

AUDIO SYSTEMS[®]

superior sound since 1978

www.aeta-audio.com

Getting started

Specifications are subject to change without notice

FRONT PANEL AND INDICATORS



- **1. USB socket** (type A) for connecting a memory key, usable for a recovery configuration.
- 2. Power indicator: green when the unit is powered.
- **3. "Ready" indicator**: this LED goes red at startup time or in case of an alarm, and goes green when the unit is ready for operation. The LED also blinks red if SIP registration is active but fails.
- **4. Decoder status:** this LED is off when idle, green when a link is established, red in case of a sync loss of the decoder.
- 5. Signal presence, left input: green for a normal audio level, orange and then red in case of an overload.
- **6. Signal presence, right input**: green for a normal audio level, orange and then red in case of an overload.
- **7. Reset button**: allows to restart the unit (use a sharp object, such as a pen).
- **8.** Area for marking: available for writing an identification text or sticking a label.

REAR SIDE AND CONNECTIONS

Analog version (µScoop A)



Digital version (µScoop D)



- 9. MAC Address of the Ethernet interface.
- **10. DC power**: 2.1 mm jack for an external power supply 10 to 16 V, 0.5 A maximum.
- **11. Ethernet / PoE** : 10/100Mbit/s Ethernet interface , with "Power over Ethernet" capability.
- **12. GPIO** : RJ11 socket with 2 relays and 2 isolated contacts. Pinout :

Contact	Function	Direction
1	Ground	
	+ GPI 1&2 common	
2	Relays/GPO 1&2 common	Output
3	Relay/GPO 2	Output
4	Relay/GPO 1	Output
5	GPI 2	Input
6	GPI 1	Input

- **13.** Line output, right: 3-pin male XLR, electronically balanced, adjustable level +4 dBu to +22 dBu.
- **14.** Line output, left: 3-pin male XLR, electronically balanced, adjustable level +4 dBu to +22 dBu
- **15.** Line input, right: 3-pin female XLR, electronically balanced, clipping level adjustable +4 dBu to +22 dBu.
- **16.** Line input, left: 3-pin female XLR, electronically balanced, clipping level adjustable +4 dBu to +22 dBu.
- 17. AES/EBU output: stereo output, AES3-2009 format.
- 18. AES/EBU input: stereo input, AES3-2009 format.

INTRODUCTION

We advise you to read this guide first to help you get familiar with the μ Scoop in a short time. For more detailed information, please consult the full user manual on our web site: <u>www.aeta-audio.com</u>

For this guide, we assume that the basic principles are known and that you have already connected audio equipment to the μ Scoop.

 μ Scoop is a full-duplex mono/stereo audio codec, designed for performing audio over IP (AoIP) links via Ethernet access. μ Scoop comes in two versions:

- Analog version **µScoop A**, with two analog audio inputs and two analog audio outputs.
- Digital version µScoop D, with one AES input and one AES output.

Various compression algorithms are available: linear 16/20/24 bits, G711, G722, MPEG Layer 2, AAC including HE-AAC and HE-AAC v2 variations, and above all OPUS coding.

The audio program is transported using the RTP/UDP protocol. The audio link can be set up with three possible protocols:

- SIP (Session Initiation Protocol): this protocol eases the configuration of the codecs to be linked, especially (but not necessarily) when using a SIP server on the network. μScoop is also compliant with the EBU Tech3326 recommendation, also known as « N/ACIP ».
- **"Direct RTP"** mode: basic mode with no signaling. This mode is compatible with devices that do not support SIP or N/ACIP, but it requires dealing directly with all the parameters of the codecs to be linked.
- **Multicast**: usable on private networks that support it, the multicast mode allows distributing efficiently a program towards several destinations on the network.

 $\mu Scoop$ can be integrated in a 19" rack, with a size of 1/3 of a rack unit.

 μ Scoop can be configured and controlled by remote through its Ethernet/IP interface. We describe here the most common management interface, which is the embedded HTML server (aka "web pages").

This guide applies to units with firmware version 1.02 or newer.

SETTING UP

 $\mu Scoop$ can be operated under an ambient temperature of 0° to 45°C (32°F to 113°F).

Installation in a 19" rack

A rack mount tray is available for integrating the units in a rack. Mount 1 to 3 units in the tray, using the provided screws (do not use longer screws!). Then set the assembled tray into a 1U space of the rack.



