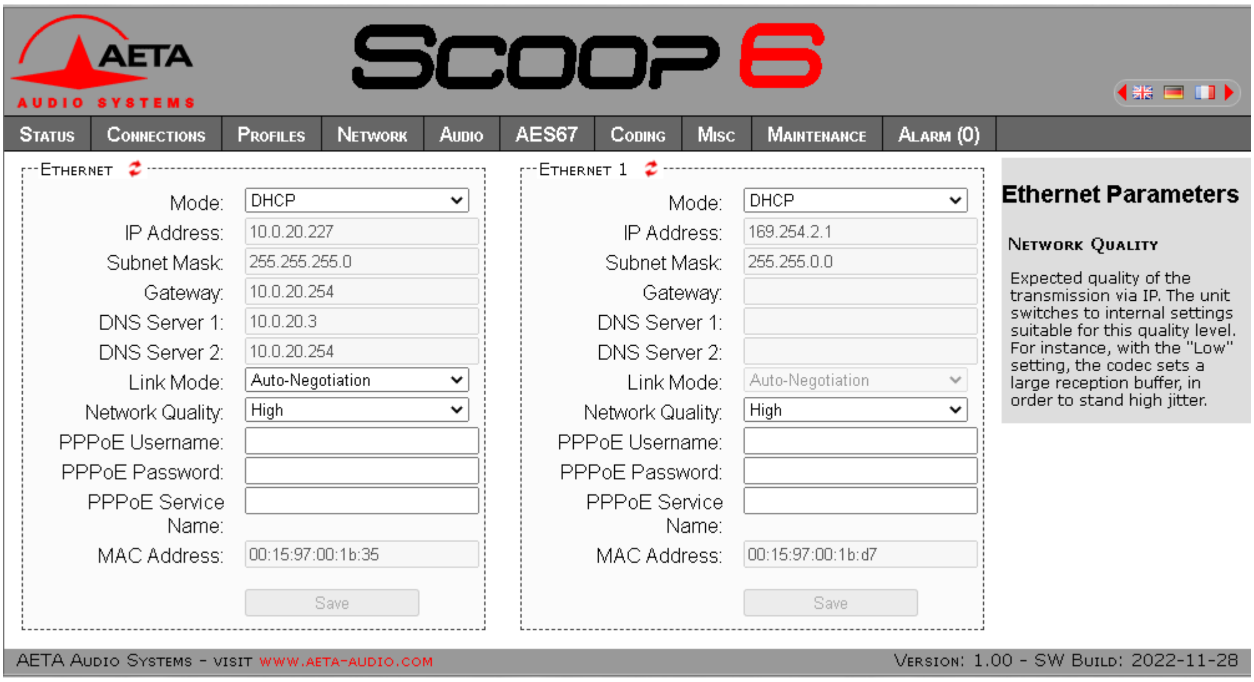


Select the desired interface and click “**Save**”.

5.6.2. “Ethernet Parameter” page

.

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



Ability to select a distinct network quality setting for both Ethernet and wireless IP interfaces.

Remember to **save the changes**, clicking the “Save” button!

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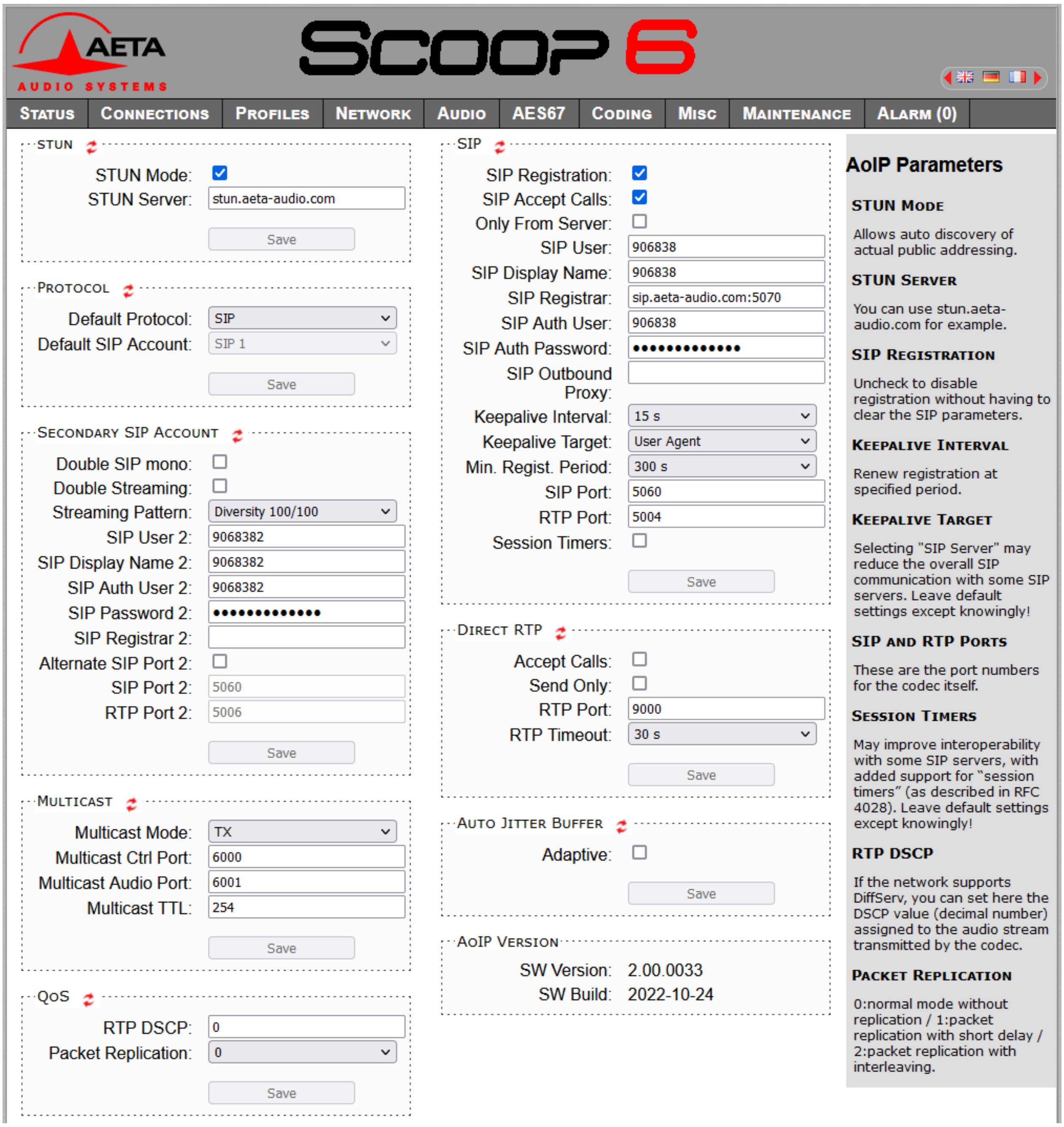
[1] Remind that the algorithm settings are memorized separately for each network interface. When switching the default interface, the codec settings for this interface are recalled.

ⓘ *Settings regarding the Ethernet interface and IP addressing are especially critical, because this is just the interface you are controlling Scoop 6 through. You can lose control because of a mistake in the settings; it might even be the normal expected consequence for a configuration change. So be very careful with the settings on this page!*

In case control is lost after an error, you must regain control and correct the settings using the front panel of Scoop 6.

