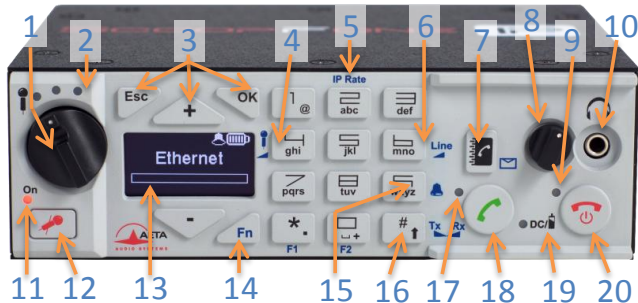# ScoopFone IP

Getting started

FRONT PANEL AND CONTROLS

1. **Microphone potentiometer**: controls the microphone level.

2. **Microphone level**: 3-segment indicator (-20 dBFS / -12 dBFS / -3 dBFS).

3. **Navigation keys**: use the **Esc** key to access the configuration menu. Use the **Up/+** and **Down/-** keys to browse the selections and use the **OK** key to validate menu selections or messages.

4. **Microphone gain**: when **Fn** is enabled, press this key and use **+** key to increase and **-** key to decrease the gain of the microphone.

5. **IP Rate**: while a link with Opus is active, press **Fn**, press this key and use **+** key to increase and **–** key to decrease the bit rate.

6. **Line input gain**: when **Fn** is enabled, press this key and use the **+** and **–** keys to increase/decrease the line input gain; the maximum (clipping) level of the input (from +4 dBu to +16 dBu) is displayed during the adjustment.

7. **Phonebook**: press the **Phonebook** key to access one of the 9 memories or to save the current phone number into a memory.

8. **Headphone volume**: handles the level of the headphone connected on the ScoopFone IP.

9. **Alarm indicator**: the red LED turns on when an alarm triggers (Ethernet disconnected, sync loss…).

10. **Headphone jack:** 3.5 mm stereo jack.

11. **Microphone indicator**: this red LED turns on when the microphone input is enabled.

12. **Microphone button**: enables/mutes the microphone input.

13. **OLED display.**

14. **Function key**: used in combination with other keys, it gives access to functions marked in blue. It is active for 5 seconds. This key is called **Fn** hereafter.

15. **Ringer shortcut**: when **Fn** is enabled, use this key to enable or disable the ringer.

16. **Shift/Balance**: switches the keypad to alphanumeric or numeric mode. When **Fn** is enabled, press this key and use **+** and **-** to adjust the send/receive balance for the headphone monitoring.

17. **Communication indicator**: this green LED turns on when the communication is established.

18. **Call key**: use the **Green phone** key to initiate a call.

19. **Power indicator**: the LED turns red when the battery is charging from an external power supply and green when it is fully charged.

20. **Hang up key**: press the **Red phone** key to release a connection. Hold it for 2 seconds to start or switch off the ScoopFone IP.

REAR PANEL AND CONNECTIONS
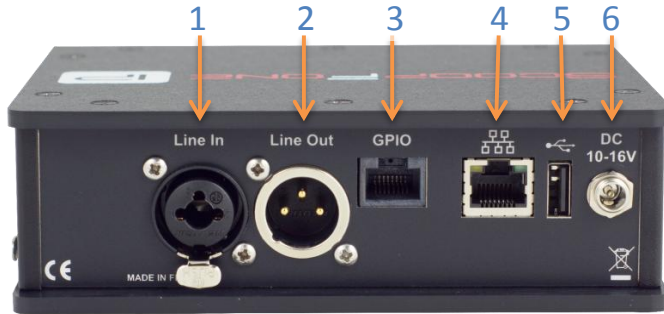
1. **Line input**: combo socket, balanced input (XLR) or unbalanced input (6.35 mm jack), +16 dBu max.

2. **Line output**: balanced output, 3-pin XLR. Level adjustment via menu from +4 dBu to +22 dBu.

3. **GPIO**: RJ45 socket with 2 relays, 2 contacts and power. Pin allocation as follows:

Contact	Function	Direction
1	+5 V / 500 mA	Output
2	Contact / GPI 1	Input
3	Contact / GPI 2	Input
4	Contact/GPI 1&2 common	Input
5	Electrical Ground	
6	Relay/GPO 1	Output
7	Relay/GPO 2	Output
8	Relay/GPO 1&2 common	Output

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

5. **USB**: Type A USB 2.0 Interface for memory import.