For a suitable heat dissipation, leave some free space (10 mm at least) at the top and bottom of the assembled trav.

Powering

 μ Scoop can be powered in two ways: either by using an external DC supply, or by connecting to a switch or router capable to provide power through the Ethernet link (PoE, Power over Ethernet).

In the former case, you can use the provided power adapter (GST25B12-P1J), or another suitable DC power supply, plugged on the [10] socket. μ Scoop runs from a nominal 12 V, but it can run from a 10 to 16 V voltage, with a maximum consumption of 0.5 A

In the latter case, the Ethernet connection on the PoE switch/router also provides the unit its power supply. μ Scoop is detected by a PoE source as a class 2 powered device.

It is possible to use both power sources at the same time; μ Scoop draws current from the PoE source as a priority, as long as the voltage of the DC source is less than 13.5 V.

Connection to the IP network

Connect the Ethernet interface (socket [11]) to the network. The LEDs on the socket show the network presence and possible activity.

For operating the μ Scoop, first it must be assigned a valid IP address. Then this address can be used for configuring the device.

Out from the factory, μ Scoop is set for using a DHCP server. Once the unit is connected to the network and switched on, there are two alternatives:

- Either a DHCP is indeed present on the network the µScoop is connected to: in this case the server provides the unit the parameters it must apply.
- Or no DHCP server is available on the network: in this case the μScoop automatically applies a "link-local" IP address, with a 169.254.x.x pattern. This address can be used to initially access the μScoop, and then configure it as needed with parameters more suitable for the network in use.

At this stage, there are several ways to find out the IP address of the $\mu S \text{coop}$:

- The network administrator can program an address reservation for the µScoop on the DHCP server;
- The network administrator can use a tool for identifying/detecting the μScoop (for example by consulting the table of address leases in the DHCP server).
- The AetaScan tool can scan you local area network, enumerate the AETA codecs and display a list with their MAC and IP addresses. You can download it from the AETA web site (www.aeta-audio.com, see the µScoop product page). AetaScan is a Java script Java which runs on every OS.

In every case, the MAC address of the μ Scoop is useful to identify it; it is recalled on the sticker [9] on the rear side of the device.

Access to the management interface, "web pages"

Once the IP address is known, enter it in an HTML browser, on a computer connected to the same LAN as the μ Scoop (all current browsers are suitable). The home page of the μ Scoop then appears:

If the automatically set address is not suitable for the subsequent operation, the first thing to do is to configure "statically" the addressing with suitable parameters provided by the network administrator:

- Click « Login » on the top right corner of the page;
- Select "Network" in the menu bar and click on "Ethernet Parameters";
- Select the "Manual" mode, enter the needed parameters and click "Save";
- At this moment you will lose control because of the IP address change! Enter now in the browser's address field the newly set IP address and check that the change has been performed successfully.
- (i) You can find more details on page 14.

	<u>I</u> Sc			Login
Aux. FUNCTIONS Relays: AoIP Default Protocol: SIP Registration: SIP Registratrar: SIP User:	RLES NETWORK AUDIO	COONG MISC MAINTENANCE	DHCP 10 0 20 38 255 255 255 0 10 0 20 254 10 0 20 3 100BaseT-FD	dBFS 0 -6 -12 -12 -18
Connection State Status: Cause: Network:	Released Ethernet			

ESTABLISHING A LINK

Establishing a link is roughly like making a phone call: a "caller" device initiates a link by soliciting a "called" unit.

Switching µScoop on/off

The unit starts up as soon as a power source (whichever it is) is connected.

To switch off the unit you must disconnect both power sources (external DC and PoE).

Open the management interface

On a computer connected to the same LAN as the $\mu Scoop$, enter its IP address in an HTML browser, as shown on the previous page. If necessary, click the flag corresponding to the language you prefer for the interface.

Click "Login" on the top right corner to access the device configuration. With factory settings or after a full reset, the password for this access is empty. Otherwise enter here the password.

Select the coding algorithm

If you want the unit to <u>receive</u> a call (or connection request) with the SIP protocol, you can skip this step, and go directly to "Preparation: for the SIP protocol") further in this guide.

Click the "CODING" tab: the displayed page includes an "ALGORITHM" frame. Use the drop down lists and set up the μ Scoop as needed. Lastly click on "Save".

(1) For details about the settings on this page, refer further to "Coding" tab, page 16.

