MultiScoop

Modular multichannel AoIP, ISDN and AES67 codec up to 16 audio connections in 3RU

User manual

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1. Presentation – Overview

The MultiScoop is a modular codec system that can manage a large number of audio links in a compact chassis.

Up to 16 mono AoIP or ISDN connections can be performed with a 3RU unit, with several available audio formats: analog, AES3 or AES67.

MultiScoop features a high operational reliability thanks to the ability to "hot" plug or replace every module, and the redundancy available for all common functions.

The Scoop Manager software is provided with MultiScoop for an easy management of the audio links.

This chapter describes the global system and the chassis housing the modules.

These are described in the following chapters 2 to 5, and also their operating mode.

The annexes bring miscellaneous additional information, including an index you can use to look for a given information topic.

A specific user manual is available separately for the Scoop Manager software.

The following table shows the main features of the MultiScoop and its modules.

Characteristics	Notes
19" modular chassis	
Eurorack 3RU format, 483x132x282 mm	
14 slots of 6TE width including:	
"Power" modules — 2 slots	
"Control" modules — 2 slots	
"AES67" modules — 2 slots	
"Codec" modules — 8 slots	
85-263 Vac mains supply — 2 sockets	
Codec module	
Ethernet/IP transmission interface (on front panel)	
Ethernet interface, 10BaseT / 100BaseT; TCP/IP, UDP/IP, RTP, RTCP protocols	
Audio transmission (AoIP) in unicast mode : SIP signalling protocol, SDP, RTP streaming	
Single or double AoIP/SIP codec	
Audio transmission in multicast mode, RTP streaming	
Net bit rate 12 to 256 kbit/s (depending on coding algorithm, linear coding excluded)	
ISDN transmission interface *only for module in "AoIP/ISDN" version*	
S0 (BRI) interface (on front panel)	
Single codec 64 or 128 kbit/s, or double 64 kbit/s codec	
5AS auto configuration on incoming calls	
Audio interfaces	
Two analog inputs and two analog outputs, with adjustable gain	a
Digital input and output, AES3 format	a
2 input channels and 2 output channels on AES67 interface	b
Audio coding algorithms (audio modes)	
G711 (standard ISDN telephony, AoIP) — Mono	
G722 SRT, H221, H242 — Mono	
MPEG Audio Layer II, Layer III — Mono, Stereo, Dual mono, Joint stereo	c
MPEG AAC-LC , HE-AAC, HE-AAC v2 — Mono, Stereo	
OPUS — Mono, Stereo	d
Linear L16, L20, L24 — Mono, Stereo	d
Auxiliary functions (availability depending on transmission interface)	
GPIO transmission: 2 GPI and 2 GPO	a
Control and supervision	
Embedded html server and/or remote control using Scoop Manager	
Access via Ethernet interface, or via "Control" module as a gateway	
Remote access via Internet (AETA Remote Access service)	Option
Power module	
85-263 Vac mains input — 1 dedicated socket for each module	
Operation with 1 module, or 2 modules in "hot" redundancy (interruption-free)	
Hot insertion/replacement (without disturbing the codecs)	
Module Control	
Supervision du châssis et passerelle pour commande des modules "Codec"	
Ethernet interface, 10BaseT / 100BaseT (on front panel)	
Embedded html server	
Ability to operate 2 modules in redundancy	
AES67 module	
16 input channels and 16 output channels linked to the inputs/outputs of the "Codec" modules	
Ethernet interface, 1000BaseT — 2 interfaces (layer 2 switch)	
Ravenna format, AES67 compliant (Dante compatible)	
Ability to operate 2 modules in redundancy	
Notes :	
a — *An optional rearmount module is required*	
b — *At least one AES67 module must be installed in the MultiScoop*	
c — *Layer III only available for AoIP transmission*	
d — *Opus and linear coding only available for AoIP transmission*	

Table 1 – Main features of the MultiScoop

1.1. Physical description of the system

The MultiScoop comes in the form of a 19" chassis in "Europe" 3RU format, able to house up to 14 3RU modules of 6TE width (30 mm).

Figure 1 – Front view of the MultiScoop (partially equipped with 3 Codecs)

The slots are reserved to the types of modules according to the following diagram:
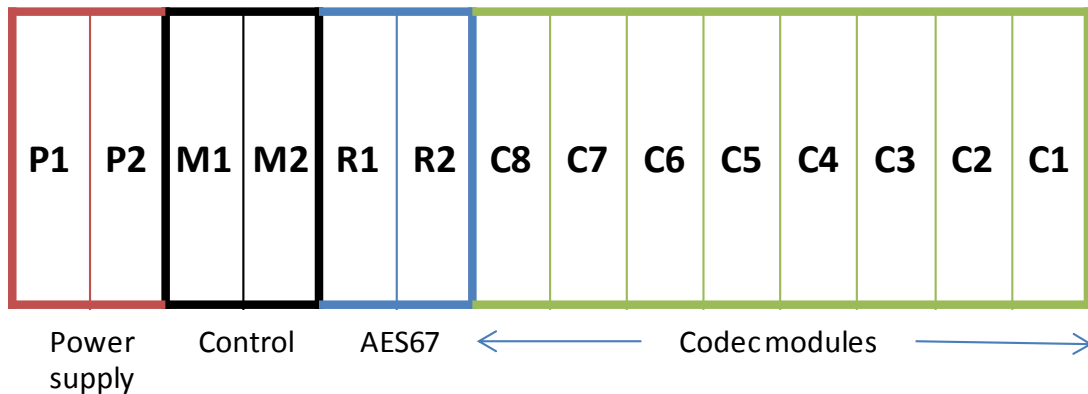
Table 2 – Assignment of the chassis slots (front view)

From left to right the slots are assigned as follows:

- P1, P2: "Power" (1 or 2 modules)
- M1, M2: "Control" (0, 1 or 2 modules)
- R1, R2: "AES67" (0, 1 or 2 modules)
- C1 à C8: "Codec" (1 to 8 modules)

The unused slots are filled by blanking plates.

On the rear side one can find following items: (from right to left, see picture below):

- "Power 1": socket and switch combo feeding the "Power" module #1 (if present).
- "Power 2": socket and switch combo feeding the "Power" module #2 (if present).
- For each installed "Codec" module, a corresponding rearmount module can be installed. It especially provides the audio interfaces of the Codec in analog and AES3 formats. *It is possible not to install this rearmount module, in which case the codec module only makes use of the AES67/Ravenna interface (an AES67 module must be present in the chassis).*
- The unused slots are closed by blanking plates.

Figure 2 – Rear view of the MultiScoop (partially equipped with 3 Codecs)

1.2. Installation and set up

A 3RU height empty space must be available in the 19" rack for installing the MultiScoop chassis.

The equipment is cooled by natural convection. Avoid obstructing the air vents on the top and bottom sides of the chassis. It is advisable to leave at least ½ RU (22 mm) free on each side. *This space will also leave room for cables from the back of the rack to the front faces of the MultiScoop modules.*

The extraction levers on the front of the modules facilitate the insertion or removal of the modules. Tighten the locking screws (top and bottom of the front panel) after inserting a module.

The mains connection (IEC outlets on the rear) must be provided for each power module installed in the chassis. Lockable cords are provided with these modules. Similarly, a mains switch and a fuse are available for each power supply module.

All modules can be inserted or extracted "hot" while the system is powered.

1.3. Technical characteristics

1.3.1. Power supply

The codec is powered by one or two mains sources (85-263Vac, 47-63 Hz). Each mains connection is protected by a fuse (T2A 230V, 5x20 mm format).

The maximum overall consumption is 50 W (depending on installed equipment) . Where appropriate, the consumption is divided between the two sources.

1.3.2. Dimensions and weight

The unit is a 19 inches chassis of 3U height (132 mm) and 282 mm overall depth.

1.3.3. 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 MultiScoop complies with "CE" directives regarding safety and EMC.

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

1.3.4. Accessories and related products

In addition to the various modules and their accessories (described in the following chapters), additional accessories or spare parts are available:

Code	Description	
80 00 273 02	6TE blanking plate (front or rear mount)	

The MultiScoop system is supplied with the Scoop Manager configuration and supervision software. This allows the management of audio links on the codec modules of the MultiScoop (from a Windows PC).

ⓘ *Other references may be available later, in which case you can find them on our web pages; consult our site www.aeta-audio.com .*
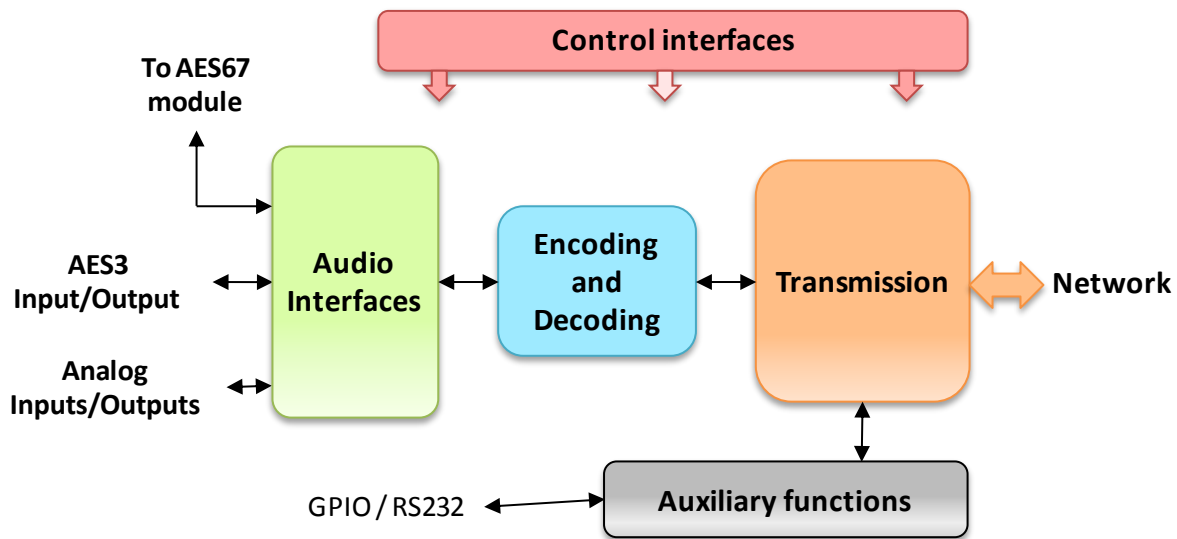
2. Codec module

The "Codec" module allows the bi-directional transmission of one or two audio signals with bit rate reduction, via an Ethernet/IP network access, or an ISDN line ("IP/ISDN" version of the module).

The module can be used either in a "single codec" mono or stereo mode, or in a "double codec" configuration. It is then equivalent to two independent mono codecs.

Table 1 (page 2) shows the main features of the Codec module.

2.1. Functions

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

Figure 3 – Functional diagram of the Codec module

The audio signals to be transmitted are selected among the available interfaces, then the encoding function reduces the bit rate, using a selectable algorithm; the resulting bit flow is sent to one of the transmission interface: Ethernet interface or ISDN line.

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

The MultiScoop can also transmit auxiliary information, usually by embedding them inside the transmitted audio streams.

2.1.1. Audio interfaces

At the input of the encoder, it is possible to select the source between:
- The analog inputs of the module
- The AES3 input of the module
- Two audio channels from the AES67 module, if there is one installed in the MultiScoop

The output of the decoder is reproduced on the three types of outputs:
- Analog outputs of the module
- AES3 output of the module
- Two audio channels to the AES67 module, if there is one installed in the MultiScoop

Analog interfaces

The module has two balanced analog inputs and two balanced analog outputs; their input or output gain is adjustable.

To use these interfaces, the optional "Audio Interface" module must be installed on the back of the Codec module.

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

Digital interfaces

The module has digital audio inputs/outputs in AES/EBU format.

To use these interfaces, the optional "Audio Interface" module must be installed on the back of the Codec module.

The digital output is locked to the digital audio input ("genlock" mode).

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 **independent** of the sampling frequency of the compression algorithm.*

Interfaces with the AES67 module

When an AES67 / Ravenna module is installed in the MultiScoop:

- Two digital audio inputs are provided (via an internal bus) by this module to the Codec module
- Two outputs are provided by the Codec module to the AES67 module.

A sampling rate conversion is automatically performed, when necessary, to adapt to the coding algorithm used in transmission.

2.1.2. Audio encoding and decoding

The Codec module features a wide range of coding algorithms. Their availability depends on the transmission network used, and on the single/double codec mode used.

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

Single codec mode

In this configuration, the selected network interface (ISDN or IP) is used to set up a link with a single remote device. The table below synthetically describes the capabilities in the different cases:

Codec	Frequency (kHz)				Net bit rate (kbit/s)	ISDN			IP		
	16	24	32	48		Mono	Dual	Stereo	Mono	Dual	Stereo
G711					**64**	green			green		
G722	●				**64**	yellow			yellow		
G722-H221	●				**64**	yellow					
G722-H242	●				**64**	yellow					
MPEG L2		●	●	●	**64**	red	red	red	red	red	red
MPEG L2		●	●	●	**128**	red	red	red	red	red	red
MPEG L2			●	●	**192**					red	red
MPEG L2				●	**256**					red	red
MPEG L3		●	●	●	**64**				red		red
MPEG L3		●	●	●	**128**				red		red
AAC-LC			●	●	**16 => 56**				red	red	red
AAC-LC			●	●	**64**	red	red	red	red	red	red
AAC-LC			●	●	**96**				red	red	red
AAC-LC			●	●	**128**	red	red	red	red	red	red
AAC-LC			●	●	**192**				red	red	red
AAC-LC			●	●	**256**					red	red
HE-AAC			●	●	**16 => 56**				red	red	red
HE-AAC			●	●	**64**	red	red	red	red	red	red
HE-AAC			●	●	**96**					red	red
HE-AAC			●	●	**128**		red	red		red	red
HE-AAC v2			●	●	**16 => 56**						red
HE-AAC v2			●	●	**64**			red			red
AAC-LC LOAS			●	●	**64**	red	red				
HE-AAC LOAS			●	●	**64**	red	red				
HE-AAC v2 LOAS			●	●	**64**			red			
OPUS				●	**12 => 192**				red		red
OPUS				●	**256**						red
Linear L16				●	**768/1536**				red		red
Linear L20				●	**960/1920**				red		red
Linear L24				●	**1152/2304**				red		red

Table 3 – Single codec - Available coding depending on network

Note: "dual mono" encoding processes two audio channels independently while stereo encoding may involve coupling (except linear encoding).

Double codec mode

In this configuration, two independent connections can be set over the selected network interface (ISDN or IP) towards possibly two remote devices. Either link is a mono link. The table below shows the capabilities for each link:

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

Table 4 – Double codec - Available coding depending on network

2.1.3. Audio transmission

The Codec module features in all versions an Ethernet interface for IP protocol networks. The AoIP/ISDN version also features an ISDN S0 interface (BRI).

AoIP transmission

The Codec module performs AoIP (Audio over IP) transmission in a wide range of possible bit rates. Audio streams are always transported under the RTP/UDP protocol.

Unicast AoIP mode

The most classical transmission mode is unicast: audio connection with a 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 three ways:

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

SIP and SDP protocols

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

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

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

With the SIP protocol, the Codec module can operate as two independent mono AoIP codecs. Each is a mono SIP codec that can deal with an audio link in two ways:

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

The available algorithms are a subset of those for the "standard" mode (single codec): see above "Double codec mode" on page 9.

Double Streaming

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

Refer to chapter 6.3 (Using the Double Streaming system), for more information on the system and how to use it.

Double Streaming mandates the use of the SIP protocol and cannot be used when the Codec module is operated as a double SIP codec.

Packet duplication

Codec module 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.

Multicast IP mode

The multicast mode allows an encoder device to transmit an audio program 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 Codec module 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 signaling 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 accounts

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

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

A second "factory" SIP account is available in the unit:

- This account can be used for implementing "Double Streaming" (see 6.3, Using the Double Streaming system).
- In the "double SIP codec" mode, this account is assigned to the second SIP codec.

Ethernet/IP interface

The IP interface is a 10BaseT/100BaseT Ethernet interfaces used for audio transmission and/or remote control (see further 2.1.5, Remote control interfaces).

ISDN interface

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

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

Network protocols

Available protocols:

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

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

5A System

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

The "5A" system makes it easier to set an ISDN connection, because the codec, on receiving a call, automatically adjusts itself, following the calling party algorithm and parameters.

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

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

Managing audio links

The audio transmission implies a link/session setup phase.

One of the transmission interfaces is selected as the *default interface* on the Codec module.

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

On the other hand, an incoming call on an interface other than the default interface may be accepted under certain conditions. These are described more precisely in 2.4.5, "Receiving a call". In such case, the module first switches to the suitable interface, and then processes the incoming call. When the link is released, it will come back to its previous state (and default interface).

Note: If the unit is in double codec mode and one codec is busy, the other codec is not available for calls on the other network interface (e.g. IP link active, no ISDN call is acceptable).

The detailed procedure for setting a link can be found in 2.4 (Managing audio links).

2.1.4. Auxiliary transmission: "relays"

In addition to the main function of the module, which is the transmission of one or two main audio channels, auxiliary functions are available for transmitting additional data or signals, included in the same connection (more generally the same session).

This feature is only compatible with AETA products, because it is not covered by independent standards.

It is available depending on the coding algorithm and the transmission context.

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

Auxiliary functions are not available in the "Direct RTP" AoIP mode.

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.

The function is available only in following cases:

- Transmission over IP (SIP / AoIP codec), regardless of the used coding (except linear coding L16, L20, L24).
- For ISDN transmission, the function is only available with G722-H242 encoding.

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.

Data channel

This function is not yet available.

2.1.5. Control and supervision

The control and supervision of the Codec modules (configuration, communication management, status monitoring), is carried out in various ways.

Embedded html server: "web pages"

The Codec module provides html pages that enable full control using a web browser, via port 80 (default port for the HTTP protocol) or port 443 (default port for the HTTPS protocol). See on chapter 2.5 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. With the "Remote access" option described further, this control mode is also usable over the Internet.

Front panel elements

LED indicators provide basis status information.

Supervision using Scoop Manager

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

Remote control interfaces

The module can be remote controlled from various interfaces:

- The Ethernet interface, used for audio streaming, can be used for remote control as well.

- A connection internal to the MultiScoop system links each Codec module to the "Control" module(s) possibly installed in the chassis. The "Control" module acts as a gateway providing access to the Codec module via the Ethernet interface of the control module. *Advantage of this technique: the module management is totally isolated from the transmission Ethernet interface.*

Remote access

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

The principle of this remote access and its mode of operation are described in Annex 6.4 ("Remote Access" service). A "Remote Access" option is required on the module.