5.6.3. "AoIP PARAMETERS" page

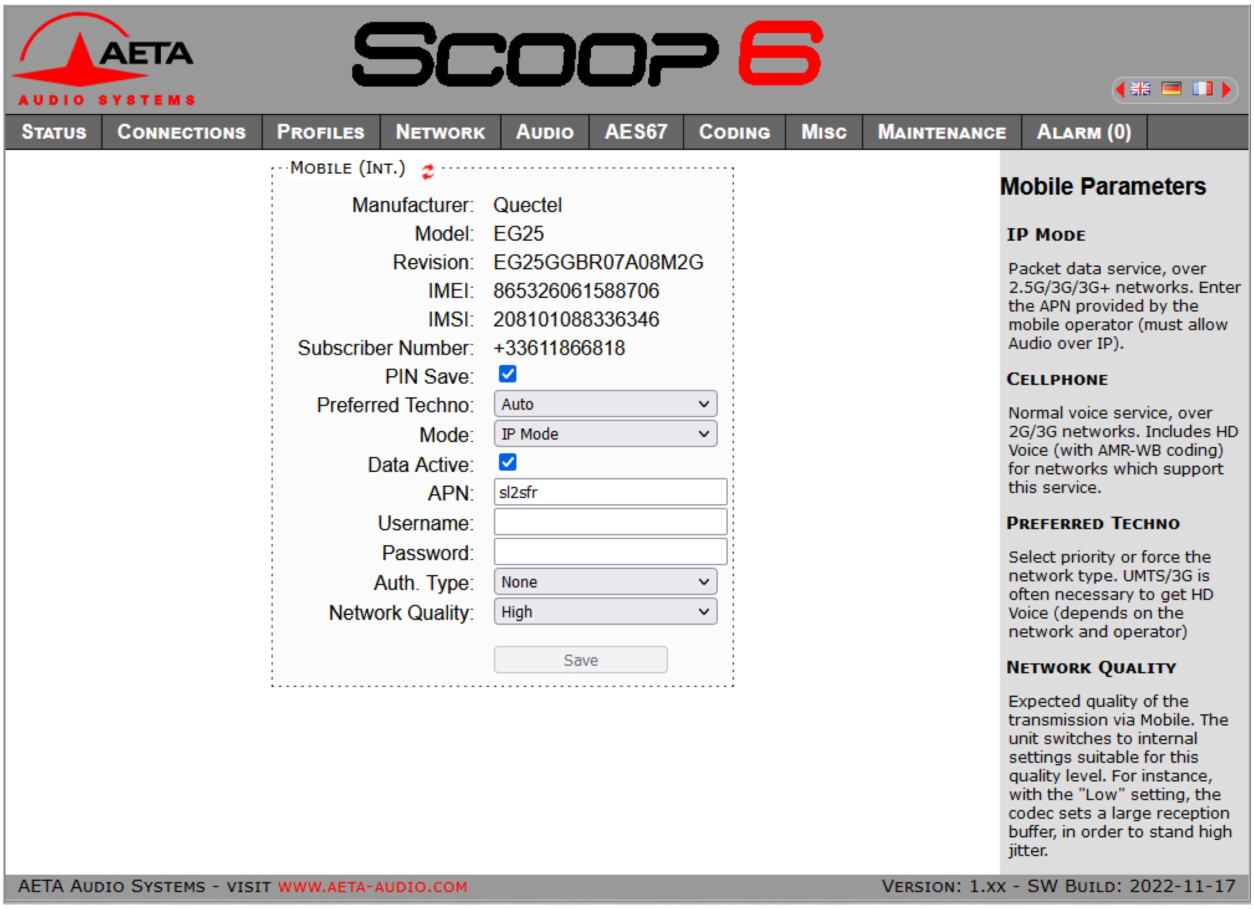


Remember to **save the changes**, clicking the "Save" button!

5.6.4. “MOBILE PARAMETERS” page

This page is only visible for units equipped with the “HD-xG” option. You cannot access all settings unless the PIN code has been entered (otherwise the view is partial only).

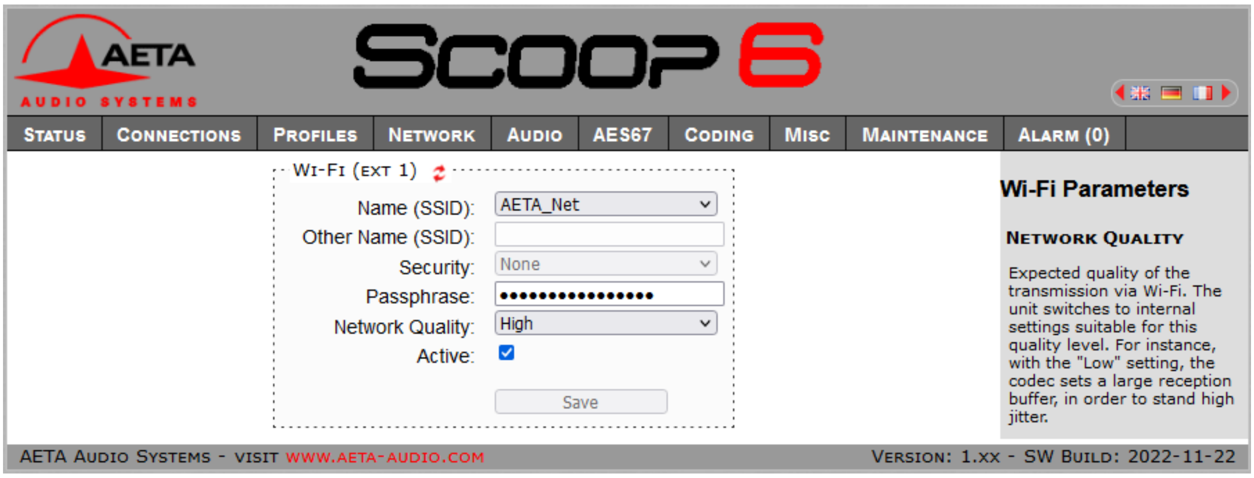
ⓘ *If you have plugged a USB mobile access device, it is also assigned a specific parameter page.*



Remember to **save the changes**, clicking the “**Save**” button!

