User manual for IQOYA SERV/LINK

Multichannel IP audio codec for distribution or remotes over IP networks

Applies from firmware version 4.12
February 2026

Date	What's new
February 2026	• Security: New IP addresses banning service (Fail2Ban) New default password based on serial number for SSH access • MPEG-TS decoding (optional licence) • RIST tunneling (optional Licence) • Path Delay explanation added in the parameters of the received IP stream.

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1 SAFETY INFORMATION

Important Safety Information: read carefully before using this equipment!
Follow these instructions and keep them in a safe place! Keep in mind that damages due to failure to observe the instructions contained in this manual are not covered by warranty.

Instructions importantes de sécurité: lire soigneusement avant d'utiliser l'équipement!
Lisez et suivez ces instructions. Conservez-les pour consultation ultérieure! Les dommages dus au non-respect des instructions contenues dans ce manuel ne sont pas couverts par la garantie.

Wichtige Sicherheitshinweise: vor Inbetriebnahme des Gerätes sorgfältig lesen!
Befolgen Sie die Anweisungen und bewahren Sie sie für spätere Fragen auf! Bei Schäden, die durch Nichtbeachten dieser Bedienungsanleitung verursacht werden, erlischt der Garantieanspruch!

Throughout this manual, the lightning bolt triangle is used to alert the user to the risk of electric shock.

The exclamation point triangle is used to alert the user to important operating or maintenance instructions.

Do Not Open the Cabinet
There are no user-serviceable components inside this product. Opening the cabinet may present a shock hazard, and any modification to the product will void your warranty. If it is necessary to open the device for maintenance or advanced configuration purposes, this is to be done by qualified personnel only after disconnecting the power cord and network cables!

Power supply
The device is to be connected only to a power supply as specified in this manual and marked on the equipment.
This equipment must be earthed!
Do not block any of the ventilation openings!

Humidity
To reduce the risk of fire or shock, do not expose this device to rain or moisture. Do not place objects filled with liquid on this device.

Installation Location
To ensure proper operation and to avoid safety hazards, the device must be installed in a 19“ rack mount chassis. The electrical installation of the building should dispose of easily accessible disconnecting means in the immediate vicinity of the device.
If rack installation is not possible, place it on a firm and level surface. The use of a supply lead with a power plug respecting the legal standards in the country of use is obligatory. The plug shall be easily accessible in case of a problem.
Avoid installation in extremely hot or cold locations, or in an area that is exposed to direct sunlight or heating equipment. Avoid moist or humid locations.
Connection of this product to an IT power supply system is only in Norway.

Cleaning
Clean only with a soft, dry cloth. If necessary, after disconnecting the unit's cables, wipe it with a soft cloth dampened with mild soapy water, then with a fresh cloth with clean water. Wipe dry immediately with a dry cloth. NEVER use benzene, aerosol cleaners, thinner, alcohol or any other volatile cleaning agent. Do not use abrasive cleaners, which may damage the finish of metal or other parts.

Refer all servicing to qualified service personnel.
Servicing is required when the apparatus has been damaged in any way, such as power supply cord or

Ne pas ouvrir l'appareil
L'ouverture du coffret peut produire un risque de choc électrique, et toute modification du produit annule votre garantie. S'il est nécessaire d'ouvrir l'appareil pour l'entretien ou la configuration avancée, cela doit être fait par du personnel qualifié, après avoir débranché le cordon d'alimentation et les câbles réseaux !

Alimentation
Il est primordial de connecter l'appareil à une alimentation électrique telle que spécifiée dans ce manuel d´utilisateur et sur le matériel même.
Cet équipement doit être raccordé à la terre !
Ne pas obstruer les ouvertures de ventilation !

Humidité
Afin de réduire les risques de feu ou de choc, n'exposez pas cet appareil à la pluie ou l'humidité. Ne placez pas d'objet contenant un liquide sur l'appareil.