Preparation: for the SIP protocol

On the menu bar select "Network" and click on "AoIP Parameters".

In the "PROTOCOL" frame, as default protocol select "SIP". Lastly click "Save".

If you want the $\mu Scoop$ to be able to receive calls with the SIP protocol, make sure that "SIP Accept Calls" is checked in the "SIP" frame.

If you use a SIP server for the links: enter in the "SIP" frame the data of the SIP account allocated to the unit, and check the "SIP registration" checkbox. Lastly click "Save" at the bottom of the "SIP" frame. Check on the home page ("STATUS" tab) that the μ Scoop is successfully registered on the server. In addition the "Ready" LED [3] blinks red if the SIP registration fails.



For a link over the Internet via an access router with NAT, it is recommended to use a STUN server. In the "STUN" frame, enter the address of a public STUN server¹ (IP address or domain name), and check the "STUN Mode" box.

Verification: you can see on the home page new information "Public IP" and "NAT Type", as detected thanks to STUN.

 For details about the settings on this page "AoIP Parameters", refer further to "AoIP Parameters" page ("Network" tab), p. 13.

¹ You can for instance use the AETA server : stun.aeta-audio.com

Preparation: for using the "Direct RTP" mode

On the menu bar select "Network" and click on "AoIP Parameters".

In the "PROTOCOL" frame, as default protocol select "Direct RTP". Click "Save".

If you want the μ Scoop to be able to <u>receive calls</u> in this mode, make sure that "Accept Calls" is checked in the "DIRECT RTP" frame. *Read* (or edit) the RTP port number in this frame, as the "caller" unit must specify this port in order to establish a link.

(1) For details about the settings on this page "AOIP PARAMETERS", refer further to page 13.

Preparation: for a multicast link

On the menu bar select "NETWORK" and click on "AOIP PARAMETERS".

In the "PROTOCOL" frame, as default protocol select "Multicast". Click "Save".

Multicast is essentially a unidirectional protocol, where a source/transmitter device sends a media stream to several receivers. In the "MULTICAST" frame, select the "Multicast Mode" as needed:

- 1. "TX" so that the unit is <u>sender</u> of an audio stream towards a multicast group.
- 2. "RX" so that the unit can be a <u>receiver</u> of a multicast audio stream.

Click "Save".

() For details about the settings on this page "AOIP PARAMETERS", refer further to page 13.

Make a call / establish a link

Click the "CONNECTIONS" tab: the "CONNECTION STATE" frame recalls the preselected coding configuration. Enter in the "Remote Number" field the call destination: either the IP address (numeric or URL) of the remote codec, or its SIP identifier (URI).

For an IP address, if a port number must be specified (other than the default value), add ":" and the port number, as in the example: 192.168.1.35:9000

Click "Dial" to trigger the call. The "Status" field monitors the call progress and the establishment of the link.

The "Dec" LED [4] on the front panel goes green when the decoder is synchronized. The quality meters on the html page monitor the quality in real time in both directions (in the transmit direction, this depends on the capability of the remote codec).

① Specificity of a multicast link: the address you enter in the Number field is a <u>multicast group address</u>. In addition, you must launch a "call" on each device that must receive the multicast program.

Receive a call (SIP or direct RTP)

You have nothing special to do; the link is initiated by the "caller" codec. The μ Scoop which receives the call automatically "unhooks" and establishes the link. In the case of the SIP protocol, it directly negotiates the coding parameters with the remote caller unit.

 In the Direct RTP case, the link will fail if both devices are not configured the same way (audio coding, mono/stereo mode, bit rate...).

The "CONNECTION STATE" frame of the home page ("STATUS" tab) monitors the call progress and the establishment of the link. The "Dec" LED [4] on the front panel goes green when the decoder is

synchronized. The quality meters on the html page monitor the quality in real time in both directions (in the transmit direction, this depends on the capability of the remote codec).

Adjust the bit rate

If the link is set using the Opus coding and the SIP protocol, you can adjust the transmission bit rate during an audio connection.

During such connection, a "CONNECTION PARAMETERS" frame appears in the "CONNECTIONS" tab. It recalls the current transmission bit rate. You can change it on the fly, without any dropout or undesired noise in the transmitted signal.

(1) Click the "Send" button to actually apply the new bit rate. If the link is set with another AETA product, the bit rate also changes on the stream received from the remote codec.

Hang up / release a link

Click on the « Release » button ("CONNECTION STATE" frame of the "CONNECTIONS" tab).