5.6.5. “Wi-Fi” page

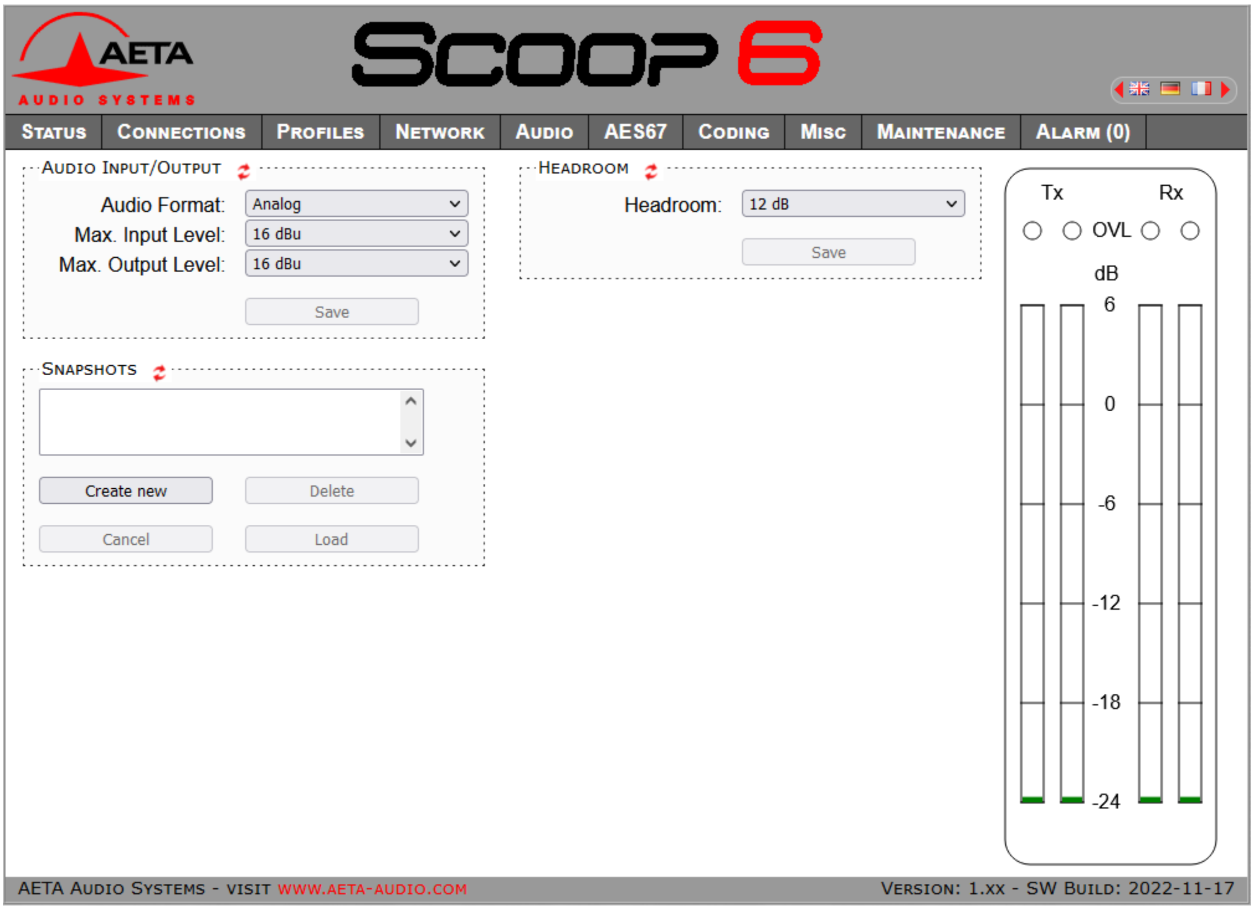
This page is only available when a USB Wifi Key is connected.



Remember to **save the changes**, clicking the “**Save**” button!

5.7. "AUDIO" tab

Like the "STATUS" page, this page shows the audio level transmitted/received to/from the network.



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.

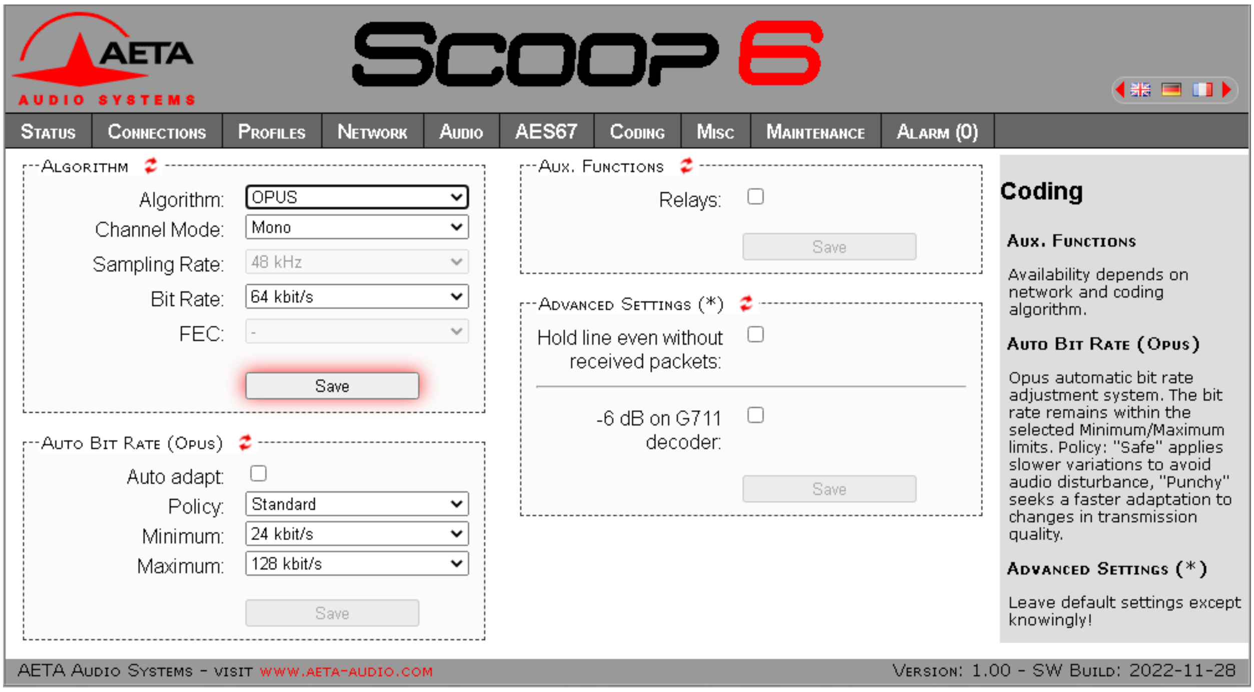
To **create** a snapshot based on the current settings, click the "**Create new**" button, and enter a name for the snapshot.

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.



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.

