SCOOP 4+

Stereo audio codec for real time audio transmission

User manual

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1. General

The SCOOP 4+ codec allows the bi-directional transmission of one or two audio signals with bit rate reduction, over digital leased lines, ISDN lines or IP protocol networks. The codec is available with the following main product versions:

- SCOOP 4+ LL, with digital leased line interfaces and an Ethernet interface for IP transmission;
- SCOOP 4+ ISDN 2B, with digital leased line interfaces, Ethernet interface and an ISDN interface;

The following table shows the main features of the product. Functions marked with ● in this table are available as options. Functions marked with □ are only available in the version equipped with the ISDN interface.

One outstanding feature of AAS codecs in ISDN mode is the 5A System®: on receiving an incoming ISDN call, the unit can automatically detect the coding algorithm and parameters of the calling codec, and then adjust itself in a compatible configuration so that the connection succeeds regardless of the initial configuration and that of the remote unit.

In IP mode, the codec features the same ease of operation thanks to the use of the SIP protocol.

The standard operation mode is the "single codec" mode, where the unit can be connected to a remote codec using any one of the listed coding algorithms.

In the dual 7 kHz codec mode (available for leased line transmission), the equipment is equivalent to two independent mono codecs, each running G722 over a 64 kbit/s leased lines.

® 5AS = Aeta Audio Advanced Automatic Adjustment System

Characteristics		Optional
Operation modes		
Single wide band codec Dual 7 kHz codec (LL mode only)		
IP transmission interface		
Ethernet interface, 10BaseT / 100BaseT; TCP/IP or UDP/IP protocol SIP signalling protocol		
Leased line transmission interfaces		
Two X24/X21/V11/V35 interfaces; 64, 128, 192, 256 or 384 kbit/s over one line, or 2x64 kbit/s over two lines		
ISDN transmission interface		
One S0 interface (U interface available for North America) 5AS automatic setting for incoming calls		□
Audio coding algorithms	(audio modes)[1]	
G711 (standard telephone)	Mono	
G722 SRT, H221, H242	Mono	
CELP 7 kHz (IP mode only)	Mono	
MPEG Audio Layer II	M, DM, S, JS	
4 sub-band ADPCM (low delay)	M, S	
TDAC (ISDN mode only)	M	□ ●
Available bit rates (depending on coding algorithm):		
Leased line transmission: 64, 128, 192, 256 or 384 kbit/s over one line, or 2x64 kbit/s over two lines		
IP transmission: 24 kbit/s to 384 kbit/s		
ISDN: 64 to 128 kbit/s transmitted via 1 interface (1 or 2 B channels)		□
Audio interfaces		
Two analogue inputs and two analogue outputs with adjustable level		
Digital audio input and output, AES/EBU format		
Auxiliary functions		
Data channel, 300 to 9600 bauds		
Relay transmission: 2 isolated inputs and outputs		
Audio coordination channel		●
Control and supervision		
Keyboard and LCD display on front panel Remote control serial port		
50 programmable set-up memories		
Ethernet remote control interface and embedded html server		

Table 1 – Main features of the SCOOP 4+

[1] M = Mono, DM = Dual Mono, S = Stereo, JS = Joint Stereo

2. Functions

The following synoptic diagram shows the basic functions of the equipment.

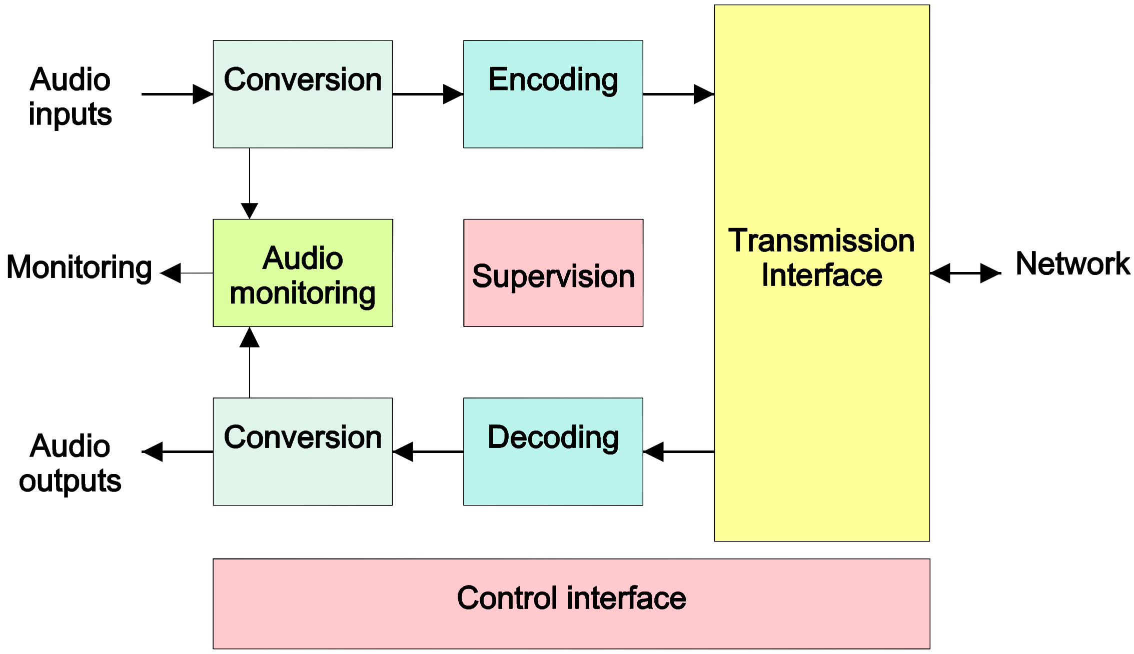


Figure 1 - Functional diagram of equipment

The audio signals to be transmitted are converted (when needed) to digital format, then the encoding function reduces the bit rate, using a selectable algorithm; the resulting bit flow is sent to one of the available transmission interfaces: permanent link data interfaces (X21/X24/V35), ISDN lines (S0 or U0), or an Ethernet interface.

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 output to both digital and analogue outputs.

2.1. Conversion of audio signals

The analogue inputs and outputs are balanced, and the input and output gains are adjustable. The sampling frequency of the analogue ⇔ digital converters depend on the operating mode.

The equipment also provides digital audio inputs/outputs in AES/EBU format. The input to the encoder is selectable between the digital audio input and the analogue stereo input. The output from the decoder is sent both to the digital output and the analogue stereo output. The digital audio interfaces are usually locked to the digital audio input ("genlock" mode), but alternatively they can be synchronised to the internal clock reference of the codec.

Having the digital samples from the audio interfaces (analogue or digital), sample rate conversion is fulfilled whenever needed to get audio data at the coding frequency F_c which is, depending on the coding type, 16, 24, 32 or 48 kHz. The coding clock is also locked to the internal clock.

2.2. Encoding and decoding

In the dual 7 kHz codec mode over leased lines, each codec uses the ITU-T G722 algorithm, running in mono at a 64 kbit/s rate.

In the normal single codec mode, the codec readily includes a wide range of coding algorithms. First, one can select among algorithms compliant with ISO and ITU-T recommendations:

- G711 (IP or ISDN mode only);
- ITU-T G722 (mono at 64 kbit/s);
- MPEG Audio Layer II at 48, 32, 24 or 16 kHz, with programmable channel mode and bit rate ;

Besides, other algorithms are available, that are so-called "proprietary" because they do not comply with enforced standards:

- CELP, running in mono at a net 24 kbit/s bit rate, and providing a 7 kHz bandwidth (only used in IP mode);
- 4SB ADPCM, running either in mono at a 128 kbit/s bit rate, or in stereo at 256 kbit/s ; the bandwidth with this algorithm is 15 kHz.
- TDAC mono, running at 64 kbit/s, with a 15 kHz bandwidth ; available as an option in ISDN mode.

The following describes some important features of the various available algorithms and protocols.

2.2.1. Notes about G711

G711 is the standard coding used for voice transmission on public telephone networks. 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.

G711 is available only for IP or ISDN transmission, not over the leased line interfaces.

2.2.2. Notes about G722

With G722 coding, three synchronisation modes are available:

- "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. This is only available for the ISDN mode.

H221 synchronisation is highly recommended when possible, as it features higher reliability and faster recovery time, while degradation (because of the bit rate used for framing) is minimal.

H242 protocol, the most flexible mode, is recommended by the ITU-T, and is included in J52. However, the mode with H221 synchronisation but without H242 protocol can be useful for compatibility with old generation codecs which did not use this protocol.

No specific synchronisation is needed for the IP mode.

2.2.3. Notes about MPEG coding and J52

The ITU-T J52 recommendation was defined in order to allow the interoperability of multimedia terminals over the ISDN[1], using common coding standards. It includes the following features:

- Framing as per ITU-T H221 recommendation, ensuring byte synchronisation and interchannel synchronisation when more than one 64 kbit/s B channel is required for the desired bit rate ;
- Interoperation procedures according to ITU-T H242 recommendation ;
- In the case of MPEG encoding, optional protection against transmission errors (Reed-Solomon error correction codes). Although J52 does not apply to leased line connections, this error protection technique is also available for leased line transmission with the SCOOP 4+.

Details about MPEG and J52 can be found in the annexes (refer to 6.1, Complements on the algorithms and protocols used).

It must be noted that, thanks to the interoperation protocol, J52 codecs, when setting up a link, can negotiate automatically and agree on a configuration that is compatible with the capability of both units (regarding bit rate, channel mode, etc.). In this way, when the units differ in their capability (or make), the resulting configuration may be different from expected beforehand, but in most cases the link will work and audio will be transmitted.
As another useful consequence, this also gives users more tolerance to mistakes when configuring the units on the two sides of the transmission links, as the codecs will adapt automatically even with differences in the initial settings of the two units.

2.2.4. MPEG coding for leased line or IP transmission

J52 is only applicable to ISDN transmission, and no inverse multiplexing is needed for leased line transmission neither for IP transmission, because a single data stream is transmitted.

For these reasons, only one MPEG format is defined for non-ISDN transmission; there is no distinction in these modes between J52-compliant or non compliant format.

2.2.5. Notes about TDAC

As an option, the codec can also include the TDAC algorithm. TDAC is for Time Domain Aliasing Cancellation ; this is a transform coding based on an MDCT (Modified Discrete Cosine Transform), encoding a 15 kHz bandwidth mono signal at a 64 kbit/s bit rate.

Some specific product versions also include "asymmetric" modes:

- G722/TDAC : G722 encoding, TDAC decoding, running both in mono at 64 kbit/s ;
- TDAC/G722 : TDAC encoding, G722 decoding (with SRT), running both in mono at 64 kbit/s ; this mode is symmetric to the previous one.

2.2.6. Symmetric or asymmetric codec modes

The codec allows two communication modes:

Symmetric communication: in this mode, the encoder and decoder both use the same coding algorithm with the same configuration (channel mode, etc.). In this case, the communication is strictly symmetric full-duplex, with exactly the same coding configuration used in both directions (local to remote and remote to local). This is usually required when using proprietary algorithms.