Installation, mise en place
Afin d'assurer le fonctionnement correct et de minimiser les risques potentiels liés à la sécurité, l'appareil doit être installé dans un châssis 19 pouces. Si cela ne vous est pas possible, placez-le sur une surface solide et plane. Prévoir dans l'installation électrique du bâtiment un dispositif de sectionnement aisément accessible et à proximité immédiate de l'appareil.
L'utilisation d'un câble d'alimentation avec une fiche de prise de courant respectant les normes en vigueur dans le pays d'utilisation est obligatoire. De plus, la fiche de prise de courant doit être aisément accessible en cas de problème.
Évitez une installation dans des endroits très chauds ou très froids ainsi que dans des lieux exposés directement au soleil. Évitez les lieux présentant un excès d'humidité.
Le raccordement de ce produit à un régime d'alimentation IT n'est possible qu'en Norvège.

Nettoyage
Nettoyez uniquement avec un chiffon doux et sec. Si nécessaire, après avoir débranché le cordon d'alimentation, essuyez-le avec un chiffon doux humidifié avec de l´eau savonneuse puis rincez-le à l´aide d' un chiffon propre et d'eau claire.
Séchez-le immédiatement avec un chiffon sec.
N'utilisez JAMAIS d'essence, de nettoyants en aérosols, d'alcool ou tout autre agent nettoyant volatile. N'utilisez pas de produits nettoyants abrasifs qui pourraient endommager les finitions métalliques ou

Gerät nicht öffnen
Öffnen des Geräts kann eine Gefährdung durch Stromschlag und Erlöschen der Garantie zur Folge haben. Reparaturarbeiten und Änderungen der Hardwarekonfiguration dürfen nur von qualifiziertem Personal nach entfernen der Strom- und Netzwerkkabel durchgeführt werden.

Stromversorgung
Das Gerät darf nur mit der in dieser Bedienungsanleitung und auf dem Gerät angegebenen Stromversorgung betrieben werden.
Erdung ist zu gewährleisten!
Belüftungsschlitze nicht verdecken!
Wasser und Feuchtigkeit
Um Brand- oder Stromschlagrisiken zu vermeiden, darf das Gerät nicht mit Feuchtigkeit in Berührung kommen.

Aufbau des Geräts
Um den einwandfreien Betrieb zu gewährleisten und Sicherheitsrisiken zu vermeiden, sollte das Gerät in einem 19-Zoll Baugruppenrahmen montiert werden. Die elektrische Installation des Gebäudes sollte über einen leicht zugänglichen Trennschalter in unmittelbarer Nähe des Geräts verfügen Nur wenn die Installation im Rack nicht möglich ist, stellen Sie das Gerät auf einen festen, waagerechten Untergrund.
Die Verwendung eines Anschlußkabels und eines Steckers, die die im Benutzungsland gültigen Normen erfüllen, ist obligatorisch. Des weiteren muß die Steckdose für einen eventuellen Problemfall leicht zugänglich sein.
Meiden Sie Standorte in der Nähe von Wärme- oder Feuchtigkeitsquellen sowie direkte Sonneneinstrahlung.
Anschluß dieses Produktes an eine spezielle IT-Stromversorgung ist nur in Norwegen genehmigt.

Reinigen des Geräts
Säubern Sie das Gerät nur mit einem weichen, trockenen Tuch. Bei Bedarf verwenden Sie ein mit mildem Seifenwasser befeuchtetes Tuch, nachdem Sie die Netzanschlusskabel aus der Steckdose gezogen haben, anschließend ein weiches, mit klarem Wasser befeuchtetes Tuch. Trocken Sie das Gerät sofort im Anschluß. Keinesfalls Benzol, Verdünner oder sonstige starke Lösungsmittel oder Scheuerreiniger verwenden, da hierdurch das Gehäuse beschädigt werden könnte.