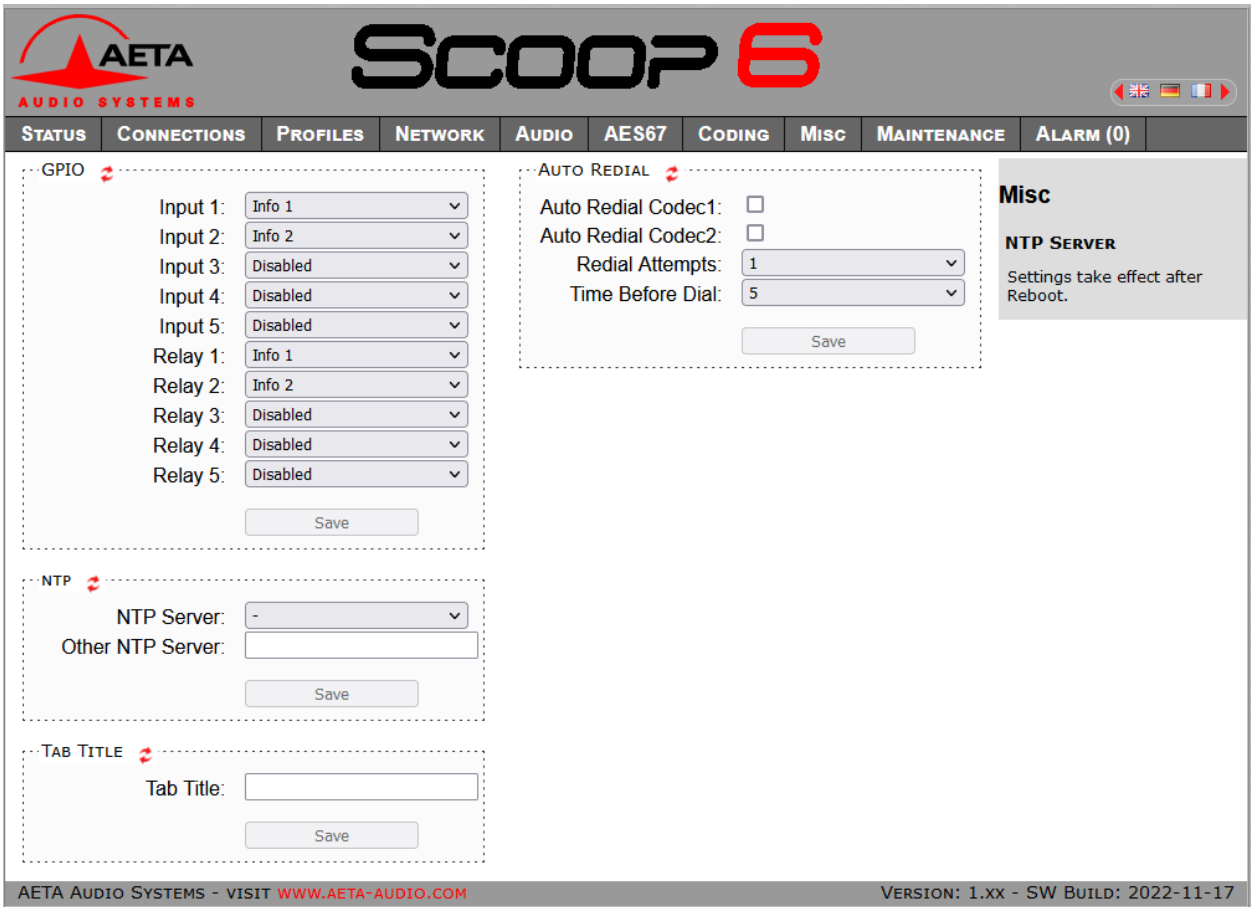
The settings are saved separately for each network interface. This means that when you change the current network interface, the coding configuration that is related to it is recalled.

Remember to **save the changes**, clicking the “**Save**” button!

ⓘ *When the Scoop 6 is set as a double codec, the “ALGORITHM” block splits as below into “CODEC 1” and “CODEC 2”. The range of allowed coding configurations is less wide than in single codec mode.*

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 : auto redialing configuration and activation, loop control activation, SMS reception enabling/disabling…



There are two additional blocks on this page:

- It is also possible to configure a **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 5.10.4, “EVENT LOG” page). If such a server is available and accessible via the IP interface, enter its address in the “NTP Server” field. A few public servers are also proposed in the dropdown list[1].

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

- 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.

Remember to **save the changes**, clicking the “Save” button!

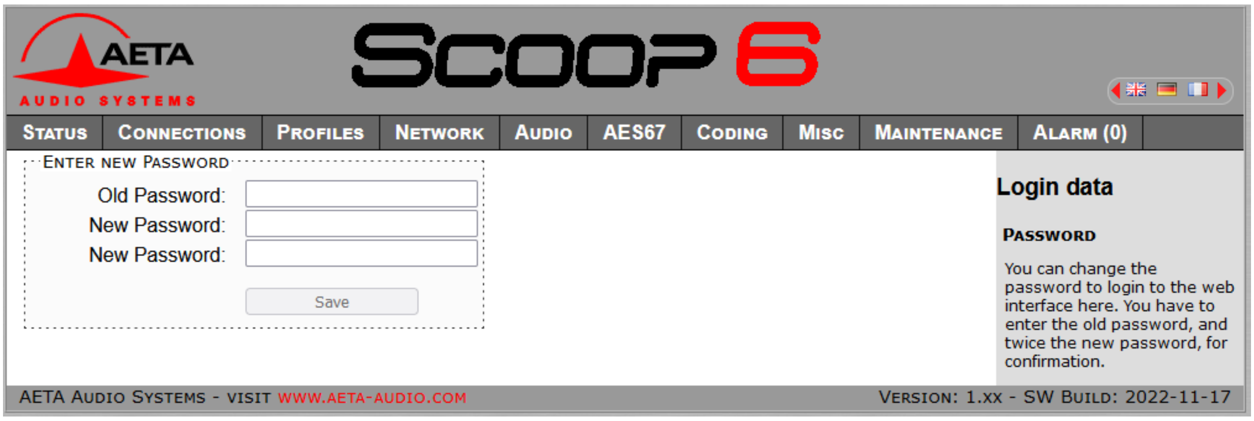
[1] Note: to be able to use these servers 1) access to the Internet from the codec must not be restricted, 2) a DNS must be set so the symbolic names can be resolved.

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, and then enter the new one and confirm it.



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

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

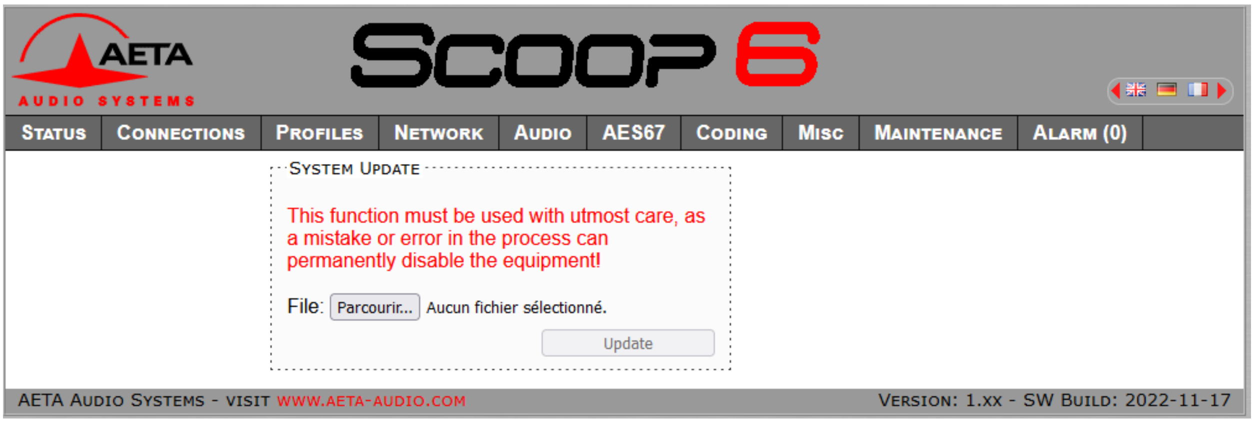
If you lose the password: using the front panel interface of Scoop 6, perform a complete reset of the settings, or a complete "factory reset". The passwords (front panel and html access) are then reset as well as the other parameters in the unit.

5.10.2. "SYSTEM UPDATE" page

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

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

- From this page, click "**Browse**...", find and select the update file.
- Click the "**Update**" button.
- The process begins, and a warning message is also displayed on the Scoop 6 front panel.
- Wait for the completion of the update. Normally the unit should reboot by itself at the end of the process.
- Once the unit has restarted, perform a complete "Factory reset".

