# SCOOPYFLEX



## AETA
### AUDIO SYSTEMS ®
Simple and safe audio links
www.aeta-audio.com

# Getting started

## Panels & controls





1.  **Ready indicator**: The LED turns red when an alarm triggers (PIN request, missing SIM, no network, sync loss…). The nature of the alarm is shown on the display.
2.  **Communication indicator**: This LED turns green when a communication is established and red if an error occurs.
3.  **Rotary Encoder**: Turn to toggle settings on screen and press to select (ie toggle L/R on monitoring).
4.  **Microphone mute button**: Enables/mutes the microphone input. The LED turns red when the microphone input is enabled.
5.  **Rec button**: Start/stop record (Red when recording).
6.  **Headphone volume**: Handles the level of the headphones connected on the ScoopyFlex.
7.  **Microphone potentiometer**: controls the microphone level.
8.  **Microphone level**: 3-segment indicator (-20 dBFS / -12 dBFS / -3 dBFS).
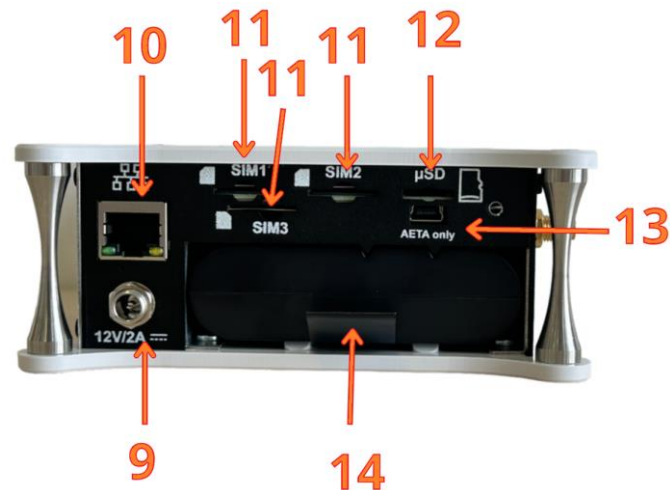
9.  **DC Power**: 2.1 mm jack for external DC power 10 to 16 V, 2A max. It can also be used for charging the battery.
10. **Ethernet**: 10/100Mbit/s Ethernet interface
11. **SIM card slot**
12. **Micro SD Card slot**: used for recording.
13. **CAUTION : Never use it**
14. **Battery slot :** Use only battery referenced by AETA

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# INTRODUCTION

We advise you to read this guide first to help you get familiar with the ScoopyFlex in a before its use.

For this guide, we assume that the basic principles are known and that you have already connected a microphone and a headphone to the unit.

The ScoopyFlex allows you to set up live mono or stereo audio connections in three possible ways:

- Standard telephone or "HD Voice" (7 kHz bandwidth) link over mobile 2G, 3G or 4G networks (Mono only)
- Audio over IP (AoIP) connections over wireless networks: mobile 3G up to 5G networks or WIFI.
- Audio over IP (AoIP) connections over an Ethernet wired access.

In IP mode, (up to 20 kHz bandwidth) link is available with many coding algorithms like OPUS (recommended algorithm).

*AoIP connections are normally based on the SIP protocol. It is possible to use a "Direct RTP" mode without the SIP protocol: for such operation please refer to dedicated application notes.*

This guide applies to units with firmware version 1.01.04 or newer.

1. **Docking Interface**: For the Docking connection.
2. **Aux 1 and 2**: SMA connector for antennas for each internal mobile ("diversity" mode). Please ensure the connection when unit is docked.
3. **USB C**: USB 2.0 Interface for audio or network extension.
4. **USB A**: USB 2.0 Interface for memory or network extension.
5. **Headphone jack:** 3.5 mm stereo jack.
6. **Microphone/line input:** Balanced input (XLR), +20dBu max
7. **Headphone jack:** 6.35 mm stereo jack.
8. **Indicators**: DC LED turns green when right DC plugged in. Battery LED flashes red when charger is on (3modes: idle/slow/fast charge)
9. **Power**: Press 2s to switch on and 3s to switch off.