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

Configuration and dialing memories

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

- "**Call profiles**", which include call parameters for a given **destination** (number(s), coding, etc.). Such a profile is similar to a phonebook entry, but in addition it can memorize the coding parameters. Recalling a profile allows to directly configure the coding parameters and/or launch an outgoing call with the parameters set as memorized in the profile.

- "**Presets**" memorize the **network access** parameters. Recalling a *preset* is a quick way to restore the configuration needed for a given line or network.

- "**Snapshots**" memorize the settings for the **audio** interfaces and processing.

These various memories can be imported/exported from/to a computer.

2.2. Physical description of the module

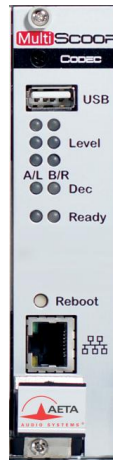The Codec module is a standard 3RU "Europe" format module, with a 6TE width (30 mm).

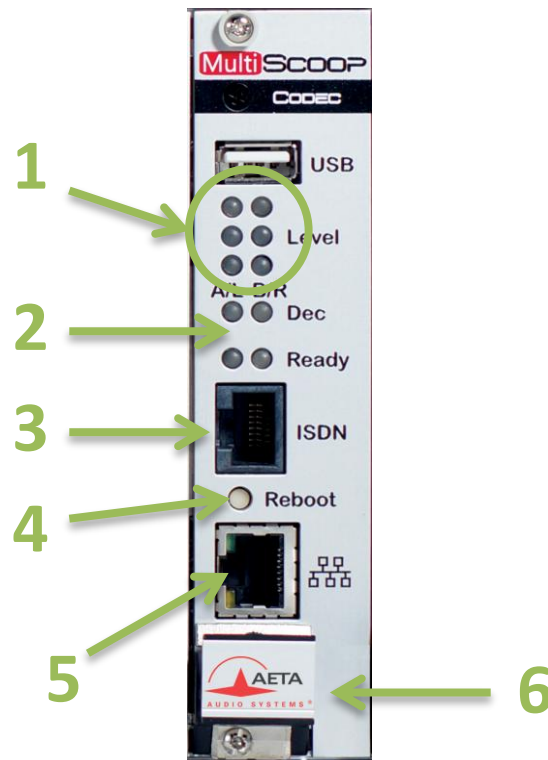**Figure 4 – Front panel of the Codec module (IP version)**


Figure 5 – Front panel of the Codec module (IP/ISDN version)

1. Indicators (3 LED) for the audio level at the input of the module
2. Status indicators: "Dec" = link status (1/2), "Ready" = codec status (1/2)
3. ISDN S0 socket, only present on "IP/ISDN" version
4. Reset button
5. Ethernet interface (10/100BaseT)
6. Extraction lever

The status indicators can be understood as follows:

Indicator	Dec	Ready
Function	Audio link status	Codec status
Off	Idel (no connection)	
Amber	Call or release in progress	
Red/Amber blinking	Call in progress, waiting for synchronization	
Green	Link established, synchronization OK	Ready or established link
Red	Link established, not synchronized	Initialization in progress
Red/Green blinking		SIP registration failure

For using audio inputs/outputs other than AES67/Ravenna, an "Audio Interface" module must be mounted on the back of the Codec module:
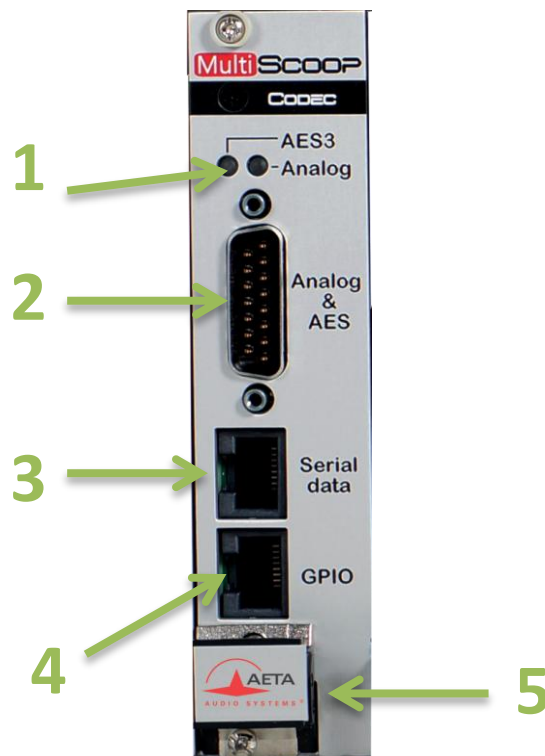
Figure 6 – Rear view of the "Audio Interface" module

1. Indicator for the audio format selected for inputs/outputs: AES3 or analog (none of the LEDs if the AES67 interface is selected instead).
2. Sub-D 15-pin male socket: audio inputs/outputs (see pin allocation further in 2.6.1, "Analog&AES" socket ("Audio Interfaces" rear mount module)).
3. RJ45 socket for an RS232 interface (*not implemented*)
4. RJ45 socket for GPIO (see pin allocation further in 2.6.1, GPIO interface).
5. Extraction lever

2.3. Installation and maintenance

2.3.1. Assembly and connections

The module can be put in place even when the chassis is powered on. It can only be mounted in one of the slots reserved for the "Codec" modules (see above 1.1, Physical description of the system).

An "Audio Interfaces" module must be mounted on the back of the Codec module when using analog and/or AES3 audio interfaces. In the opposite case, make sure to close the rear side by mounting a blanking plate.

The pinout of the connectors is indicated in 2.6.1. In addition, an adapter cable is available as an accessory to split the audio inputs/outputs to 3-pin XLR plugs.

2.3.2. Initial set up

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

This setting up is done using the html interface of the module; two cases arise:

- It is possible to access these pages through the Ethernet interface of the module: you must first configure it and detect/note its IP address. This preliminary operation is described in 2.3.3 below (Initial setup of the Ethernet interface).

- If a "Control" module is present in the MultiScoop, the simplest method is to access the Codec module via the Control module. This access is operative even if the Ethernet interface of the Codec module is not configured properly.

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

2.3.3. Initial setup of the Ethernet interface

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

- Connections to 10BaseT or 100BaseT interfaces are both suitable, as the module automatically switches to the adequate 10 Mbit/s or 100 Mbit/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 can be done in a variety of ways depending on your network and system configuration, as described in the following chapters.

Case 1: availability of a "Control" module in the MultiScoop

This case is the most convenient one: access to the html pages of the module is readily possible through the Control module.

1. Open the html pages of the "Control" module.

2. Click the "Codecs" tab: the installed Codecs modules are visible. In the location of the Codec module concerned, you can see its current IP address (whether it is appropriate or not).

3. Click on this address and you access the html pages of the Codec module (in fact not by its IP address, but via the connection between modules, internal to the chassis).

4. On these pages, you can access the configuration page of the Ethernet interface: NETWORK > ETHERNET PARAMETERS. Adjust these parameters if necessary and/or read the IP address of the module. In particular, you can enable or disable DHCP on the module as needed.

Case 2: no "Control" module available, DHCP server available

This case is also quite simple; DHCP is enabled by default on the Codec module. If a DHCP server is active on the network to which the module is attached, the server assigns a valid IP address and provides the unit with the appropriate settings.

To know the IP address allocated by the DHCP server, a tool in your network may provide this information (DHCP server lease table, for example).

If you do not have this type of tool, you can then use the "Aetascan" tool (see detail in 6.2, Using the "Aetascan" tool). When you launch this utility on a computer that is connected to the same local area network as the Codec module, it searches for and enumerates the AETA codecs and modules that are present, including their IP addresses.

ⓘ *If, instead of the default setting, the module is previously set with a static address (with DHCP disabled), refer to the following method.*

Case 3: no "Control" module available, no DHCP server

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 you must get the current IP address of the codec: either this is a previously set static setting, or a "link local" address that the module has auto-configured because it found no DHCP server. The "AetaScan" tool will allow you to detect this address, whatever the current settings (see 6.2, Using the "Aetascan" tool). The module is listed with "MultiScoop Codec" as its *Device Type*.

If this address is reachable from the computer, you can enter it in the address bar of the web browser and access the html pages of the module. Then you can set up the addressing as required, via the "ETHERNET PARAMETERS" of the "NETWORK" tab (see 2.5.6, "ETHERNET PARAMETERS" page).

If this address is not reachable from the computer, you should go through the following steps:

- Change the computer address (temporarily) so it belongs to the same sub-network as the module.

- Access the html pages of the module by entering its IP address in the browser's address bar.

- Setup the addressing of the module as required, via the "ETHERNET PARAMETERS" of the "NETWORK" tab (see 2.5.6, "ETHERNET PARAMETERS" page).

- Setup the computer back to its original addressing.

Checking the IP configuration

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

To check the setting, from a computer connected to the same network, open a web browser and enter the module's IP address in the address bar: the home page should be displayed.

ⓘ *If a problem in the IP addressing makes the module unreachable, a backup procedure should be used, as described in 2.3.4, "Backup reset".*

Optional configuration of the "link" Ethernet layer

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

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

However, in the rare cases where this is needed, it is possible to set the desired mode manually: in the "ETHERNET PARAMETERS" of the "NETWORK" tab, use the "Link Mode" parameter (drop-down list). Click "Save" to apply the setting.

ⓘ *Be aware that you may lose control over the module if this setting is inappropriate!*

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

2.3.4. Maintenance

The replacement of a module can be carried out hot, without cutting the power supply and without disturbing the other modules in the chassis.

For ease of maintenance, it is possible to export and save all the configuration of the module, which makes it easy to restore it, even possibly on a replacement module.

Saving and restoring the configuration

It is possible to export/save all the equipment settings into a file, and conversely to import/restore a complete configuration from a file previously recorded in such way. Such transfers can be done using the embedded html interface; refer to 2.5.10 ("SETUP TRANSFER" page) for the operating mode.

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

Clearing all settings

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

To clear the whole configuration and come back to the factory settings, normally you should go to the "MAINTENANCE" tab, "RESET" page, and select "**Factory Reset**", or "**Reset Settings**" (the latter does not delete the memorized profiles).

The default (html) password is blank.

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

Backup reset

This procedure should be applied to recover full control over the module if it is in a status, in which it is not possible to access the html pages.

After this reset, the MultiScoop will be brought back to its "factory" configuration with all parameters in their default value, especially: blank html 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 the MultiScoop).

- Copy this file *without changing its name* onto a USB memory stick.

- Plug this stick into the USB host socket on the front panel of the module (while it is off).

- Power on the module. Alternatively, plug the stick with power on, and then use the reset button of the module to reboot it.

- After the initialization phase, the module will go back to its "factory" settings.

- Remove the USB stick (no matter with or without power on), <u>before restarting</u> the module, otherwise all your possible new settings will be deleted again.

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

Firmware update

This is done using the html interface of the Codec module. The procedure for updating the firmware is described further in 2.5.10, "SYSTEM UPDATE" page.

2.4. Managing audio links

Generally you should go through the following steps:

1. Select the **basic options** that will essentially determine the operation: audio operation (mono/stereo, single/double codec), network interface. This is described below (2.4.1, Basic options).

2. Prepare or check the settings for the **network interface** you will use. See below (2.4.2, Preparing the network interface).

3. Select the suitable **coding configuration** for the link. See below (2.4.3, Setting the coding configuration).

4. Select or dial the destination address/number and **launch a call** to the desired destination. This is described below (2.4.4, Launching an outgoing call).

5. Alternatively, the unit can **receive calls** from remote units. See below (2.4.5, Receiving a call).

6. The use of multicast brings over specific aspects described in 2.4.6 (Special case of multicast).

7. Using the Double Streaming system implies specific settings and operation: see annex 6.3, Using the Double Streaming system.

2.4.1. Basic options

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

- What will be the main network for the transmission? For the IP/ISDN Codec module, there is a choice between ISDN and Ethernet.

- Mono or stereo transmission, single or double codec?

The table below summarises the possible combinations available:

Case	Codec mode	Audio mode	Transmission	
			ISDN	**Ethernet**
1	Single codec	Mono		
2		Stereo		
3	Double codec	Mono + Mono		

Table 5 – Basic operation modes

- **Case 1:** only one link is used for a mono programme.

- **Case 2:** only one link is used for a stereo programme (or variations dual mono, joint stereo).

- **Case 3**: the double codec mode allows setting two independent links over the same transmission interface.

- Using **multicast** is only possible in mono or stereo "single codec" mode, and only on the Ethernet interface.

2.4.2. Preparing the network interface

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

ISDN interface

Connect the ISDN line to the ISDN socket on the front of the module.

The network interface must be configured depending on the local ISDN line that is used: see "ISDN SETTINGS" page.

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

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

Protocol

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

Codec mode

ⓘ *This setting is important as it defines the basic behavior: single codec and single connection, or double codec with two independent connections.*

Local address

In some cases, it may be necessary to set the local address (or local ISDN number) of the line, and/or it is possible to assign a sub-address to the codec.

The local number allows "multiple subscriber numbering" or MSN. This number is usually the number remote equipment must dial to call your equipment. Configuring this number in the equipment is not mandatory if the equipment is directly connected to the public network. On the other hand, if the equipment is connected to a PABX, the number(s) are often required. The PABX may also impose a unique number for each B channel within the same BRI interface. In such a case, refer to the characteristics and configuration of the PABX.

ⓘ *Proper configuration of the local numbers is essential, and many problems in setting up links originate from mistakes or misunderstandings regarding this configuration. In doubt, leave this number blank! This is usually appropriate for public lines.*

Sub-address SA

This number differentiates several terminals connected to the same ISDN bus, which are allocated the same call number(s). Thus it can be useful in case other devices are connected with the Codec module on the same line.

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

ⓘ *Most often, the best setting is to leave this blank, except in double codec mode where this setting is useful (see further in 2.4.5 the processing of incoming calls in double codec mode).*

SPID

This "Service Profile Identifier" is only relevant for the NI-1 protocol (used by operators in North America). This number identifies the services of the ISDN line, and usually includes the subscriber number or part of it. This number is mandatory to operate the line, and must be provided by the network operator.

Note: the module only supports one SPID, therefore setting connections beyond 64 kbit/s may not be possible.

AoIP settings

These settings apply to **all IP protocol connections.** They can be found in 2.5.6, "NETWORK" tab, "ETHERNET PARAMETERS" page, along with details about these settings.

The usual and most convenient operating mode uses the SIP protocol: select "SIP" as **Default Protocol**.

If a SIP server is used, the Codec module must be registered on this server. Use the html page for entering the SIP account data and activate the SIP registration. *Check that the codec is successfully registered by looking for the indication on the home page "*STATUS*". A possible registration failure is also shown by the "Ready" LED blinking red on the front panel of the module.*

Besides, for links over the Internet, most often the link passes through a NAT router which "hides" to the codec the characteristics of the public access: public IP address and RTP ports actually used. This can make the SIP transactions fail, or get a non-optimal route through the network. To avoid this, a STUN server is often used; the corresponding settings are also found in the AOIP PARAMETERS page.

*Check that the public address discovery has succeeded: "*STATUS*" home page,* **Public IP** *information (its mere presence shows that the discovery has succeeded). You can also see there the type of NAT that the codec has detected (***NAT Type***).*

If you need to use the module as a double mono codec, check **Double SIP mono**. The module then manages two independent SIP connections.

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

ⓘ *The "Direct RTP" mode is not compatible with the operation as double codec.*

Ethernet settings

Use an RJ45 cable to connect the unit to a network switch or router. A couple of LEDs on the socket show the state and activity of the Ethernet connection.

The network interface may have to be configured for the local installation, especially if there is no DHCP server available. The relevant settings are accessed in the ETHERNET PARAMETERS page.

The settings are described further in 2.5.6, "NETWORK" tab, "ETHERNET PARAMETERS" page. In the most frequent case, the following settings are suitable:

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

2.4.3. Setting the coding configuration

ⓘ *Note: the goal here is to define a coding configuration for outgoing calls. On receiving a call, the codec will normally "follow" the calling party (AoIP/SIP or 5A System with ISDN).*

On the AoIP/ISDN version, you should first select the network interface to be used: "NETWORK" tab, "CHANGE NETWORK" page.

Make sure to enable double mono codec operation if necessary: this belongs to the preliminary network settings (see above 2.4.2, Preparing the network interface).

The coding configuration is then accessible though the "Coding" page. You first have to select the coding algorithm, and then adjust its various parameters:

- The list of algorithms depends on the current default network.
- The possible choices for the parameters depend on the algorithm; it is not possible to build an invalid combination.
- Especially, the number of parameters depends on the coding. Some algorithms just need no additional parameter (example: G711).

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

2.4.4. Launching an outgoing call

Once the preliminary settings are done, the procedure for a call on a codec is straightforward: dial a destination "number" and launch the call. This is done via the "CONNECTIONS" page.

Preliminary note about the "dial number"

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

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

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

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

Direct dialing

ⓘ *In case of a SIP call, remember that in most cases the unit must be registered on a SIP server!*

From the "CONNECTIONS" page, enter the call number in the **Number** field. Then click the "**Dial**" button.

ⓘ *Whenever an ISDN connection requires two B channels (128 kbit/s coding), a second number is required. The field may be left empty if the same number is valid for both channels.*

Releasing a connection

The "**Dial**" button becomes "**Release**" when a link is active. Click this button to release the connection.

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

Recalling the same number

Once you have called a number (or a couple for an ISDN call with 2 B channels), you can easily call it again without having to type it: browse the "call history" using the drop-down list and pick a previous call.

Using the phonebook

The "call profiles" of the phonebook provide a quick method for setting calls: see how to use this feature in 2.5.5, "CALL PROFILES" page.

ⓘ *With a Call Profile you can at once: select the network interface, select a coding configuration, and of course recall the destination number(s). The flexible structure of the profiles allows you to include or not each of these elements in a profile.*

2.4.5. Receiving a call

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

ISDN incoming calls

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

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

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

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

- Use a distinctive sub-address on each codec.

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

ⓘ *If an ISDN line is connected, the module will receive and accept an incoming ISDN call even if its default transmission interface is not the ISDN line (provided that it is not already busy with an AoIP connection). In such event the unit will switch to the ISDN Interface when receiving the ISDN call. Once the connection is released, the module will come back to its previous default interface.*

AoIP incoming calls

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

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

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

ⓘ *For being able to receive a "Direct RTP" incoming "call", the "**Accept Calls**" parameter in the "DIRECT RTP" section must be checked.*

2.4.6. Special case of multicast

On a network that can support it[1], the multicast mode can optimize the resource usage when an audio stream has to be distributed simultaneously to several destinations. Contrary to the normal bidirectional unicast mode, this mode is unidirectional: one *sender* encoder sends a stream towards a multicast *group address*, and one or several decoders receiving the stream pick up the packets sent to this group address and decode the audio stream.