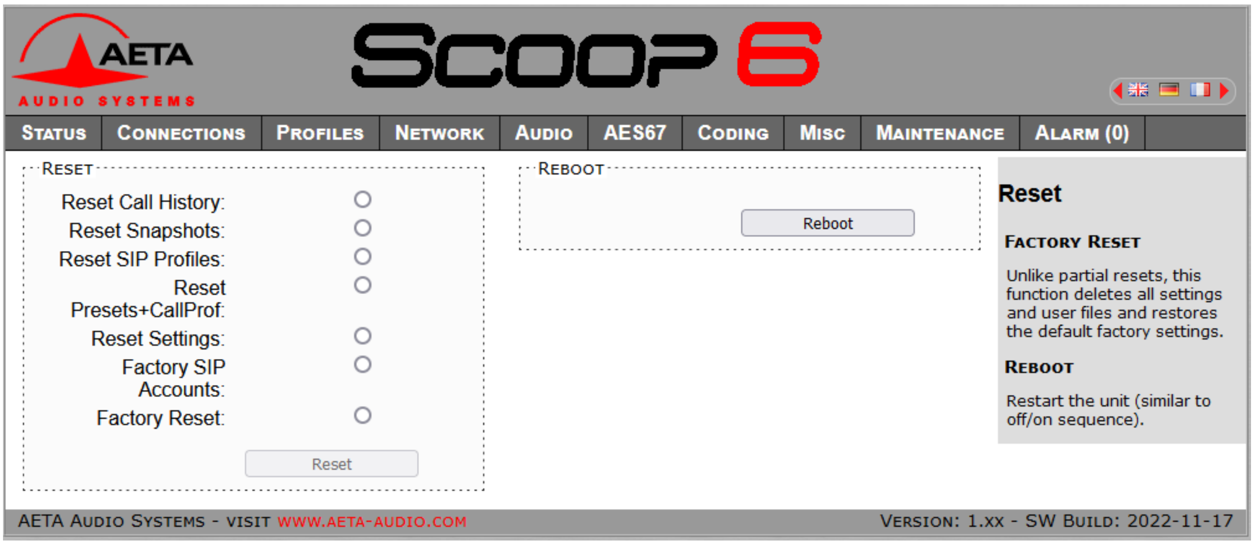


This procedure requires maximum care, because such update always implies some risk. Please note these additional recommendations:

- 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 Scoop 6 is stable (no undesirable interruption during the process).
- Make sure the Scoop 6 is not powered off and its Ethernet interface is not disconnected during the process, do not use its keypad during the process.
- Re-import your profiles and snapshots after the update. *But be aware that, although we always try to keep a maximum upward compatibility, AETA cannot guarantee the re-usability and consistency of these memories after a firmware update.*

5.10.3. "RESET" page

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



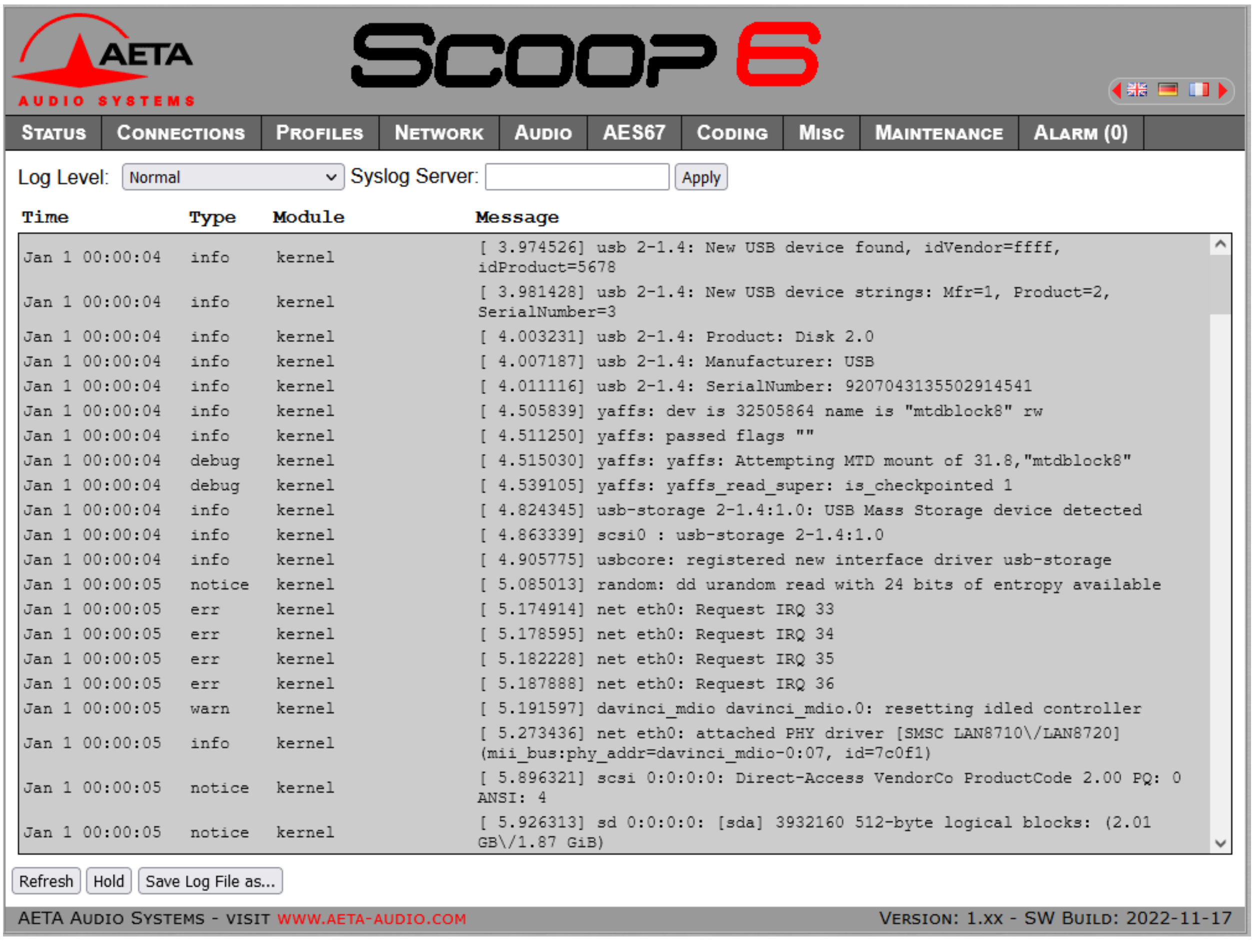
You can reset the unit completely or only clear a category of data; the available options are:

- "**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 snapshots**": all snapshots are cleared form the memory.
- "**Reset Presets+CallProf**": call profiles and presets are cleared from the memory.

The page also allows you to reboot the unit by remote, clicking the "**Reboot**" button. The device is actually rebooted when you confirm by clicking "OK" in the dialog box which then shows up. This is equivalent for Scoop 6 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.