# Main screen

The "Home" page is the default page for regular operation



1. Status bar
2. Status panel, including
   a. SIP registration status
   b. 6-segment indicators, showing respectively the quality of the transmission and reception on the network
   c. Audio bar graphs (send and receive) for the Stream 1 &2
3. Current state of the Global Purpose "Contacts" and "Info" data
4. Apps launcher: Stream (AoIP connections), audio, recording, editor, settings…
5. Assignment of the virtual pushbuttons on the ScoopyFlex and pushbuttons below the screen on the docking
6. Tools bars

# Apps

Apps available from the « Home page »

Stream Apps: **Stream 1 & Stream 2**
   AoIP application to establish IP connection trough one or two available networks. The Apps support SIP and direct RTP protocols. The audio link can be mono or stereo.
Phone call App: **Voice**
   Phone call application to establish a voice call trough one of 2 internal mobiles. HD voice (7kHz bandwidth) quality mode is supported by all AETA internal mobile. This App is available only with an integrated wireless module.
**Mixer** App:
   Routing application for assigning audio inputs (mic, lines…) to outputs (streamers, headset…).
**Audio** App:
   Application to manage all analog and digital input audio settings. A complete audio monitoring is also available.
**Record** App:
   Application for recording set-up and operations. Storage is possible on many types of memory.
**Editor** App:
   Simple audio editor.
**Media** App
   Manager of recorded files and player with play list.

# SET UP

*ScoopyFlex can be operated under an ambient temperature of 0° to 45° C (32°F to 113°F).*

## Powering

Before switching ScoopyFlex on, you need to have smart battery inserted in the rear compartment, or an external DC power supply connected via the DC jack.

## Battery

ScoopyFlex works with rechargeable Smart Battery Pack Li-Ion 7.20V / 6.90Ah / 49.70Wh. The battery provides power for more than 5 hours of transmission. The battery has an integrated protection circuit which ensures complete operational safety. This includes protection against over temperature, over charge/discharge or excessive current flow situations



⚠ **THERE IS A RISK OF EXPLOSION IF THE BATTERIES ARE OF A WRONG TYPE, OR ARE INSERTED THE WRONG WAY.**

## External power supply *(indoor use)*

The external DC supply is also used to charge the battery. You can use the provided LPS power adapter (GS25B12 or GST25B12), or another adequate LPS DC power source. ScoopyFlex runs from a nominal 12 V, but it can run with a voltage from 10 to 16 V, with a maximum consumption of 2 A.

## Antennas

ScoopyFlex has dedicated internal antennas for each mobile and WIFI module. Most of the time, it is enough to establish network connection. A second antenna is however recommended to optimize the mobile operation. For the best efficiency, place the second antenna as far as possible from the ScoopyFlex unit.

## SIM card

To establish mobile links, you must have a SIM card from a mobile telecom operator. ScoopyFlex accepts standard size SIM cards; use a card adaptor in case of a micro-SIM or nano-SIM card (Contact your dealer for more details).



SIM1 & SIM3: For the mobile module 1 (SIM3 only usable if mobile module 1 is a 5G module)
SIM2: For the mobile module 2

⚠ Switch off the unit before inserting the SIM card

To insert the SIM card into the slot, follow the orientation marked on the front panel (contacts facing down, cut mark forward). As the SIM card holder is a push-push holder, you should push the SIM card as well to extract it.

Warning: for a new SIM card requiring activation (e.g. prepaid SIM card), you can use a mobile phone to perform the activation procedure.

# MAKING A CALL

## Switching ScoopyFlex on / off

To switch the ScoopyFlex on, you should press **on/off key** (**[9]** on page 3) for 2 seconds. After the splash screen, the "login" page is displayed, as shown below. Select the desired access level (Administrator / Expert / Operator) and tap the "Log in" button.
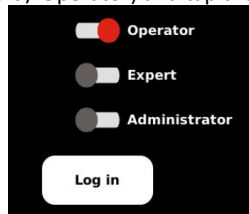


If a password is needed for further access, you must enter it at this time. A "virtual" keyboard is displayed; enter the password and press the "OK" button. The "Home" page is then displayed.

To switch it off, press **on/off key** for 3 seconds; a confirmation message is displayed. Press **OK** to confirm or **Esc** to cancel.

## With AoIP Apps

From the home page, select one of the two stream Apps. From the user interface, you can apply appropriate set-ups to establish an audio link.

At first, you should select one or two networks. The network is operational when highlighted in green (data connection enable). All available networks and their status are present in the status bar.