6. **DC Power**: 2.1 mm jack for external DC power 10 to 16 V, 1 A max. Can also be used for charging the batteries.

7. **Headphone**: 6.35 mm jack.

8. **Microphone input:** balanced input, gain switchable from 0 to 48 dB by 16 dB steps.

MAIN SCREEN
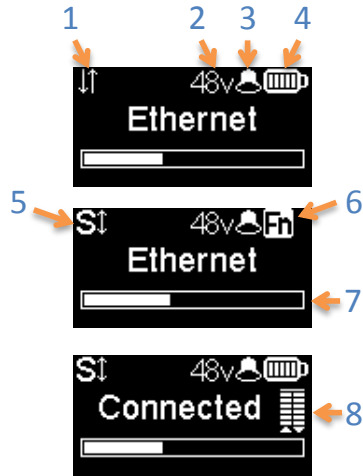
1. **Ethernet connection** active.
2. **Microphone power** indicator: 48V phantom.
3. **Buzzer indicator**.
4. **Battery charge** level: 5-segment display.
5. **SIP registered**: on when the unit is registered on a SIP server.
6. **Function key indicator**: flashing when functions marked in **blue** are enabled by the **Fn** key.
7. **Audio send level**: from -30 dBFS to 0 dBFS.
8. **Transmission quality**: 6-segment indicators, showing respectively the quality of the transmission and reception on the network.

INTRODUCTION

We advise you to read this guide first to help you get familiar with the ScoopFone IP in a short time. For more detailed information, please consult our web site: www.aeta-audio.com

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 ScoopFone IP allows you to set up live mono audio connections over IP (AoIP), via an Ethernet wired access. Up to 20 kHz bandwidth is available with the OPUS algorithm (default algorithm). In case the remote unit does not support this algorithm, the connection will be established automatically with G722 or G711 coding.

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

- **SIP** (*Session Initiation Protocol*) : this protocol eases the configuration of the codecs to be connected, especially (but not only) if using a SIP server on the network. ScoopFone IP is also compliant with the EBU Tech3326 recommendation, also known as "**N/ACIP**". *This is the recommended mode for operating the ScoopFone IP.*
- **"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. *Please refer to the detailed user manual for this operation mode.*

This guide applies to units with firmware version 1.04 or newer.

SETTING UP

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

Powering

Before switching on the ScoopFone IP, you need to have 6 AA batteries inserted in the bottom compartment, or an external DC power supply connected via the DC jack.

Batteries

ScoopFone IP works with either disposable or NiMH batteries. Six 2500 mA.h NiMH batteries provide power for more than 7 hours of transmission.

In order to get an accurate charge level indication with rechargeable batteries, it is necessary, at least once, to leave the ScoopFone IP on with the external DC power until it is fully charged.

⚠️ **CAUTION : THERE IS RISK OF EXPLOSION IF THE BATTERIES ARE OF A WRONG TYPE, OR ARE INSERTED THE WRONG WAY.**
DISPOSE OF USED BATTERIES ACCORDING TO INSTRUCTIONS.

External power supply *(indoor use)*

The external DC supply is also used to charge the batteries. Only NiMH batteries are supported by the internal charger. You can use the provided power adapter (GS25B12 or GST25B12), or another adequate DC power source. ScoopFone IP runs from a nominal 12 V, but it can run with a voltage from 10 to 16 V, with a maximum consumption of 1 A.

⚠️ **IMPORTANT: DO NOT CONNECT AN EXTERNAL DC SOURCE WHEN DISPOSABLE BATTERIES ARE INSIDE THE BATTERY COMPARTMENT**

Connection to the IP network

Connect the Ethernet Interface ([4] on page 3) of the ScoopFone IP to the network. The LEDs on the socket show the network detection and its activity.

MAKING A CALL

Switching ScoopFone IP on / off

To switch the ScoopFone IP on, press ![power button] for 2 seconds.

To turn it off, press ![power button] for 2 s; a confirmation message is displayed. Press **OK** to confirm or **Esc** to cancel.

Adjust the microphone level

Turn ON the mute LED indicator, using the mute key ![mute key] if needed.