5.10.4. “EVENT LOG” page

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



Time	Type	Module	Message
Jan 1 00:00:04	info	kernel	[3.974526] usb 2-1.4: New USB device found, idVendor=ffff, idProduct=5678
Jan 1 00:00:04	info	kernel	[3.981428] usb 2-1.4: New USB device strings: Mfr=1, Product=2, SerialNumber=3
Jan 1 00:00:04	info	kernel	[4.003231] usb 2-1.4: Product: Disk 2.0
Jan 1 00:00:04	info	kernel	[4.007187] usb 2-1.4: Manufacturer: USB
Jan 1 00:00:04	info	kernel	[4.011116] usb 2-1.4: SerialNumber: 9207043135502914541
Jan 1 00:00:04	info	kernel	[4.505839] yaffs: dev is 32505864 name is "mtdblock8" rw
Jan 1 00:00:04	info	kernel	[4.511250] yaffs: passed flags ""
Jan 1 00:00:04	debug	kernel	[4.515030] yaffs: yaffs: Attempting MTD mount of 31.8,"mtdblock8"
Jan 1 00:00:04	debug	kernel	[4.539105] yaffs: yaffs_read_super: is_checkpointed 1
Jan 1 00:00:04	info	kernel	[4.824345] usb-storage 2-1.4:1.0: USB Mass Storage device detected
Jan 1 00:00:04	info	kernel	[4.863339] scsi0 : usb-storage 2-1.4:1.0
Jan 1 00:00:04	info	kernel	[4.905775] usbcore: registered new interface driver usb-storage
Jan 1 00:00:05	notice	kernel	[5.085013] random: dd urandom read with 24 bits of entropy available
Jan 1 00:00:05	err	kernel	[5.174914] net eth0: Request IRQ 33
Jan 1 00:00:05	err	kernel	[5.178595] net eth0: Request IRQ 34
Jan 1 00:00:05	err	kernel	[5.182228] net eth0: Request IRQ 35
Jan 1 00:00:05	err	kernel	[5.187888] net eth0: Request IRQ 36
Jan 1 00:00:05	warn	kernel	[5.191597] davinci_mdio davinci_mdio.0: resetting idled controller
Jan 1 00:00:05	info	kernel	[5.273436] net eth0: attached PHY driver [SMSC LAN8710\/LAN8720] (mii_bus:phy_addr=davinci_mdio-0:07, id=7c0f1)
Jan 1 00:00:05	notice	kernel	[5.896321] scsi 0:0:0:0: Direct-Access VendorCo ProductCode 2.00 PQ: 0 ANSI: 4
Jan 1 00:00:05	notice	kernel	[5.926313] sd 0:0:0:0: [sda] 3932160 512-byte logical blocks: (2.01 GB\/1.87 GiB)

Events are logged by the codec in its internal memory card; 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.

Scoop 6 can also send event messages to a SYSLOG server is such server is available on the network: enter the address or name of this server and click the “**Apply**” button. From then on, Scoop 6 sends the designated server <u>all</u> events, regardless of the detail level selected as described above. The two “histories” run in parallel: on one hand messages to the SYSLOG server, on the other hand the “filtered” events (depending on the detail level selected) recorded in the Scoop 6 log file.

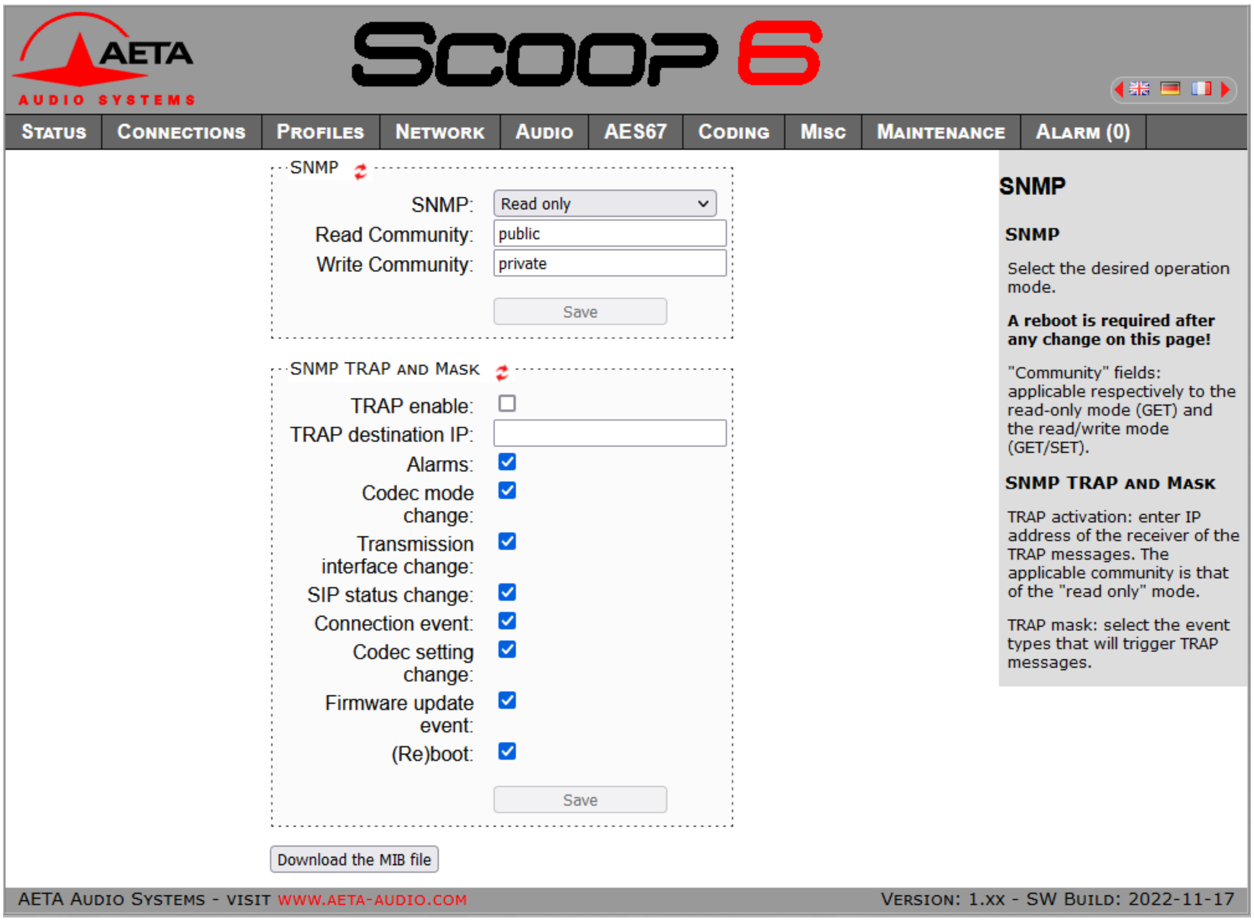
Events are time-stamped (date, hour, minute, second) from the internal clock of the unit. This clock is not backed (no battery inside the unit), but the unit can synchronize at boot time using the NTP protocol. The address of such server must be set in the “MISC” page (see 5.9 above).

ⓘ *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…

5.10.5. SNMP page



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

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

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

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

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

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

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

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

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

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 Scoop 6 (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.

ⓘ *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

This page I used to set up the remote access settings.