On the Codec module the operating mode stays quite similar to the "normal" (unicast) AoIP mode, with mainly the following differences in multicast mode:

- SIP is not used and hence the SIP configuration is not relevant.
- A codec must be set as sender or receiver device.
- The double codec mode is not available with multicast.

For the operation it is assumed the network "statically" supports UDP multicast, i.e. routers of the network recognize and deal with routing the packets with multicast group addresses. First you should select or assign a multicast address for the session.

For obvious reasons the coding setup is entirely decided at the source. On the codec on the audio source side (hence *sender* of the encoded stream), the procedure for setting up a multicast stream is the following:

- Set the codec in "multicast send" (TX) mode. To do this, refer to the "AOIP PARAMETERS" page.
- Select on this codec the desired audio coding parameters.
- Start the streaming as for setting up a call in normal mode ("CONNECTIONS" tab): enter as "Number" the IP group address, then click the "**Dial**" button.

ⓘ *Note that the codec decodes its own stream on its audio output; this can be useful for audio monitoring.*

For each codec that has to decode the stream, the procedure is simple:

- Set the codec in "multicast receive" mode. To do this, refer to the "AOIP PARAMETERS" page.
- Start the streaming as for setting up a call in normal mode ("CONNECTIONS" tab): enter as "Number" the IP group address, then click the "**Dial**" button.

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[1] This does not include the Internet; multicast cannot be used over the Internet.

2.4.7. "Network quality" setting (AoIP)

Depending on the quality of service provided on the IP network, especially its jitter performance, it is possible to change the stability/latency compromise used by the Codec module. For this purpose, a "**Network Quality**" setting is available among the settings of the IP interface ("ETHERNET PARAMETERS" page). Six choices are proposed:

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

On a LAN and/or private network with a controlled quality, the "Very High" quality setting is recommended, as it yields minimum latency. On the contrary, it should be avoided for a link via the Internet, as it can only tolerate a low jitter. One solution can be to start with a "Middle" setting, and move to the "Low" setting, or further, if too much audio disturbance is heard.

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

Coding	Very High	High	Middle	Low	Bad	Very Bad	Tolerance
G711	85 ms	120 ms	220 ms	320 ms	420 ms	780 ms	± 10 ms
G722	85 ms	120 ms	215 ms	315 ms	415 ms	775 ms	± 10 ms
MPEG L2 @48kHz	95 ms	140 ms	260 ms	355 ms	450 ms	815 ms	± 10 ms
MPEG L2 @32kHz	155 ms	179 ms	285 ms	395 ms	505 ms	827 ms	± 10 ms
MPEG L2 @24kHz	190 ms	190 ms	320 ms	415 ms	510 ms	850 ms	± 10 ms
AAC-LC @48kHz	135 ms	175 ms	275 ms	385 ms	470 ms	830 ms	± 10 ms
AAC-LC @32kHz	175 ms	202 ms	325 ms	420 ms	515 ms	870 ms	± 10 ms
HE-AAC @48kHz	245 ms	275 ms	355 ms	485 ms	570 ms	910 ms	± 15 ms
HE-AAC @32kHz	370 ms	375 ms	475 ms	535 ms	665 ms	986 ms	± 25 ms
HE-AAC v2 @48kHz	285 ms	320 ms	400 ms	530 ms	615 ms	955 ms	± 10 ms
HE-AAC v2 @32kHz	425 ms	430 ms	530 ms	595 ms	725 ms	1045 ms	± 20 ms
OPUS	85 ms	125 ms	220 ms	320 ms	420 ms	780 ms	± 10 ms
Linear	80 ms	135 ms	230 ms	330 ms	425 ms	770 ms	± 15 ms

Tableau 6 – Latency as a function of the coding algorithm and the "Network Quality" setting

2.4.8. Auto-redial feature

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

- The codec can 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 during a link, after rebooting the codec will automatically redial and set-up the link back.

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

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

The activation of this function and the configuration of its parameters can be found in the "**MISC**" page ("AUTO REDIAL" section). In double codec mode, the function can be activated separately for each codec.

2.4.9. GPIO control

In normal operation, outgoing calls are sent or released using the **CONNECTIONS** page.

It is also possible to assign functions to the GPI so that these can partially manage the connections: "Redial" action or "Release Call" action. The GPO can be set for indicating the link status.

These capabilities are described in 2.5.9, "MISC" tab.

2.5. Operating mode – Embedded html interface

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

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

2.5.1. Accessing the html pages

There are several ways to connect on these pages:

- Via the Ethernet interface of the module (computer connected to the same network): enter the IP address of the Codec module in the "address" or "URL" field of the browser. *You can also use HTTPS, provided that you override or bypass the security warning regarding the site certificate of the module.*

- Via a "Control" module installed in the MultiScoop (thus via the Ethernet interface of the Control module, possibly isolated from the network the Codec module is connected on): enter the IP address of the Control module in the "address" or "URL" field of the browser, and then in the "Codecs" tab click the IP address of the desired Codec module.

- By remote access via the AETA server: see further in 6.4.2, How to use the remote access service.

The home page that is displayed is similar to the following picture:
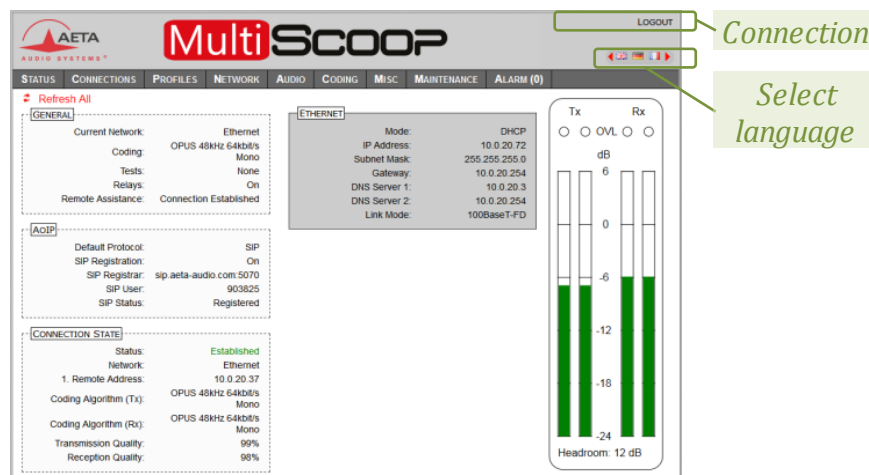
If needed, select another language by clicking the suitable flag.

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

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

To log in, enter the html password and click the connection button. The initial password is blank: click directly on the button. To set a 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.

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

2.5.2. Principles of operation of the html pages

The picture below shows a typical page of the interface:

On top you find a bar of tabs corresponding to the various categories of functions and parameters for the module. 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 or sections (each surrounded with a frame). On the right side, a text area provides additional help and hints.

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

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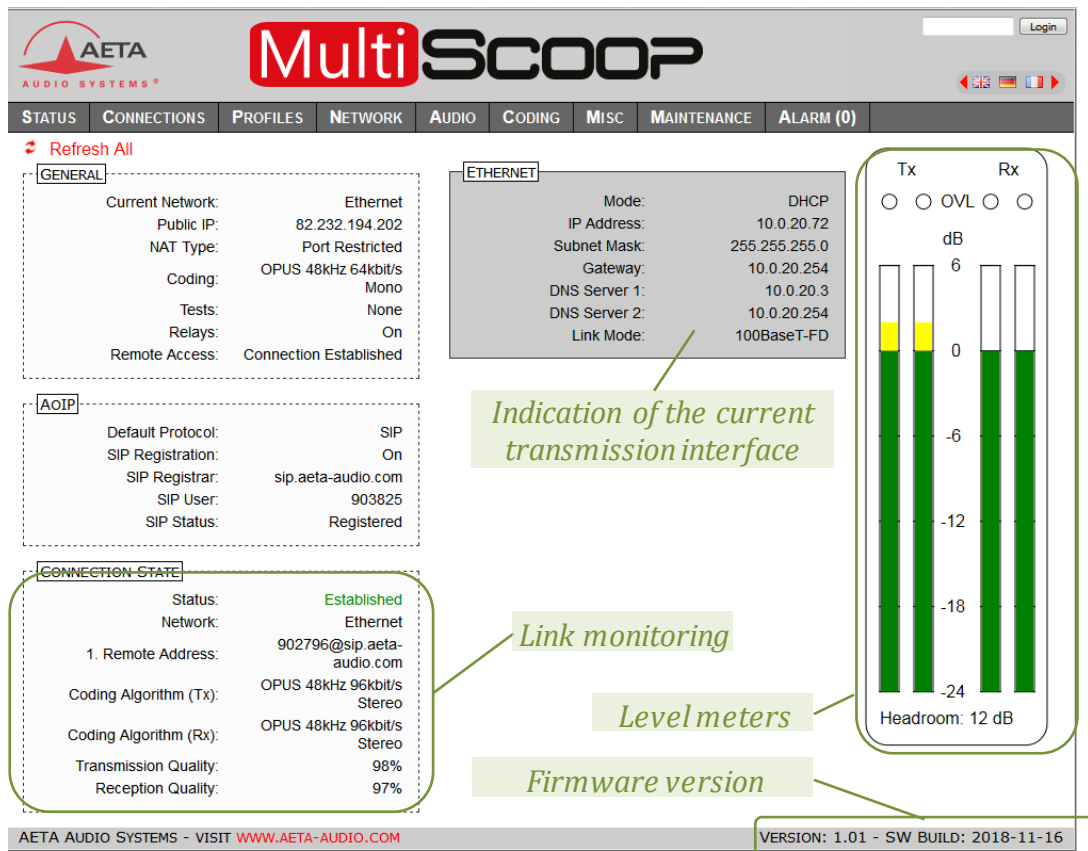
Two tabs have a specific behavior:

- "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.

2.5.3. "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 another operator is managing.

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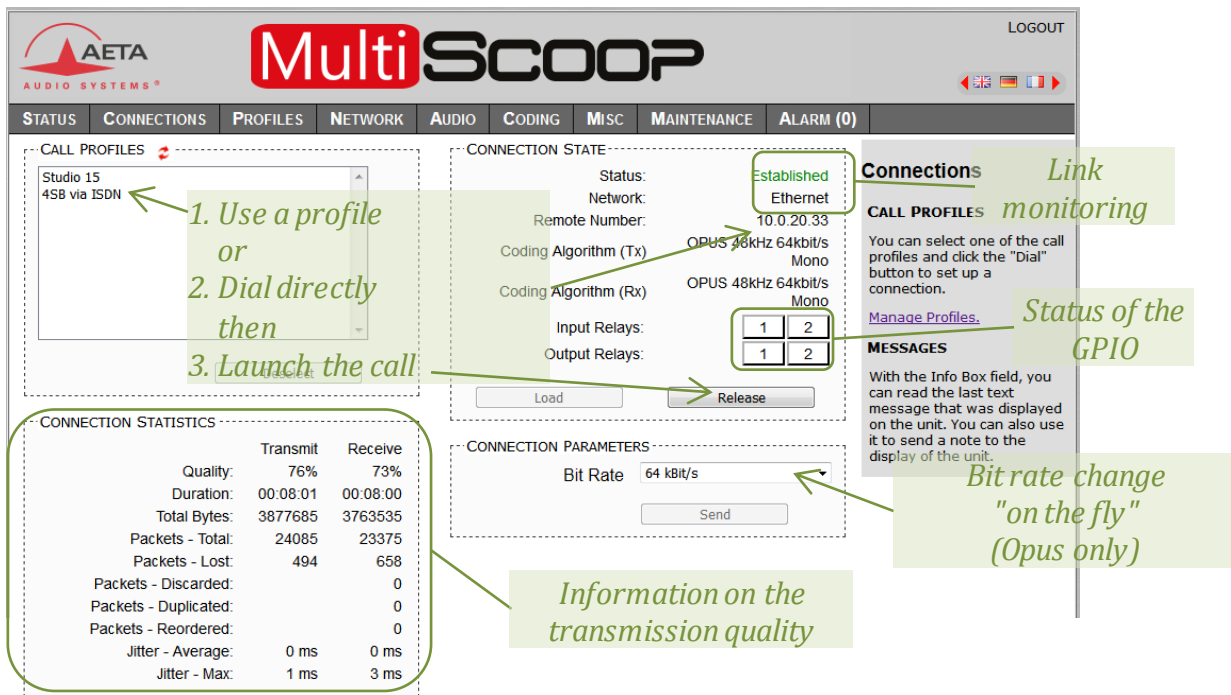
ⓘ *Note: the goal of these bargraphs is to provide an indication on the presence and level of the audio modulation. The measurement is accurate for stationary signals (1 dB accuracy and resolution), but the update rate is rather slow. As a consequence these bargraphs are not appropriate for precisely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

One can find in the "CONNECTION STATE" section global transmission and reception quality indicators (for IP links only). These are an estimation, with 100% corresponding to perfect quality, that takes into account various relevant data: packet loss rate, jitter, jitter buffer variations… More detailed data are provided on the "CONNECTIONS" page.

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

2.5.4. "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 "Remote Number" field and pick a number in the dropdown list.*

- For using a call profile (see further "PROFILES" tab, "CALL PROFILES" page): 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.

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

During a connection, 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 changed setting is applied at least in the sending direction to the remote codec. If this device is also an AETA codec, it will "follow" by changing its bit rate as well, thus the selected new bit rate is applied in both transmission directions.*

During an AoIP connection, transmission statistics and quality indicators are available in the "CONNECTION STATISTICS" section like on the "STATUS" page; in addition detailed data are provided in each transmission direction:

- "Duration": time elapsed since the beginning of the link (hh:mm:ss)
- "Total Bytes": total number of bytes sent/received since the beginning of the link
- "Packets – Total": total number of packets sent/received since the beginning
- "Packets – Lost": number of packets lost since the beginning of the link
- "Packets – Discarded": total number of packets rejected (reception only)
- "Packets – Duplicated": total number of packets received twice (reception)
- "Packets – Reordered": number of packets received out of their normal sequence (reception)
- "Jitter – Average": average jitter (in ms)
- "Jitter – Max": maximum jitter since the beginning of the link (in ms)

All these measurements are cumulated from the beginning of the running connection.

ⓘ *Beware that information on the transmitted stream is not always available, depending on the remote device that receives the stream. It must support the RTCP protocol for providing relevant data on the sending direction.*

In double SIP codec mode, a second section provides information on a possible second connection.

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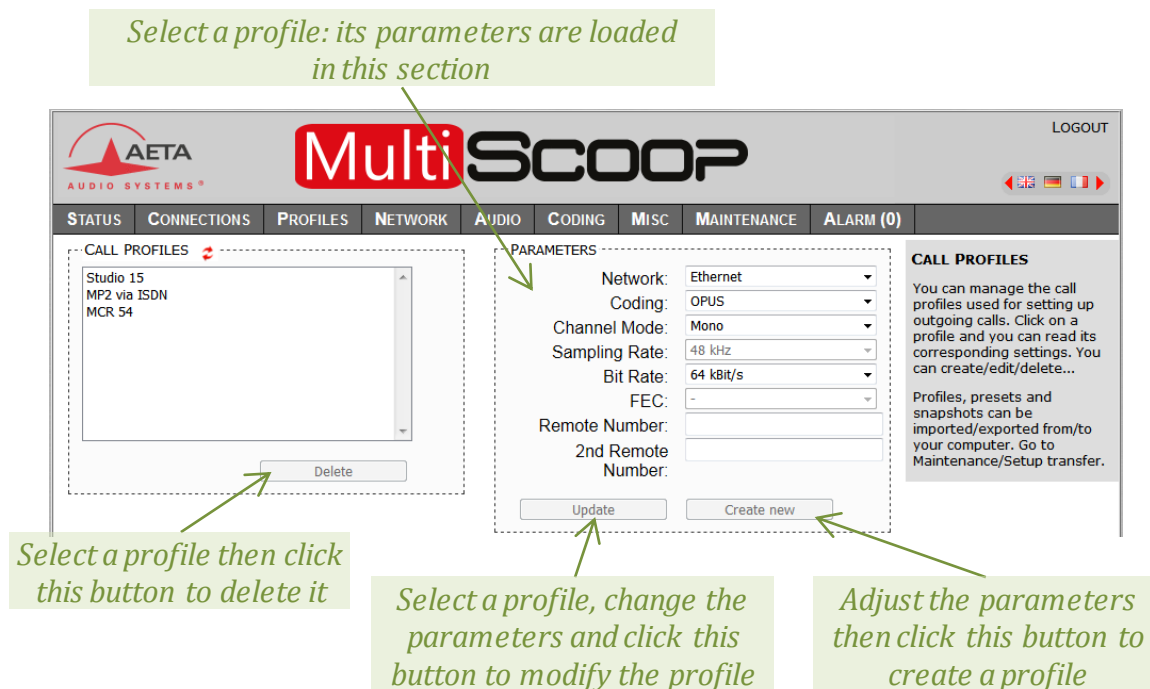
This tab proposes three secondary pages, one for each profile type.

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

- "**Call profiles**", which include call parameters for a given **destination** (number(s), coding, etc.). Such a profile is similar to a directory entry, but in addition it can memorize the coding parameters.
- "**Presets**" memorize the **network access** parameters. These are the settings found in the pages of the NETWORK tab. Recalling a *preset* is a quick way to restore the configuration needed for a given line or network.
- "**Snapshots**" memorize the settings for the **audio** interfaces. These are settings found in the AUDIO page (except the "Headroom" setting).

These various memories can be imported/exported from/to a computer (see further 2.5.10 "SETUP TRANSFER" page.

"CALL PROFILES" page

Select a profile: its parameters are loaded in this section

Select a profile then click this button to delete it

Select a profile, change the parameters and click this button to modify the profile

Adjust the parameters then click this button to create a profile

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

Each call profile includes the following elements:

- *Name* assigned to the profile
- *Number(s)*: ISDN call number(s), or IP URI, IP address, depending on the transmission mode.
- *Coding configuration* associated with the profile

A call profile may have no number included; in such case it is used for quick and safe recall of a given coding configuration.

Conversely a call profile may have no associated coding configuration included; in such case it is just like a directory entry, used for instant dialling a known destination.

You can *Load* an existing profile or directly *Dial* this profile from the page described previously (2.5.4, "CONNECTIONS" tab).