[1] J52 is only relevant for ISDN connections

Asymmetric communication: this mode is used for applications requiring different coding configurations in the two directions. The J52 protocol allows such mode. To give some examples, it is possible to transmit MPEG Layer II in one direction and G722 in the other one, or MPEG stereo in one direction and MPEG mono in the other one, etc.

Specific product versions also allow asymmetric modes wherein one direction is G722 coded while the other one is TDAC coded. Such mode is useful e.g. in order to get a low delay return path encoded in G722 while the send path is encoded with higher quality but a higher delay.

2.2.7. 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.

5A stands for Aeta Audio Advanced Automatic Adjustment. This 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, stereo…), 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 programme. 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 active for ISDN connections.

2.2.8. SIP protocol and SDP

The SIP protocol is a signalling protocol, used for IP connections, which allows the SCOOP4+ to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Details about the SIP protocols can be found in the annex (refer to 6.2, Overview of the SIP protocol).

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 need not know how the remote unit is configured before initiating a link.

2.3. Transmission interface

The codec includes an Ethernet interface for IP protocol networks, interfaces for transmission over leased lines, and an ISDN interface is available in some versions.

2.3.1. Ethernet interface

The IP interface is a 10BaseT/100BaseT Ethernet interface allowing transmission of the audio programmes in a wide range of possible bit rates. The SCOOP 4+ implements the SIP protocol, which allows it to interoperate with IP phones and other SIP compatible audio codecs, in a way similar to ISDN or POTS connections. Links can be set up in two ways:

- “Peer to peer” connection between two compatible units
- Use of a SIP proxy server to set up the link

Details about the SIP protocols can be found in the annex (refer to 6.2, Overview of the SIP protocol).

The audio coding algorithm can be selected depending on the required quality and the available network bandwidth. The following algorithms are currently available:

Codec	Bit rate[3]	Audio bandwidth	Typical use, main features
G711	64 kbit/s	3 kHz	Voice, telephony Compatible with IP phones
CELP	24 kbit/s	7 kHz	Suitable for high quality speech Low network bandwidth consumption
G722	64 kbit/s	7 kHz	High quality speech
MPEG Layer II	64 or 128 kbit/s	up to 20 kHz	Highest quality, suitable for speech and music

In addition, the Ethernet interface can be used for remote controlling the unit via a TCP/IP connection (TCP port 6000).

2.3.2. Leased line interfaces

For transmission over leased lines, the codec includes two X24/V11 ports which can run at 64 kbit/s, 128 kbit/s, 192 kbit/s, 256 kbit/s and 384 kbit/s bit rates.

With most coding modes, only one X24/V11 port is used. In the "2*64" dual mono G722 mode, the two ports provide two independent interfaces; the equipment is similar to two mono codecs.

When transmitting in the "leased line" mode, the codec synchronises onto the network clock provided by the X24/V11 interface. In the specific "2*64" mode where the two ports are used, the codec initially synchronises on port #1, but it changes the synchronisation port in case of a fault.

If no valid clock is available on the X24/V11 interfaces, the system folds back to an internal clock.

2.3.3. ISDN Interface

On the ISDN side, the codec includes one BRI interface (S0 or U0 depending on equipment version), allowing transmission over one or two 64 kbit/s B channels. Thus, the total available bit rate ranges from 64 to 128 kbit/s (1 to 2 B channels).

The codec synchronises itself onto the ISDN network clock when a link is active.

2.4. Supervision and control interface

These functional modules fulfil the control and supervision of the equipment (configuration, communication management, status monitoring), thanks to a keyboard, an alphanumeric display, LED indicators, and remote control interfaces.

The remote control is possible either via a serial data port or through the Ethernet interface with a TCP/IP connection.

In order to allow easy and quick programming of the codec for specific operational configurations, the equipment features configuration memories (or "profiles"). When recalling a profile, the codec is directly reconfigured with parameters that were stored beforehand in this profile by the operator.

3 This is the "net" encoded audio bit rate; the actual network occupation is higher because of the protocol overhead

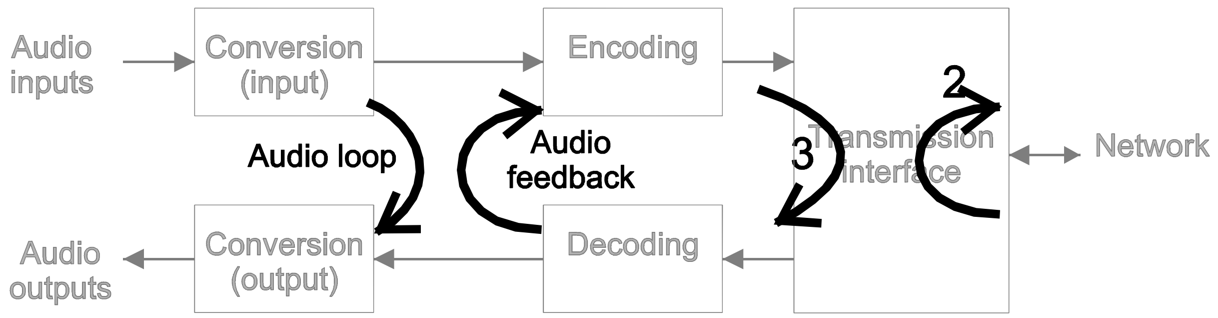
Besides configuring the equipment operating mode, this module monitors its status (detection of alarm conditions). On detecting operation or transmission faults, the equipment switches on indicators and relay contacts. Two alarm classes are defined:

- "Internal" alarm ; corresponds to a major fault internal to the equipment ;
- "External" alarm ; corresponds to a fault whose origin is deemed external to the equipment (for example, transmission fault);

For maintenance purposes, some test loops can be activated:

- "Audio loop": uncompressed audio data are looped from the input of the encoder to the input of the output conversion functional block. This loop redirects the audio input to the audio outputs;
- "Loop 3", or "Codec" loop: compressed audio data are looped just before the network interface ;
- "Loop 2", or "Network" loop: this loop sends the received data back to the network ; for the remote codec, the effect is the same as a loop 3 when the transmission works correctly ;
- "Audio feedback" loop (audio output to audio input) ; this allows the codec to send back to the remote codec the signal it receives, after decoding and re-encoding.

The following drawing schematically shows the test loops:



2.5. Audio monitoring

This function enables the monitoring of the audio input (before encoding) and the audio output (after decoding the received signal), and provides:

- A display of the signal level both at the encoder input and the decoder output ;
- A test output on a stereo headphone jack, monitoring either the encoder or decoder audio signals.

☞ *Note: as the audio output is monitored <u>immediately after decoding</u>, this monitoring position is <u>not</u> sensitive to the possible activation of the audio test loop (see above diagram), contrarily to the physical audio outputs (both analog and digital).*

2.6. Auxiliary functions

2.6.1. Data channel

This function is currently only available in leased line transmission mode.

In leased line mode, a bi-directional data channel can be transmitted along with the compressed audio signals, by reserving a fraction of the transmitted bit rate. The equipment includes a serial asynchronous port for this purpose. The data are transparently transmitted end-to-end; hardware signalling is not available.

This function is only available when the main audio programme is MPEG or ADPCM encoded.

The interface speed is programmable at 300, 1200, 2400, 4800 or 9600 bauds. However, the actual transmission capacity depends on the coding algorithm, as indicated by the table hereunder.

Coding type	Possible transmission rate (bit/s)				
	300	1200	2400	4800	9600
MPEG Audio					
4SB ADPCM					

Table 1 – Capacity of data channel depending on type of coding

2.6.2. Relay transmission

This function is currently only available in leased line transmission mode.

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.

This function is only available when the main audio programme is MPEG or ADPCM encoded.

A typical application is the transmission of an “on air” signal; the contact closure may be used for e.g. switching on a lamp or starting other devices.

When using MPEG coding, relay transmission can be activated along with other auxiliary functions. For ADPCM, relay transmission is activated in place of the data channel.

2.6.3. Coordination channel

This function is currently only available in leased line transmission mode.

This function is available as an option. It enables the transmission of an auxiliary audio channel (or coordination or “order-wire” channel), along with the compressed audio, by reserving 8 kbit/s from the transmitted bit rate. This channel uses a compression algorithm of CELP-HLTP type and provides a “voice grade” channel (300-3400 Hz pass-band).

This function is only available when the main audio programme is MPEG or ADPCM encoded.

With ADPCM, the coordination channel cannot be used along with other auxiliary functions (i.e. data channel or relay transmission).

When using MPEG coding, all three auxiliary functions can be activated at the same time. Note that relay transmission and the coordination channel are only compatible with AAS products, as these functions are not covered by independent standards.

3. Operation

3.1. General principles

The equipment control and supervision (configuration, status monitoring) is possible in two ways:

- "Local" mode: front panel keyboard and display, status indicators ;
- "Remote control" mode, thanks to an asynchronous serial port or the Ethernet interface.

As a general rule, the configuration parameters are saved in non-volatile memory, and restored when the unit is powered-on.

Local mode operation is described in detail in chapter 4 (Detailed operating mode).

Thanks to the remote control mode, the codec can be operated from a computer with supervision software. The supervision station is a PC computer running Windows (95, 98, ME, NT, 2000, XP or Vista), equipped with the TeleScoop™ configuration and monitoring software. This optional software gives full access to the codec functions (configuration and status monitoring) with a graphical interface, and several units can be controlled from the same computer.

Details about this supervision software can be found in the documentation and user manual of the TeleScoop software.

In addition to this, some parameters related to the Ethernet/IP interface and transmission can be set by using an embedded HTML server; these are described further in 3.5.2, "Use of the embedded html server".

The SCOOP 4+ can be remote controlled by third-party codec management software and systems. Please consult us for more information on the available offer in this field.

3.2. Physical description of the equipment

The SCOOP 4+ codec is housed in a 19 inches chassis of 1U height (44 mm or 1.75"); it includes a universal mains power supply.

3.2.1. Front panel

All the elements needed for local control are located on the front panel (see picture on page 13 below).
This panel can be roughly divided in three areas:
On the left-hand side, one can find an LCD and the basic navigation and dialling keys. The central area of the panel includes several status LEDs and a keypad for the entry of dialling numbers and/or text data.
The right hand side groups audio monitoring elements.

> ☞ ***The "Esc" key is also used to power the unit on and off****:*
> *When the unit is in standby (the blue LED besides the Esc key is on), hold the key depressed for at least 3 seconds to switch the unit on;*
> *When the unit is in operation, press the key down for more than 3 seconds to switch it off.*

In addition to this "soft" switch, the unit automatically switches on when AC power is applied to its power socket.

LCD and basic control keypad

This part is used for configuration and call set up; details cane be found in chapter 4, dealing with the operating mode.