Lassen Sie etwaige Reparaturen nur von qualifizierten Fachleuten durchführen!
Sollten das Netzkabel oder der Netzstecker beschädigt

plug is damaged, liquid has been spilled, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped. **Moving the device** Before moving the unit, be certain to disconnect any cables that connect with other components.	d'autres pièces. **Réparation** Lorsque l'appareil a été endommagé quelle qu'en soit la cause ou qu'il ne fonctionne pas normalement, toute réparation doit être effectuée par du personnel qualifié. Avant de transporter l'unité, assurez-vous d'avoir bien déconnecter le cordon d'alimentation ainsi que tous les câbles le reliant à d'autres appareils.	sein, oder sollte das Gerät selbst beschädigt worden sein (z. B. durch Eindringen von Feuchtigkeit durch Fall auf den Boden), oder sollte es nicht ordnungsgemäß funktionieren oder eine deutliche Funktionsabweichung aufweisen, so ist es von qualifizierten Fachleuten zu reparieren.

INFORMATION FOR THE USER

"This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense."

Warning
This is a class A product. In a domestic environment this product may cause radio interference in which case the user may be required to take adequate measures.

CAN ICES-3 (A) / NMB-3 (A)

User safety	**EMC**
European Directive *2006/95/EC "Low Voltage Directive* ***Europe****: EN60950-1 (2006+A11/2009+A1/2010+A12/2011+A2/2013)* ***International****: IEC 60950-1 (2005+A1/2009+A2/2013)*	***European Directive****: EMC 2004/108/EC* ***Radio disturbance*** *:* *International: CISPR22 (2008) Class A* *IEC 61000-6-3 (2006+A1/2010)* *European : EN55022 Class A (2010) Requirements for Information Technology Equipment (ITE)* *EN 61000-6-3 (2007+A1/2011)* ***Immunity****: International : CISPR24 (2010)* *IEC 61000-6-2 (2005)* *European : EN55024 (2010) (ITE)* *EN 61000-6-2 (2005)* ***Harmonics****: International : IEC 61000-3-2 (2005 + A1/2008 + A2/2009)* *European : EN61000-3-2 (2006 + A1/2009 + A2/2009)* ***Voltage changes*** *: International : IEC 61000-3-3 (2013)* *European :EN 61000-3-3 (2013)* ***United States****: CFR 47, FCC Part 15, Subpart A (Class A Digital Device) & Industry Canada ICES-003 (Issue 5/2012)*
RoHS European directive 2011/65/EU aka "RoHS"	**Note: to comply with standard EN55024, use shielded network cables!**

In order to guarantee compliance with the above standards in an installation, the following must be done:

- the provided cables must not be modified.

- additional cables used must have their respective shield connected to each extremity.
- Attach a ground wire to the chassis (ideally the ground wire has a ring terminal). Connect the other end of the ground wire to a good electrical ground point.

The limits specified in the standards are designed to provide reasonable protection against harmful interference in an industrial installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation.
If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- reorient or relocate the receiving antenna.
- increase the separation between the equipment and the receiver.
- connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- consult the dealer or an experienced audio/television technician for help.

Note:
Connecting this device to peripheral devices that do not comply with CLASS A requirements or using an unshielded peripheral data cable could also result in harmful interference to radio or television reception. The user is cautioned that any changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment. To ensure that the use of this product does not contribute to interference, it is necessary to use shielded I/O cables.

2 INTRODUCTION

You have just acquired a Digigram IQOYA SERV/LINK and we congratulate you!

IQOYA SERV/LINK is a multi-channel IP audio codec of the IQOYA range based on Digigram FluidIP technology. The manual at hand will guide through installation, configuration, and operation. For any software related issue, please refer to the specific documentation provided in its online help.

IQOYA SERV/LINK features two modes of use (“Programs distribution” and ‘Remote Broadcasting’) that are described in this document.