Redial

To recall easily the last destination called, in the "CONNECTIONS" tab click the arrow at the right of the "Remote Number" field and select the number on top of the list. Click "Dial" to launch the call.

 In a similar way, you can see a list of the last calls and pick one for a quick recall.

Use the directory (Call profiles)

If call profiles are recorded in the $\mu\text{Scoop},$ you can use them to establish a link quickly.

in the "CONNECTIONS" tab, select the desired call profile in the "CALL PROFILES" frame: the corresponding coding configuration is displayed

in the "CONNECTION STATE" frame. Click on the "Dial" button to trigger the link immediately.

(1) However, make sure that the default protocol (SIP, Direct RTP or Multicast) is suitable for the called destination.

To create a call profile, select "PROFILES" in the menu bar and click "CALL PROFILES". In the "PARAMETERS" frame edit the adequate parameters for the link with the remote unit, including the IP address or identifier on a SIP server. Click "Create new" and enter a name for this new profile.

(1) You can also edit call profiles, or import/export profiles from/to your computer: see further on page 11 and page 20.

PRESENTATION OF THE MANAGEMENT INTERFACE PAGES

Home page "STATUS"

This page is always accessible unconditionally (no login needed), and provides an overview of the µScoop status.

AETA	^µ Scc			
US CONNECTIONS F	PROFILES NETWORK AUDIO	Coding Misc Maintenance		
Refresh All				dBFS
ENERAL		ETHERNET		
Current Network:	Ethernet	Mode:	DHCP	
Public IP:	82.232.194.202	IP Address:	10.0.20.18	
NAT Type:	Port Restricted	Subnet Mask:	255.255.255.0	
Coding:	OPUS 48kHz 96kbit/s	Gateway:	10.0.20.254	
ooung.	Stereo	DNS Server 1:	10.0.20.3	
		DNS Server 2:		
IX. FUNCTIONS		Link Mode:	100BaseT-FD	
Relays:	Off			└ - - -12 -
Default Protocol: SIP Registration: SIP Registrar: SIP User: SIP Status:	SIP On 10.0.20.40 1006 Registered			24
DNNECTION STATE	,			
Status	Established			
Network:	Ethernet			
1. Remote Address:	1005			TX F
Coding Algorithm (Tx):	OPUS 48kHz 96kbit/s Stereo			
Coding Algorithm (Rx):	OPUS 48kHz 96kbit/s Stereo			
Transmission Quality:	98%			
Reception Quality:	87%			

() Note: some information data visible on this screen copy are available depending on the context.

"CONNECTIONS" page

					Logout
AUDIO SYSTEMS*	μSc	C			<₩ ■ □ >
STATUS CONNECTIONS	PROFILES NETWORK	Audio	Coding Misc Mainte	INANCE	
Profi2 STUDIO3	Deselect		CONNECTION STATE	Established Ethernet 10.0.20.15 OPUS 48kHz 96kbil/s Stereo OPUS 48kHz 96kbil/s Stereo 98% 87% Release	Connections CALL PROFILES You can select one of the call profiles and click the "Dial" button to set up a connection. Manage Profiles,
-	only shows up US coding	→	Connection Parameters Bit Rate	96 kBit/s •	

"CALL PROFILES" page ("PROFILES" tab)

AUDIO SYSTEMS*	μ	50	C		2			Logout
STATUS CONNECTIONS	PROFILES	Network	Audio		Misc Mai	NTENANCE		
Call Profiles 💈				PARAMETERS				CALL PROFILES
Prof12		*			Network:	Ethernet	۲	
STUDIO3					Coding:	OPUS	T	You can manage the call profiles used for setting up
				Char	nnel Mode:	Stereo	•	outgoing calls. Click on a pro
				Sam	pling Rate:	48 kHz	Ŧ	and you can read its corresponding settings. You
					Bit Rate:	96 kBit/s	•	can create/edit/delete
				Remot	e Number:	10.0.20.15		Profiles, presets and snapsho can be imported/exported
		-		Up	date	Create new		from/to your computer. Go t Maintenance/Setup transfer.
	Del	ete						

AUDIO	SYSTEMS®	_	50					 Image: A state of the state of
S TATUS	CONNECTIONS	PROFILES	NETWORK	Audio	CODING	Misc	Maintenance	
		SIP 1	5008			*		PRESETS A preset includes the s for the local network a conditions. You can sa current settings in a n preset, or load an exis preset for a quick setu Profiles, presets and sr can be imported/expor from/to your compute Maintenance/Setup tr

Some use cases for the Presets:

"PRESETS" page ("PROFILES" tab)

- Using more than one SIP server and/or account: the preset allows to switch quickly and without error from one account to another.
- Changing the usage context (with or without the SIP protocol, for example).