The "Call Profiles" page allows to manage the profiles with the following possibilities:

- *Create* a profile, assigning it a name, possibly call number(s), and possibly an associated coding configuration.
- *Delete* a profile previously recorded.
- *Update* a profile previously recorded, either for renaming it or changing its content.

"PRESETS" page

This page allows to manage the presets with the following possibilities:

- *Create* a preset, which will memorize all the current network settings, assigning it a name.
- *Load* a preset: the codec then restores the settings included in the preset.
- *Delete* a preset.

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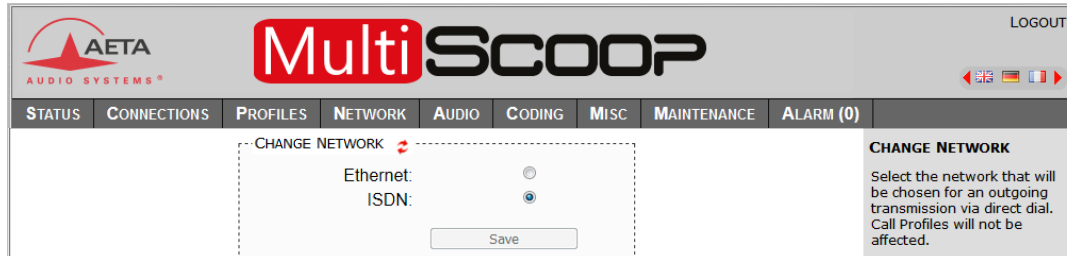
Snapshots are in fact managed on the "AUDIO" page, and this "SNAPSHOTS" choice on the "PROFILES" tab just redirects to the "AUDIO" page. *See further the description of the audio page.*

2.5.6. "NETWORK" tab

This tab leads to two or three pages, depending on the version of the module (IP or IP/ISDN).

"CHANGE NETWORK" page: default interface selection
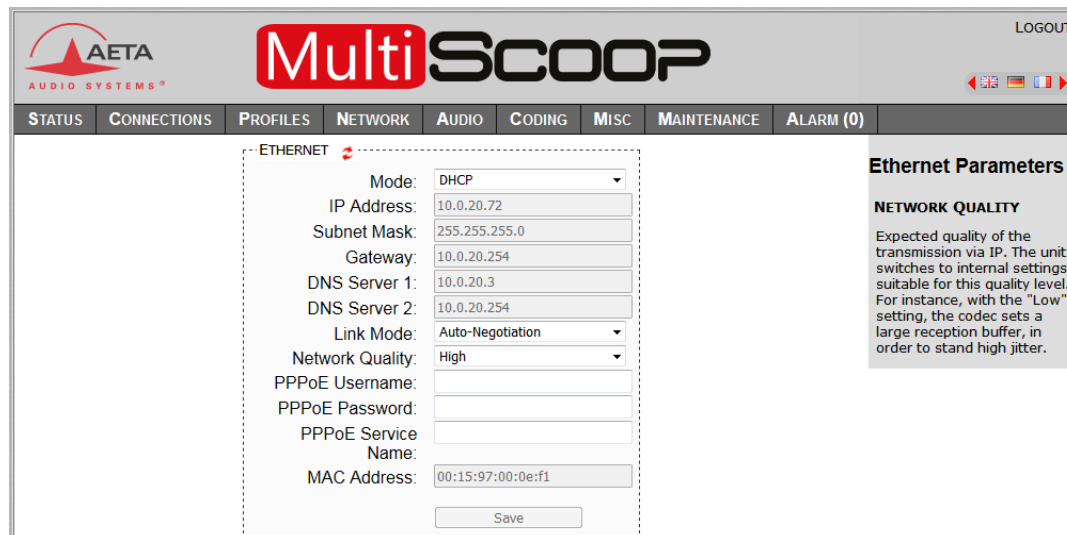
This page exists only on the IP/ISDN version, and it is used for selecting the default interface: this is the one that is implicitly used when dialing to set an outgoing call. Besides, codec algorithm settings (via the suitable html page) are applicable to this interface.

Select the desired interface and click "**Save**".

"ETHERNET PARAMETERS" page

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

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

ⓘ *Settings regarding the Ethernet interface and IP addressing are especially critical, because this is the interface you are controlling the module 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!*

Notes:

- **Mode**: enable or disable DHCP.
  This selection conditions most of the other settings.

- If DHCP is not used, several fields are available for entering the necessary parameters, such as **IP Address**, **Gateway**, etc. These are shaded (read only) if DHCP is active.

- **Link Mode**: concerns the physical Ethernet interface; the default setting "Auto-Negotiation" is suitable for almost all situations.

- **Network Quality**: select here the *expected* transmission quality via the Ethernet interface. The unit switches to internal settings suitable for this quality level.
  For instance, if you select "Bad", the codec sets a *large* size for the reception buffer, in order to stand a higher jitter. As a counterpart, the latency will be higher as well.
  Conversely, if you select "High", the buffer is *smaller* and the latency is lower, but the system is more vulnerable to possible jitter (fluctuations of the network transmission latency).
  Resulting latency depending on the setting: see 2.4.7, "Network quality" setting (AoIP).

"AoIP SETTINGS" page

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

Notes:

PROTOCOL and STUN sections:

- **Default Protocol**: this main setting selects the protocol used for an "outgoing call" (more precisely, an AoIP session initiated by the operator of the unit). Here the three possible types of AoIP sessions can be found: *SIP* (unicast), *Direct RTP* (unicast) or *Multicast*. SIP is the standard choice; "Direct RTP" is a mode without signaling used for instance for compatibility with devices that do not support SIP.
- **STUN Mode**: enable or disable STUN. In this way you can disable STUN without erasing the STUN server address.

- **STUN Server**: enter the address of the STUN server, either in numeric form or with a symbolic name. This kind of server allows the module to discover its public IP address when it accesses the Internet through a router with NAT. You can for instance use AETA's server: **stun.aeta-audio.com**.

SIP section

- **SIP Registration**: enable or disable the registration on a SIP *registrar* server. The following menus **SIP xxx** are used to enter the parameters of the account (or the accounts) on this server, when using such service. These parameters should be provided by the administrator of the SIP server. With this **SIP Registration** parameter, you can disable the registration and hence stop using a SIP server, without having to erase account parameters; these stay ready for later use.

- **SIP Accept Calls**: when this box is checked, the codec accepts incoming SIP calls (regardless whether the codec is registered or not on a SIP server). If it is not checked, incoming SIP calls are rejected, with a "busy" indication.

- **SIP User**: enter the user ID as provided with the account. This field corresponds to a number that a remote party can call to initiate a link with the codec when it is registered on the server. *If this field is left empty, codec 1 does not register on the SIP server.*

- **SIP Display Name**: this (optional) parameter may be displayed by the remote agent during a link; you can enter any text you wish. *If this field is left empty, the actual value is the SIP URI of the unit.*

- **SIP Registrar**: URL of the SIP server, in case one is used. You can also enter a numeric IP address. Like in the above example, you can append ":" followed by a number if you need to specify the SIP port *of the registrar*. If you don't specify (example sip.aeta-audio.com), the port is automatically set to 5060, standard port for the SIP protocol.

- **SIP Auth User**: enter here the authentication ID as provided with the account. *If this field is left empty, the actual parameter is the same as the* **SIP User** *field, which is appropriate in most cases.*

- **SIP Auth Password**: password provided with the SIP account.

- **SIP Outbound Proxy**: IP address or domain name of a proxy server through which SIP transactions and audio streams are relayed. This parameter is optional, to be filled only is such server is used in the organization.
  *Note: contrary to the SIP registration data, this parameter is active even if "**SIP Registration**" is disabled.*

- **Keepalive Interval**: whenever necessary, the codec performs transactions to "keep alive" a translation through a NAT router. This setting defines the maximum time left between two transactions with the SIP server. It should be set on a value lower than the NAT translation time out of the router.

- **Min. Regist. Period**: sets the interval between two re-registrations on the SIP server. Actually the server always imposes a maximal period; therefore this setting is optionally used to make the registration renewal more frequent, if its value is lower than the time set by the SIP server.

- **SIP Port**: the default port for SIP signalling is 5060 (UDP), you can enter a different value here.
  Note: this port number is valid for the module *itself, not the server. For the server side, if a non-standard port is used, e.g. 5070, the "**SIP Registrar**" field must specify the port after a colon like in "mySIPserver:5070".*

- **RTP Port**: the default port for RTP/SIP audio streams is 5004 (UDP), but you can enter a custom value here if desired.

- **Double SIP mono**: when this box is checked, the module operates as a double SIP codec. *This choice is incompatible with a default protocol other than SIP.*
  Note: codec 2 is registered on the same SIP server as codec 1. SIP data other than the individual account parameters apply to both codecs.

- **Double Streaming:** enable or disable the AoIP transmission with redundancy. For operating with this system, refer to annex 6.3 (Using the Double Streaming system). *This system cannot be used in double codec mode.*

- **SIP User 2**, **SIP Display Name 2**, **SIP Auth User 2**, **SIP Password 2**:
  enter there the data for a second SIP account. This account is used either for codec 2 in double AoIP codec mode, or possibly for setting links with "Double Streaming" (see above).

- **RTP Port 2**: this port number for the second SIP connection is shown for reference and it can't be edited. Its value is always that of the main RTP/SIP port plus 2.

DIRECT RTP section

- **Accept Calls**: when this box is checked, the codec accepts incoming RTP streams. If it is not checked, incoming RTP streams are ignored.

- **Send only**: if this is activated and the operator sets an outgoing call, the unit does not expect to receive an RTP stream and rather decodes its own stream. The device keeps sending data until the user releases the link on purpose. In this use case the "RTP Timeout" parameter (see further) is irrelevant.
  *This parameter should not be checked at the same time as "**Accept Calls**".*

- **RTP Port**: port allocated to the transmission in Direct RTP mode; *must not be the same as the RTP port for SIP*.

- **RTP Timeout** (default 30 seconds): after a link is established, if no data is received within this time period, the device "releases" the link (i.e. stops sending its own RTP stream) and comes back to the idle state (**Accept Calls** is unchanged).

MULTICAST section

- **Multicast Mode**: if the selected mode is Multicast, the transmission is unidirectional. Select with this menu if the codec must be *sender* of a multicast stream, or rather *receiver* of such stream.

- **Multicast Ctrl Port**: this port number (UDP) is used for sending control packets accompanying the multicast stream. Default value is 6000.

- **Multicast Audio Port**: this port number (UDP) is used for transporting the multicast audio stream. Default value is 6001.
  Note: this setting is not related to the RTP Port settings valid for RTP/SIP or Direct RTP.

- **Multicast TTL**: TTL value applicable to the RTP multicast audio stream. Default: 254.

QoS section

- **RTP DSCP**: if the transmission network supports DiffServ to manage the media priority, you can set here the type of service which is assigned to the stream transmitted by the codec.
  You should enter here a decimal value, e.g. 46 for the maximal priority code "Expedite Forwarding". The default value is 0, "Best Effort".

- **Packet Replication**: (see also 2.1.3 regarding this feature) you can select here the setting for packet replication:
  0 => standard mode without replication (default setting)
  1 => packet replication without interleaving
  2 => packet replication with interleaving: the second packet is delayed; robustness is improved but latency is slightly increased.

"ISDN SETTINGS" page

This page is only visible for modules in IP/ISDN version.

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

Notes:

- **Protocol**: NI-1 is used in North America (also compatible with NI-2 networks), NTT is the protocol of the Japanese operator NTT, and Euro ISDN (or ETSI) can be used in the majority of the other countries (*see more details on page 12, Network protocol*).

- **Local Number** and **Sub-address**: these settings are irrelevant for the NI-1 protocol. *Their meaning and use is detailed in 2.4.2, ISDN interface.*

- **SPID**: this setting only applies to the NI-1 protocol, and is not presented for the other protocols. More about this setting in 2.4.2, ISDN interface.

- **Codec Mode**: selection of either the normal "single codec" mode or the "Double codec" mode. In this latter case, it may be important to set a distinctive local number + sub-address combination for each of the two first channels. This is detailed in 2.4.5, ISDN incoming call.

- **5AS**: by default the 5AS system for ISDN (cf. page 12, 5A System) is active. If required, it can be disabled using this menu entry.

2.5.7. "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.

Notes:

- **Input Level** and **Output Level**: gain setting for (respectively) the analog inputs and outputs. The parameter is the <u>maximum</u> level (absolute level expressed in dBu) of the interface. For an input this is the clipping level, which when applied at the input produces a transmitted signal at 0 dBFS. For an output this is the maximum output level, produced when receiving a signal at maximum level, i.e. 0 dBFS.

- **Input Source**: select the audio source to be used by the codec, between:
  - "Analog": analog inputs (an "Audio Interfaces" module must be installed on the back of the Codec module).
  - "Digital": AES3 digital input (same remark as above).
  - "Ravenna": audio channels from the AES67/Ravenna module (if one is integrated in the MultiScoop), assigned to this Codec module.

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

ⓘ *On the display, the "OVL" clipping indicator shows that the absolute maximum level is reached (0 dBFS), regardless of this headroom setting. This is not necessarily the top of the bargraph scale, which is +6 dBr (relative), and whose absolute level depends on the "headroom" setting.*
Example, if the headroom is set at 10 dB: then the reference is -10 dBFS. The bargraph can go up to +6 dBr, which is -4 dBFS. At this stage the "OVL" indicator does not yet trigger, as it reacts at 0 dBFS.

⚠ Beware of the differences between the three level measurements!

- Absolute digital levels in dBFS for the AES3 or Ravenna inputs and outputs, internally (encoder/decoder) and for transmission.
- Displayed levels in dBr, relative to an adjustable 0 reference. The relationship between the display and the dBFS level is controlled by the "**Headroom**" setting.
- Absolute analog levels in dBu for the analog inputs and outputs. The relationship between the absolute analog and digital levels is controlled by the "**Input Level**" and "**Output Level**" settings.

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 (**Delete**, **Load**, **Cancel**) 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 section at a time*), "**Load**" and directly apply the snapshot (*also possible using the "Save" buttons, but one section at a time*), or "**Delete**" to remove the selected snapshot from the list of snapshots.

2.5.8. "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 type of network interface. This means that when you change the current network interface, the coding configuration that is related to it is recalled. Example:
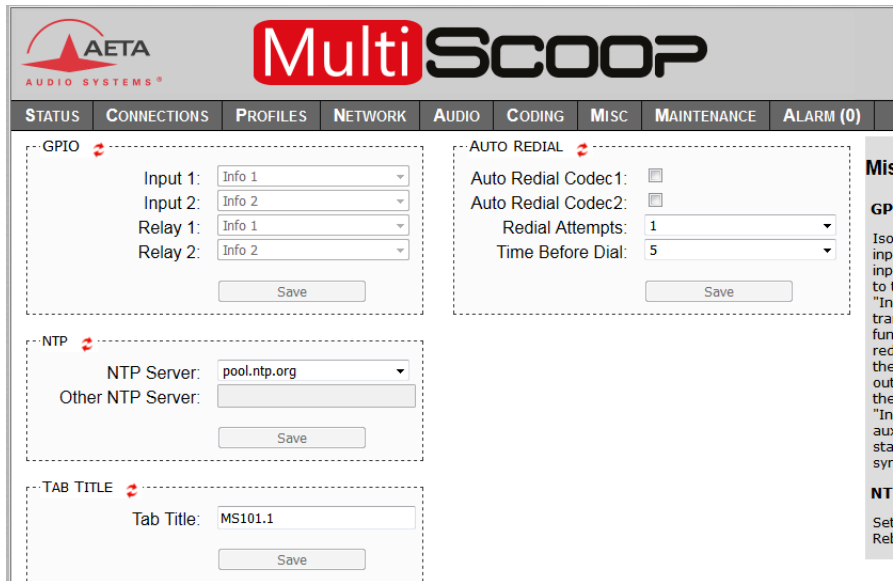
- Current interface = ISDN, selected configuration MPEG L2 mono, 24 kHz and 64 kbit/s.

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

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

- Switch the network interface and select the Ethernet interface: the Opus configuration is restored.

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

ⓘ *When the module is set as a double codec, an additional "CODEC 2" section is available. The range of allowed coding configurations is less wide than in single codec mode (see 2.1.2, Audio encoding and decoding).*

2.5.9. "Misc" tab

This **Misc** tab includes several settings.

Among other features, this page includes the configuration and activation of the auto redial feature:

- **Auto Redial**: activation of the feature, whose operation is described in 2.4.8 above. For the double codec mode, auto redialing can be enabled separately for each codec.

- Redial Attempts: you can adjust from 1 to 20 the number of attempts to reset the link in case of a drop, or even "Infinite" for redialing indefinitely.

- **Time before dial**: (in seconds) adjust between 1 and 30 seconds, after a link loss, the waiting time before redialing. It is often worth leaving such a pause, so that the cause of the link loss is possibly cured, or the remote unit gets back to its normal state, etc.

There are two additional items on this page:

- It is 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 2.5.10, "Maintenance" tab, "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 changes done**, clicking the "Save" button!

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[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.

2.5.10. "MAINTENANCE" tab

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

"LOGIN DATA" page

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

Note: this password has no relationship with the password used for remote access (see further).

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

If you lose the html password, you must perform a complete reset of the settings, using the backup reset procedure: see in 2.3.4, Backup reset.

"SYSTEM UPDATE" page

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

For an update, the file must be provided to you by AETA; usually you can download it from the AETA web site www.aeta-audio.com. The procedure is rather simple:

- From this page, click "**Browse**…", find and select the update file.
- Click the "**Update**" button.
- The process begins, and on the module's front face the status indicators warn of this by a specific sequence.
- Wait for the completion of the update. Normally the unit should reboot by itself at the end of the process.
- Usually the memorized settings stay valid after an update. However, for maximum safety, it is recommended to apply 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 module is stable (no undesirable interruption during the process).
- Make sure the module is not powered off during the process.
- Re-import your profiles and snapshots after the update. But be aware that, although we always try to keep a maximum upward compatibility, AETA cannot guarantee the re-usability and consistency of these memories after a firmware update.

"RESET" page

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

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

- "**Factory Reset**": all parameters are brought back to their default setting, and all memories (call profiles, presets, snapshots) are deleted. *Afterwards, the module 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 from the memory.

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

- "**Factory SIP Accounts**": reload the SIP account data integrated in the unit, and register the module on the AETA SIP server.

The selected action is performed after you click the "**Reset**" button.

The page also allows you to reboot the unit by remote, clicking the "**Reboot**" button. For the Codec module this is equivalent 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.

"Event log" page

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

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

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

- "Coarse": only essential events are logged, such as reboots, link set up, alarms… This type of history can be useful as a link record, and gives a very synthetic overview.