3 KEY HARDWARE FEATURES

3.1 All IQOYA SERV/LINK models

- 1U rack
- Hot swappable redundant power supply units (220 VAC, optionally -48VDC)
- 2 x RJ45 10/100/1000 Mbits/s ports
- Front panel display and keypad

3.2 Models with analog and AES/EBU I/OS (SERV/LINK882, SERV/LINK16162)

- 8/8 or 16/16 balanced analog inputs/outputs, +24 dBu Max
- 4/4, 8/8 balanced stereo AES/EBU inputs/outputs
- AES11 synchronization input
- Word Clock synchronization input
- Video Reference synchronization (black burst)
- 1 Sub-D 62 pin connector and 1 associated breakout cable per group of 8/8 mono analog channels and 4/4 stereo AES/EBU stereo channels

3.3 Models with AES67 connectivity

- from 8/8 up to 64/64 mono I/O channels on AES67 for SERV/LINK88-AES67
- from 72/72 up to 128/128 mono I/O channels on AES67 for SERV/LINK7272-AES67
- Synchronization on PTP, WordClock
- 2 Gbit/s R-J45 dedicated to AES67
  Note the previous version that supports 128 I/O channels through 4 Gbit/s R-J45 is replaced by a version with 2 Gbit/s Eth ports.
- 2 x RJ45 10/100/1000 Mbits/s Ethernet ports for WAN
- Option for 4 additional Eth ports.

3.4 Models with DANTE connectivity

- from 8/8 up to 128/128 mono I/O channels on DANTE
- Synchronization on PTP
- 2 Gbit/s R-J45 dedicated to DANTE, Primary port and Secondary ports
- 2 x RJ45 10/100/1000 Mbits/s Ethernet ports for WAN

- Option for 4 additional Eth ports.

3.5 Models for transcoding only

- from 8/8 up to 128/128 mono channels
- 2 x RJ45 10/100/1000 Mbits/s Ethernet ports for WAN
- Option for 4 additional Eth ports.

3.6 Discontinued versions

IQOYA SERV/LINK with MADI I/OS (base version: SERV/LINK88-MADI)

- from 8/8 up to 128/128 mono I/O channels on MADI
- Synchronization on MADI, Word Clock
- 2 x RJ45 10/100/1000 Mbits/s Ethernet ports

IQOYA SERV/LINK with AES67 and MADI connectivity

- from 8/8 up to 64/64 mono I/O channels on AES67 & MADI, for SERV/LINK88-AES67/MADI
- from 72/72 up to 128/128 mono I/O channels on AES67 & MADI, for SERV/LINK7272-AES67/MADI
- Synchronization on PTP, WordClock, MADI
- 2 Gbit/s R-J45 dedicated to AES67 (64 mono channels),
  or 4 Gbit/s R-J45 dedicated to AES67 (128 mono channels),
- 2 x RJ45 10/100/1000 Mbits/s Ethernet ports for WAN

Models with AES/EBU only I/OS (SERV/LINK881, SERV/LINK16161, 24241, 32321*)

- 4/4, 8/8, 12/12 or 16/16 balanced stereo AES/EBU inputs/outputs
- AES11 synchronization input
- Word Clock synchronization input
- Video Reference synchronization (black burst)
- 1 Sub-D 62 pin connector and 1 associated breakout cable per group of 4/4 AES/EBU channels

