Direct RTP
Operation instructions

1. Comparison of SIP and Direct RTP

SIP is the protocol used by default in AETA codecs for AoIP (Audio over IP) transmission. This protocol is also part of the "N/ACIP" recommendation (EBU Tech3326).

"Direct RTP" is a more basic protocol that is compatible with many devices that do not support SIP. It differs from SIP in the following:

- SIP includes signaling, which means that one unit can call the other unit, negotiate coding algorithm parameters and exchange necessary addresses and ports. Few characteristics of the remote device have to be known beforehand. In contrast, no signaling is involved with Direct RTP. All needed parameters must be known beforehand.

- With SIP, a caller unit makes a call to a remote unit (just like a telephone call). With Direct RTP, the initiating unit directly sends an RTP stream to a remote device. In this model, the initiating unit behaves as a "client" sending the stream to a "server". When it receives a stream, the server sends its RTP stream to the address which initiated the link.

- With Direct RTP, the two units to be linked must be set beforehand with the suitable audio coding configuration, because there is no signaling for the units to negotiate it and do it automatically.

2. Principle of operation for establishing a link

On AETA codecs the operation mode is made similar to making a call, so for simplicity we will say that the initiating/client unit "calls" the passive/server unit. When the link is stopped, we will say that a unit "releases" the link as it stops to send and/or accept RTP data.

Each unit has got a reserved UDP port for Direct RTP; it sends and/or receives an RTP stream from/to this port. The default port number is 9000. This port is different from the RTP port used in the SIP mode of transmission. *Note that AETA codecs also exchange RTCP on the next port (default 9001).*

3. How to make a "call"

On the AETA codec, the unit that initiates the link must:

- Select as default network an IP network (Ethernet or mobile)
- Select the desired coding configuration
  (*must be the same on the remote unit!*)
- Select the Direct RTP mode: on the front panel, Setup/Network/AoIP Parameters/Default Protocol and select "Direct RTP".
  This selects Direct RTP for subsequent AoIP "outgoing calls".
  Important notice for the ScoopFone 4G: this is the only parameter accessible from the front panel of the ScoopFone 4G. Other parameters can only be reached via the web interface.
- Come back to the root menu and "dial" the IP address and RTP port number of the remote destination, as in this example: **12.34.56.78:9000**
  (*you may omit ':' and the port number if it is the same as the unit's own RTP port for Direct RTP*).

At this moment the unit starts to send the stream.
If it receives no stream in return, it will give up and stop streaming after a 30 second time-out period. This behavior can be adjusted, see in further chapters).

To stop the link, hang up just like for any call. The unit stops sending the RTP stream. The remote unit will stop after a period of time that depends on many conditions.
If both units are AETA and neither the RTP nor the RTCP ports are blocked, the remote unit will stop immediately.

4. How to set up a codec for "receiving calls"

First of all set the coding parameters as desired (*must be the same afterwards on the "calling" unit*).

On an AETA codec, the unit must be set for accepting Direct RTP "calls": from the front panel, Setup/Network/AoIP Parameters/Direct RTP Accept.
There is a shortcut for quickly enabling/disabling this parameter: dial 0 and "make a call" (green phone key). A pop-up message confirms you that the unit accepts (or not) Direct RTP incoming calls.

When the unit detects the reception of a valid RTP stream, it also sends its encoded stream to the address and port that are source of the received stream.

To stop the link, hang up just like for any call. The unit stops sending the RTP stream. It also stops accepting incoming streams until "Direct RTP Accept" is reactivated.

If the received stream is interrupted for more than a 30 second time-out period, the units stops sending the RTP stream but comes back to an idle state, and stays ready to receive a new incoming stream. This behavior can be adjusted, see in further chapters.

5. Miscellaneous parameters and details

All the parameters for the Direct RTP ports are located in the menu Setup/Network/AoIP Parameters.

The screen copy of the web page (below) also shows all these settings.

- **Default Protocol**: selects the protocol that will be used for an outgoing call in AoIP mode (Ethernet or mobile IP network).
  This selection does not prevent the unit to accept either protocol for an "incoming call".

- **Direct RTP / Accept Calls**: check to enable incoming Direct RTP streams. If unchecked, incoming RTP streams on the Direct RTP port are ignored.

- **Direct RTP / Send Only**: if checked and if the unit makes an outgoing call, the unit does not expect an RTP stream in return but rather decodes its own stream. The unit keeps on sending data until the user releases the link on purpose. The "RTP Timeout" parameter (see further) is ignored in such case.
  This parameter should not be checked together with the "Accept Calls" parameter.

- **Direct RTP / RTP Port**: port allocated to the Direct RTP transmission; *must be different from the RTP port for SIP*.

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

Note that, in the SIP parameters, there is also a "SIP Accept Calls" parameter. If unchecked, incoming SIP SIP calls are rejected.