The 2x20 character alphanumeric display is surrounded by the following keys, (*from left to right*):

Key	Function
"Hang up"	Release a link in IP or ISDN transmission mode
"Unhook"	Start a link or accept an incoming call (in IP or ISDN transmission mode)
Navigation keys	Menu-dependent keys; used to scroll options and/or select an option in a menu. The bottom line on the LCD shows the function of each key.
"OK" key	Confirm a selection or enter data
"Esc" / ⏻ key	*Short pressure*: Escape to higher menu level; *Long pressure*: Switch on or switch off the unit[4]

The blue LED besides the Esc/Power key is off in operation, but lights on when the unit is in standby.

Status LED indicators

The LEDs have the following meaning (*from left to right*):

Marking	*Color*	Function
⏻	*blue*	On when unit is in standby
Line 1	*Green*	On when interface n°1 is active / connected
Line 2	*Green*	On when interface n°2 is active / connected
Dec 1	*Green*	On when the decoder is synchronised on "line 1"
Dec 2	*Green*	On when the decoder is synchronised on "line 2"
INFO 1	*Amber*	Displays the status of the received "relay info" n°1
INFO 2	*Amber*	Displays the status of the received "relay info" n°2
Test	*Red*	On whenever a test loop is active
Alarm / Ext	*Red*	On in case of an external alarm
Alarm / Int	*Red*	On in case of an alarm with internal cause

[4] Note: some specific versions of the product **cannot** be switched off; in such case a long pressure has no effect!

Audio monitoring

Two pairs of LED bargraphs display the level of the audio signals, both on the transmission and reception directions. The top bargraphs display the level of the audio channels on the transmitter (encoder), while the bottom bargraphs display the level of the received channels (decoder side). The 0 dB mark is a reference level that can be adjusted (*relatively to digital full scale; the reference level can be set in the menu, SETUP / Audio / Level Meter / HEADROOM)*. The "OVLD" LED at the right end of each bar shows when the signal reaches maximum digital level (or clipping level), regardless of the reference level setting.

☞ *"OVLD" always reacts to absolute full scale level, while the bargraph level indication depends on the reference level setting*

The "HEADROOM" setting in the menu defines (in dB) the difference between the *maximum level* (or digital full scale) and the *reference level*, for which 0 dB is displayed on the level meters. Here are some examples:

- If the headroom is set at 0 dB, then the maximum displayed level is 0 dB; note that OVLD will light on whenever the signal reaches this level (or exceeds it on the analogue input).
- If the headroom is set at 10 dB, then "0 dB" is displayed when the signal is 10 dB below maximum level, or -10 dBFS. The display in such case can reach up to +5 dB. OVLD lights up when the signal reaches maximum level (but not before!).

The audio signals can also be monitored with a headphone connected on the front panel (1/4" or 6.35 mm stereo jack). The headphone volume is adjustable thanks to a potentiometer, and the source select key toggles the listening between transmission (**Tx** indicator) or reception (**Rx** indicator).

Actions dealing with this area (connecting or disconnecting the jack, Tx/Rx selection, volume adjustment) never affect the transmitted or received signals.

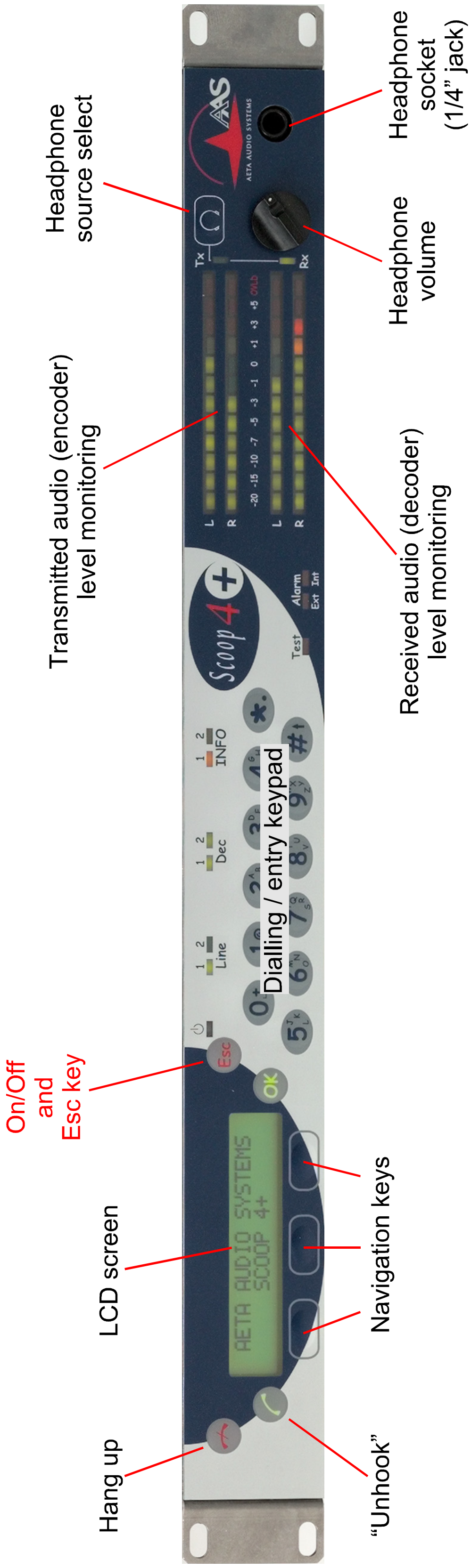


Figure 2 - Front panel of SCOOP 4+

3.2.2. Rear panel

All connections are done on the rear panel of the codec. The characteristics of the interfaces and layout of the sockets are detailed in chapter 5.1. Characteristics of interfaces.

The following elements are available on the rear panel (refer to following Figure 3 - Rear panel):

Mains power socket

This is an IEC type power socket.

Audio inputs/outputs

- Analog inputs/outputs: at the input, plug the audio cables into the female XLR sockets. At the output, plug the audio cables into the male XLR sockets. In mono mode, only “A” channel is used.
- Digital inputs/outputs: a digital input (mono or stereo) in AES/EBU format (or SPDIF) can be connected on the female XLR socket, and a digital output in AES/EBU format is available on a male XLR socket.
- It is possible to select which input (analog inputs or digital input) is fed to the encoder for transmission. On the receiving side, the decoded signals are output both on the analog and digital outputs.

X24/V11/V35 interfaces (labelled “X24/V11/V35” and “ALARM + X24/V11”)

These sockets are used for the connection to data transmission equipment in the “leased line” mode.

The connectors are 15-point male, Sub-D type. In the standard single codec mode, only one port is used. This is normally the main port “X24/V11/V35”, but it is possible to select the other port.

In the dual codec mode, both ports must be used. In this mode, audio channel A is transmitted on the main port “X24/V11/V35”, and audio channel B is transmitted on the additional port labelled “ALARM + X24/V11”.

Alarm indicator and contacts

This “ALARM + X24/V11” port also includes two “form C” relays, providing isolated contacts, which can signal alarm conditions:

- Internal alarm contact;
- External alarm contact;

A red LED indicator also indicates that an alarm relay is activated. In the factory setup, every alarm cause sets the LED on, but by setting jumpers on the motherboard it is possible to program the indicator to react to only one type (internal or external alarm).

The pin-out of the socket and the detailed characteristics of the alarm relays can be found in chapter 5.1.6: “Alarm + X24/X21” interface (p. 37).

USB socket

This host USB port is currently not used.

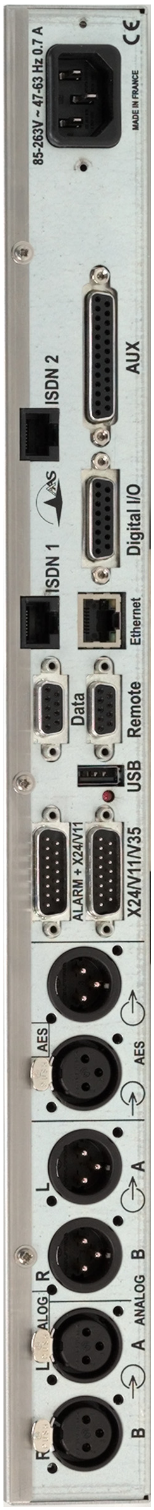


Figure 3 - Rear panel

Remote control (Remote)

This 9-pin female sub-D socket is an asynchronous serial interface port, usable for remote controlling the equipment thanks to a control and supervision PC.

Data

This 9-pin female sub-D socket is an asynchronous serial interface port, usable for transmission of a bi-directional data channel (refer above to 2.6.1, Data channel).

Ethernet interface

This socket is a 100BaseT/10BaseT port, used for audio transmission over IP and/or for remote controlling the unit via a TCP/IP connection (TCP port: 6000). This RJ45 socket is devised for a normal "straight" cable to an Ethernet hub or switch. The two integrated LEDs show the presence and activity of the network (green LED) and the interface mode: half-duplex (yellow LED off) or full-duplex (yellow LED on).

The configuration of the interface is described in 3.5.1, IP configuration (Ethernet interface).

"ISDN 1" and "ISDN 2" sockets

These RJ45 sockets allow the connection to the ISDN, for the product versions which include this capability. Their layout is standard. The sockets must be used according to their number, i.e. #1 must be used if one line only is needed, #1 and #2 if two lines are needed. *Currently, the product can only be equipped with one ISDN interface.*

"Digital I/O" socket

Reserved for future use.

"AUX" socket

This 25-pin female sub-D socket groups the interfaces for the relay transmission function and the (optional) coordination audio channel.

It also includes loop interfaces for the loop control function (*to be described later*), as well as a +5 V power supply that can be used to provide current for the loop and relay interfaces.

3.3. Equipment configuration parameters

The parameters may be divided into the following categories:

- Coding configuration parameters, which include audio coding type, coding frequency Fc (and subsequently the nominal bandwidth), audio channel mode and transmission bit rate. Besides, in case of MPEG coding, it is possible to select the error protection mode.
- Configuration of the audio interfaces, including: selection of analogue or digital format for the audio input, maximum level for the analogue inputs and outputs, and synchronisation mode for the AES/EBU interfaces.
- Parameters of the auxiliary functions: possible activation of a data channel, bit rate of this, possible activation of the relay transmission, possible activation of the auxiliary audio channel (if this option is available).
- Parameters of the network access: type of network interface (Ethernet/IP, leased line or ISDN), interface parameters, etc.
- Parameters of the keyboard/display interface (as an example, selection of the language for the display messages).

Chapter 4 (Detailed operating mode) describes these two last categories.

The parameters dealing with the audio interfaces are programmable independently from the others. On the other hand, the auxiliary functions depend on the current transmission mode and coding type.

The following table is a summary, for each coding type, of the allowed values for the various parameters of the coding configuration and auxiliary functions.

Meaning of abbreviations in the table:

- Channel mode : M = Mono, S = Stereo, JS = Joint stereo, DM = Dual Mono
- Coding : H221 = H221 synchronisation, SRT = Statistical Recovery Timing
- X / — = function available / not available with this type of coding
- FEC : Forward Error Correction = Reed-Solomon error correction

Only MPEG can be configured with all three auxiliary functions (data, auxiliary audio, relays). For other algorithms, each function, when available, can only be used alone. Auxiliary functions are only available for codec 1 when in double codec configuration.