"AoIP Parameters" page ("Network " tab)

Notes about parameters on this page:

STUN: using STUN is recommended for a SIP link via an access router with NAT. The public address and type of NAT are detected and displayed on the STATUS home page (see p. 10).

Default Protocol: the one used for an outgoing call.

SIP-Registrar: IP address or domain name. To specify a port other than the standard 5060, add ":" and port number, as in the following example: sip.aeta-audio.com:5070.

SIP Port, RTP Port (*SIP*): these are the <u>local</u> ports of the codec.

Keepalive Interval (SIP): reduce if necessary to be lower than the translation time out in the NAT router.

Send Only (Direct RTP): if this box is not checked, the codec is full-duplex, and expects to receive a stream during a link. If not receiving data, it releases the link after the "RTP Timeout" period. If the box is checked, the codec sends data but does not expect any data in return.

RTP Port (Direct RTP): must not be same as the **SIP Port**. This is the <u>local</u> port; the same number is used for the destination of a call, unless it is explicitly specified in the call destination.

								Logout
AUDIO SYSTEMS*	μ	50	C		1			
STATUS CONNECTION	NS PROFILES	Network	Audio	Coding Misc	Maint	ENANCE		
STUN 💈	de: 🗹			SIP 💈	ion:	ø		AoIP Parameters
STUN Serv		om		SIP Accept C SIP U	ano.	✓		STUN MODE Allows auto discovery of actual
PROTOCOL				SIP Display Na SIP Regist	trar:	microScoop_1 10.0.20.40		public addressing.
Default Protoc			•	SIP Auth U SIP Auth Passw SIP Outbo	ord:	1006		You can use stun.aeta- audio.com for example.
	S	ave			oxy:			SIP REGISTRATION
Multicast 2	de: TX		•	Keepalive Inter Min. Regist. Per	iod:	15 s 300 s	T	Uncheck to disable registration without having to clear the SIP parameters.
Multicast Ctrl Pe Multicast Audio Pe	ort: 6000			SIP F RTP F	0.11.	5060 5004		KEEPALIVE INTERVAL Renew registration at specified
Multicast T						Save		period.
	S	ave		-Direct RTP 💈				SIP AND RTP PORTS These are the port numbers for the codec itself.
;QoS 💈			;	Accept C Send O	ano.			
RTP DS0	CP: 0			RTP F		9000		If the network supports
Packet Replicati		ave	T	RTP Time		30 s Save	•	DiffServ, you can set here the DSCP value (decimal number) assigned to the audio stream transmitted by the codec.
								PACKET REPLICATION

0:normal mode without replication / 1:packet replication with short delay / 2:packet replication with interleaving.

"ETHERNET PARAMETERS" page ("NETWORK " tab)

	007	Logout 🏼 🗮 💷 📦
STATUS CONNECTIONS PROFILES NETWORK AUG	DIO CODING MISC MAINTENANCE	
:Етнеклет 🕏		
Mode:	DHCP	Ethernet Parameters
IP Address:	10.0.20.30	IP NETWORK QUALITY
Subnet Mask:	255.255.255.0	Expected quality of the
Gateway:	10.0.20.254	transmission via IP. The unit
DNS Server 1:	10.0.20.3	switches to internal settings suitable for this guality level.
DNS Server 2:		For instance, with the "Low"
Link Mode:	Auto-Negotiation 🔻	setting, the codec sets a large reception buffer, in order to
IP Network Quality:	High 🔹	stand high jitter.
PPPoE Username:		
PPPoE Password:		
PPPoE Service		
Name:		
MAC Address:	00:15:97:00:0a:01	
	Save	
	<u> </u>	

() Note: you can read on this page the MAC address of the μ Scoop.



Handle carefully the IP addressing parameters, because a mistake there can set the $\mu Scoop$ out of reach.

In such event, see further « Reset the μ Scoop without Ethernet », page 22.

"AUDIO" page or "SNAPSHOTS" ("PROFILES" tab)

(i) This page is not available on the μ Scoop D.