- "Normal": gives more details, and this is the default setting, suitable for most operation needs.

- "Debug": absolutely all events are logged; this makes a very "verbose" and technical history, reserved for testing and fixing issues.

The Codec module can also send event messages to a SYSLOG server is such server is available on the network: enter the IP address of this server and click the "**Apply**" button. From then on, module 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 module's log file.

Events are time-stamped (date, hour, minute, second) from the internal clock of the unit. This clock is not backed up (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 2.5.9 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 reading the logs.*

Under the log window you can find some control buttons:

- "**Refresh**": click this button to update the display and show the most recent events. This button is relevant if the display has been frozen (with "Hold"), in order to refresh the event display on demand.

- "**Hold**": normally the event window scrolls at any time while new events occur. Click this button to stop this and freeze the display. The button becomes "**Continue**". *The events are still logged, only the page update is interrupted.*

- "**Continue**": resumes the automatic updating of the event display.

- "**Save Log File as…**": allows you to download on the control computer the history file. This takes place as a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

ⓘ *If you want to submit to AETA information about a possible issue or bug, you should first set the log level at "Debug", then save a log file right after the relevant event or scenario, and send to AETA the log file.*

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.

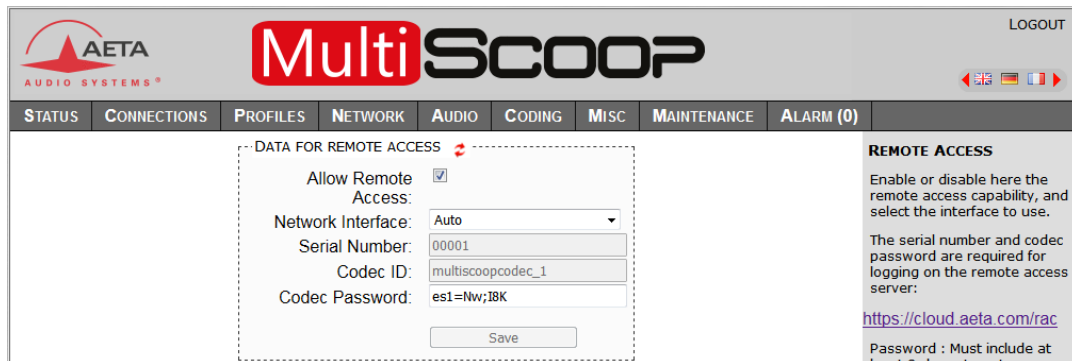Partial exports are also possible, selecting the items you wish to include: call profiles, presets, snapshots, settings…

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 "**Import**". The file must be a file previously exported from a Codec module (same unit, or another with a compatible hardware configuration). The settings or memories included in the file are respectively applied to the codec, or recorded in its memory of profiles/snapshots.

ⓘ *Warning: the export files can be imported into a device with the <u>same firmware version</u>, but there is no guarantee of "portability" of the configuration files from one firmware version to another. In other terms, the outcome is not guaranteed when importing into a unit files which were exported from a unit with a different firmware version. We recommend in such case to carefully check the settings after importing the file, although this works correctly in most cases.*

"REMOTE ACCESS" page

You find on this page the settings related to the remote assistance and/or remote access.

There are some differences depending on the installed options:

- "Allow Remote Access" becomes "Allow Remote Assistance" if no "remote access" option is installed:

- The "Codec Password" must be known for getting remote access to the html pages, so it protects the unit against undesired access. No such password exists if no "remote access" option is installed.

ⓘ *Be aware, if you are currently on this page using remote access, that any change here can make you lose the connection. In the best case (password change), you will have to re-connect on the remote access server, using the new password.*

You can find the procedure for using remote access in annex 6.4.2, How to use the remote access service.

2.5.11. "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.

Clicking the tab, you reach this "Alarm" page showing all the alarm conditions. All possible issues are

listed, but only those actually detected are highlighted with the icon.

The issues are grouped in two areas:

- Alarms related to "TRANSMISSION": essentially issues related to the decoder synchronization. In addition, for ISDN links, there may be encoder or decoder fallback cases ("fallback" is the situation when the encoder or decoder configuration actually set on the link differs from the one which was initially programmed or expected).

- Alarms related to the AUDIO interfaces: audio overload, AES input errors.

2.6. Technical characteristics

2.6.1. Interface characteristics

Analog audio inputs

The rear mount module "Audio Interfaces" is required for having these balanced inputs, available on the "Analog&AES" socket. See the wiring below: "Analog&AES" socket ("Audio Interfaces" rear mount module).

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

Maximum input level:	adjustable from +4 dBu to +22 dBu
Nominal input impedance:	10 kΩ

Analog audio outputs

The rear mount module "Audio Interfaces" is required for having these balanced outputs, available on the "Analog&AES" socket. See the wiring below: "Analog&AES" socket ("Audio Interfaces" rear mount module).

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

Maximum output level:	adjustable from +4 dBu to +22 dBu	
Nominal load impedance:	600 Ω or 10 kΩ	
Output impedance:	< 50 Ω	
Symmetry:	> 40 dB	(ZL = 150 Ω)

Digital audio input and output

The rear mount module "Audio Interfaces" is required for having these interfaces, available on the "Analog&AES" socket. See the wiring below: "Analog&AES" socket ("Audio Interfaces" rear mount module).

These interfaces comply with recommendations:

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

They support a sampling rate from 28 to 96 kHz.

"Analog&AES" socket ("Audio Interfaces" rear mount module)

This 15-pin male sub-D socket groups the audio inputs/outputs of a Codec module (the AES67 inputs/outputs are not handled by this module).

The pinout is shown by the following table:

Broche	Fonction
1	Input 1 (+), analog or digital
9	Input 1 (-), analog or digital
2	Ground
10	Input 2 (-), analog
3	Input 2 (+), analog
11	Ground
4	Output 1 (+), analog or digital
12	Output 1 (-), analog or digital
5	Output 2 (+), analog
13	Output 2 (-), analog
6	Ground
14	Ground
7	
15	
8	

Input 1 and output 1 have an analog or digital (AES3) format depending on the selection done for the audio part (see 2.5.7, "AUDIO" tab).

Ethernet Interface

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

The installation and operation of these interfaces is detailed in 2.3.3, Initial setup of the Ethernet interface.

ISDN interface

This interface is available only on the IP/ISDN version of the module.

The socket is an ISDN S0 interface (BRI), the supported protocols are ETSI (Euro RNIS), NI-1 / NI-2 or NTT.

"Serial data" interface (reserved)

The rear mount module "Audio Interfaces" is required for having this interface.

Reserved for future use

GPIO interface

The relay transmission interface (refer to 2.1.4, "Transmission of isolated relays", about this feature) is available on the "GPIO" RJ45 socket. It includes two isolated current loop inputs and two relay outputs.

The following table shows the pinout of this socket:

Pin	Function	
1	+5V, provided by the module, protected by a 0.5 A fuse	Output
2	GPI 1 Input loop (a)	Entrée
3	GPI 2 Input loop (a)	Entrée
4	GPI 1 and GPI 2 Common (b)	Entrée
5	Ground	
6	GPO 1 relay output loop (a)	Output
7	GPO 2 relay output loop (a)	Output
8	GPO 1 and GPO 2 Common (b)	Output

All loops are opto-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: ca 870 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 500 mA current consumption), e.g. to power the input loops, or LED indicators connected to the relay outputs. *Do not draw too high current from this line!*

2.6.2. 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.

Section to be defined.

2.6.3. IP network protocols and ports

The Codec module implements or complies with the following protocols (non exhaustive list):

- Physical and link layers: Ethernet, 100BaseT, 10BaseT
- Network/Transport layers (IPv4): TCP/IP, UDP/IP, RTP/IP
- Application: DHCP, DNS, HTTP, HTTPS, NTP, STUN, SYSLOG
- Audio transmission: SIP signaling, SDP, RTP, RTCP, RFC3550/3551, RFC3640
- Compliant with EBU recommendation Tech 3326 (interoperability of audio codecs for contribution)

The TCP ports used by the device are the following:

Type	Port		Designation	Direction	Notes
	Local	Remote			
TCP	22		SSH	⇆	Debug interface
	80		HTTP	⇆	Embedded html server
	443		HTTPS	⇆	Embedded html server
	6000		Control	⇆	Remote control ("command line" mode)
	7001		Control	⇆	Remote control; used by Scoop Manager
		234	Control	⇆	Connection to cloud.aeta.com; used for the remote access function

None of these ports is involved in audio links.

The UDP ports used by the device are the following:

Type	Port		Designation	Direction	Notes
	Local	**Remote**			
UDP		53	DNS	⇆	Domain name resolution
		123	NTP	⇆	For querying an NTP server
		514	SYSLOG	→	For sending messages to a SYSLOG server
	2382		AETA enumeration	⇆	Used by "AetaScan"
		3478	STUN	⇆	For querying a STUN server
		3479	STUN	⇆	For querying a STUN server
	5004		RTP	⇆	For audio streaming (SIP)
	5005		RTCP	⇆	For audio streaming (SIP)
	5006		RTP	⇆	For audio streaming (double SIP codec or Double Streaming)
	5007		RTCP	⇆	For audio streaming (double SIP codec or Double Streaming)
	5060		SIP	⇆	SIP signalling
	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** are default values that can be edited (**AoIP Parameters** page).
However, these are local ports of the module; the corresponding remote ports do not have a fixed value and they only depend on the remote device. For instance, the remote SIP port is by default 5060, but it may have another value.

The grey background highlights the ports possibly involved in an audio connection implementing SIP.

2.6.4. Dimensions

The Codec module has a 3U "Europe" format, with a width of 6TE (30 mm).

2.6.5. Versions

The basic version of the Codec module features an Ethernet interface for audio over IP transmission. Another "IP/ISDN" version also includes an S0 ISDN interface.

Code	Version	Description
80 00 271 10	IP Codec Module	Mono/stereo AoIP codec, switchable to double mono codec
80 00 271 11	IP/ISDN Codec Module	Mono/stereo AoIP or ISDN codec, switchable to double mono codec

2.6.6. Accessories and related products

Code	Version	Description
80 00 271 40	Rear mount audio interface module	Rear mount module for a Codec module, audio interfaces on a Sub-D 15M socket
80 00 273 01	Sub-D <-> XLR splitter cable	Optional addition to the rear mount Codec module, audio I/O split over XLR plugs (2 female + 2 male).

3. "Control" module

The "Control" module allows to supervise the whole MultiScoop system, and manage all the modules of the chassis. Specifically it acts as a gateway for accessing the Codec modules across its Ethernet/IP interface.

Table 1 (page 2) shows the main features of the module.

3.1. Functions

The module detects the modules installed and their possible failures.

It is controlled using its embedded html pages, via an Ethernet interface.

The module also fulfils a gateway function providing access, via this interface, to all the "Codec" and "AES67" modules installed in the MultiScoop chassis. This brings several advantages:

- A single physical interface allows to control all the Codec modules of the system;
- This control interface can be totally separated from the Ethernet interfaces of the Codec modules.

This characteristic applies both to the html pages and to the control protocol used by the Scoop Manager software.

Ethernet interfaces of the codecs

Control   C8 C7 C6 C5 C4 C3 C2 C1

Codec modules

Ethernet interface of module
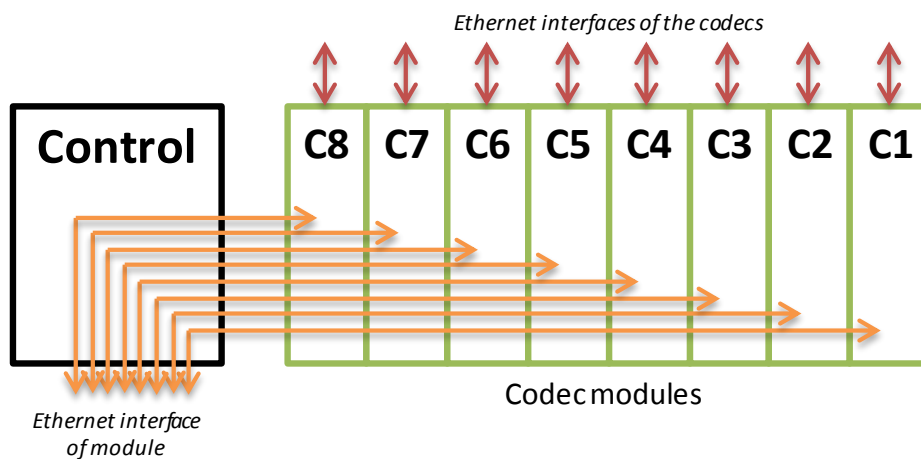
The "Control" module can be backed up by a redundant module that takes over the control in case it is defective or it is removed from the chassis.

3.2. Physical description of the module

The Control module is a standard 3RU "Europe" format module, with a 6TE width (30 mm).
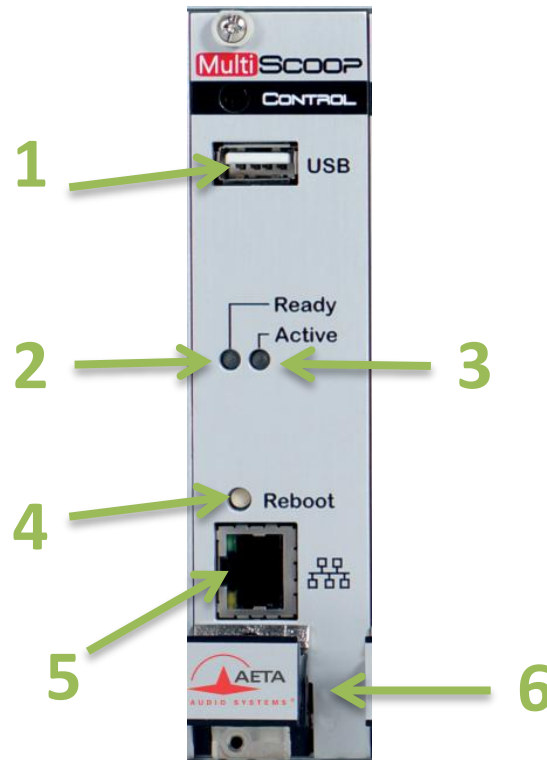
Figure 7 – Front panel of the "Control" module

1. USB socket (reserved)
2. "Ready" status indicator, module is operative
3. "Active" status indicator, module is operative and active
4. Reset button
5. Ethernet interface (10/100BaseT)
6. Extraction lever

The status indicators can be understood as follows:

Indicator	Ready	Active
Function	Status of module	Actual control
Off		Module in standby (no control over the system)
Green	Module is operative	The module is operative and actually in control of the system
Red	Booting phase	

Only one module is active and actually controls the system. When two "Control" modules are present (redundancy), module 1 (leftmost) is the active one. In such case however, if module 1 is extracted, module 2 automatically takes control.

3.3. Installation and maintenance

3.3.1. Assembly and connections

The module can be put in place even when the chassis is powered on. <u>It can only be mounted in one of the two slots reserved for the "Control" modules</u> (see above 1.1, Physical description of the system).

3.3.2. Initial setup of the Ethernet interface

For setting into operation, first connect the Ethernet interface to the network, using CAT5 wiring.

- Connections to 10BaseT or 100BaseT interfaces are both suitable, as the module automatically switches to the adequate 10 Mbit/s or 100 Mbit/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 can be done in a variety of ways depending on your network and system configuration, as described in the following chapters.

Case 1: DHCP server available

This case is quite simple; DHCP is enabled by default on the Control module. If a DHCP server is active on the network to which the module is attached, the server assigns a valid IP address and provides the unit with the appropriate settings.

To know the IP address allocated by the DHCP server:

- Either a tool in your network may provide this information (DHCP server lease table, for example).
- Either you do not have this type of tool, you can then use the "Aetascan" tool (see detail in 6.2, Using the "Aetascan" tool). When you launch this utility on a computer that is connected to the same local area network as the Control module, it searches for and enumerates the AETA devices and modules that are present, including their IP addresses. The Control module is listed with as *Device Type* "MultiScoop Control".

ⓘ *If, instead of the default setting, the module is previously set with a static address (with DHCP disabled), refer to the following method.*

Case 2: no DHCP server available

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

First you must get the current IP address of the codec: either this is a previously set static setting, or a "link local" address that the module has auto-configured because it found no DHCP server. The "AetaScan" tool will allow you to detect this address, whatever the current settings (see 6.2, Using the "Aetascan" tool). The module is listed with "MultiScoop Control " as its *Device Type*.

If this address is reachable from the computer, you can enter it in the address bar of the web browser and access the html pages of the module. Then you can set up the addressing as required, via the "ETHERNET PARAMETERS" of the "NETWORK" tab (see 3.4.4, "ETHERNET" tab).

If this address is not reachable from the computer, you should go through the following steps:

- Change the computer address (temporarily) so it belongs to the same sub-network as the module.

- Access the html pages of the module by entering its IP address in the browser's address bar.

- Setup the addressing of the module as required, via the "ETHERNET PARAMETERS" of the "NETWORK" tab (see 3.4.4, "ETHERNET" tab).

- Setup the computer back to its original addressing.

Checking the IP configuration

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

To check the setting, from a computer connected to the same network, open a web browser and enter the module's IP address in the address bar: the home page should be displayed.

ⓘ *If a problem in the IP addressing makes the module unreachable, a backup procedure should be used; this one is the same as for a Codec module, as described in 2.3.4, "Backup reset".*

Optional configuration of the "link" Ethernet layer

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

This setting is suitable for almost all situations and usually you don't need to change it. However, in the rare cases where this is needed, it is possible to set the desired mode manually: in the "ETHERNET PARAMETERS" of the "NETWORK" tab, use the "Link Mode" parameter (drop-down list). Click "**Save**" to apply the setting.

ⓘ *Be aware that you may lose control over the module if this setting is inappropriate!*

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

3.3.3. Maintenance

The replacement of a module can be carried out hot, without cutting the power supply and without disturbing the other modules in the chassis.

- If this is the only Control module installed: of course, the control is lost while the module exchange is being done. However, <u>none of the other modules</u> in the chassis is disturbed in its operation.

- If two Control modules are installed while the module *in standby* is replaced: the module exchange just has no impact on the rest of the system.

- If two Control modules are installed and the *active* module is replaced: when extracting the module, the other Control module becomes the active module, without disturbing the other modules. When the module is inserted back and ready, it can take control again.

For ease of maintenance, it is possible to export and save all the configuration of the module, which makes it easy to restore it, even possibly on a replacement module. Such transfers can be done using the embedded html interface; refer to "MAINTENANCE" tab for the operating mode.