☞ *The auxiliary functions are currently only available in leased line mode.*

Coding	Channel mode	Coding frequency Fc	Bandwidth	Bit rate	FEC mode	Data channel	Aux. Audio	Relays
		kHz	kHz	bit/s		bit/s		
G711[5]	M	8	3.4	64k	—	—	—	—
CELP[6]	M	16	7	24k	—	—	—	—
G722 SRT	M	16	7	64k	—	—	—	—
G722 H221	M	16	7	64k	—	—	—	—
G722 H242	M	16	7	64k	—	—	—	—
MPEG Layer II	M DM S JS	16 24 32 48	7 to 20 depending on Fc	64k 128k 192k 256k 320k 384k	0 to 3	300 to 9600	X	X
4SB ADPCM	M	32	15	128k	—	300 to 4800	X	X
4SB ADPCM	S	32	15	256k	—	300 to 4800	X	X

Table 2 – Possible values for configuration parameters

5 This coding mode is not available in the leased line mode

6 This coding mode is available in IP mode only

3.4. Installation and set up

3.4.1. Mounting and connections

Natural convection cools the equipment. Avoid obstructing the openings on the flanges.

To operate the codec, the minimum necessary connections to set up are (see details in the rear panel description):

- Power supply ;
- Audio inputs and outputs (XLR sockets) ;
- Network interface: depending on the networks used, Ethernet interface, ISDN line and/or X24/V11/V35 interface(s) ;

Whenever needed, the “ALARM + X24/V11” socket must be connected to an external supervision system (alarm relay contacts).

The pin out of the connectors is indicated in chapter 5.1**:** Characteristics of interfaces.

3.4.2. Initial set up

Before the first link, the equipment must be configured according to the desired operation mode (audio input/output format, coding type and parameters, etc.) and the local conditions (network interface parameters…).

For using the keyboard, a password may have to be entered. After factory setting or after total configuration erasure, the password is blank (no password needed). Afterwards, a password can be programmed by the user if one is needed.

For more details about the codec configuration, see chapter 3.3 (Equipment configuration parameters, p. 17) and chapter 4 (Detailed operating mode). The setup of the Ethernet interface is described in 3.5 below (Initial setup of the Ethernet interface).

3.4.3. Notes about the use of AES/EBU interfaces

When using digital audio interfaces, it must be decided whether the codec is “***master***” or “slave” regarding audio sampling clock synchronisation. In the first case, the codec derives the sampling clock from the network clock or an internal source, and the device(s) connected to the codec must synchronise to the same clock source.

The most common choice is rather the “slave” mode, to be used when it is not possible (or not desired) to synchronise the external equipment onto the clock of the transmission link or the codec. In this case, the AES/EBU interfaces should be set in the so-called “***genlock***” mode. When in this mode, the codec derives the sampling clock of the digital audio interfaces from its AES input (in other terms is “gen-locked” onto the incoming AES signal), and sampling rate conversion (SRC) is used for interfacing to the coding parts.

> ☞ *It is mandatory in such situation to provide the codec input with an AES signal featuring the same sampling frequency as the external equipment, even if the codec is used only as a decoder.*

If this requirement is ignored, the unit will actually fall back to “master” mode. In such situation, clicks in the audio programme might be heard, especially when the resulting sampling rate is very different from that of the external device.

If, on the contrary, it is decided to synchronise the external equipment (at 32, 48 or 96 kHz) onto the transmission clock of the leased line interface, the codec must be configured in “master” mode. In this case, the output is locked onto this clock, and it can be used as a reference to synchronise the equipment connected to the codec output. The digital audio signal at the codec input must then come from a device synchronised by this way.

3.5. Initial setup of the Ethernet interface

The SCOOP 4+ includes a 100BaseT / 10BaseT Ethernet interface, and the audio transmission can take place over an IP network through this interface. In addition, it is always possible to use the Ethernet interface for remote controlling the unit via a TCP/IP connection (TCP port 6000).

For IP audio transmission, SCOOP 4+ uses the SIP protocol, which eases the setting up of a link. The operation is similar to setting a call over the ISDN or PSTN. The transmission can be done in two modes:

- Direct “peer to peer” transmission between two compatible units.
- Use of a SIP proxy server for the call setup

3.5.1. IP configuration (Ethernet interface)

As a very first step, the Ethernet interface must be assigned an IP address, and related parameters. This phase is very simple when a DHCP server is available in the network.

Before going further, connect the Ethernet interface to the network, using CAT5 wiring.

- Connection to 10BaseT or 100BaseT interfaces are both suitable, as the SCOOP 4+ automatically switches to the right 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 is very simple when a DHCP server is available in the network. The menu to use is reached by TOOLS / Maintenance / Ethernet Setup.

☞ *When Ethernet/IP is the current interface for audio transmission, an alternate path in the menu is SETUP / Net / Param / Network Setup.*

DHCP server available

This is the simplest case, because the server will allocate a suitable IP address and give the unit the right settings. Select “DHCP” in the menu (TOOLS / Maintenance / Ethernet Setup). The unit will then automatically find the DHCP server and automatically set the parameters.
You can read the IP address (allocated to the unit by the DHCP server) in the “About” menu (TOOLS / Maintenance / Ethernet Setup).

Note that, as an additional advantage with DHCP, you do not need to change this setting later, even if you move the SCOOP 4+ to another network, as long as it is still connected to a DHCP server.

"Static" IP configuration

When there is no DHCP server, you have to enter the settings manually, using the menu (TOOLS / Maintenance / Ethernet Setup / Manual / etc.). The IP address must be "free", i.e. not already assigned to other equipment. Ask support from the network administrator(s) as needed. The following has to be entered:

Parameter	Notes
IP address	Must be unique on the network
Network mask	A typical value is 255.255.255.0
IP Gateway	
DNS	Domain Name Server

All addresses are in the form n.p.q.r. Examples: 192.168.0.12, 10.0.54.123.

☞ *Note: in contrast to the configuration with DHCP, the "static" setting has to be reviewed each time you move the unit to a new physical site/network, as the previous IP addressing is probably not valid for the new location.*

Checking the IP configuration

The above configuration is kept in the unit's memory, and reloaded at each start. It is recommended to restart the unit right after the initial setting, to ensure that everything is OK.

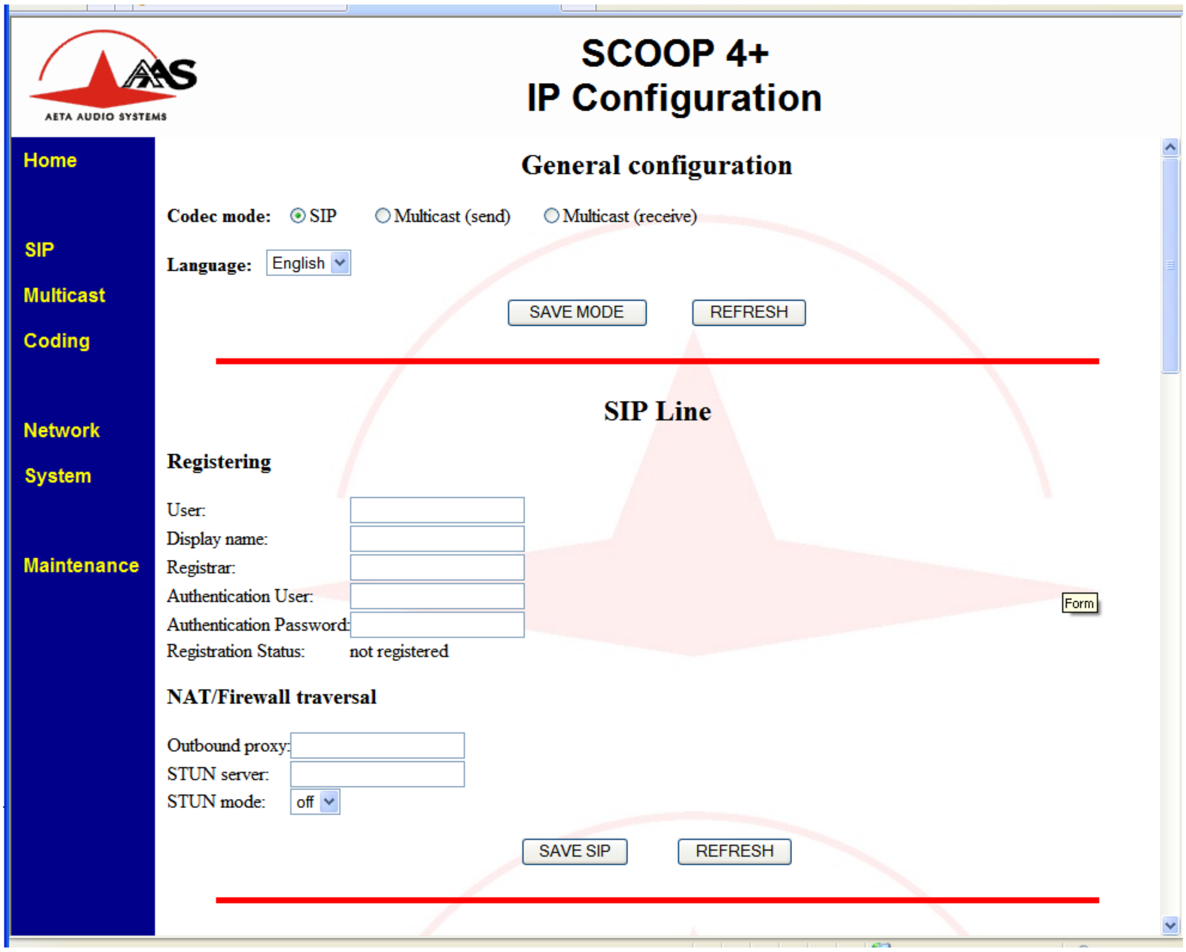
To check the setting, you can read the IP address in the "About" menu (TOOLS / Maintenance / About).

You can then also check that the unit is seen on the network and at the right address: from a computer connected to the same network, enter (in the command mode, or console mode depending on the OS) "ping *ipaddr*", where *ipaddr* is the IP address of the SCOOP 4+.

If the response is positive, then you can proceed with the rest.

3.5.2. Use of the embedded html server

From a computer connected to the same network, open an html browser window and enter the IP address of the SCOOP 4+ in the “address” or “URL” field. This gives access to the html server that is embedded in the SCOOP 4+. The page typically looks like the following picture:



If you click “Network” on the left, you can get a display of the IP addressing data. It is possible to change settings, and click the “SAVE” button[7] to write them into the SCOOP 4+. “REFRESH” reloads the page from the unit to update the display.