Notes about the parameters on this page:

AUDIO INPUT: the selected value is the input <u>clipping</u> <u>level</u>; a larger value corresponds to a lower input gain, and vice versa.

AUDIO OUTPUT: the selected value is the maximum output level; this level increases with the output gain.

The **SNAPSHOTS** allow to memorize alternate settings for the audio inputs/outputs.

	OGOUT
STATUS CONNECTIONS PROFILES NETWORK AUDIO CODING MISC MAINTENANCE	
Auno Ourpur Concelle Load	dBFS 0 -6 -12 -12 -18 -24 -30 Tx Rx

"CODING" page	e									
		AETA	μ	50	C		7			
	S TATUS	CONNECTIONS	PROFILES	Network	Audio	CODING	Misc	MAINTENANCE		
	ALGOR	тнм 💈				- Aux. Fun	CTIONS 💈	•		
		Algorithm:	OPUS		•		Re	lays:		Check to activate the
		Channel Mode:	Stereo		•				Save	"relay transmission"
		Sampling Rate:	48 kHz		Ŧ			3	bave	function
		Bit Rate:	96 kBit/s		•					Junetion
			Sa							



Notes about the parameters on this page:

NTP Server: useful for time-stamping the events in the log.

Tab Title: you can enter here a label that will appear on the tab of the html browser. This helps identifying the unit among several devices controlled via many tabs in the browser. You should refresh the display to actually apply the label change.

AUDIO SYSTEMS*	μSc				Logout 🕂 🖼 💷 🚺
-GPIO CONNECTIONS	PROFILES NETWORK		NG MISC MAI REDIAL 2		Misc
Input 1: Input 2: Relay 1: Relay 2:	Info 2 Line Status Sync Status Save	•	Redial Attempts: Time Before Dial:	5 • • • • • • • • • • • • • • • • • • •	GPIO Isolated optocoupler/relay inputs/outputs. Each GPI inpu either can be allocated to the transmission of an "Info" stat ("Relay transmission" auxiliar
NTP 2	pool.ntp.org	•			function), or on a pulse it can redial the last call, or release the current link. Each GPO output can be allocated to th indication on a received "Info ("Relay transmission" auxilia function), or the line status, the connection sync status.
TAB TITLE 2 Tab Title:	microScoop_1				NTP SERVER Settings take effect after Reboot.
	Save				

"LOGIN DATA" page ("MAINTENANCE" tab)

	μ	S	C					Locout
STATUS CONNECT	IONS PROFILES	Network	Audio	CODING	Misc	MAINTENANCE		
ENTER NEW PASSWO	0RD						-	
Old Pass	word:							Login data
New Pass	word:							PASSWORD
New Pass	word:							You can change the password
	S	ave						to login to the web interface here. You have to enter the old password, and twice the new password, for confirmation.

"SYSTEM UPDATE" page ("MAINTENANCE" tab)





	AETA	μ	50	C					Losout
STATUS	CONNECTIONS	PROFILES	Network	Αυdio	CODING	Misc	Maintenance		
Pre F	set Snapshots: Reset ssets+CallProf: Reset Settings: Factory Reset:	Reso	et		Reboot		R	eboot	Reset FACTORY RESET Unlike partial resets, this function deletes all settings and user files and restores the default factory settings. Restart the unit (similar to off/on sequence).

"EVENT LOG" page ("MAINTENANCE" tab)