Clearing all settings

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

To clear the whole configuration and come back to the factory settings, normally you should go to the "MAINTENANCE" tab, "RESET" page, and select "**Factory Reset**".

The default (html) password is blank.

ⓘ *If it is not possible to access the menu to perform the factory reset (administrator password lost), a backup reset should be done: the procedure is the same as for a Codec module, described in 2.3.4, "Backup reset", and using the same file.*

Firmware update

This is done using the html interface of the module. The procedure for updating the firmware is described further in 3.4.7, "SYSTEM UPDATE" page.

3.4. Operating mode – Embedded html interface

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

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

3.4.1. Accessing the html pages

There are two ways to connect on these pages:

- Via the Ethernet interface of the module (computer connected to the same network): enter the IP address of the module in the "address" or "URL" field of the browser. *You can also use HTTPS, provided that you override or bypass the security warning regarding the site certificate of the module.*

- By remote access via the AETA server: see further in 3.4.7, "REMOTE ACCESS" page.

The home page that is displayed is similar to the following picture:
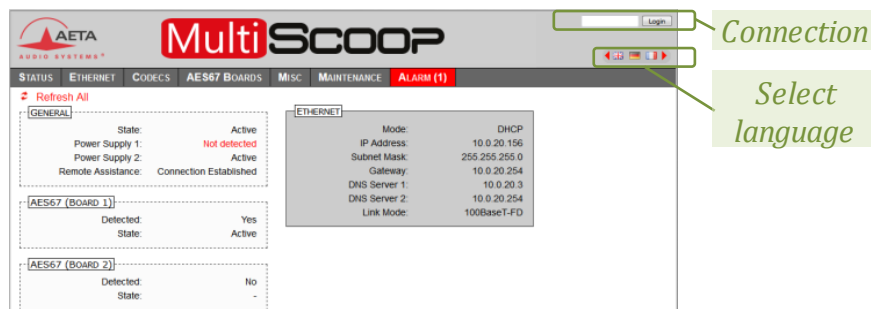
Select the language by clicking the suitable flag.

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

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

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

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

---

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

3.4.2. Principles of operation of the html pages

The picture below shows a typical page of the interface:

On top you find a bar of tabs corresponding to the various categories of functions and parameters for the module. 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 or sections (each surrounded with a frame). On the right side, a text area provides additional help and hints.

To force a refresh, click the ⟳ icon: the data in the area or frame are read and refreshed.

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.

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.

3.4.3. "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.
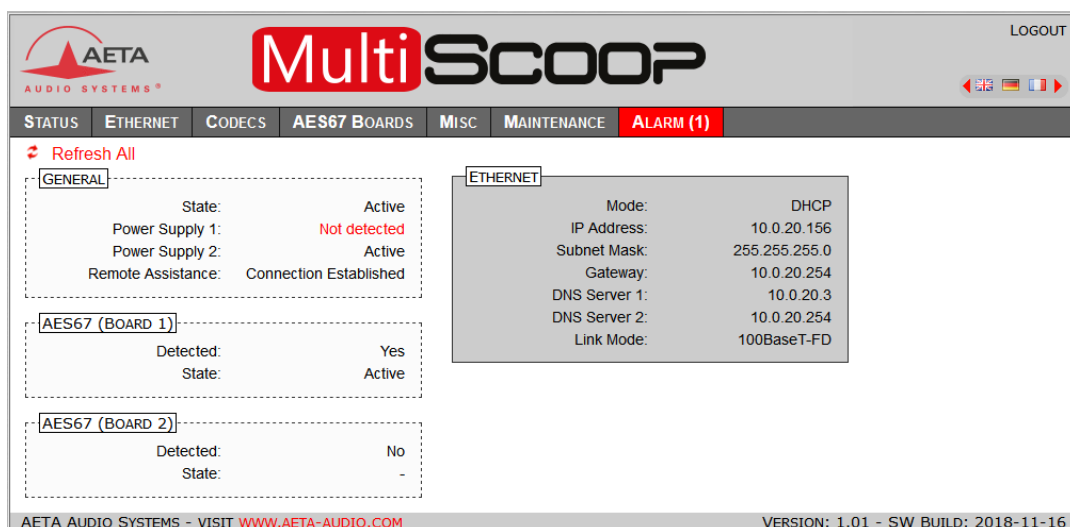
This page is accessible without a log in, and without a password.

The "GENERAL" section shows:

- The state (active or standby) of the module.
- The presence or not of a power supply module is each of the power slots.

An "AES67" section for each AES67 slot shows whether a module is installed, and whether it is active or not.
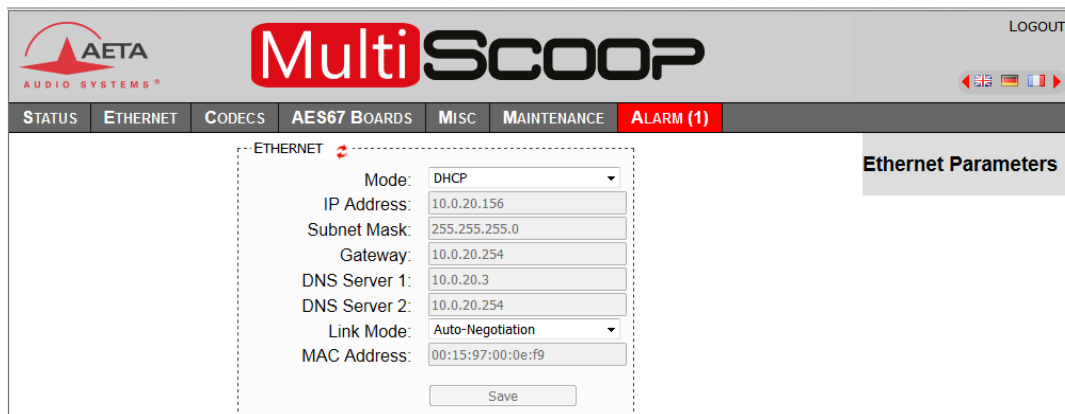
The "ETHERNET" section reminds the configuration of the Ethernet interface.

The page footer shows the firmware version of the module.

3.4.4. "ETHERNET" tab

This tab allows to configure the Ethernet interface of the module.

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

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

ⓘ *Settings regarding the Ethernet interface and IP addressing are especially critical, because this is the interface you are controlling the module 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!*

Notes:

- **Mode**: enable or disable DHCP.
  This selection conditions most of the other settings.

- If DHCP is not used, several fields are available for entering the necessary parameters, such as **IP Address**, **Gateway**, etc. These are shaded (read only) if DHCP is active.

- **Link Mode**: concerns the physical Ethernet interface; the default setting "Auto-Negotiation" is suitable for almost all situations.

3.4.5. "CODECS" tab

This page fulfils two purposes:

- Detection of the Codec modules available in the system.
- Direct access for controlling the Codec modules, wherein the Control module acts as a gateway.

For every Codec module detected, a "CODEC" section is displayed with the number (from 1 to 8) corresponding to its physical location in the chassis (see 1.1, Physical description of the system). In the above example, all 8 slots are busy. Basic information on each Codec module is displayed:

- **Type** of the module: IP or IP/ISDN.
- **Audio format** used on the module.
- **IP address** of the module.
- **Status** of the connection, for each "sub-codec" if the module is et as a double mono codec.
- **SIP User**: SIP identifier for each "sub-codec".

The IP address includes an hyperlink: click this link and you can access directly the html pages of the concerned module, through the internal communication links of the MultiScoop. It is not necessary for the module's Ethernet interface to be reachable from the computer.

ⓘ *When the Control module is on standby (inactive), the module can't access the Codec modules, and the "CODEC" tab is not available.*

ⓘ *You can also go directly to the html pages of a Codec module by specifying, after the IP address of the Control module, a port number according to the following rule:*

Codec module	Port to specify
CODEC 1	8081
CODEC 2	8082
CODEC 3	8083
CODEC 4	8084
CODEC 5	8085
CODEC 6	8086
CODEC 7	8087
CODEC 8	8088

For example, if the IP address of the Control module is 10.0.10.20, enter the URL 10.0.10.20:8085 and you directly get the html pages of the Codec in slot 5.

3.4.6. "MISC" tab

This MISC tab includes two settings.

- It is possible to configure an **NTP server**: this kind of server provides a time reference (from the Internet or a server on the local network), useful and recommended for the absolute time stamping of the events in the log (see further 3.4.7, "MAINTENANCE" tab). If such a server is available and accessible via the IP interface, enter its address in the "Other 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 changes done**, 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.

3.4.7. "MAINTENANCE" tab

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

"LOGIN DATA" page

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

Note: this password has no relationship with the password used for remote access (see further).

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

If you lose the html password, you must perform a complete reset of the settings, using the backup reset procedure: see in 2.3.4, Backup reset.

"SYSTEM UPDATE" page

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

For an update, the file must be provided to you by AETA; usually you can download it from the AETA web site https://www.aeta-audio.com. The procedure is rather simple:

- From this page, click "**Browse**…", find and select the update file.
- Click the "**Update**" button.
- The process begins, and on the module's front face the status indicators warn of this by a specific sequence.
- Wait for the completion of the update. Normally the module should reboot by itself at the end of the process.

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

- Make sure the link between the control computer and the module is stable (no undesirable interruption during the process).
- Make sure the module is not powered off during the process.

"RESET" page

This page provides two functions: clearing the settings saved in the unit, and rebooting the unit.

You can reset the unit completely by selecting "**Factory Reset**".

The page also allows you to reboot the unit by remote, clicking the "**Reboot**" button. The reboot actually starts once you confirm by clicking "OK" in the confirmation dialog box that pops up. This is equivalent 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.

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

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

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

- "Coarse": only essential events are logged, such as reboots, alarms… This type of history 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.

The module can also send event messages to a SYSLOG server is such server is available on the network: enter the IP address of this server and click the "**Apply**" button. From then on, module 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 module's log file.

Events are time-stamped (date, hour, minute, second) from the internal clock of the unit. This clock is not backed up (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 3.4.6 above).

ⓘ *Important notice: time stamps are universal time (UTC), hence they take no account of geographical location, neither any DST (daylight saving time).*

Under the log window you can find some control buttons:

- "**Refresh**": click this button to update the display and show the most recent events. This button is relevant if the display has been frozen (with "Hold"), in order to refresh the event display on demand.
- "**Hold**": normally the event window scrolls at any time while new events occur. Click this button to stop this and freeze the display. The button becomes "**Continue**". *The events are still logged, only the page update is interrupted.*
- "**Continue**": resumes the automatic updating of the event display.
- "**Save Log File as…**": allows you to download on the control computer the history file. This takes place as a typical download. Depending on the browser and its settings, you can possibly select the file destination, name…

ⓘ *If you want to submit to AETA information about a possible issue or bug, you should first set the log level at "Debug", then save a log file right after the relevant event or scenario, and send to AETA the log file.*

"SETUP TRANSFER" page

This page provides a means to backup all the settings of the unit into a file, and reciprocally to restore a complete configuration from a file previously saved in this way.

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. Depending on the browser and its settings, you can possibly select the file destination, name…

The right side is for "importing" the device configuration: click "**Browse…**" to select the file to be imported, and launch the process by clicking "**Import**". The file must be a file previously exported from a MultiScoop Control module (same unit or another). The settings included in the file are then applied to the module.

ⓘ *Warning: the export files can be imported into a device with the <u>same firmware version</u>, but there is no guarantee of "portability" of the configuration files from one firmware version to another. In other terms, the outcome is not guaranteed when importing into a unit files which were exported from a unit with a different firmware version, although this works correctly in most cases.*

"REMOTE ACCESS" page

You find on this page the settings related to the remote assistance and/or remote access.

There are some differences depending on the installed options:

- "Allow Remote Access" becomes "Allow Remote Assistance" if no "remote access" option is installed:

- The "Codec Password" must be known for getting remote access to the html pages, so it protects the unit against undesired access. No such password exists if no "remote access" option is installed.

ⓘ *Be aware, if you are currently on this page using remote access, that any change here can make you lose the connection. In the best case (password change), you will have to re-connect on the remote access server, using the new password.*

You can find the procedure for using remote access in annex 6.4.2, How to use the remote access service.

Important: for this module, the type of product to select for the remote access interface is "MultiScoop/Control".

3.4.8. "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.

3.5. Technical characteristics

3.5.1. Ethernet Interface characteristics

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

The installation and operation of these interfaces is detailed in 3.3.2, Initial setup of the Ethernet interface.

3.5.2. IP network protocols and ports

The Control module 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
- Application: DHCP, DNS, HTTP, HTTPS, NTP, SYSLOG

The TCP ports used by the device are the following:

Type	Port		Designation	Direction	Notes
	Local	Remote			
TCP	22		SSH	⇄	Debug interface
	80		HTTP	⇄	Embedded html server
	443		HTTPS	⇄	Embedded html server
	7071		Control	⇄	Codec 1 remote control ; used by Scoop Manager
	7072		Control	⇄	Codec 2 remote control ; used by Scoop Manager
	7073		Control	⇄	Codec 3 remote control ; used by Scoop Manager
	7074		Control	⇄	Codec 4 remote control ; used by Scoop Manager
	7075		Control	⇄	Codec 5 remote control ; used by Scoop Manager
	7076		Control	⇄	Codec 6 remote control ; used by Scoop Manager
	7077		Control	⇄	Codec 7 remote control ; used by Scoop Manager
	7078		Control	⇄	Codec 8 remote control ; used by Scoop Manager

(continued)

Type	Port		Designation	Direction	Notes
	Local	Remote			
TCP		234	Control	⇆	Connection to cloud.aeta.com; used by the remote access function
	8081		Control	⇆	Access to embedded html server, Codec 1
	8082		Control	⇆	Access to embedded html server, Codec 2
	8083		Control	⇆	Access to embedded html server, Codec 3
	8084		Control	⇆	Access to embedded html server, Codec 4
	8085		Control	⇆	Access to embedded html server, Codec 5
	8086		Control	⇆	Access to embedded html server, Codec 6
	8087		Control	⇆	Access to embedded html server, Codec 7
	8088		Control	⇆	Access to embedded html server, Codec 8

The UDP ports used by the device are the following:

Type	Port		Designation	Direction	Notes
	Local	Remote			
UDP		53	DNS	⇆	Domain name resolution
		123	NTP	⇆	For querying an NTP server
		514	SYSLOG	→	For sending messages to a SYSLOG server
	2382		AETA enumeration	⇆	Used by "AetaScan"

3.5.3. Dimensions

The Control module has a 3U "Europe" format, with a width of 6TE (30 mm).

Its weight is around 0.16 kg.

4. "AES67" module

The "AES67" module provides the MultiScoop chassis 16 audio inputs and 16 audio outputs with two AES67/Ravenna interfaces.

Table 1 (page 2) shows the main features of the module.

4.1. Functions

The module can receive 16 audio channels on the AES67 interfaces. The input channels can be extracted from 1 to 16 AES67 input streams.

At the output, the module transmits a 16 channel stream on the AES67 interface.

The module communicates with the other parts of the MultiScoop:

- An internal link allows the active "Control" module to configure and supervise the AES67 module.
  This function is not yet implemented.

- The module casts its 16 input channels to the Codec modules, 2 audio channels per Codec module.

- The module gathers the 16 audio outputs from the Codec modules (2 per Codec module) for building the AES67 output streams.

The AES67 module can be backed up by a redundant module that takes over in its place if it has a failure or is extracted from the chassis.

4.2. Physical description of the module

The AES67 module is a standard 3RU "Europe" format module, with a 6TE width (30 mm).
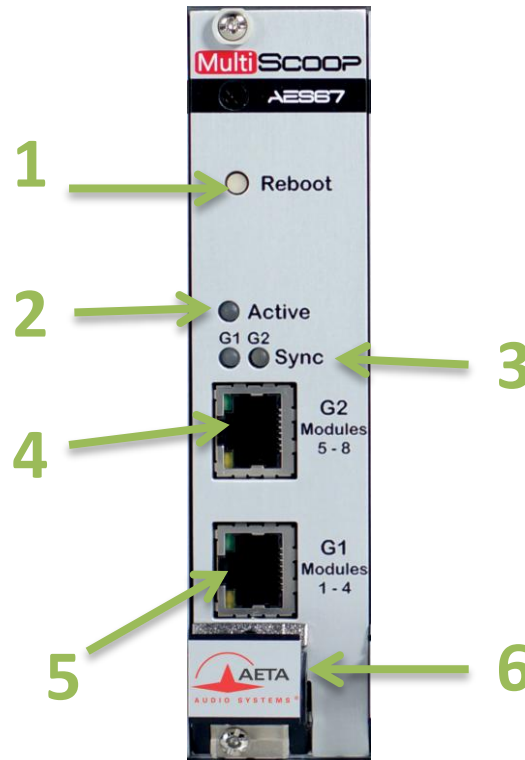
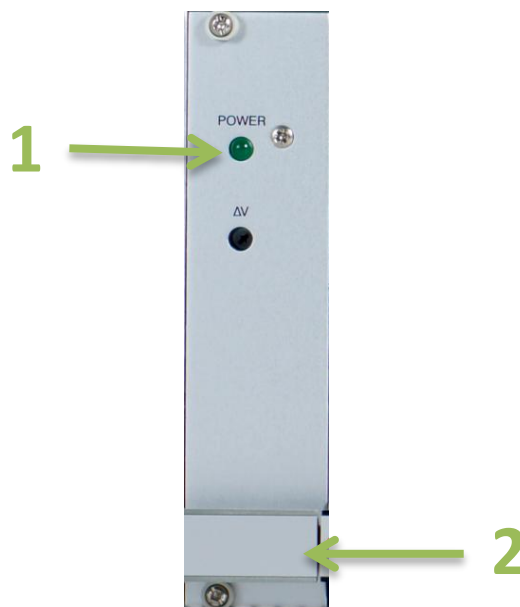**Figure 8 – Front panel of the "AES67" module**

1. Reset button
2. "Active" status indicator, module is operative and active
3. "Sync" status indicator, module is synchronized (PTP clock)
4. AES67 Ethernet socket, port 2
5. AES67 Ethernet socket, port 1
6. Extraction lever

The status indicators can be understood as follows:

Indicator	Active	Sync 1	Sync 2
Function	Actual control	Sync status of port 1	Sync status of port 2
Off	Module in standby (no audio exchanges)		
Green	The module is operative and actually processing the AES67 interfaces	Clock and synchronization OK on port 1	Clock and synchronization OK on port 2 (*reserved*)
Red	Booting phase	Clock fault or loss of synchronization	Clock fault or loss of synchronization

Only one module is active and actually controls the AES67 inputs/outputs. When two "AES67" modules are present (redundancy), module 1 (leftmost) is the active one. In such case however, if module 1 is extracted, module 2 automatically takes over.