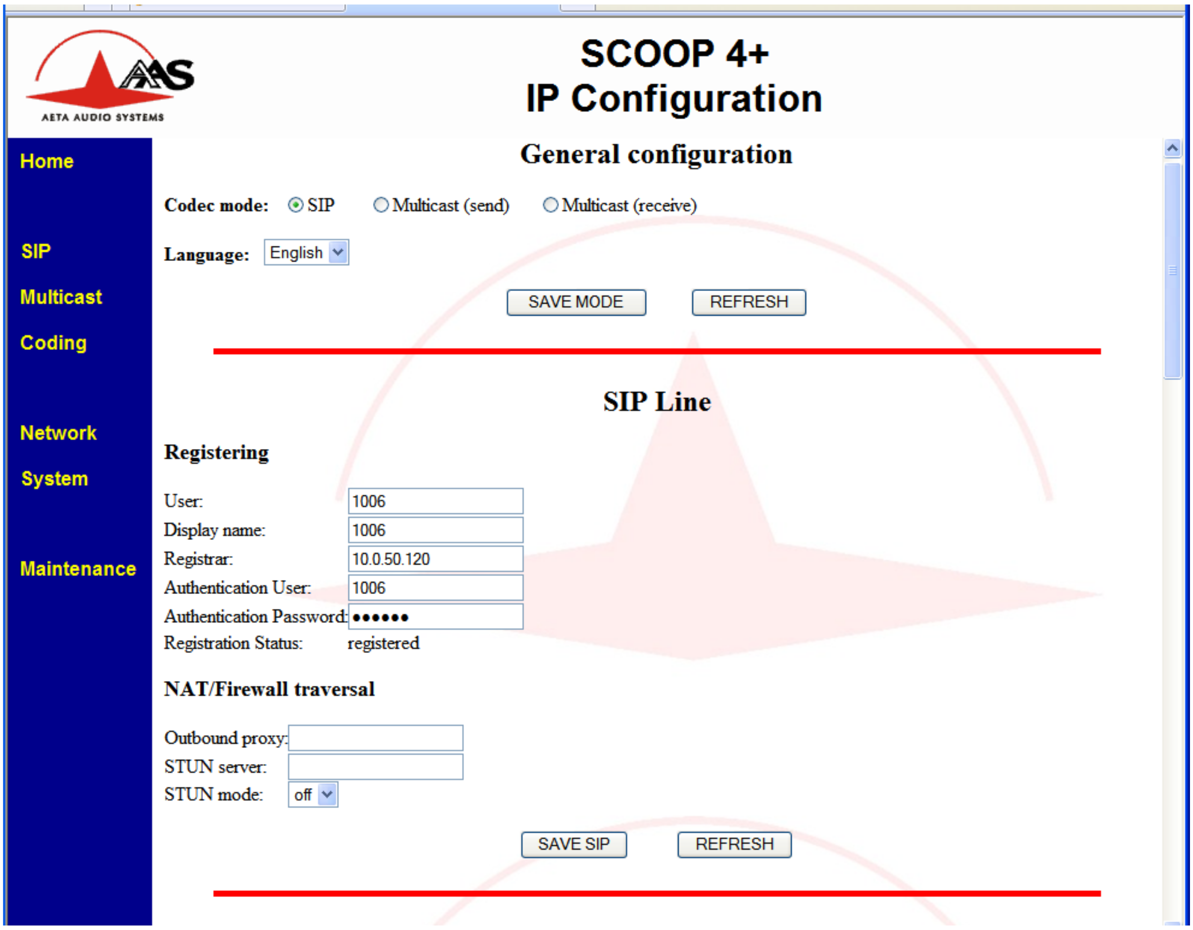
The network settings can be updated from this page, **but**:

- Obviously it is not usually possible to do the initial setting in this way!
- Be careful before changing these settings, as a wrong setting here can make you loose control over the unit… (In such event, go back to 3.5.1 above)

[7] Important notice : the SAVE button only uploads a section (enclosed between two bold horizontal lines), unlike the REFRESH button, which refreshes the whole page.

3.5.3. SIP registration and configuration data

You can access these parameters if you click the "SIP configuration" button on the embedded server html page. This is the only way to configure these settings, and most cannot be set from the keypad (except those mentioned in the following). The following is an example screen copy, and some comments about the displayed data:



Item	Notes
User, Display name, Authentication user	Refer to the network administrator and/or the administrator of the SIP server; Often these three parameters have the same value, as here, but they may be different.
Authentication password	Refer to the network administrator and/or the administrator of the SIP server
Registrar	IP address of the SIP registrar; a symbolic name (e.g. sipsrv.mycomp.com) is accepted, if recognised by the DNS. *Can be also read from the menu* (TOOLS / Maintenance / About)
Registration status	(read only data) Shows that the unit is (or is not) successfully registered on the server. *Can be also read from the menu* (TOOLS / Maintenance / About)
Outbound proxy	An outbound proxy is one way of getting access through a NAT router or a firewall; Refer to the network administrator and/or the administrator of the SIP server for this setting
STUN server	A STUN server is also one means of getting access through a NAT router. If such server is available, enter here its IP address or domain name.
STUN mode	Enable or disable the use of the STUN server. This allows to keep the address of the STUN server even when the function is disabled. *This setting is available from the menu (SETUP / Net / Param / STUN Mode)*

Make sure to click the "SAVE SIP" button located at the bottom of this section if you want to actually write your changes into the SCOOP 4+.

In the "Coding" section, you can define the desired encoding for outgoing calls. However, this is usually done from the keypad and LCD.

☞ *Note that the unit adjusts automatically for incoming calls, so no preliminary codec setting is needed for receiving calls.*

The registration data do not have to be changed often in normal operation. In fact, they may be still valid even after the unit moves to another location, even though its IP configuration changes.

3.5.4. Other information and settings

The html page includes various other settings. Many of these are reserved at the moment.

- *Do not change the following parameters from their initial settings: "Codec mode" (leave SIP selected), "Multicast mode parameters" (not implemented yet).*
- The system information includes the Ethernet MAC address (unique and fixed for a given unit), and the current IP address. *Be careful with the security passwor!.* This optional feature is left blank in the initial factory setting.

3.6. First level maintenance

3.6.1. Internal description

To be added later

3.6.2. Internal configuration

☞ *All the configuration is done in the factory, and/or it can be changed by means of the keyboard/display interface, without having to open the unit.*

However, a jumper may be set to prevent one alarm type to light on the red alarm LED on the back of the equipment.

Please consult us for such operation!

3.6.3. Analysis of malfunctions

The following table indicates the detected alarm conditions and their classification:

Alarm condition	Internal	External	Minor[8]
Power or fuse fault	X		
Bad start-up of a microprocessor, or interface fault detected on start-up	X		
Overload on an audio input			X
Fault on AES/EBU audio input		X	
Decoder synchronisation error		X	
Network clock fault[9]		X	

Table 3 - List and classification of alarm conditions

Excluding the case when an internal failure disables the management micro-controller, messages are displayed to indicate the anomaly, or the fault can be searched using the menu.

The test loops accessible from the "TESTS" menu can help improve the analysis of a problem:

- In order to check if the audio part functions correctly, use the "Audio" loop and check if the audio is OK at the output.
- To check if the coding part functions correctly, activate "Loop 3" and check if the alarm disappears (and the decoding indicators come back to normal), and if the audio is present at the output.
- "Loop 2" sends back to the remote codec the compressed data received from the network (see the description of test loops in 2.4, Supervision and control interface, page 8). This way, it is possible to test the integrity of the transmitted data and/or check that the remote codec works properly.

The decoder out to encoder in loop ("Audio feedback" loop) can be used for overall functional check, and also for aligning the overall chain.

---

8 Minor alarms are readable on the display, but do not trigger alarms (contacts and LEDs)

9 Fault of the network clock source currently used for synchronisation (X21/X24 main port or secondary port)

In leased line mode, a clock fault is one typical cause of an external alarm. This can be due to:

- complete loss of the X24/V11 interface, due to a failure of the transmission line;
- a failure of the transmission device connected to the codec;
- an incorrect clock frequency (i.e. incompatible with the codec configuration).

On the other hand, in case of a decoder alarm with no clock error, possible causes are :

- lack of signal received from the X24/V11 interface, due to a failure of the transmission device connected to the codec, or a transmission failure in the network ;
- a fault in the remote codec, or else the remote codec has an incompatible configuration ;
- transmission errors causing erratic alarms.

Errors such as "AES error" and "AES sync loss" can frequently be seen, even when the unit is configured to use the analog inputs. This is because the AES output is always active, and by default "genlocked" to the AES input. To avoid such undesired alarms:

☞ *When not using digital audio interfaces, set the digital audio sync in "Master" mode (SETUP / Audio / Digital / Synchro / Maste)*

4. Detailed operating mode – User interface

In local mode, the unit is operated thanks to a keyboard and display on the front panel. The display is an alphanumeric backlit LCD with two 20-character lines.

Operating from the keyboard can be protected by a password (8 digits maximum). In such case, the password must be entered to start a session and get access to the user menus. The password can be changed or deleted by the user.

4.1. Main operation modes

There are two parameters which have a major impact on the operation of the unit and on the user interface.

First, the unit features three transmission modes: transmission over Ethernet/IP, “leased line” mode, and transmission over the ISDN.
In comparison with the permanent leased line connection, IP and ISDN modes are “dial up” modes and bring a number of additional parameters to be controlled:

- dial number and/or full SIP URI for the destination of a call;
- call set up and control;
- device SIP registration data, or local ISDN number and sub-address;
- miscellaneous network operation parameters.

The status display is slightly different in order to recall the transmission mode currently in use.

Second, in the leased line mode the unit can be operated either as a normal “single codec”, or as a “dual codec” capable to transmit two independent 7 kHz bandwidth audio channels. This aspect has a big influence on the way the device is installed, set up and monitored.

In the following, the main operation modes are shortly designated as: “IP mode”, “ISDN mode” or “LL mode” (for leased line mode), and “Single codec” or “Dual codec”.

4.2. Equipment start-up

During start-up, the unit displays temporary messages. This initialisation lasts around one minute. Then the main menu is displayed.

At this stage, if the configuration includes a non-blank password, the keypad is locked and the password must be entered in order to access the menus: just enter the password (1 to 8 numbers), and the unit is unlocked as soon as the last digit is entered. On factory setting or after erasure of the unit memory, the password is blank so this step is skipped.

4.3. Description of the keyboard

Please refer to the description in 3.2.1 above(Front panel, LCD and basic control keypad).

4.4. Description of the menus

The unit features a tree-structured menu, and the three function keys on the bottom of the LCD are used to navigate through the menus. The OK key is used to confirm some settings or enter data, and the “Esc” key allows to go back to the upper menu level. Pressing this key several times makes sure you come back to the main default menu.

From the top menu, you can directly enter one of the three main menus by hitting the function key just beyond:

- TOOLS: maintenance and housekeeping functions, and access to status information
- DIR: access to the directory (IP mode only)
- SETUP configuration of the codec

This “SETUP” menu is itself divided into three sub-menus:

- Net selection and configuration of network interface and parameters
- Audio configuration of audio interfaces and parameters
- Cod selection and configuration of the coding algorithm

The following diagrams show the various sub-menus and accessible parameters.

Note that the “*” character in these diagrams shows the default and/or factory reset value for a given parameter.