		μSc	
STATUS CONNEC	_	ROFILES NETWORK	Audio Coding Mesc Maintenance
.og Level: Norma	al	 Syslog Ser 	Apply
fime	Type	Module	Message
Jan 4 03:03:02	notice	core	OUTGOING CALL on eth.0
Jan 4 03:03:04	notice	core	CALL ESTABLISHED on Ethernet with Tx: OPUS 48kHz 96kbit\/s Stereo Rx: OPUS 48kHz 96kbit\/s Stereo
Jan 4 03:05:34	notice	core	CALL RELEASED on Ethernet (Normal) by local
Jan 4 03:05:34	notice	core	Rx Quality Summary [General]: duration:00:02:26 , bytes total:1760601
Jan 4 03:05:34	notice	core	Rx Quality Summary [Jitter]: min:0 , max:1 , avg:1
Van 4 03:05:34	notice	core	<pre>Rx Quality Summary [Packets]: total:7311 , discarded:0 , duplicated:0 , lost:0 , reordered:0</pre>
Jan 4 03:05:34	notice	core	Tx Quality Summary [General]: duration:00:02:26 , bytes total:1764361
Jan 4 03:05:34	notice	core	Tx Quality Summary [Jitter]: min:0 , max:1 , avg:1
Jan 4 03:05:34	notice	core	<pre>Tx Quality Summary [Packets]: total:7321 , discarded:0 , duplicated:0 , lost:0 , reordered:0</pre>
Jan 4 03:09:19	err	core	Failed to request all parameters from acip : 200
an 4 03:09:19	err	core	Failed to request all parameters from eth.0 : 200
Jan 4 03:10:17	err	core	Failed to request all parameters from acip : 200
Jan 4 03:10:17	err	core	Failed to request all parameters from eth.0 : 200
Jan 4 03:25:31	notice	core	OUTGOING CALL on eth.0
Jan 4 03:25:33	notice	core	CALL ESTABLISHED on Ethernet with Tx: OPUS 48kHz 96kbit\/s Stereo Rx: OPUS 48kHz 96kbit\/s Stereo
Jan 4 03:28:38	err	core	Change requests to network controllers are not allowed
Jan 4 03:29:05	err	dsp_controller	Communication not synchronized for at least 30 seconds
Jan 4 03:29:05	err	core	Timeout during connection: release call $\backslash/$ stop connection
Jan 4 03:29:05	notice	core	CALL RELEASED on Ethernet (Timeout) by local
Jan 4 03:29:05	notice	core	Rx Quality Summary [General]: duration:00:03:31 , bytes total:2197938

One application for the event log is investigating in case of operation issues. For such use:

- Select the "debug" log level;
- Execute the scenario that triggers or shows up the issue;
- Click "Save Log File as..." and save the log into a file on the computer; this file can be transmitted to AETA along with accurate information about the reported issue.

"SETUP TRANSFER" page ("MAINTENANCE" tab)

AUDIO SYSTEMS*	μS	200			Logout
STATUS CONNECTION	IS PROFILES NETWORK	AUDIO CODING I	Misc Maintenance		
EXPORT CONFIGURATIC Call Profil Press Snapshc Settin	es:	File: Chois	IGURATION	Update	Setup Transfer EXPORT CONFIGURATION Select the data categories to be exported and click "export" to save the data into a file on the computer.
					IMPORT CONFIGURATION Click the button to import a configuration file previously saved on the computer.

"EXPORT CONFIGURATION" allows saving in a file on the computer the data categories which are selected (checkboxes). You can then use this file either for later restoring these settings into the μ Scoop, or for copying them onto other units.

Conversely, "IMPORT CONFIGURATION" allows to set up the µScoop by using the data included in the file on the computer.

 <u>All the included data categories</u> are used and will overwrite (replace without keeping the pre-existing data) the corresponding categories of the μScoop.

НОЖ ТО...

Transmit "relays" to the remote unit

It is possible to transmit/receive two Boolean signals to/from the remote codec. To do that you should activate the feature on <u>both</u> <u>units</u> on the link:

- On the "CODING" tab, check the "Relays" box.
- On the "Misc" tab, select for the inputs/outputs the choice "Info 1" (resp. "Info 2").
- You may, if you wish, assign only one signal, and keep the other input/output of the µScoop for other uses.

Increase the robustness of the transmission

- To increase the resilience to packet losses, activate the packet duplication: page "AoIP PARAMETERS", frame "QoS", "Packet Replication" parameter. Mode 2 provides the highest protection. Be aware that this setting influences the transmission (you should apply it on the remote unit for an effect on the reception).
- To stand a high jitter level at the <u>reception</u>, change the "IP Network Quality" setting (page "ETHERNET PARAMETERS"). When this quality is declared bad, the reception buffer size is increased for a higher tolerance to jitter (but traded for a latency increase).

Secure a permanent link between two codecs

For this, use the auto redial feature of the μ Scoop: if for any reason the link is dropped, the codec re-launches the call and sets the link up again.

- While idle and not connected, activate "Auto Redial" in the "AUTO REDIAL" frame of the "MISC" page.
- Select the number of redial attempts in case of a link loss ("Redial Attempts"), and the waiting time before each attempt ("Time Before Dial").
- Save the settings ("Save" button)
- Launch the call to the remote unit, just like a regular call.

This feature is also effective in case of a power interruption; the μ Scoop recall the remote unit as soon as it restarts.