Set the **Potentiometer** ([1] on page 2) to its middle position (white dot of the button in the topmost position). Adjust the headphone balance (send/receive): enable the **Fn** mode and press the **#** key, then set the balance to the average position with the **+** and **–** keys.

Make a test with your microphone while watching the level indicators (above the potentiometer). If the level is too high or too low, enable **Fn** mode + key **4** and press key **+** to increase or key **-** to decrease the gain of the microphone, until you get the appropriate level.

From then on, you can use the potentiometer for fine adjustment.

Preparation

Connect the Ethernet Interface ([4] on page 3) of the ScoopFone IP to the network.

ⓘ *The alarm indicator ([9] on page 2) blinks as long as the Ethernet interface is disconnected.*

If you use the ScoopFone IP with a SIP server, you should have the ![SIP icon] icon displayed on the screen.

![warning triangle] For links over the Internet via an access router with NAT, we recommend using a STUN server. To enable it, enter the **IP / Use STUN** menu.

Optional check: the type of NAT detected is displayed in the IP / NAT type menu.

Making a call

Enter directly with the keypad the call destination: IP address, or SIP identifier (URI).

![Display showing "Dial: abc" and "1234abcd"]

Press **UP** or **DOWN** to erase the last digit(s)/character(s).
If needed, press the **#** key to access the alphabetic characters.

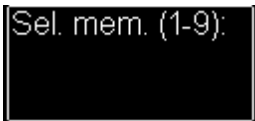Press the ![call key] key to make the call. *Otherwise cancel using Esc or*
![power button] .

Using the phonebook

Enter a number as described above, and press the ![key] key to save this number into one of the 9 memories of the current mode.

To recall a number from the phonebook, press the ![key] key and enter the number of the desired memory from 1 to 9.

![Sel. mem. (1-9):]

The number is displayed and you can:

- Set the call by pressing the ![call key] key.

- Cancel using **Esc** or ![hangup key] .
- Edit the number using **+** or **−**

During the call

To adjust the headphone balance, enable the **Fn** mode and press the **#** key. Adjust the balance as desired using the **+** and **−** keys.

Adjust the transmission bit rate

You can adjust the transmission bit rate during an audio connection[1]. Enable the **Fn** mode + **2 (IP Rate)** key. Using the **Up** and **Down** keys you can increase or decrease the bit rate.

Notes:

- The quality bars on the screen (see [8] on page 4) provide real time information on the transmission quality for both directions (but for the transmit direction, this depends on the remote unit's capability).
- If the connection is set with an AETA product at the other end, the bit rate also changes in the receiving direction, from the remote codec.

Hang up

Press ![hangup key] twice to hang up.

Redial

Press the key ![call key] twice to redial the last called number.

How to turn on/off the ringer

Enable **Fn** mode and press **9** key to change the ringer state.

When the ringer is off, the ringer icon on the display shows muted.

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[1] If the link is set with the Opus coding algorithm.

IMPORTING MEMORIES

You can import a phonebook into the ScoopFone IP using a USB memory stick.

To do this, place a "book-scoopfone.txt" import file on the root of the USB drive and plug it into the ScoopFone IP after it has started.

The import file must have the following format:

Memory number:1,IP address or SIP number or SIP URI

The memory number is a number from 1 to 9.

Example:
1:1,3000045
2:1,15000@sip.aeta-audio.com
5:1,10.0.20.123:5063

ⓘ *Importing a file completely clears the previous phonebook; all the memories are reconfigured or erased.*

MAIN MENU

Audio: menu for configuring the audio interfaces

Ethernet: configuration of the Ethernet interface

IP: configuration of the IP transmission

Tools: user interface configuration, reset settings

About: shows information such as:
- Firmware version number
- SIP account and server address
- Ethernet interface IP address

AUDIO MENU

Microphone phantom power
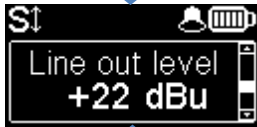
You can enable 48 V phantom power for the microphone.

Selection of the microphone button operation
 Flip-flop: switch the input between enabled and disabled.
 Push to talk: keep the key pressed to enable the microphone input.
 Cough key: keep the key pressed to mute the microphone input.