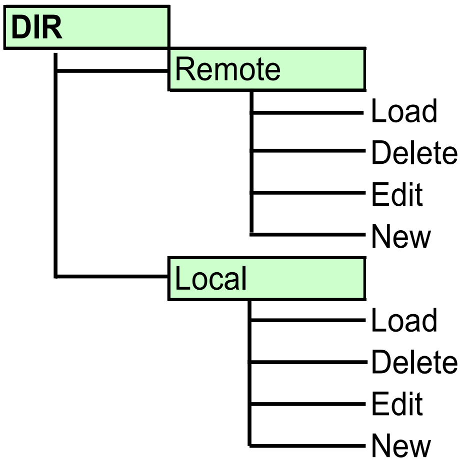
4.4.1. TOOLS menu

Notes

- **TOOLS**
  - Status
    - <Alarm 1>
    - …
    - <Alarm n>
  - Maintenance
    - About
      - IP ADDRESS — *displays IP address*
      - VERSION — *displays version info*
      - SOFTWARE NS — *displays "software" serial number*
      - SIP REGISTRAR — *displays SIP registrar's address, if any*
      - REGISTRAR STATUS — *displays status (OK or NOK)*
    - Ethernet Setup
      - DHCP
      - Manual
        - IP Address
        - Mask
        - Gateway
        - DNS
    - User Access — *switch Full ↔ Reduced mode*
      - Confirm
    - Password — *blank after "factory reset"*
      - Enter new password
        - Confirm password
    - Tests — *
      - No test loop
      - Audio loop
      - Loop3-codec
      - Loop2-Network
      - Audio Feedback
  - Misc
    - General Reset
      - CONFIRM
        - Yes
        - No
    - Aux. Functions
      - DATA CHANNEL
        - Off — *
        - On
      - RELAYS
        - Off — *
        - On
      - COORD. CHANNEL
        - Off — *
        - On
    - LCD Backlight
      - Off
      - On — *
    - LED Brightness
      - Low
      - Medium — *
      - High
    - Language
      - ENGLISH — *
      - FRENCH
      - …
    - 5A System — *concerns ISDN mode only!*
      - Off
      - On — *

4.4.2. DIR menu

Notes



This menu is not implemented for the moment.

4.4.3. SETUP menu

- **SETUP**
  - Net
    - Other [to select other network]
    - Param [for Leased Line mode]
      - Interface — *Selection of the active port in normal mode*
        - LL1 * — *Main port, labelled "X24/V11/V35"*
        - LL2 — *Secondary port, labelled "ALARM + X24/V11"*
      - Mode
        - Normal *
        - DualCodec
    - Param [for IP transmission mode]
      - Network Setup
        - DHCP *
        - Manual
          - IP Address ↕ Mask, Gateway, DNS
      - Network Quality — *Selection of the preferred stability / latency compromise (see "Setting up a link in IP mode")*
        - Low ↕ Middle * ↕ High
      - STUN Mode — *Activate/disable use of STUN*
        - OFF *
        - ON
    - Param [for ISDN]
      - Protocol — *Selection of local ISDN protocol*
        - ,,, ↕ Euro ISDN * ↕ ,,,
      - Address — *Configuration of local line (see "Setting up a link in ISDN mode")*
        - Local Number 1
        - Sub-Address 1
  - Audio
    - Analog
      - Input Impedance
        - High *
        - 600 Ohm
      - Max Output lvl — *Defines the maximum analog level reached for a maximum level decoded signal (0 dBFS)*
        - 0 dBu ↕ … ↕ +16 dBu * ↕ … ↕ +22 dBu
      - Max Input lvl — *Defines the analog input clipping level, sending a maximum level signal to the encoder (0 dBFS)*
        - 0 dBu ↕ … ↕ +16 dBu * ↕ … ↕ +22 dBu
    - Digital
      - Synchro
        - Master
        - Genlock *
      - Sampling — *Note: valid for "Master" mode ("Genlock mode follows AES input)*
        - 32 kHz ↕ 48 kHz * ↕ 96 kHz
    - Input Source
      - Analog *
      - AES
    - Level Meter /HEADROOM — *Refer to "Audio monitoring" section for this setting*
      - 0 dB ↕ … ↕ 12 dB * ↕ … ↕ 40 dB
  - Cod → ***Refer to "COD" sub-menu***

4.4.4. COD sub-menu

Notes

- **SETUP**
  - Cod
    - [Previous 1]
    - [Previous 2]
    - [Previous 3]
    - [Previous 4]
    - [Previous 5]
    - Other
      - G711 — *Not available in LL mode*
      - G722
        - SRT
        - H242 — *H242: only for ISDN mode*
        - H221
      - CELP — *Only for IP mode*
      - 4SB ADPCM
        - MONO
        - STEREO
      - L2 J52 — *Only for ISDN mode, MPEG L2 with J52*
        - CHANNEL MODE
          - MONO
          - DUAL MONO
          - STEREO
          - JOINT STEREO
        - SAMPLING RATE
          - 16 kHz
          - 24 kHz
          - 32 kHz
          - 48 kHz
        - BIT RATE
          - 64 kb/s
          - 128 kb/s
          - 192 kb/s
          - 256 kb/s
          - 384 kb/s
        - ERROR CORRECTION
          - MODE 0
          - MODE 1
          - MODE 2
          - MODE 3
      - MPEG L2
        - [same parameters as L2 J52] — *but max 128 kb/s in ISDN mode*
        - ,,,
      - TDAC — *OPTIONAL*

☞ *Important notice: limitations in bit rate depend on the transmission mode, the equipment version and the software version. This is especially true for MPEG coding.*

4.5. Handling the configuration profiles

To be added later

4.6. Setting up a link in IP mode

A link is set up in a similar way as an ISDN link. The difference is mainly that instead of the telephone number, we use either an IP address, or a SIP URI (Uniform Resource Identifier).

4.6.1. Directly call an IP address

This is the most basic way of setting the link. It is suitable only if:

- The other unit is "directly" reachable, i.e. there is no NAT Router or firewall blocking the connectivity. The simplest case is when both units are on the same LAN.
- The IP address of the other unit is known.

To set the link, first set the desired encoding format (MENU / SETUP / Cod).

Then enter the IP address and press the "green phone" button.

☞ *When operating in this way, it is preferable to leave blank the SIP registering data.*

4.6.2. Calling via a SIP server

This is the technique when both units are registered on a SIP proxy server. In this case, each unit is identified by its SIP URI, in the form username@sipservername, like an email address. There is no need to know any IP address (and hence there is no problem if the IP address of a unit changes for whatever reason).

To set the link, first set the desired encoding format (embedded server, or simply from the keypad MENU / SETUP / Cod).

Then enter the SIP URI of the unit to call, and press the "green phone" button.

4.6.3. Receiving calls

This is very simple, in both cases (direct peer to peer link or 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 unit will "follow" the calling unit.

4.6.4. "Network quality" setting

Depending on the quality of service provided on the network, especially its jitter performance, it is possible to change the stability/latency compromise used by the Scoop 4+. For this purpose, a setting is available in the menu (SETUP / Param / Network Quality). Three choices are proposed:

- "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 (and default) 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.

On a LAN and/or private network with a controlled quality, the "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 switch to the “LOW” setting if too much audio disturbance is heard.

4.6.5. Links with IP phones

SCOOP 4+ is compatible with IP phones that use the SIP protocol (many on the market do). The algorithm used in this case is G711, but a few IP phones can also accept G722.

Note that “IP phones” include software SIP phones implemented on computers.

4.6.6. Notes about the keypad

- Use the “up arrow” key to switch between low case and capital letters.
- Once a number or SIP URI has been called, it is easy to recall it without having to type it again: press the “green phone” key, then you can scroll through the “history” (last dialled numbers) using the arrows. Press the phone key when the desired number is displayed. This is especially useful for quickly redialling the previous number or URI.

4.7. Setting up a link in ISDN mode

4.7.1. Preliminary setup

The network interface must be configured depending on the local ISDN line that is used.

Protocol

First, the protocol should be set appropriately (SETUP / Net / Param / Protocol). The default setting is “Euro ISDN”, also known as ETSI protocol. Change this setting if another protocol is needed in your location.

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 Scoop 4+ 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!*

4.7.2. Call an ISDN number

To set the link, first set the desired encoding format (MENU / SETUP / Cod).

Then enter the ISDN number of the destination and press the “green phone” button.

If a sub-address is needed, after the number enter the “*” character and the sub-address (4 digits max.). The number then has the form nnnn*ssss, e.g. 0912345678*32.

When the selected encoder needs two B channels, the units asks for a second number. If the same number is suitable, just press a second time the “green phone” button.

An error code is displayed in case of a failure of the link establishment. Refer to annex 6.4 (ISDN error causes) to find the corresponding meaning.

4.7.3. Receiving calls

When the 5A System is active, receiving calls is simple. When a call is received, the codec automatically “unhooks” and the units negotiate automatically a commonly acceptable coding algorithm, and finally set the link automatically. On the receiving side, the unit will “follow” 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. *However, if J52 is used by both parties, the link will succeed even without 5AS active.*

4.7.4. Quick redialling

Once a number (or a couple of numbers) has been called, it is easy to recall it without having to type it again: press the “green phone” key, then you can scroll through the “history” (last dialled numbers) using the arrows. Press the phone key when the desired number is displayed. This is especially useful for quickly redialling the previous number.

4.8. Erasing and resetting the configuration

In some cases like e.g. if the password is forgotten, it may be necessary to restart from the factory default setting.

To erase the entire configuration and load the factory default settings, you should normally go to the “General Reset” sub-menu (TOOLS / Misc / General Reset).

☞ ***In case the keyboard is locked and you have lost the password, hold the “hang up” key pressed for 10 seconds; this forces the system directly to the “General reset” sub-menu. You can then confirm that you really want to erase all the settings, including the password…***

The factory default password is blank.

5. Technical characteristics

5.1. Characteristics of interfaces

5.1.1. Analogue audio inputs

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

The inputs are balanced type, using 3-pin female XLR sockets.

Maximum input level:	adjustable from 0 to +22 dBm ± 0.3 dB
Nominal input impedance:	600 Ω or 10 kΩ (menu setting, SETUP / Audio / Analog / Input Impedance)
Impedance balance:	*TBD*
Common mode rejection ratio:	> *TBD* dB (measured with Z = 600 Ω)

5.1.2. Analogue audio outputs

Audio characteristics are measured over a 20 to 20 000 Hz bandwidth except when differently stated. The outputs are balanced type, using 3-pin male XLR sockets.

Maximum output level:	adjustable from 0 to +22 dBm ± 0.3 dB.
Nominal load impedance:	600 Ω or 10 kΩ
Output impedance:	<50 Ω
Symmetry:	> *TBD* dB (Z_L = 150 Ω)

5.1.3. Digital audio input and output

These interfaces comply with recommendation AES3-1992. They support a sampling rate from 28 to 96 kHz.

5.1.4. Headphone output (front panel)

This output (6.35 mm jack on front panel) is for the connection of a 32 Ω headphone. It is also possible to plug a high impedance headphone; however, the maximum available power will be lower.