For stopping permanently a link secured in such way, you must imperatively release it <u>from the unit which has initiated it</u>.

Update the firmware of the µScoop

- You can download from AETA web site the most recent version of the firmware.
- Use the HTML interface to perform the update (see page 17).
- During the update both the "Ready" and "Dec" LED blink red, until the process is complete.
- For more details, refer to the page dedicated to the μScoop on our web site <u>www.aeta-audio.com</u>.

Reset the µScoop without Ethernet connectivity

You may need to bring the unit back to its standard settings (including activating DHCP client) even without being able to control it through a LAN. To do this:

- On the AETA web site, go to the µScoop product page and download the "recovery script".
- Save the file into a USB memory key.
- Plug the memory key on the µScoop.
- Switch on (or reboot) the µScoop: at the end of the startup ("Ready" LED goes green), all settings have come back to their "factory" value (but existing call profiles, presets and snapshots are not deleted).
- Lastly, *remove the USB key*, otherwise this full reset will take place again on the next reboot.

TROUBLESHOOTING

The "Ready" LED stays red permanently

• Check the proper connection of the RJ45/Ethernet cable; check also the LEDs on the RJ45 socket on the rear side of the unit.

The unit is not detected by the AetaScan tool

- Check the proper connection of the RJ45/Ethernet cable, and check that the "Ready" indicator is green.
- Check that the computer is connected to the same LAN as the µScoop (no router which might block the messages).
- There might be a static IP addressing configuration that is not adequate and prevents the connection. In such case you

should apply a recovery procedure: see above "Reset the μ Scoop without Ethernet connectivity".

You cannot access anymore the control interface

- Does µScoop react to a "ping"? Is it detected by AetaScan?
- It may be necessary to apply the recovery procedure as described above ("Reset the µScoop without Ethernet connectivity").

The registration on the SIP server fails

The "SIP Status" does not show "Registered", and the "Ready" LED [3] blinks red:

- Check that the SIP account parameters are entered properly, especially the password.
- Make sure that a firewall is not blocking the SIP protocol.

The link seems to establish, but the decoder does not sync or the link is dropped after a while

- Disable STUN, or conversely enable it.
- Make sure that a firewall is not blocking the protocol or the RTP port.

The audio stream is very disturbed

If the quality meters show a poor transmission quality:

- Use the settings proposed above (page 21, "Increase the robustness of the transmission").
- Decrease the transmission bit rate (this can be efficient when the network access has a limited capacity).

Changing the Opus bit rate during the connection has no effect on the received stream

• This feature is only available when the remote unit is an AETA codec.

The "Transmission Quality" meter does not show up

- During the link you can see the "Reception Quality" meter but the "Transmission Quality" meter is absent or blank: this latter measurement requires the remote device to support the RTCP protocol.
- The transmission quality measurement is not available in multicast mode.
- Check also that the RTCP port is not blocked by a firewall (its number is that of the RTP port + 1).

PORTS USED

Туре	Numéro	Protocole	Utilisation		
ТСР	80	HTTP	Embedded html server		
	7001	Specific	Used by Scoop Manager		
	123	NTP	Requests to a time server		
	514	SYSLOG	SYSLOG server		
	2382	Specific	AETA enumeration		
	3478	STUN	Requests to a STUN server		
	5004 5005	RTP (local) RTCP (local)	Ports of the codec for audio streaming		
UDP	5004 5005	RTP (distant) RTCP (distant)	Remote ports for audio streaming		
	5060	SIP (local)	SIP port of the µScoop		
	5060 SIP (distant)		SIP port of the server or the remote codec		
	9000	RTP (local)	Streaming, "Direct RTP" mode		
	9000	RTP (distant)	Streaming, "Direct RTP" mode		

Bold ports are initial values that can be changed by configuring the codec (local ports), or by explicit indication in the remote address. However:

- Port RTCP = port RTP + 1 in every case
- In "Direct RTP" mode, the local port number is also the default value for the remote port (applied whenever the remote port is not explicit in the remote address).

ACCESSORIES



Spare AC/DC power adapter

AETA test numbers/addresses

IP: 200@82.232.194.202 [1] IP: 15000@sip.aeta-audio.com [2]

Your numbers

Rack mount tray



Blanking plate for rack mount

[1] identifier valid at the date of issue of this document[2] can be called only from a device registered on the AETA SIP server



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