Line output level adjustment

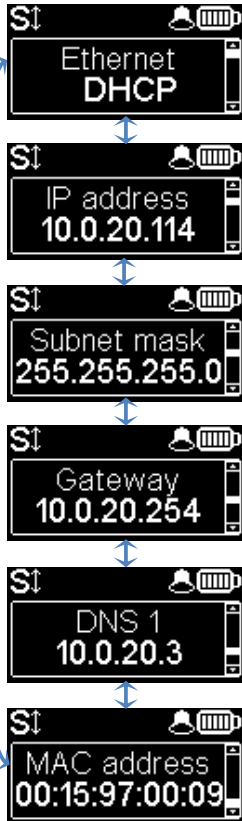The maximum output level can be set from +4 dBu to +22 dBu.

Line output source selection

You can select between the transmit signal (Transmit), the received signal (Receive) and the headphone balance (Balance).

ETHERNET MENU

ⓘ *Changes made using this menu are only effective after exiting the menu (**Esc** key to go back to the upper level).*

Set Ethernet IP address allocation

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

IP address

For static IP addressing, enter here the IP address.

Set network mask

For static IP addressing, set/change here the network mask.

Set Ethernet gateway

For static IP addressing, set/change here the default gateway address.

Set DNS 1

For static IP addressing, set/change here the numeric address of the domain name server.

Display the Ethernet MAC address

Press **OK** to see the complete address.

IP MENU

ⓘ *Changes made using this menu are only effective after exiting the menu (Esc key to go back to the upper level).*

Set the default IP transmission bit rate (OPUS)

Here you can set the preferred bit rate for OPUS coding, from 12 kbit/s to 192 kbit/s.
Note: this is the initial bit rate, when the link is established; afterwards the bit rate can be changed during the transmission if the connection is done with OPUS.
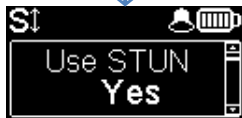
Enable STUN

Using STUN is usually recommended when using a SIP server; however this is not suitable in some cases. As a general rule, enable STUN. If you meet connectivity issues (such as failure to sync when starting the connection, or link loss after a few seconds), disable the function.

Type of NAT (Network Address Translation)

When STUN is used, the ScoopFone IP can detect the type of NAT performed by the router through which it accesses the Internet. *In case the router applies symmetric NAT, it is usually recommended to disable STUN.*

Default AoIP protocol

AoIP connections normally use the SIP protocol. By selecting here "Direct RTP" it is possible to make calls without the SIP protocol. The various parameters of this "Direct RTP" mode are available only via the embedded html pages. *The operation mode is described in the detailed user manual.*
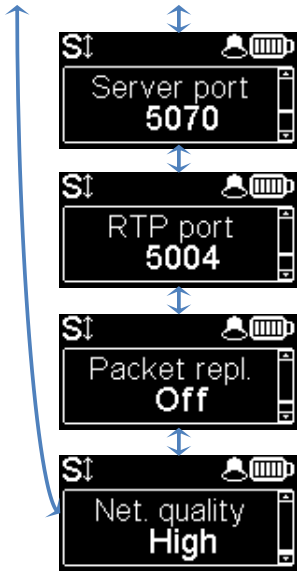
Change displayed SIP name

You can enter a name that will be displayed on the remote unit (depending on its type and settings).

Display your SIP account number

The SIP account data are normally entered/edited by using the web interface of the ScoopFone IP.

Server SIP port

The default SIP port for the server is 5060. In some cases it is useful to use the alternate port 5070.
You can also set any other port value by using the web interface, or the extended IP menu (see below).

RTP port for the audio stream (display only)

This port can be changed through the web pages or the extended IP menu (see below).

Activate packet duplication (double stream)

When this feature is enabled, a double stream is sent, for a more robust transmission.

Line quality setting

This setting adapts the size of the reception jitter buffer depending on the expected transmission quality (the lower quality, the bigger buffer for compensation).
"Low" is recommended for a poor quality data link. The "Very good" setting provides the minimum latency, but is also *more sensitive to possible jitter on the network.*

Extended IP menu

Some parameters of the Audio over IP (AoIP) operation are not normally accessible via the IP menu, or they can only be read but not edited. However, there is a mode, reserved for advanced users, that provides full access to all the AoIP parameters.