4.3. Installation and maintenance

4.3.1. Assembly and connections

The module can be put in place even when the chassis is powered on. <u>It can only be mounted in one of the slots reserved for the "AES67" modules</u> (see above 1.1, Physical description of the system).

4.3.2. Initial setup of the Ethernet interface

To be defined.

4.4. Detailed operating mode

On this firmware version, the configuration of the AES67 interfaces is performed thanks to an html server embedded on the AES67 module, or preferably using the configuration application ANEMAN from Merging Technologies.

5. "Power" module

5.1. Functions

The Power module provides to the various modules of the system the needed power sources, from an AC mains input. Table 1 (page 2) shows the main features of the module.

The module can be backed by a redundant module:

- The two modules run in active redundancy mode, with load sharing.
- If a module has a failure or is extracted from the chassis, the other module provides alone 100% of the needed energy. The transition is done instantly and without disturbing the other modules.

As each module draws its power from a separate mains input, the system also enables, whenever needed, the operation from two mains sources.

5.2. Physical description of the module

The Power module is a standard 3RU "Europe" format module, with a 6TE width (30 mm). It weighs around 0.54 kg.

Figure 9 – Front panel of the "Power" module

1. Mains presence and operation indicator
2. Extraction lever

5.3. Assembly and connections

The module can be put in place even when the chassis is powered on. It can only be mounted in one of the slots reserved for the "Power" modules (see above 1.1, Physical description of the system).

6. Annexes

6.1. Details on the coding algorithms

6.1.1. G711 coding

Application: telephony, coordination. Low latency.

G711 is the standard coding used for voice transmission on public telephone networks, and features 300 to 3400 Hz audio bandwidth. This algorithm is typically used for links over IP networks with IP telephones or VoIP gateways. Via ISDN, G711 is used for links with telephones or hybrid devices.

6.1.2. G722 coding

Application: commentaries, coordination. Low latency.

This mono coding algorithm at a 64 kbit/s bit rate is a reference for commentaries, and features a 50-7000 Hz bandwidth.

It is available for ISDN or IP networks (single codec, or double ISDN or AoIP codec).

For ISDN, three synchronisation modes are possible:

- "Statistical recovery" byte synchronisation method (alias SRT);
- H221 synchronisation; in this case, 1.6 kbit/s from the compressed data are used for this;
- H221 synchronisation and H242 protocol.

No specific synchronisation is required for the IP mode.

6.1.3. MPEG Audio Layer 2 coding

Application: mono or stereo music, high quality.

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

The two channel modes come in three varaiations:

- Stereo: the two audio channels are bit-rate compressed independently.
- Dual mono: the encoding is just the same as the previous case, but this choice applies to channels without any acoustic relationship, such as for example commentaries in two languages.
- "Joint stereo": applies to a stereo program, but here the encoding process takes advantage of the inter-channel correlation. To be used only for a stereo program.

The 24 kHz sampling rate features a moderate bandwidth (10 kHz) and is rather useful for commentaries.

ⓘ *The latency is rather high with this sampling rate.*

MPEG L2 is available both for ISDN and IP networks.

6.1.4. MPEG AAC algorithms

Application: music, low capacity transmission channels.

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

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

AAC codecs are available both for ISDN and IP networks.

6.1.5. OPUS coding

Application music, low capacity transmission channels. Moderate latency.

Opus is an open source coding algorithm (under BSD license), described by RFC6716 from the IETF. Like MPEG coding algorithms, it is a wide band audio codec, but it features distinctive characteristics:

- Low latency; the algorithmic delay of Opus is very low compared to MPEG coding (Layer 2, AAC, HE-AAC...).
- Adaptability: Opus features many parameters influencing its performance, but it is possible to adapt them automatically depending on the external constraints and the signal characteristics. In the MultiScoop, the configuration stays very simple: the user just imposes the desired bit rate, and the codec automatically adjusts the other parameters (for instance the audio bandwidth).
- Capacity to apply changes "on the fly": the coding allows changing its operation parameters on the fly, with no interruption and no artefacts when switching. In this way the bit rate can be changed during the transmission, without any audio noise or dropout.
  In addition, when the remote device is another AETA codec, it will automatically apply the same bit rate change to the stream it is transmitting.

Opus coding is only available for IP networks.

6.1.6. 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).

6.2. Using the "Aetascan" tool

This software can be downloaded from AETA's web site https://www.aeta-audio.com.

When launching the tool, the following window is displayed:
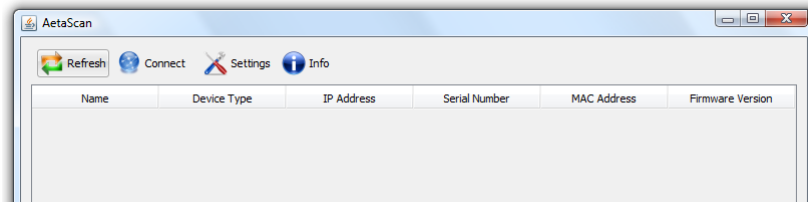
Click the "Refresh" icon: the tool scans the local network, detects the AETA codecs and lists them with their IP and MAC address:
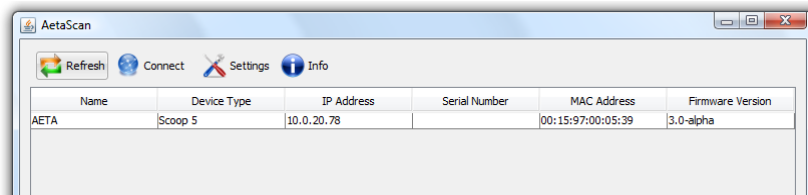
ⓘ *At this stage, you can click/select a codec in the list, and click the "Connect" icon or double-click the codec's row to open directly the html interface of the selected device (using the computer's default browser).*

By default, the tool scans only the LAN. It is possible to search devices on another sub-network (given, of course, that it is reachable through an authorized network path), changing the settings behind the "Settings" icon:
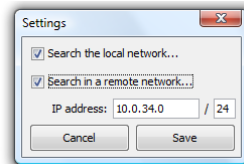
(Warning: the result is not guaranteed, because such scan can only work if the involved routers can relay messages to the "broadcast" address of the remote sub-network).

6.3. Using the Double Streaming system

6.3.1. Context

Audio transmission over IP (AoIP) is sometimes subject to data losses for various reasons, as for instance:

- In case of network congestion, the overloaded routers happen to drop packets without forwarding them.
- On mobile radio links, radio signal loss or fading cause temporary interruptions, that can last for a rather long time.

Such problems are more common on connections via the Internet.

Packet losses cause disturbances, ranging from light artefacts to the complete muting of the signal for several seconds or minutes.

The AETA "Double Streaming" system is an efficient way of strongly increasing the robustness of an AoIP link, thanks to a redundant link that can compensate for losses on the main link.

6.3.2. Presentation of the "Double Streaming" system

Let us consider a classical AoIP link between two audio codecs:
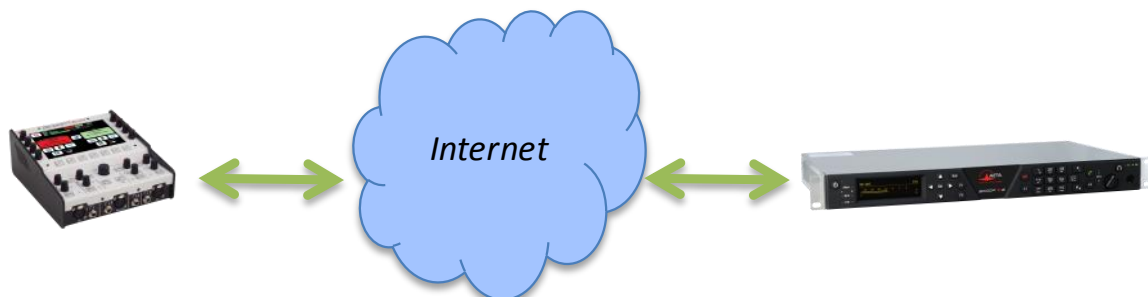
At each end the codec accesses the Internet through an interface which can be of various types:

- Ethernet connection
- Mobile access (integrated or USB device)
- Wi-Fi access

Packet losses can be experienced in various locations on the path between the two units.

General principle of operation

Double Streaming relies on setting up two SIP links, simultaneous but via two separate routes:
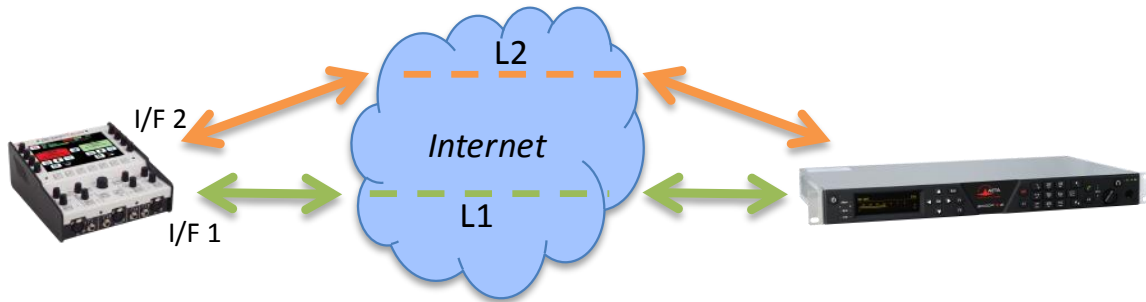
In addition to the "main" SIP link, a "secondary" SIP link is set up between the units. This link transports a copy of the audio stream of the main link.

On each codec, a specific transmission interface can be selected for each of the two AoIP connections. In this way two independent routes are created, whose possible losses are not correlated.

The codec which receives the two (normally) identical streams recombines them, discarding the duplicates:
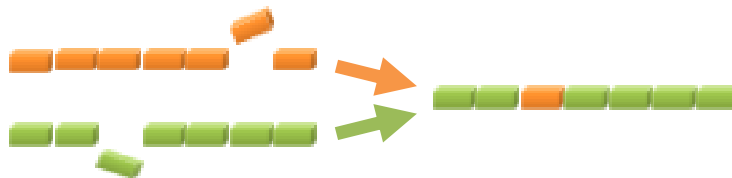
Whenever a packet is lost on the transmission, its copy from the other link compensates the loss, and a 100% lossless stream is delivered at the end of the process.

In an extreme case, if one of the two links is totally interrupted, the complete original stream is still available at the output. The few losses on the remaining link are no more compensated for, but they are much less noticeable than a complete loss.

The usage of SIP is a specific advantage of AETA's Double Streaming:

- The system works on a normal SIP architecture, and is compatible with any SIP server.
- The system is easily "portable", specifically it allows using mobile links.
- Like for a normal SIP link, an acceptable security level can be reached.
- Like for a normal SIP link, it is also possible to make links without a SIP server (with same limitations).

Double Streaming is available with all AETA codecs of the following product ranges:

ScoopTeam

Scoopy+ S

Scoop5 S and Scoop5 S-IP

µScoop

MultiScoop

Chapter 6.3.3 describes the operating instructions for using it.

Some examples of operation setups

Case 1: Redundancy on one end

In this first case, a redundant network access is used on one end of the link only, and the two redundant links are merged on a single network access on the other end.

The end with redundancy is protected against faults encountered on the main link, either at the local access level or on the path over the network towards the other end.

Conversely, the network access on the other end gets no additional protection.
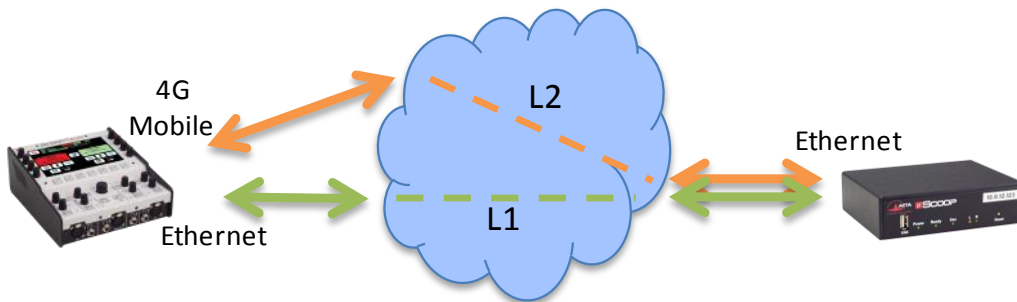
Example 1: Remote report, wired connection secured by a mobile link.

In the above example, in the field (left side on the picture) a 4G network connection is used for securing a main link via an Ethernet connection. On the control room side, the two streams are merged on a high reliability wired network that needs no redundancy.

Case 2: End-to-end redundancy

In this case, a redundant network access is used at each end.

A maximum protection is then available against faults experienced either at the network access level or inside the network.
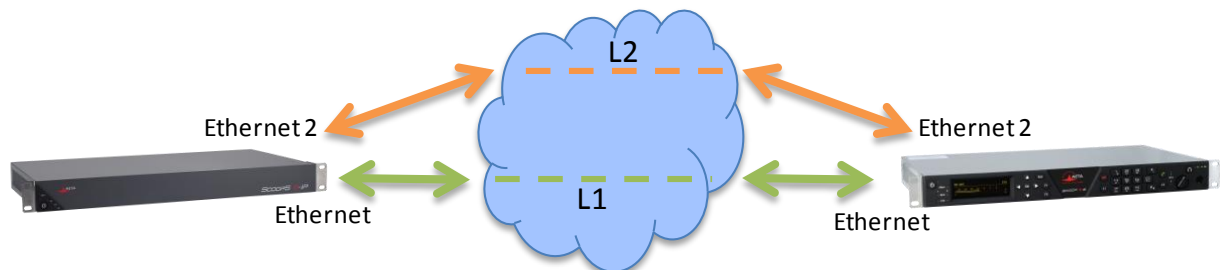
Example 2: Wired link, secured end-to-end by a secondary link

In the above example, the secondary link may for instance rely on cheap ADSL connections for providing high-grade availability at a moderate cost.

6.3.3. Setting up a link with Double Streaming

Pre-requisites

Each codec involved must have:

- Recent firmware, supporting the "Double Streaming" feature. *The feature must be enabled on both codecs.*

- Two network interfaces (depending on the unit: Ethernet, auxiliary Ethernet, integrated mobile, USB mobile device, Wi-Fi…). However it is possible on one end to use the same interface for both connections, like for instance in *Case 1* above.

- For links with a SIP server, one or two accounts on a SIP server (same server for both accounts). The first account is assigned to the main link, the second one is for the secondary link (it is optional if both links use the same interface, like on the non secured end in *Case 1* above). *It is possible not to involve a SIP server, but with same drawbacks as for regular links (i.e. without redundancy).*

Recent[1] AETA codecs are fitted with "factory" SIP accounts on the AETA SIP server, usable for Double Streaming. Other accounts and servers can be used as well.

Preparing the codecs

The following is to be done on each of the codecs involved in the link:

1. **Enable "Double Streaming"**

ScoopTeam, via menu:	**Network > AoIP settings**
Scoopy+ S and Scoop5 S, menu:	**Setup > Network > AoIP Parameters**
All products, html pages:	**NETWORK** tab, **AoIP PARAMETERS** page

   Warning: you must, where appropriate*, disable the "double codec" mode, as it is not compatible with Double Streaming.*

2. **Set up the network interfaces** that will be used (refer to the instructions of the user manual).

3. **Set up the two SIP accounts**, main and/or secondary (except if you do not intend to use a SIP server).

ScoopTeam, via menu:	**Network > AoIP settings > Account 1** / **Account 2**
Scoopy+ S and Scoop5 S, menu:	**Setup > Network > AoIP Parameters**
All products, html pages:	**NETWORK** tab, **AoIP PARAMETERS** page

   For reloading the factory AETA accounts, if needed:

ScoopTeam, via menu:	***Tools > Troubleshooting > Reset**,* "Reload the factory SIP Accounts" option
Scoopy+ S and Scoop5 S, menu:	***Tools > Misc > Reset**,* "Factory SIP Accounts" option
All products, html pages:	*MAINTENANCE **tab**, RESET page,* "Factory SIP Accounts" option

4. **Select the main interface** as for a single link (without redundancy):

ScoopTeam, via menu:	**Network > Network for Outgoing Calls**
Scoopy+ S and Scoop5 S, menu:	**Setup > Network > Change Network**
All products, html pages:	**NETWORK** tab, **CHANGE NETWORK** page

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[1] Scoop5 S, Scoopy+ S, µScoop, ScoopTeam and MultiScoop product ranges, manufactured since May 2017.

5. **Select the secondary interface**:

Note: it is possible to select the same interface as for the main link: end without redundancy in Case 1. For a μScoop or a MultiScoop Codec module this is always the case, because these products only feature one single interface.

Establishing the link

Once the above preparation is performed, normally the situation should be as follows:

- The interfaces used (possibly just one) are connected and active.
- The codec registers on the server[1] with the main SIP account (SIP 1) via the main interface.
- The codec registers on the server with the secondary SIP account (SIP 2) via the secondary interface. *If it is the same as the main interface (Case 1, end without redundancy), this secondary registration is not required.*

Check that the SIP registration indicators are OK:

- On the ScoopTeam display, the secondary indicator is the one on top right.
- On Scoop5 S or Scoopy+ S display, you should see the "S_1" and/or "S_2" indicators.

The unit is now ready for receiving a call in Double Streaming mode.

For initiating a call (launching an outgoing call), the procedure is very similar to a regular call, using the two identifiers of the destination codec: SIP URI or IP addresses.

Call from a ScoopTeam:

Enter the two identifiers, separated by a comma: *dest_1,dest_2*   (example 902942,9029422)

If you enter just one identifier, it is used for both links (in such case only one interface is involved on the remote codec, as in Case 1 on the end without redundancy).

Launch the call as for a regular call.

Call from a Scoopy+ S or a Scoop5 S:

Enter the first identifier and press the call key 🔄. A prompt for entering a second identifier is displayed: enter it and press the call key 🔄 again.

If you press the key without entering a second identifier, the first one is used for both links. In such case only one interface is involved on the remote codec, as in Case 1 on the end without redundancy.

Call from the embedded html CONNEXIONS page:

Enter the two identifiers in the two adequate fields: "Remote Number" and "2nd Remote Number". Click the "Dial" button.

If you leave the second number blank, it is automatically copied from the first one (in such case only one interface is involved on the remote codec, as in Case 1 on the end without redundancy).

What happens then: the codec calls destination 1 via the main interface, and makes a second call to destination 2 via the secondary interface. The link is established and secured.

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[1] Except if a SIP server is not used.