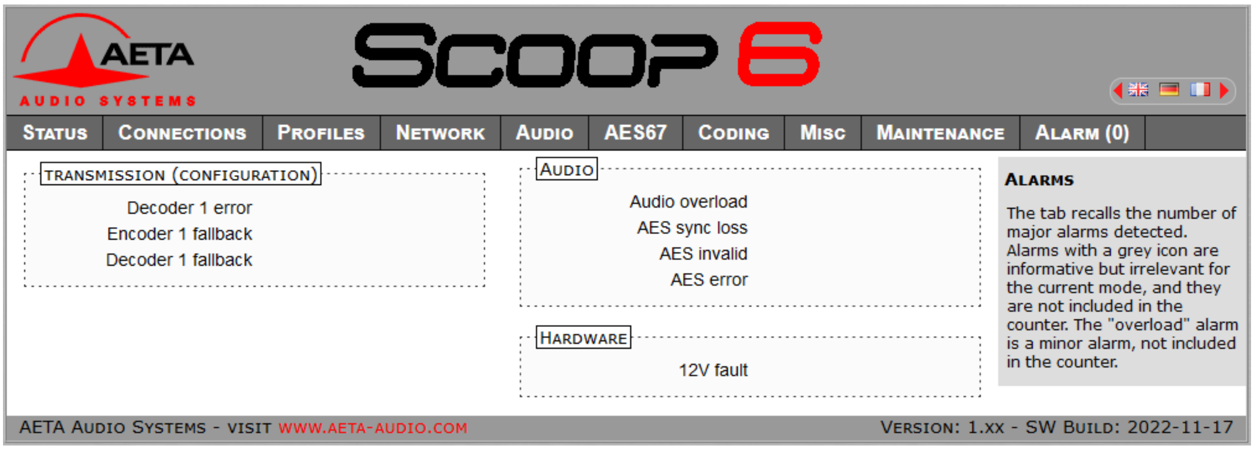
There are a few differences between the options present:

- "DATA FOR REMOTE ACCESS" becomes "DATA FOR REMOTE ASSISTANCE" if the "Remote access" option is not installed
- "Codec password" field : this password must be known to obtain remote access to the html pages, and thus protects the device against unwanted access. This codec password is not present if the "Remote access" option is not installed.

ⓘ If you are accessing this page using remote access, be aware that any changes to this page may cause you to lose this access. In the best case (change of password), you will have to reconnect to the remote access server, using the new password.

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, like in the example below with two alarm conditions detected:



Clicking the tab, you reach this "Alarm" page showing all the alarm conditions. All possible issues are listed, but only those actually detected are highlighted with the  icon.

The issues are grouped in three areas:

- Alarms related to "**Transmission**": essentially issues related to the decoder synchronization. In addition, 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 (minor alarm), AES input errors.
- Hardware alarms (internal alarms): power failure.

6. Technical characteristics

6.1. Characteristics of interfaces

6.1.1. Analogue audio 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.

Maximum input level:	adjustable from 8 to +20 dBu ± 0.3 dB
Nominal input impedance:	10 kΩ

6.1.2. Analogue audio 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 8 to +20 dBu ± 0.3 dB	
Nominal load impedance:	600 Ω or 10 kΩ	
Output impedance:	<50 Ω	
Symmetry:	> 40 dB	(ZL = 150 Ω)

6.1.3. Digital audio input and output

These interfaces comply with recommendations:

- AES3-2009
- EBU Tech. 3250-E
- CCIR Rec. 647

They support (in genlock mode) a sampling rate from 28 to 48 kHz. In master mode, the unit can be set in one the following sampling rates: 32 and 48 kHz.

6.1.4. Headphone output (front panel)

This output (3.5 mm jack on front panel) is 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.

6.1.5. Ethernet Interfaces

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 second interface can also operate at 1000baseT.

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

6.1.6. Antenna sockets (mobile networks)

Available on units equipped with the "HD-xG" mobile network option. See also chapter 2.1.2: Mobile network access.

These sockets are used to connect one or two antennas (one multiband antenna is included with the mobile network option).

The interfaces are male SMA (female contact), 50 Ω impedance.

The internal mobile access module can operate in following bands:

Frequency band	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/5G							

6.1.7. Relay transmission interface ("AUX" socket)

The relay transmission interface (refer to 2.4.1, Transmission of isolated relays) is available on the RJ45 "AUX" Socket. It includes two isolated current loop inputs and two dry contact outputs.

The following table shows the pinout of the socket for this function:

Pin	Function
6	Output loop n°4 (b)
8	Output loop n°4 (a)
7	Output loop n°1 (b)
8	Output loop n°1 (a)
2	Input loop n°1 (b)
4	Input loop n°1 (a)
3	Input loop n°4 (b)
4	Input loop n°4 (a)
1	+5V of internally supplied power supply
5	0V of power supply

All loops are isolated and bi-directional (free polarity).

The characteristics of the input loops are:

Input loop control current:	6 mA	(max. 100 mA)
Resistance of input loop:	~ 560 Ω	(current limiting series resistor)
Input loop isolation:	> 1500 V_{RMS}	

A +5V to +12V source may be connected directly on an input loop, because the internal series resistor is dimensioned for this purpose. For a higher voltage source, it may be necessary to limit the input current.

The characteristics of the output loops are:

Maximum switching voltage:	350 V peak
Maximum switching current:	120 mA
Resistance of output loop:	< 35 Ω
Output loop isolation:	> 1500 V_{RMS}

The 5V power supply is available from the unit to power a low-consumption device (maximum 300 mA current consumption), e.g. to power the input loops or LED indicators connected to the output loops.

6.1.8. “Digital I/O” interface

This 15 pin female sub-D socket provides the GPIO signals described in 2.4.2, Transmission of GPIO. The pinout is shown in the following table:

Pin	Function
1	+5V of internally supplied power supply
2	Input 2
3	Input 5
4	Input 3
5	Signal and power ground
6	Output 2
7	Output 5
8	Output 3

Note: the GPIO index starts at 3 because indexes 1 and 2 are allocated to the relays (see above in 6.1.7).

The GPIO signals do not have galvanic isolation as the relays. They feature the following characteristics:

Characteristics	Min	Nominal	Max	Notes
GPI: voltage (low level, active)	-0.5 V	0 V	1 V	
GPI: current (@ 0 V)			110 µA	
GPI: voltage (high level, inactive)	3 V	5 V	7 V	[1]
GPO: voltage (low level, active)	0 V	0 V	0.55 V	
GPO: current (low level, active)			32 mA	
GPO: voltage (high level, inactive)		5 V	6.5 V	[2]

[1] An input pull-up ensures a high level if the GPI are “open” (not connected).

[2] The GPO are “open drain”, and must be “pulled up” (to 5 V nominal) for the high level, but an internal pull-up ensures a 5 v voltage when they are inactive, even with no external polarization.

6.1.9. AC power supply

The unit is connected to mains via an IEC 3 pin socket, and accepts a 85-263 Vac, 47-63 Hz AC source. Protection is provided by a resettable fuse.

6.1.10. DC power supply (option)

When the 12V DC option is installed, the unit features, in addition to the mains socket, a 2.1mm jack socket for connecting a 12 V DC supply (non isolated). The unit includes overvoltage and overcurrent protection (resettable fuse on the DC input).

The DC supply is redundant with mains power, with priority to mains whenever it is present [1].

Characteristics	Min	Nominal	Max	Notes
Supply voltage	10 V	12 V	17 V	[1]
Supply current	0.5 A		1 A	[2]

[1]: Beyond 15 V the DC supply takes precedence, and provides power to the unit even if AC power is available.