To activate this extended IP menu:

- Hit the following key sequence: **1 Up # # @ A E T A @** (display : @AETA@)
  ("Up") is for the + key above the display)

- Validate with the call key

From then on, the extended IP menu is accessible until the ScoopFone IP is switched off.

TOOLS MENU
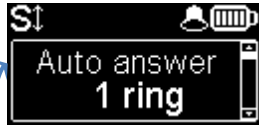
Automatic or manual answer

You can specify the number of rings before automatically answering calls, or select manual answering.

Auto redial

"On" activates the auto redial feature. In such case, when the ScoopFone IP calls a remote unit, it automatically redials the number whenever the link is dropped unexpectedly. The parameters of this function are available via the embedded html pages.
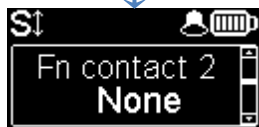*Caution: you must hang up from the ScoopFone IP for releasing the link once and for all.*
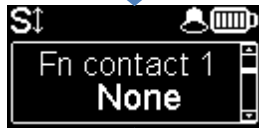
Function: contact 1

Can be used to make a call using memory 1.
Note : the contact can operate on a transition (Switch) or a pulse (Pulse) in order to call/hang up a call.
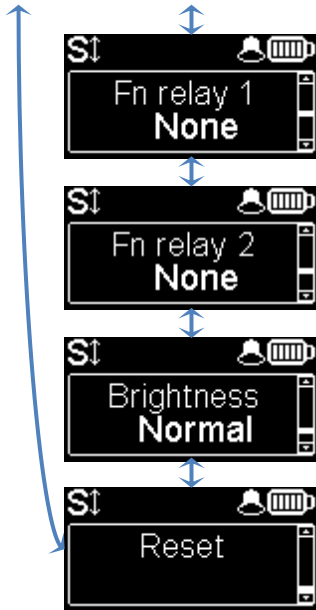
Function: contact 2

Can be used to make a call using memory 2, or control the microphone input mute.
Note : the contact can operate on a transition (Switch) or a pulse (Pulse) in order to call/hang up a call.

TOOLS MENU (CONTINUED)

Function: relay 1

Relay 1 signals the connection state :
connected (closed), on hook/idle (open), call in progress (blinking open/closed).

Function: relay 2

Relay 2 provides a selectable status: None, Ready, SIP ready.

Display brightness adjustment

3 brightness levels are available: Low, Normal, High.

Reset settings

- Reload the factory SIP account.
- Reset all settings, *including the SIP account data*.

WEB INTERFACE

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

HOW TO…

Update the ScoopFone IP firmware

First download from the AETA web site the most recent version of the firmware. Then updating the unit can be done using two alternate methods.

Update using the html interface of the ScoopFone IP

Use the menu **Maintenance** / **System update.**

- On this page, click "Browse…", select the update file on the computer;
- Click the "Update" button: the process starts, and a message can be seen on the device's display;
- Wait for the end of the process. The displayed text is then "Rebooting" on the html page. At this moment the unit should normally reboot by itself.

Update the firmware using a USB memory key

- Copy the file onto the root of the USB key;
- Switch on the ScoopFone IP and wait for the complete start;
- Plug the USB key on the rear side of the unit;
- Press the **OK** button to acknowledge the "Update Unit from USB?" message;
- The firmware update starts. It lasts for around 5 minutes, then the unit automatically reboots.

ⓘ *For more details, refer to the web page dedicated to the ScoopFone IP on our web site www.aeta-audio.com .*

Increase the robustness of the transmission

- To increase the resilience to packet losses, activate the packet duplication: menu **IP / Packet repl**. Be aware that this setting influences the transmission (you should apply the same setting on the remote unit for an effect at the reception on your side).
- To stand a high jitter level at the reception, change the "IP Network Quality" setting: menu **IP / Net quality**. When this quality is declared bad, the reception buffer size is increased for a higher tolerance to jitter (but traded for a latency increase).

TROUBLESHOOTING

The ⬆⬇ icon is not displayed

⇨ Check that the Ethernet cable is connected

The 🅢🅘 icon is not displayed

⇨ Check that the SIP account parameters are entered properly, especially the password.

⇨ Make sure that a firewall is not blocking the SIP protocol; try to change the "Server SIP port" (AETA server).

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 17) to increase the robustness of the transmission.
- Decrease the transmission bit rate (this can be efficient when the network access has a limited capacity).