If you use the SIP protocol (recommended mode), you must set-up a valid SIP account. In double streaming mode (using 2 IP networks for 1 stream), you must select a second SIP account. At this stage, SIP

status should be lighted in green. If it isn't the case, check the network configuration or select another network.

Depending on the desired audio quality, you can select different algorithms. AETA recommends OPUS as most versatile algorithm, enabling many useful functionalities such as auto bit rate.
Before establishing the connection, you need to enter a destination number. This can be done from the memory book, history calls or the virtual keypad.



1.  In the SIP accounts configuration area, tap setting icons to set the desired SIP accounts parameters.

2.  Networks area: tap on each network to select the desired network. For wireless network, data connection must be enabled. Do not forget to set-up needed SIM pin or Wi-Fi password.

3.  "Connection pad" for codec 1: this area is both a call/release connection. Tap the pad when the codec is "ready" (no running connection), and the unit calls the destination that is preselected and displayed in the pad.

If the codec is on line (red), tap the pad and the unit hangs up (after confirmation). The colour of the pad also reflects the state of the connection (light green when no input number is set).

"Book" shortcut button: tap the button to call the menu, and go directly to the **Book** / **Call Profiles** sub-menu. There you can select a call profile and load it. The loaded parameters are then shown in the connection pad, ready for launching an outgoing call.

"Call history" button: tap the button for past called numbers.

"Dial" button: tap the button and the dial keypad pops up. You can edit the last number or enter a new number. Once you validate with OK, the edited number is set inside the connection pad, ready for calling out.

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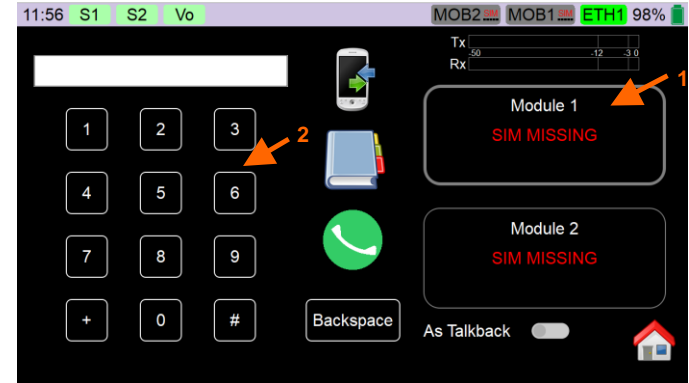
"Network" shortcut button: tap the button to call the menu, and go directly to the **Network** main menu branch.

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

## With Voice Call App

You can use this application to establish a HD voice call trough one of integrated mobile module.



1. Mobile module selection areas: tap the mobile area to select the module to be used for the upcoming call. If the mobile module isn't fully set, a popup appears to enter the SIM pin code.

2. Dial key pad.

"Call" button: Once you have entered or loaded a valid phone number and the selected mobile area is set in green, tap the button to establish the connection.

Same feature as on AoIP Apps

Note: "+" is supported for international call.

## How to get an HD Voice connection

To get an HD Voice connection, some conditions must be met:
- Both devices involved in the connection must support HD voice.
- HD voice must be supported by the mobile network.
- Usually, both units should be connected on the same mobile network (same carrier).
- Most often, 2G networks do not support HD voice. This depends on the country or area, and the network carrier.

For this last reason, the radio technology indication (2G, 3G, 4G, 5G display on the screen) is a useful indication that HD Voice is possible.

If an HD Voice connection is not possible the network operator will automatically establish a narrow band (3 kHz bandwidth) voice connection.

# SET AUDIO INTERFACES

Audio interfaces are managed through two Apps.

The first App, Audio App, manages physical configuration of each input and output.

The second App, Mixer App, manages routing between audio inputs, outputs and streams.

From this application, the user can set audio level for each available input. Unavailable audio interfaces are disabled automatically, i.e. docking extension or AES67 module not present.

To set volume, you need to tap on the control (blue if selected) and adjust the value with the encoder ([3] on page 2).

An audio level meter is present on each input to help you to adjust its volume.