5.1.5. Main X24/X21/V11/V35 interface

The X24/V11 interface uses a 15-pin male Sub-D connector. The following table shows the pinout.

Signal		Pin		Signal	
Frame ground		1			
Transmitted data	Ta	2	9	Tb	Transmitted data
		3	10		
Received data	Ra	4	11	Rb	Received data
Indication	Ia	5	12	Ib	Indication
Received clock	Sa	6	13	Sb	Received clock
		7	14		
Electrical ground		8	15		

The codec does not transmit a C signal.

The codec can also be connected to V35 interfaces; a specific adaptation cable is needed in such case. The connection is described in Annex 6.5, (V35 interface adaptation).

5.1.6. "Alarm + X24/X21" interface

This interface uses a 15-pin male Sub-D connector. The following table shows the pinout.

Signal		Pin		Signal	
Internal alarm - Common	**IA-Com**	1			
Transmitted data	Ta	2	9	Tb	Transmitted data
Internal alarm - NC	**IA-C**	3	10	**IA-O**	**Internal alarm - NO**
Received data	Ra	4	11	Rb	Received data
Indication	Ia	5	12	Ib	Indication
Received clock	Sa	6	13	Sb	Received clock
External alarm - NO	**EA-O**	7	14	**EA-C**	**External alarm - NC**
Electrical ground		8	15	**EA-Com**	**External alarm - Common**

The bold text refers to the alarm contacts. Both are form-C type. The "NO" terminal is open when the alarm is set, otherwise it is connected to the "Common" terminal. The "NC" terminal is connected to the "Common" terminal when the alarm is set, otherwise it is open.

The current and voltage handling capabilities of the relays are:

Maximum output current: 120 mA

Maximum output voltage: 350 V peak

Resistance of output loop: < 35 Ω

The codec does not transmit a C signal.

The codec can also be connected to V35 interfaces; a specific adaptation cable is needed in such case. The connection is described in Annex 6.5, (V35 interface adaptation).

5.1.7. Remote control interface

This interface uses a 9-pin female Sub-D connector on the rear panel. This is a V24/RS-232 type interface with only Tx and Rx signals (no flow control). The following table indicates its pinout (DCE type pinout).

Pin		Function	
2	Rx	V24 data to the PC	Output
3	Tx	V24 control data, from the PC	Input
5		Ground	
Other		Not connected	

The interface is configured as follows: 4800 bauds, 8 bits, no parity, one stop bit.

5.1.8. Data interface (« data »)

This V24 interface uses a 9-pin female Sub-D connector on the rear panel. Like for the remote control interface, only Tx and Rx are used, there is no flow control, and the pinout is of DCE type.

Pin		Function	
2	Rx	Received V24 data	Output
3	Tx	Transmitted V24 data	Input
5		Ground	
Other		Not connected	

The data interface is configured as follows: 8 bits, no parity, one stop bit. It is possible (see 4.4.1 TOOLS menu, TOOLS / Misc / Aux. Functions / DATA CHANNEL) to activate the interface and to configure its baud rate (300 to 9600 bauds). However, the maximum allowed baud rate depends on the audio coding used (see 2.6.1 - Data channel).

5.1.9. Ethernet interface

This RJ45 socket has standard Ethernet pinout (for use of a normal “straight” cable to an Ethernet hub).

The installation and operation of this function is detailed in 3.5, Initial setup of the Ethernet interface.

5.1.10. “Digital I/O” interface

TBD

5.1.11. Relay transmission interface ("AUX" socket)

The relay transmission interface (refer to 2.6.2, Relay transmission) is available on the 25 pin female sub-D "AUX" Socket. It includes two isolated current loop inputs and two dry contact outputs.

The following table shows the pinout of the socket for this function:

Pin	Function
13	Output loop n°2 (b)
25	Output loop n°2 (a)
12	Output loop n°1 (b)
24	Output loop n°1 (a)
11	Input loop n°1 (b)
23	Input loop n°1 (a)
10	Input loop n°2 (b)
22	Input loop n°2 (a)
9	+5V of internally supplied power supply
21	0V of power supply

All loops are 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:	~ 560 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 (output):	350 V peak
Maximum switching current (output):	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 300 mA current consumption), or e.g. to power the input loops or LED indicators connected to the output loops.

5.1.12. Coordination channel interface (“AUX” socket)

In addition to the loop control and relay transmission interfaces, the (optional) coordination channel input and output are available on the 25-pin female sub-D connector (“AUX” Socket on the rear panel), with pinout as indicated hereunder.

The input and output are balanced floating signals, transformer isolated.

Maximum level:	9 dBm
Impedance:	600 Ω
Nominal bandwidth:	300 – 3400 Hz

Pin	Function
1	Coordination channel output (-)
14	Coordination channel output (+)
2	Frame ground
15	Coordination channel input (+)
3	Coordination channel input (-)
16	Frame ground

5.1.13. Loop control interface ("AUX" socket)

This function is not yet implemented

The following table shows the wiring of the socket for this function:

Pin	Function
17	Input loop n°2 (a)
5	Input loop n°2 (b)
18	Input loop n°1 (a)
6	Input loop n°1 (b)
19	Output loop n°2 (a)
7	Output loop n°2 (b)
20	Output loop n°1 (a)
8	Output loop n°1 (b)
21	0V of power supply
9	+5V of internally supplied power supply

All loops are 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:	~ 560 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 (output):	350 V peak
Maximum switching current (output):	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 300 mA current consumption), or e.g. to power the input loops or LED indicators connected to the output loops.

AETA AUDIO SYSTEMS

5.2. Audio performance

The audio performance in this part applies to the system without coding/decoding, and excluding the coordination channel. The additional effect of the audio encoding and decoding on audio performance depends on the coding algorithm used and its parameters.

Except when differently stated, the following measurements are done at a +6 dBm input level and on the AD/DA path, with maximum input and output level set at +16 dBu.

5.2.1. Transmission gain

The drift in time of the gain from the input to the output of the codec is less than ± 0.3 dB.

5.2.2. Amplitude-frequency response

All measurements are done with a +6 dBm input signal, and a reference frequency of 1020 Hz. The measurements are done with a loopback before coding/decoding, so the possible effect of compression has no influence.

To be detailed

5.2.3. Group delay distortion

Taking the minimum group delay as reference, the group delay distortion on the AD/DA path is always less than 1 ms.

5.2.4. Idle channel noise

Background noise is measured with no audio modulation (idle channel), with maximum input and output level set at +16 dBu, through the whole encoder-decoder chain (wide band coding, with 48 or 32 kHz coding frequency).

Maximum noise level[10]: (quasi-peak detection, CCIR weighting)	- 56 dBm (or - 62 dBq0ps)

This result in a signal to noise ratio (SNR) of more than 72 dB.

When the maximum input and output level is set at another level, both the signal and noise levels are shifted but the SNR remains in the same range.

5.2.5. Total distortion vs. frequency and level

Total distortion relative to maximum level (or THD + N) is less than –82 dB over the whole audio bandwidth (20 – 20 000 Hz). This performance holds for audio signals from –80 dB to –1 dB relative to the maximum level (+16 dBu).

5.2.6. Crosstalk

Crosstalk is less than -80 dB over the whole bandwidth.

[10] Worst case for all types of algorithms; MPEG performs better than the others

5.2.7. Gain and phase difference between channels

The gain difference between channels is less than ± 0.3 dB over the whole bandwidth, for any sampling frequency.

The phase difference between channels is less than ± 3 degrees over the whole bandwidth, for any sampling frequency.

5.3. Power supply

The codec operates from mains 85-263Vac, 47-63 Hz. Protection is provided by a resettable fuse. The maximum power consumption is 15 W (without the coordination channel option).

5.4. Dimensions and weight

The unit is a 19 inches frame of 1U height (44 mm or 1.75") and 252 mm depth (12.5").

Its weight is about 3.2 kg.

5.5. 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 SCOOP 4+ complies with "CE" directives regarding safety and EMC.

- Safety: compliance with EN60950
- Susceptibility: compliance with EN50082-1
- EMI: radiated emissions complying with EN55022 (class A); conducted emissions complying with EN55022 (class B).

5.6. Versions - Options

The various versions available for the SCOOP 4+ are the following:

- SCOOP 4+ with Ethernet/IP and leased line interfaces
- SCOOP 4+ with Ethernet/IP, leased line interfaces and one ISDN interface

Besides, certain functions are available as options:

- Audio coordination channel ;

5.7. Accessories and related products

The SCOOP 4+ is delivered with a mains cord and a CAT5 Ethernet cable.

Along with the coordination channel option, a specific cable is delivered, which provides XLR plugs for the coordination channel input and output (input on a female plug, output on a male plug).

For remote controlling SCOOP 4+ units from a PC, the TeleScoop™ supervision software is available separately.

6. Annexes

6.1. *Complements on the algorithms and protocols used*

6.1.1. Auxiliary data in the MPEG frames

The auxiliary data are used for the following purposes:

- Reed-Solomon error detection and correction (J52 standard)
- Data channel
- Other auxiliary information: relay transmission, and/or auxiliary audio channel.
  The insertion of this auxiliary information is an extension (AAS proprietary format) to MPEG. However, the frame structure remains compliant.

6.1.2. Reed-Solomon encoding

In order to cope with possible transmission errors in the network, Reed-Solomon error correction coding can be added, compliant with J52 recommendation. Four correction modes are available in the SCOOP 4+:

- Mode 0 : no error correction, Reed-Solomon coding disabled
- Mode 1 : protection of only the control information and scale factors in the MPEG frame, low redundancy (so-called “unequal protection”)
- Mode 2 : protection of the whole frame, moderate (2.5 %) redundancy (so-called “low equal protection”)
- Mode 3 : protection of the whole frame, high (10 %) redundancy (so-called “high equal protection”)

Higher redundancy increases the protection against errors, but slightly degrades the audio quality, as redundancy takes up part of the bit rate that could be allocated to audio coding.

Most often, for a normal quality transmission link, mode 1 is sufficient and it consumes little bit rate from the compressed data, so it hardly impacts the audio quality. Although J52 does not apply to leased line connections, this error protection technique is also implemented in the SCOOP 4+ in leased line mode.

6.1.3. H221 framing

H221 defines a framing structure that allows byte synchronisation recovery in leased lines, and the transmission of control data along with the main data.

6.1.4. Proprietary coding algorithms

The “proprietary” coding algorithms are algorithms which are not standardised by the ITU-T but have distinctive features that make them useful for some applications:

- Low coding-decoding delay (4SB ADPCM);
- Low bit rate wide band speech coding (CELP)
- TDAC (Time Domain Aliasing Cancellation, MDCT-based algorithm)

6.2. Overview of the SIP protocol

6.2.1. What is SIP?

SIP is for Session Initiation Protocol, a protocol specified by the IETF for establishing media transmission sessions. SIP is considered the communication protocol of the future by most vendors, and as such it has deep influence on the VoIP applications.