You can't change the bit rate during the connection (shortcut Fn+2 (IP/Rate))
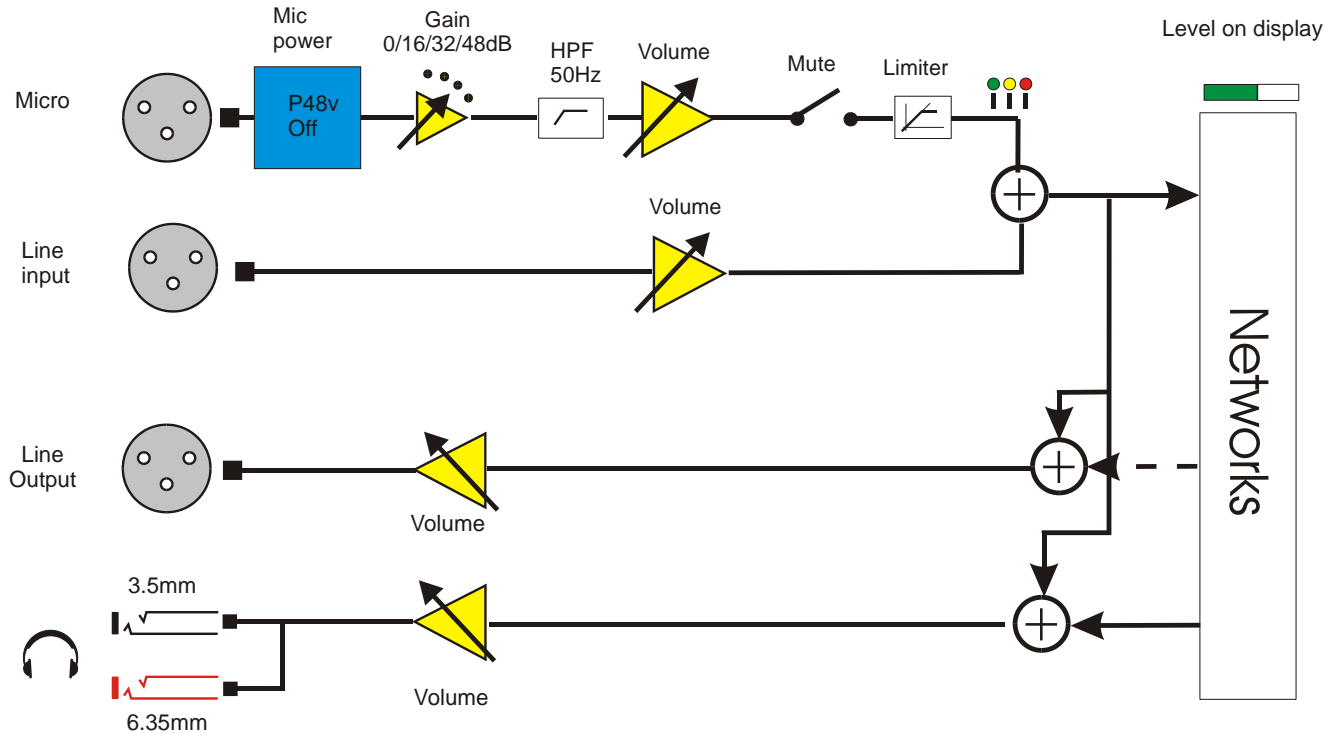
- Setting the bit rate "on the fly" is a feature only available with OPUS. This implies that the remote unit must support this coding algorithm.
- Besides, only another AETA codec is able to "follow" that during the connection, and change on its side the bit rate it is transmitting.

The "Transmission Quality" meter does not show up

You can see the "Reception Quality" indicator but the "Transmission Quality" meter is absent ("?" instead of the bargraph):

- This measurement requires the remote device to support the RTCP protocol, and this might not be the case.
- Check also that the RTCP port is not blocked by a firewall (its number is that of the RTP port + 1).

AUDIO DIAGRAM

Micro

Mic power

P48v Off

Gain 0/16/32/48dB

HPF 50Hz

Volume

Mute

Limiter

Level on display

Line input

Volume

Line Output

Volume

3.5mm

6.35mm

Volume

Networks

ACCESSORIES

Car battery adapter

Spare AC/DC adapter

Carrying bag

Carrying bag with accessory pouch