When the lock is open, each microphone gain is controlled by the hardware knob. When the lock is close, the microphone gain is controlled by the virtual knob/encoder.
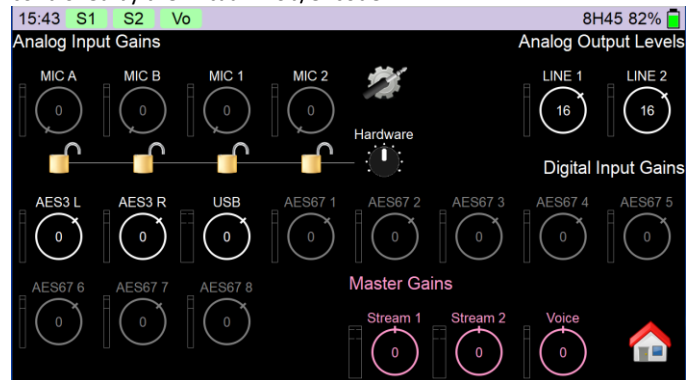


Tap on the settings icon to have access to microphones configuration. Filter, phantom power, limiter and PAD configuration are available from a popup box.
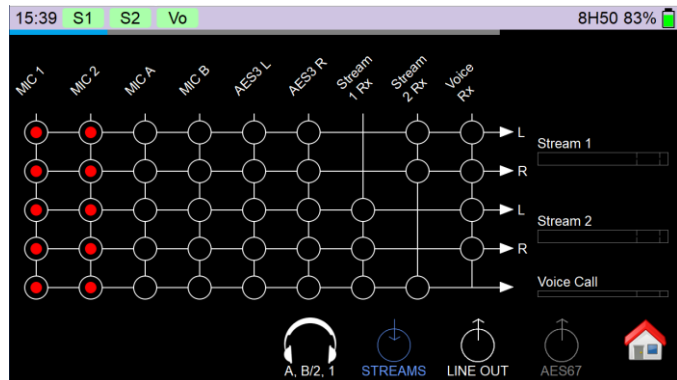
Don't forget to enable the input. Turn ON the mute LED indicator, using the mute button ([4] on page 2) if needed.

## Audio App

# Mixer App

From this application you can route all sources to all destinations. Tap on cross circle to enable or disable the connection.  Don't forget to slide the mixer area to see all inputs.



The mixer is split on 4 pages.

1. **"AES67 mixer"**: route audio inputs and streamer outputs on 8 channels AES67 (Dante or Ravenna) stream. AES67 module should be present in the Docking and enabled in the Audio menu.
2. **"Line Out mixer"**: route audio inputs and streamers on the docking line outputs and USB outputs.
3. **"Streams mixer"**: route audio inputs and stream talkbacks into a streamer.

4. **"Headphone Mixer"**: two versions are offered to the user : a simple "radio" version with balance and "full volume monitoring" version where the gain of all inputs can be adjusted. Both versions follow the mono or stereo configuration of streams.

## *Radio Version*
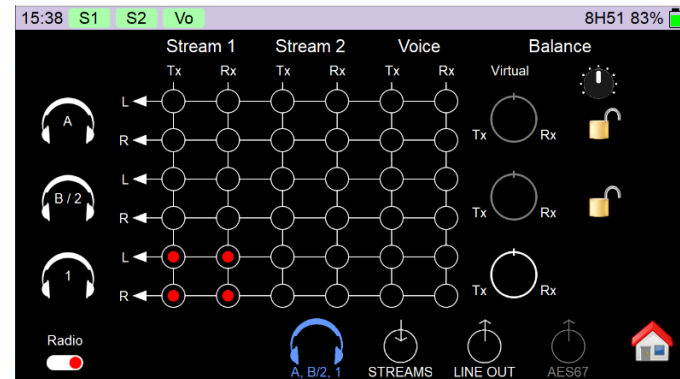


On the radio version, only streams can be monitored. The balance between send (Tx) and receive (Rx) can be adjusted. When the docking is connected, the balance can be adjusted by the physical potentiometer (lock icon shows open). When it is not the case, virtual balances are always accessible.

Tap on the Radio switch to swap to the "full volume monitoring" version.

## *Full volume monitoring Version*



On this version, you have access to separated configuration for the 3 headphones. To avoid double configuration, you can copy the mix of another headphone using appropriate switch *(coming soon)*.