As a signalling protocol, SIP brings methods and techniques to solve the issues related to the establishing of an audio link. Almost as important, it is a recognised standard, implemented on many network devices and systems. Using SIP helps you build modular and really evolutive systems, not being tied to a single vendor.

6.2.2. Setting a link with SIP

Let us look at an example (diagram below): a reporter on the move with a Scoopy[11] wants to make a call to a SIP compliant codec located in the main station. The reporter may be at home, or at another location, not necessarily known in advance.

Once the Scoopy is on and connected to the network, it will register itself ❶ to a SIP "registrar". This registrar can be located on the LAN of the radio house, but it may as well be elsewhere in the network. Then the registrar "knows" where the Scoopy is, what its IP address is. On the radio house side, a similar process takes place ❷.

To call the codec of the radio house (e.g. a SCOOP 4+), the reporter just needs to know its SIP address, which can be like studio12cod@radiomcr.com (indeed very similar to an e-mail address). To call the unit, the reporter has to select the preferred audio coding mode on the Scoopy (e.g. mono G722), then call the remote unit, simply using this SIP address (SIP URI).

What happens then on the network: the Scoopy sends the request ❸ (INVITE in SIP protocol) to a proxy server (often the same device is also the registrar). To make things simple, this proxy then relays and routes the request ❹ to its destination. Resolving the SIP URI to a physical network and address uses mechanisms similar to those used for resolving URLs. Several proxys may be involved in cascade to finally reach the desired destination, but this does not have to be known or dealt with by the end devices. The following is like the initiation of a phone call: the IP codec "rings"; if it accepts the call, this is notified to the Scoopy.

At this stage, the proxy(s) provide the Scoopy and the IP codec with all the address data they need for the link, then the actual audio streams can be exchanged ❺ between both units. As a very important feature, the end devices now can exchange data directly; the proxys do not have to be on the path, they are only involved in the setting (and later the ending!) of the session. The codecs will automatically exchange their coding capabilities, and agree on a coding mode, with no further user intervention.

Alternatively, the call can be done from the station to the reporter, in a way very similar to the above. In contrast with ISDN links, the operators at the station do not even need to know where the reporter is located! This is because the registrar deals with this issue.

Note that it is also possible to set a link with a SIP-compliant VoIP phone instead of another codec. This is one of the numerous advantages of using a standard.

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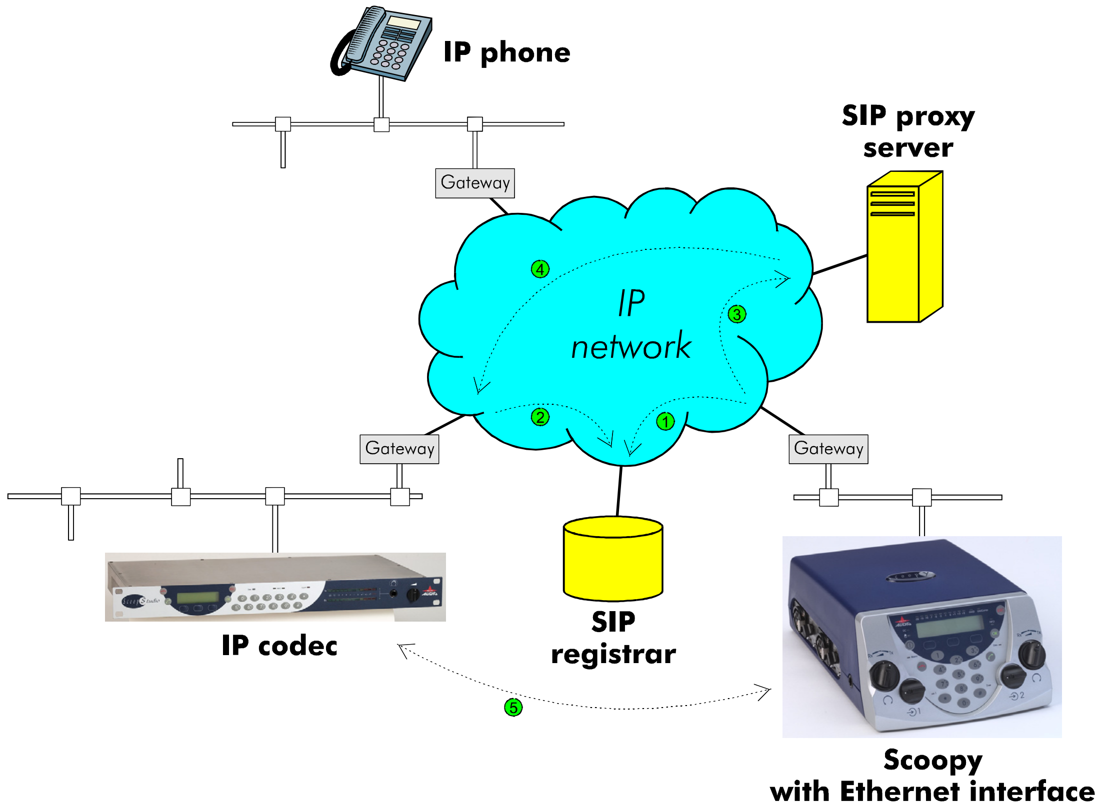
[11] Scoopy is a portable audio codec from AETA AUDIO SYSTEMS; the description here applies to both Scoopy and the Scoop 4+, as they are both SIP compliant and mutually compatible



6.2.3. Supported communication protocols

The SCOOP 4+ IP supports and/or implements the following protocols:

- Ethernet (IEEE802.1)
- IP
- DHCP (client)
- HTTP (embedded html server)
- TCP/IP, UDP/IP transport protocols
- SIP (Session Initiation Protocol), SDP (Session Description Protocol)
- RTP, RTCP

6.3. Some methods to deal with NAT routers and firewalls

This problem arises when the desired connection has to go through a NAT router, and/or a firewall, that blocks a direct IP communication.

This is a very common issue, especially if one needs to set up a transfer via the Internet. It is impossible here to describe in details the possible ways to deal with this problem, but the following just shortly discusses some typical solutions. One should decide the most suitable solution depending on the specific conditions. Most probably, a network administrator should be consulted for support, and for granting adequate network authorisations and/or privileges.

6.3.1. DMZ

The SCOOP 4+ can be set in the "DMZ" of the router/firewall. In this way, it is possible to reach directly the SCOOP 4+ from "outside" the router.

However, such a solution is not recommended for network security reasons, and it should only be used as a temporary test configuration.

6.3.2. Port forwarding

This solution exposes less directly the unit to external attacks. However, it is more complicated, as a number of dedicated ports have to be opened and routed.

6.3.3. Use of a SIP proxy

A SIP proxy often can deal with the issue. There are various options, e.g.:

- Proxy on the external "public" side: efficient when the router does not block outgoing traffic. Such solution can be implemented e.g. by registering on public SIP servers.
- Proxy inside the LAN area: efficient if the proxy server is allowed to receive and send dedicated traffic. This solution is compatible with restrictive firewalls. Port forwarding has to be set for the proxy to be able to receive calls from outside the LAN.

6.3.4. STUN server

In many cases, a STUN server can be used instead of a dedicated proxy server. STUN is a network protocol allowing a client behind a NAT router (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that might be behind NAT routers.

If a STUN server is available, a SCOOP 4+ located behind a NAT router can use the server to complete successfully its call setup with a remote unit outside the NAT router.

6.4. ISDN error causes

The following table lists the call clearing causes. The error message is typically CLEARED: hh (dd), where hh is an hexadecimal number and dd its decimal value. The message meaning is given for an ETSI ISDN. Causes with values greater than 80 hex are generated internally.

Code	Meaning
01 (1)	unallocated (unassigned) number
02 (2)	no route to specified transit network
03 (3)	no route to destination
06 (6)	channel unacceptable
07 (8)	call awarded and being delivered in an established channel
10 (16)	normal call clearing
11 (17)	user busy
12 (18)	no user responding
13 (19)	no answer from user (user alerted)
15 (21)	call rejected
16 (22)	number changed
1A (26)	non-selected user clearing
1B (27)	destination out of order
1C (28)	invalid number format
1D (29)	facility rejected
1E (30)	response to STATUS ENQUIRY
1F (31)	normal, unspecified
22 (34)	no circuit/channel available
26 (38)	network out of order
29 (41)	temporary failure
2A (42)	switching equipment congestion
2B (43)	access information discarded
2C (44)	requested circuit/channel not available
2F (47)	resources unavailable, unspecified
31 (49)	quality of service unavailable
32 (50)	requested facility not subscribed
39 (57)	bearer capability not authorized
3A (58)	bearer capability not presently available
3F (63)	service or option not available, unspecified
41 (65)	bearer capability not implemented
42 (66)	channel type not implemented
45 (69)	requested facility not implemented
46 (70)	only restricted digital information bearer capability is available
4F (79)	service or option not implemented, unspecified
51 (81)	invalid call reference value
52 (82)	identified channel does not exist
53 (83)	a suspended call exists, but this call identity does not
54 (84)	call identity in use
55 (85)	no call suspended
56 (86)	call having the requested call identity has been cleared

58 (88)	incompatible destination
5B (91)	invalid transit network selection
5F (95)	invalid message, unspecified
60 (96)	mandatory information element is missing
61 (97)	message type non-existent or not implemented
62 (98)	message not compatible with call state or message type non-existent or not implemented
63 (99)	information element non-existent or not implemented
64 (100)	invalid information element contents
65 (101)	message not compatible with call state
66 (102)	recovery on timer expiry
6F (111)	protocol error, unspecified
7F (127)	interworking, unspecified
91 (145)	no signaling data link establishment
A2 (162)	no line activation
FF (255)	call clearing, unspecified

6.5. V35 interface adaptation

This annex indicates the proper connections to use when interfacing the SCOOP 4+ codec to DCE equipment using a V35 interface.

6.5.1. Connection table

The three leftmost columns show the pin allocation on the 15-pin connector of the codec.

The two columns on the right indicate the pinout on a 34-pin V35 connector or a 37-pin sub-D connector. Consult the DCE documentation for other connector types.

Only the bold indicated signals need be connected; leave other pins unconnected. However, the frame ground (pin 1) may be used for connecting the shield of the connection cord.

X24 signals	**Pin**	**Signal direction**	**Function**	**V35 signal**		**Pin number**	
						34-pin connector	**37-pin connector**
G	**8**		**Signal ground**	**102**	**SG**	**B**	**19**
	15						
	7						
	14						
Sa	**6**	←	**Data clock**	**115a**	**RETA**	**V**	**8**
Sb	**13**	←	**Data clock**	**115b**	**RETB**	**X**	**26**
Ia	5	←					
Ib	12	←					
Ra	**4**	←	**Received data**	**104a**	**RDA**	**R**	**6**
Rb	**11**	←	**Received data**	**104b**	**RDB**	**T**	**24**
	3						
	10						
Ta	**2**	→	**Transmitted data**	**103a**	**TDA**	**P**	**4**
Tb	**9**	→	**Transmitted data**	**103b**	**TDB**	**S**	**22**
	1		Frame ground		FG	A	1