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

Connections on the Jack socket: Ground is connected external pin, and the central pin must provide the +12 V voltage.

6.2. 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.

6.2.1. Transmission gain

The drift in time of the gain from the input to the output of the codec is less than ± 0.3 dB.

6.2.2. 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	-∞	0 dB
20 Hz	300 Hz	-2 dB	0.1 dB
300 Hz	20 000 Hz	-0.1 dB	0.1 dB

6.2.3. 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.

6.2.4. 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 kHz coding frequency).

Maximum noise level[1]: - 55 dBm (quasi-peak detection, CCIR weighting)
(or - 61 dBq0ps)

This result in a signal to noise ratio (SNR) of more than 71 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.

6.2.5. Total distortion vs frequency and level

Total distortion relative to maximum level (or THD + N) is less than –75 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).

6.2.6. Crosstalk

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

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

6.2.7. Gain and phase difference between channels

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

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

6.3. Network protocols and ports

The Scoop 6 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, Telnet, 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
	6000	Control	⇆	Remote control ("command line" mode); used by Scoop4Man and TeleScoop
	7001	Control	⇆	Remote control; used by ScoopManager
UDP	123	NTP	⇆	For querying an NTP server
	514	SYSLOG	→	For sending messages to a SYSLOG server
	2382	AETA enumeration	⇆	Used by "AetaScan"[1]
	3478	STUN	⇆	For querying a STUN server
	5004	RTP	⇆	For audio streaming (SIP)
	5005	RTCP	⇆	For audio streaming (SIP)
	5006	RTP	⇆	For audio streaming (double SIP codec)
	5007	RTCP	⇆	For audio streaming (double SIP codec)
	5060	SIP	⇆	SIP signalling
	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 ports are mandatory for creating AoIP links.*

[1] AetaScan is a tool that scans a network to identify AETA codecs and e.g. detect their IP addresses.

The following must be set on the firewall:

- Allow the agent to access the SIP port (UDP 5060 or 5070) of the sip.aeta-audio.com server (85.214.119.212), and accept returning SIP/UDP packets. Block SIP requests (UDP port 5060) towards any other server, and SIP requests from any other source than the server's address.

- For the audio streams (RTP/RTCP), allow the agent to send packets to the port range 55004 to 59999 on AETA proxy server, and accept returning UDP packets. Block RTP streams to/from any other address.

6.4. Power supply

The codec operates from mains 85-263Vac, 47-63 Hz.

As an option the product can also be powered from 12V DC source.

The maximum power consumption is 12 W (depending on version and installed options).

6.5. Dimensions and weight

The unit is a ½ 19 inches frame of 1U height (44 mm or 1.75”) and 265 mm overall depth (12.5”).

Its maximum weight is about 3.8 kg (depends on installed options).

6.6. 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 Scoop 6 complies with “CE” directives regarding safety and EMC.

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

6.7. Options

The basic version of Scoop 6 includes an Ethernet interface for IP transmission. Many available options can be added to complement the capabilities of the Scoop 6.

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

Code	Option	Description
80 00 291 01	HD-4G mobile	Addition of the mobile network access LTE/4G
80 00 291 02	HD-5G mobile	Addition of the mobile network access LTE/5G
80 00 291 03	AES67 2CH	Add access to Ravenna's AES67
80 00 291 04	Dante 2CH	Add access to dante's AES67
80 00 291 05	Option Wi-Fi	Wifi using a USB device included in the option
80 00 291 06	Power redondance	Secure plug preventing accidental unplugging (Option 12V + power supply)
80 00 291 07	Option 12V	DC 12V
80 00 292 01	Remote access	

6.8. Accessories and related products

The Scoop 6 is supplied with a mains lead and a CAT5 Ethernet patch lead.
The "HD-4G" option is supplied with two multiband antennas, fitted with 3 m cables with SMA plug.

Additional accessories or spare parts are available:

Code	Description	
ZAAS00056	µSIM adaptator	
80 00 294 01	External AC/DC with 2.1 mm Jack connector	
80 00 294 02	Rack mount kit	
80 00 294 03	Blanking plate ½ U	

7. Annexes

7.1. Overview of the SIP protocol

7.1.1. 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).

7.1.2. 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).

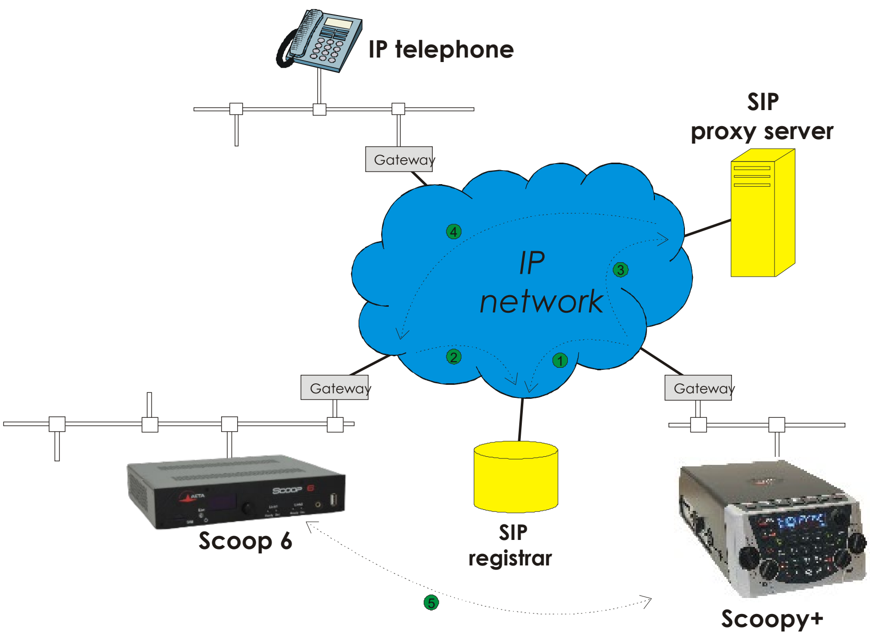


Figure 4 – Setting up an SIP session with server(s)

[1] Scoopy+ is a portable audio codec from AETA; the description here also applies to both Scoopy+ S and the Scoop 6, as they are both SIP compliant and mutually compatible

What happens then on the network: the Scoopy+ sends the request ❸ (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 ❹ to its destination. Resolving the SIP URI to a physical network and address uses mechanisms similar to those used for resolving URLs. Several proxys 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 ❺ 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.

7.1.3. Setting a link without a SIP server

Contrary to a commonplace but wrong idea, the SIP protocol does <u>not</u> 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.*

7.2. 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.

7.2.1. 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.

7.2.2. 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/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 successful[1] 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 Scoop 6 or 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.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 router[2]:

- UDP 5060 (= SIP port)
- UDP 5004 (RTP port) and 5005 (RTCP port), and if needed 5006 and 5007 (double codec)

---

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

[2] If needed you can change these numbers on the Scoop 6

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 registered[1] 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	

ⓘ *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.

7.2.3. 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

ⓘ *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.

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

ⓘ *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.

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

ⓘ *Reminder: the SIP protocol (always used by AETA codecs) 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.*

7.3. 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 software also includes the PJSIP stack covered by the GPL license.

The Opus encoder and decoder are covered by the BSD license.

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NOTES

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