To enable a source on the headphone, tap on its virtual volume control and adjust the level with the encoder. By pressing on the encoder you can toggle the destination of the headphone (Left & Right or mute).
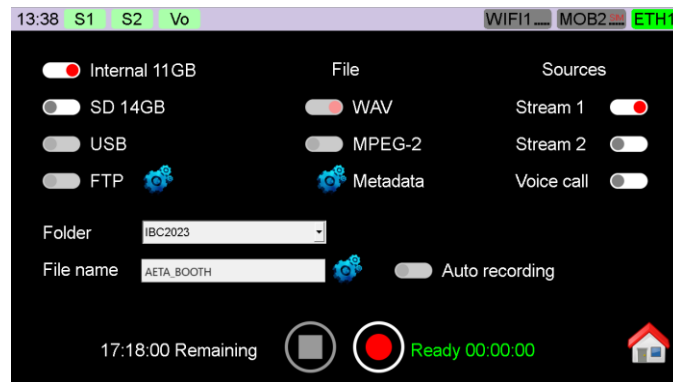
⚠️ Physical balance are disabled in this monitoring configuration

# RECORDING

## Record App

ScoopyFlex has an optional recording App. This application can be used for setting-up and processing recordings. Storage is possible on many types of memory. This app manages recorded sources, format and metadata.



1. "Disks selection": ScoopyFlex has an internal disk with limited space. To increase capability, it is recommended to use a µSD card. Enable FTP to backup automatically records on the configured FTP server *(coming soon)*.
2. "File": Select between the 2 available formats: PCM and MPEG2 *(future improvement)*
3. "Source": Select one of streams available. In Mono, the Tx is recorded on the left audio channel and the Rx on the right channel. In stereo Tx and Rx are mixed together.
4. "Folder": Record directory on disk. Use media App to change it.
5. File Name: set your own record file name.
6. "Meta data": To set user Meta data *(coming soon)*.

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7. "Auto recording": The recorder starts recording automatically when an audio link is established (*Coming soon*).
8. "Recording button": Tap record button to start recording and twice stop button to stop it. Recording time and remaining time near controls are updated during recording.

Note:
- Each record has a unique index.
- You can change the disk between two recording sessions, a new record index will be taken for this disk.
- Start and stop record can be done also through the Rec button ([5] on page 2 *Coming soon)*.

# Media App

This app is mainly a player but it also manages storages and records. You must select a drive and a folder before starting to work.



**Media Management**

Drive configuration
With the Drive combo list, you can select the storage drive. Tap its tools button  to access the following format, repair functions.

Folder configuration
With the Folder combo list, you can select an existing folder from selected drive. Tap tools button to access to create and delete folder, export and import functions (*import coming soon*).

**Records Management**

Records available in the folder are displayed in the records list. Select one of them to apply following actions:
- Play
- Edit
- Edit Meta Data (*Coming soon*)
- Delete
- Share
- Add/Remove to/from the playlist

**Playlist Management**

Records sent to the playlist are sorted chronologically. You can change the list order with the up and down buttons below the list. Select a record from the list to remove it, the record still be visible in the list of records.

Tapping the Play button on the playlist initiates the reading from the first record or the one that is selected. The reading will stop automatically at the end of the list. You can pause, skip or rewind with buttons on the right side of the list.

# Editor App

This application can be used as a simple audio editor. Original record is never deleted.
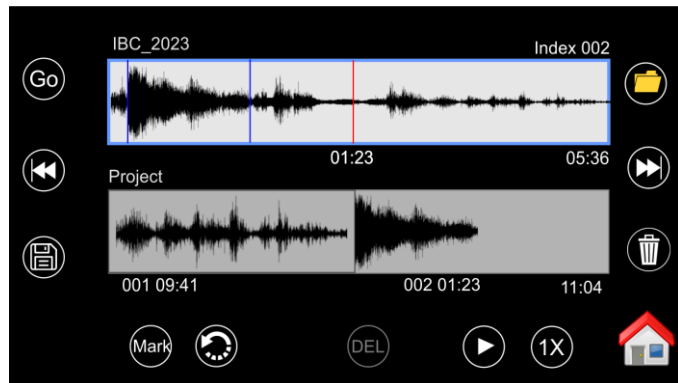


## Record edition

Tap on the displayed record you want to edit, if it isn't yet selected. Turn the encoder to set the red line (current position) to the start of the desired selection and press 'Mark' button to mark the entry. Proceed the same way to mark the end of the selection. You can cancel selection at any time by 'Cancel' button. Set the red line on this selected sample and press 'Go' to transfer this selection to the end of the project display or press 'Play' to listen from current position.