Releasing the link

The operation is just the same as for a regular non redundant link. The codec automatically releases the two redundant connections of the Double Streaming link.

6.3.4. Implementation examples

With single-ended redundancy

sip: 9012342   4G (USB)   L2   Ethernet

sip: 901234   4G Mobile   L1   sip: 905678
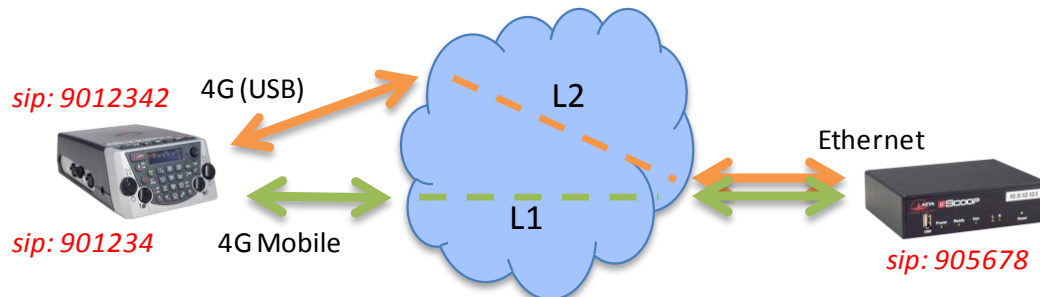
Outside remote, double mobile connection

This example involves:

- A Scoopy+ S on the outside location with 4G network access (integrated module), secured by a second mobile access (4G USB key with a SIM card from a different operator).
- A µScoop on the MCR side, with a high reliability wired Internet connection.

On the Scoopy+ S, two SIP accounts are used, for instance the integrated "factory" accounts.

In the MCR, a SIP account is used, for instance the integrated "factory" account. "Double Streaming" is enabled.

For a **"regular" call from the Scoopy+ S** without redundancy via the main 4G interface:

- Double Streaming disabled.
- Check the SIP 1 registration (integrated 4G interface).
- Call the SIP number of the µScoop: 905678 in this example.

For a **Double Streaming call from the Scoopy+ S**:

- Double Streaming enabled.
- Check the SIP 1 registration (integrated 4G interface) and the SIP 2 registration (4G mobile via USB).
- Call the SIP number of the µScoop: 905678 in this example.

For a **Double Streaming call from the µScoop**:

- Double Streaming enabled on both units.
- Check the SIP registration.
- Call the SIP numbers of the Scoopy+ S: in this example, 901234 and 9012342.

With end-to-end redundancy
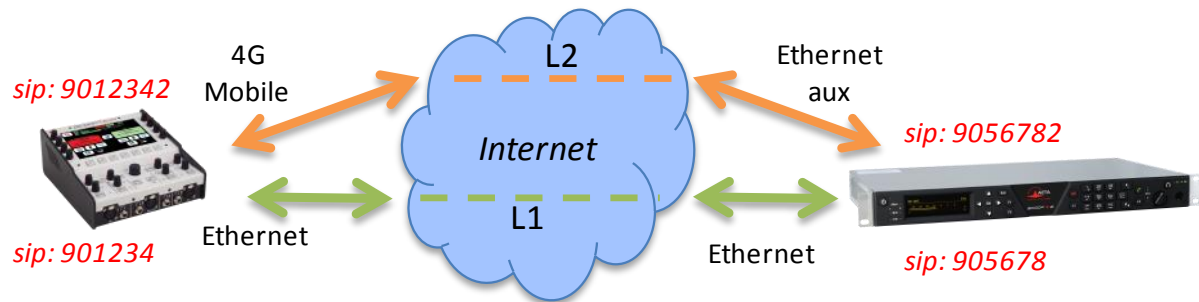
Outside remote, wired connection secured by a mobile link.

This example involves:

- A ScoopTeam on the outside location with 4G network access for securing a main link via Ethernet access.
- A Scoop5 S on the MCR side, with a main wired link, secured by a cheap ADSL connected through a second Ethernet interface (USB adapter).

On the ScoopTeam, two SIP accounts are used, for instance the integrated "factory" accounts.

In the MCR, two SIP accounts are needed, for instance the integrated "factory" account. "Double Streaming" is enabled.

For a **"regular" call from the ScoopTeam** without redundancy via the Ethernet interface:

- Double Streaming disabled.
- Check the SIP 1 registration (Ethernet interface).
- Call the SIP 1 number of the Scoop5 S: 905678 in this example.

For a **Double Streaming call from the ScoopTeam**:

- Double Streaming enabled.
- Check the SIP 1 registration (Ethernet interface) and the SIP 2 registration (4G mobile).
- Call the SIP numbers of the Scoop5 S: 905678,9056782 in this example.

For a **Double Streaming call from the Scoop5 S**:

- Double Streaming enabled on both units.
- Check the SIP 1 and SIP 2 registration.
- Call the SIP numbers of the ScoopTeam: in this example, 901234 and 9012342.

6.3.5. Additional information

Notes about the network settings

AoIP settings

Protocol: only the SIP protocol is supported with Double Streaming.

QoS: it is possible to use packet duplication, like for a regular link. This results in both "temporal" redundancy inside each link (packet duplication) and "spatial" redundancy between the two links (Double Streaming). Of course, the global cumulative bit rate is then quadrupled.

Parameters of the IP interfaces

The **Network Quality** setting, as it affects the size of the reception buffer, can have a significant impact on the performance of Double Streaming.

For correct operation of Double Streaming, this setting must be identical on both interfaces involved. If the settings are different, the codec will automatically apply the higher buffer size of the two for both links (corresponding to the poorer quality).

For better performance in terms of protection against losses, it is advisable to avoid the "High" and "Very High" quality settings, which are normally applicable to transmissions of... very good quality.

In particular, if the latency is very different between the two means of transmission (this can be encountered with networks of very different technology), a low depth buffer cancels the benefit of the redundant stream.

Transmission quality monitoring

Indicators and transmission quality measurements apply to the primary link only. As a result, there may be an apparent inconsistency between these indications and the quality actually observed on the audio program:

- When the transmission is of poor quality on the main link but correct on the secondary link, the quality indications report a poor quality, while the overall audio result remains without defect, thanks to the redundancy.

- On the other hand, if only the secondary link is faulty, the indicators remain on an excellent quality level as long as the main link maintains a good quality.

Interoperability

If a codec configured for Double Streaming receives a regular SIP call from any SIP device: the call is processed normally on the interface corresponding to the identifier that has been called.

Conversely, if a codec configured for Double Streaming calls an unsupported unit, or a unit not configured for Double Streaming: the call fails with rejection.

6.4. Remote access

6.4.1. Remote access: principle

Background

A computer connected on the same LAN as a codec (via Ethernet or Wi-Fi) can easily take control over the unit:
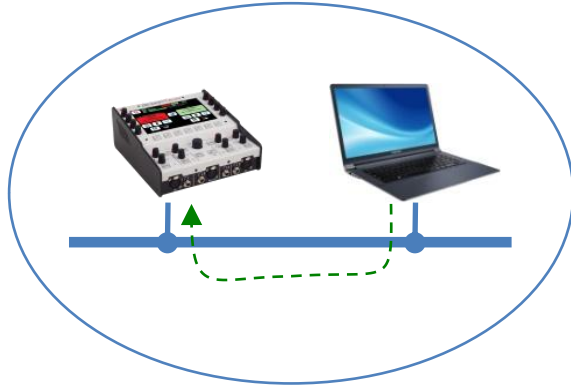
That is a configuration where the computer can access the html for controlling the unit.

When the computer is on a remote location, connected via the Internet, there is most often a need to go through NAT router(s) and/or firewall(s):
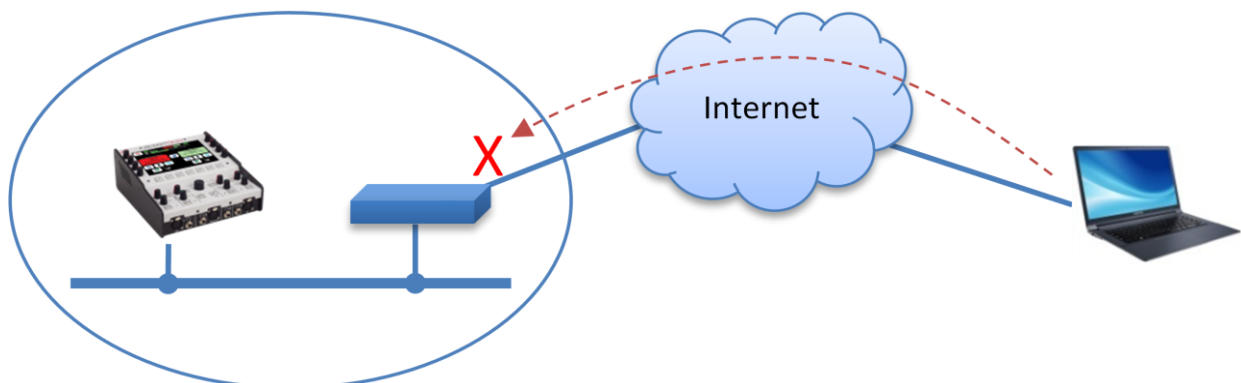
In such situation, it is not possible to get a connection as directly as over a LAN.

The AETA Remote Access service is designed for working around this obstacle and enabling to take control over a device even in such case[1].

---

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

"Remote Access" service

The system relies on using a remote access server, as an intermediary between the codec and the control device. A control session is performed in two phases:
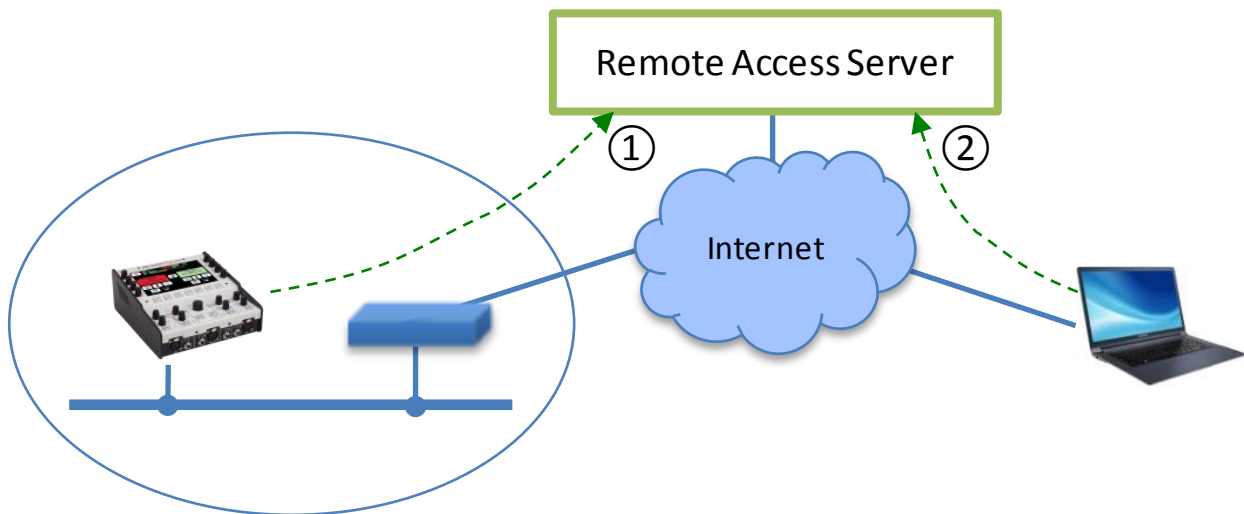
1. The codec connects to the server, and gets available for a possible remote access session.

2. A user who wants to remote control the codec sets a connection to the server, and the server sets a virtual link between the control device and the codec, identical in its operation to a direct link via a LAN.

Both connections to the server are secured and encrypted. The step 1 above is performed when the codec is connected to the network and when remote access is active on the unit. However this can be disabled by the user.

Three types of service are proposed:

- **Remote assistance**: this service allows the AETA support staff to access the codec, for help or possible investigation. *Your explicit agreement is required for such operation.*

- **Remote html access**: provides access to the html pages of the codec from remote. The control device can be any device with a web browser, without the need to install any application. *A "Remote Access" option must be installed on the codec for using this service.*

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

6.4.2. How to use the remote access service

Using the remote assistance

This service allows the AETA support staff to access the codec, for help or possible investigation.

You need on the module (Codec or Control) Internet access via the Ethernet interface.

Step 1: connect the module to the server

MAINTENANCE tab, REMOTE ACCESS page, check "Allow Remote Assistance".

If a "Remote Access" option is installed, check "Allow Remote Access".

Click the "**Save**" button.

Check the successful connection: STATUS tab, GENERAL section (after a while for the connection to be set to the AETA remote access server).

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

Step 2: open the remote session

Tell the AETA technician the MAC address of the codec; this is the identifier for the device (the device type and serial number are also suitable).

AETA can then set a control session over the codec. The link is secure and encrypted.

You may disable remote assistance once the intervention is over. *This is recommended for avoiding useless data traffic.*

Using the remote html access

The remote html access allows to access the html pages of the codec from remote.

What you need on the module (Codec or Control):

- The "Remote access" option.

- Internet access via the Ethernet interface.

The control device can be any device with a web browser, without the need to install any application.

Step 1: connect the module to the server

You first need to connect the module to the AETA server to make it accessible. In order to restrict the access to the module, you can define a "codec password" that will authorize its remote access.

Note: for security reasons, a password that is empty or too short is not accepted by the system (more about this in chapter xx further).
This password has no relationship with the one that possibly controls access to the html pages (see above).

Tab MAINTENANCE, page REMOTE ACCESS, check "Allow Remote Access" and click the "Save "button.

Check the successful connection: STATUS tab, GENERAL section (after some time needed for the connection to be established with the AETA remote access server).

You can also change the password, with following rules:
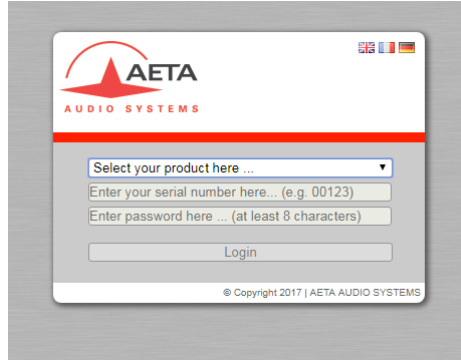
- The password must be at least 8 characters long (see above note and chapter 6.4.2 further).

- If you leave the password blank or enter one with less than 8 characters, a random password is automatically generated by the codec.

Step 2: open the remote session

The module to be controlled must be connected to the server, and you must know the serial number and the "Codec password" of the unit.

Open a web browser, and go to the following URL: https://cloud.aeta.com/rac

You reach the AETA remote html access portal:

1. Select the product type from the drop-down list: MultiScoop/Codec for a "Codec" module, MultiScoop/Control for a "Control" module.
2. Enter the serial number (leading "0" don't care).
3. Enter the codec password.
4. Click the "Login" button.

After a while you get the home page of the codec, almost identical to that you would get with a direct connection over a LAN:
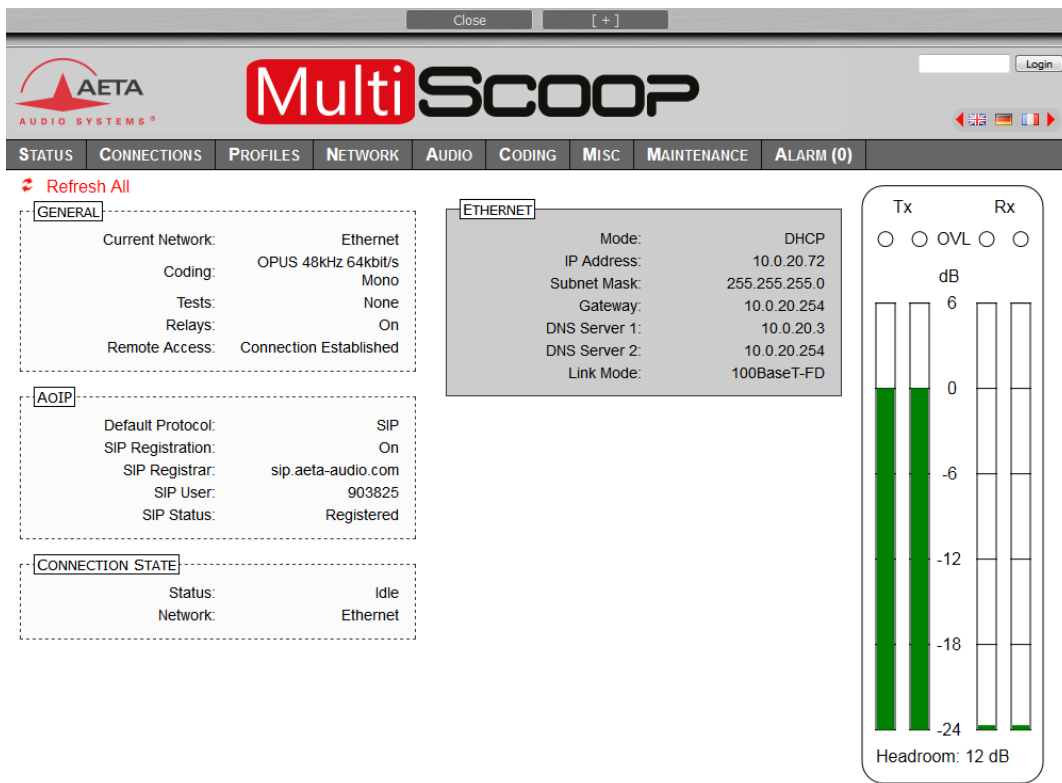
The difference is in the small bar at the top ("Close" and "[+]" buttons). Click "Close" when you wish to disconnect from the server. *"[+]" opens a new tab for connecting to another device, which is then accessed at the same time as the previous one.*

Navigating the pages follow the same rules as a direct access through a LAN.

ⓘ *Warning: in this remote control mode, be aware that any change on the "REMOTE ACCESS" page is prone to make you lose the connection. In the best case (password change), you will have to re-connect on the remote access server (using the new password), but the loss can be definitive if the module can't reconnect successfully to the server. In such case a local intervention would be necessary.*

Do not confuse the "password codec" that you used for this remote connection, and the password possibly configured to restrict the access to the html pages in a general way (i.e. regardless if the access is local or remote). These passwords are independent of each other.

The link is secured and encrypted.

You may disconnect the module once the session is over. *This is recommended for avoiding useless data traffic.*

Security aspects

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

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

Protection measures

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

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

Tips for firewalls

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

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

6.5. Notice regarding open source code

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

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

7. Index

AETA

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