You can extract another part of the same record or open another record from the same folder to complete the project.

## Project editor

Tap on the displayed project, if it isn't yet selected. Turn the encoder to set the red line positions to put marks as previously described. Once done, position the red line between two marks, press DEL to remove the selected part of the project. Press 'Cancel' to delete last mark that has been positioned.

To reset project press the trash button.

## Save work

Press 'Save' button to save the project as a new record file. The file will have an "e" extension standing for "Edition" and it is automatically added to the list of records.

Notes:

-   'Rewind' and 'skip' keys help you to reach positions: start, end, marks.
-   You can change the navigation speed by 4X, 16X using the '1X' key.
-   Remaining Marks in a project will be considered as a stop when played used to stop the player during its final used (live transmission).

# MENU
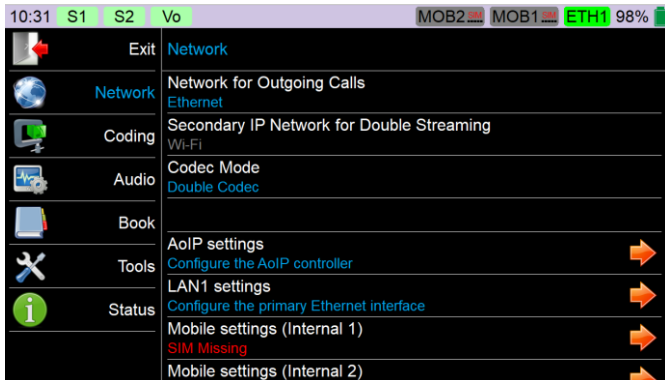


**Network**: Change networks for the streamers; enable the second streamer (double codec); configure the AoIP controller (SIP server, SIP account…), configure each available network interface (Ethernet, mobile and Wi-Fi).

**Coding**: to select the algorithm for each codec and to configure auto bit rate for Opus algorithm.

**Audio**: Set and enable a limiter/compressor on streams. If an AES67 is present, you can set here the configuration of the AES67 module and its audio routing (only on the Ravenna version).

**Book**: Create and manage call profiles, presets, SIP profiles and snapshots. At operator level, only the loading of such presets is possible.

**Tools**: Reboot the system; access level selection; settings of the softkeys; configure the GPIO, Auto redial and NTP server; reset settings and configure event logs; configure remote access service; set up passwords for the ScoopyFlex (admin level only).

# WEB INTERFACE

Once the ScoopyFlex connected to an IP network, the first step is to get its IP address, from the front panel: **About,** or the menu **Network / LAN settings / IP Address.** On the control computer, launch an html browser and enter the IP address of the ScoopyFlex in the "URL" field. This gives access to the html server integrated in the ScoopyFlex, for extended status and control features
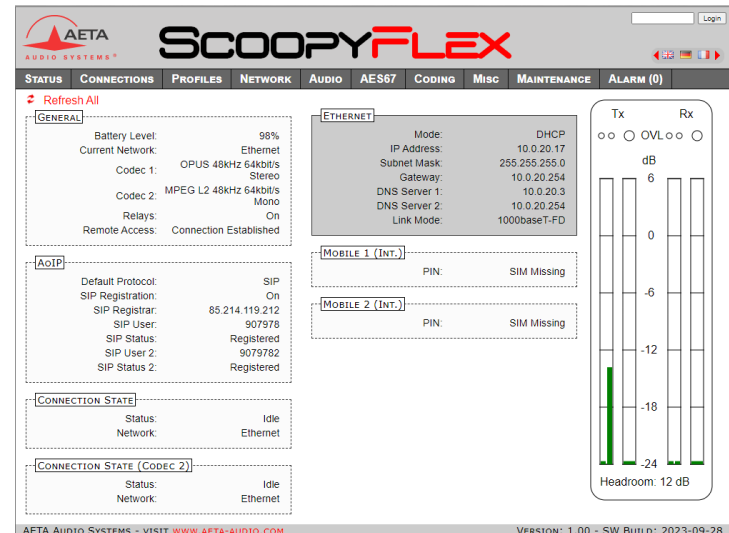
# DOCKING

Docking can be buying with its flight case as an accessory for the ScoopyFlex.

To connect it, slide the ScoopyFlex on the Docking top panel it reaches the docking connector. Don't forget to fix the ScoopyFlex with the provided lock.

⚠️ **THE DOCKING STATION IS NOT HOT PLUG**



Mounting            ScoopyFlex clamp



1. **Input potentiometer**: controls the microphone level.
2. **Headphone volume / Balance**: Handles the level of the headphones connected on the Docking.
3. **Microphone mute button**: Enables/mutes the microphone input. The red LED turns on when the microphone input is enabled.
4. **Talkback button**: switch microphone input and the headphone on the talkback streamer.
5. Programmable key
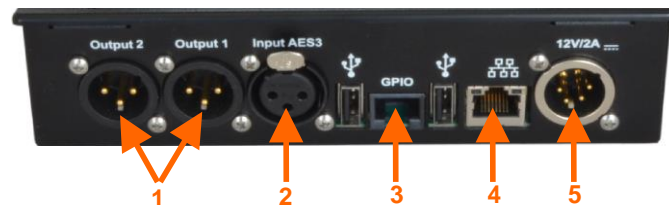


1. Line Output: **:** Balanced output (XLR), +22dBu max
2. AES3 input : Supports only up to 48kHz sample rate
3. **GPIO:** RJ45 socket with 2 relays, 2 contacts and LPS power Pin assignment:

| Contact | Function |
|---------|----------|
| 1 | +5 V / 500 mA |
| 2 | Contact 1 |
| 3 | Contact 2 |
| 4 | Contact common 1&2 |
| 5 | Ground |
| 6 | Relay 1 |
| 7 | Relay 2 |
| 8 | Relay common 1&2 |

4. **Ethernet**: 10/100Mbit/s Ethernet interface

5. **DC Power**: XLR4 connector for external DC power 10 to 16 V, 2A max. It can be used as a redundancy power input for the ScoopyFlex + Docking.



1. **Microphone/line input:** Balanced input (XLR), +20dBu max
2. **Headphone jack:** 6.35 mm stereo jack
3. **SFP:** Second AES67 interface
4. **AES67\* Ethernet**: 1Gbit/s Ethernet interface
5. **Docking interface**: Interface for the ScoopyFlex
Note: 1 & 2 also present on the other side of the Docking.

**AES67**

Optional Ravenna or Dante 8/8 audio channels can be integrated in the Docking. A second Ethernet or optical interface for the SFP port (provided as an accessory) can be added for AES67 redundancy transmission.

Note: A POE option can be added on the Docking Ethernet interface (default) or on the Ethernet AES67 (**on request)**.

# TROUBLESHOOTING

## The status bar ETH1 icon is not green
⇨ Check that the Ethernet cable is connected

## The status bar MOB2 icon is not green
⇨ Check/correct the APN; if needed try to reload the default APN (see **Mobile** menu)

## The SIP icon on the home page is not green
⇨ Change the «Server SIP port» (AETA server)

## The SIP icon is green, but the audio does not go through, or stops after several seconds.
⇨ Enable or disable STUN
⇨ Change the RTP port

## The audio stream is corrupted in mobile IP mode
⇨ Check whether a VoIP service is available at the mobile operator
⇨ If the transmission quality bars are low :
  o Decrease the network quality, Menu **Mobile / Net quality** (also affects the *reception* quality)
  o Enable the packet duplication, Menu **AOIP/Packet repl.** (affects the *send* direction)
  o Enable auto jitter buffer **Network / Jitter buffer mode / adaptative**

## You can't change the bit rate during the connection
⇨ This feature is only available with OPUS, so the remote unit must support this codec.

# ACCESSORIES

**Battery**

**Spare AC/DC adapter**

**Multiband Antennas**

**Carrying bag**

**Docking**

## AETA Test numbers *

IP : 15000@sip.aeta-audio.com
IP : 200@siptest.aeta.com

## APN codes (France) *

ORANGE : orange
BOUYGUES : mmsbouygtel.com
SFR : sl2sfr
FREE : free

_____
_____
_____

## Your numbers

_____
_____
_____

*\* Information valid at the date of issue of this document, and possibly depending on the terms of the subscription to mobile data networks*

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