4 KEY SOFTWARE FEATURES

4.1 Standard features for distribution

- Simultaneous encoding, decoding, transcoding
- Multi-format encoding and multi-protocol streaming of each input.
- Streaming protocols: RTP, UDP, Icecast/Shoutcast, HLS multi-bitrate
- Supported Audio algorithms & streaming protocols
- Support of unicast, multi-unicast, multicast, multi-multicast addressing
- Support of IGMPv3
- MPEG-TS/IP SPTS and MPTS streaming, with DVB information tables, and insertion of associated program data (serial data and triggers). MPEG-TS encoding is included, whereas MPEG-TS decoding is optional.
- VLAN Tagging + DSCP
- Bonding of Eth interfaces: mode 1 (active-backup) and mode 4 (802.3ad passive mode)
- Support of DHCP
- Asymmetric algorithmic encoding/decoding
- 3 decoding priorities
- Automatic switching to a lower decoding priority in case of upper priority failure
- 3 decoding priorities per output program, with choice of the audio source on each priority: IP service (RTP, UDP, MPEG-TS, HTTP), file, playlist and Audio input
- Possibility to disable/enable any defined priority
- Possibility to stop streaming upon input silence detection with adjustable silence threshold and duration (for raw RTP streams).
- Decoding of a stereo source to a mono output, with possibility to mix left and right channels
- Dual port redundant streaming with time diversity up to 1 second
- Selectable FECs for ACIP RTP streams (from +15% to +100% IP bandwidth)
- Pro MPEG Cop#3 FEC for MPEG-TS streams (line, columns)
- Automatic audio format detection on the decoder for ACIP streams.
- Real-time metrics on network path quality (jitter, lost packets, duplicated packets, disordered packets) for the primary stream as well as for the FEC stream.
- Adjustable jitter buffer
- Management of lost packets, disordered packets, duplicated packets, and AAC error concealment
- RS232 tunnelling
- GPIO tunnelling
- WEB user rights management
- Banning of source IP addresses upon multiple faulty connection attempts
- NTP synchronization (date and time)
- Save / load full codec configuration
- Save / load audio configuration
- Remote firmware update
- Audio still active during firmware upload
- Firmware version N and N-1 locally stored
- SNMPv2c and v3 (SET, GET, Traps)
- WEB Service API

4.2 Optional features for distribution

- aptX^{TM} audio compression
- Optional audio synchronization on NTP clock
- Number of audio I/Os supported for MADI, AES67, and Dante base versions
- Number of audio buses for IP streams transcoding
- 1+1 redundancy
- MPEG-TS/IP SPTS and MPTS decoding
- RIST tunneling with encryption

4.3 Features for remote contributions

- Support of DHCP
- VLAN tagging
- Support for SIP signalling protocol
- Up to 64 full-duplex SIP communications (mono and/or stereo)
- Dual SIP registration
- Support of SIP, Direct SIP, and symmetric RTP
- Asymmetric algorithmic encoding/decoding
- Contact list management
- Call profiles management
- Automatic answering
- Automatic redialing
- Integration to IQOYA CONNECT (centralized configuration, control and monitoring)

4.4 Supported Audio algorithms & streaming protocols

Included	**Streaming protocols**
• PCM, 16/20/24-bit • ITU G.711/722 • Opus, Opus+INBANDFEC • AAC-LC, HE-AACv1, HE-AACv2, AAC-LD, AAC-ELD	• UDP, RTP, without MPEG-TS encapsulation
• ISO MPEG-1/2 Layer II, Layer III	• UDP, RTP (w/wo MPEG-TS encapsulation) Icecast/Shoutcast HLS
• AAC ADTS and ADTS-CRC flavours AAL-LC, HE-AAC, HE-AACv2, AAC-LD, AAC-ELD	• Icecast/Shoutcast HLS
Optional	
• Enhanced aptX^{TM} 16/24-bit	• UDP, RTP, without MPEG-TS encapsulation

5 PHYSICAL INTERFACES

5.1 Front Panel

5.1.1 LEDs

Power	Status	Network	Error
Off: IQOYA is not connected to the mains **Green**: IQOYA is started **Orange**: IQOYA is connected to the mains but not started.	**Green**: No issue **Red**: temperature or fan alarm	**Green**:Ok **Orange:** No connection detected on one of the two configured Eth ports. **Red**: No connection detected on all the configured Eth ports.	**Off:** No error **Red**: Audio engine error

5.1.2 LCD display and keypad

 Cancel settings

 Confirm settings

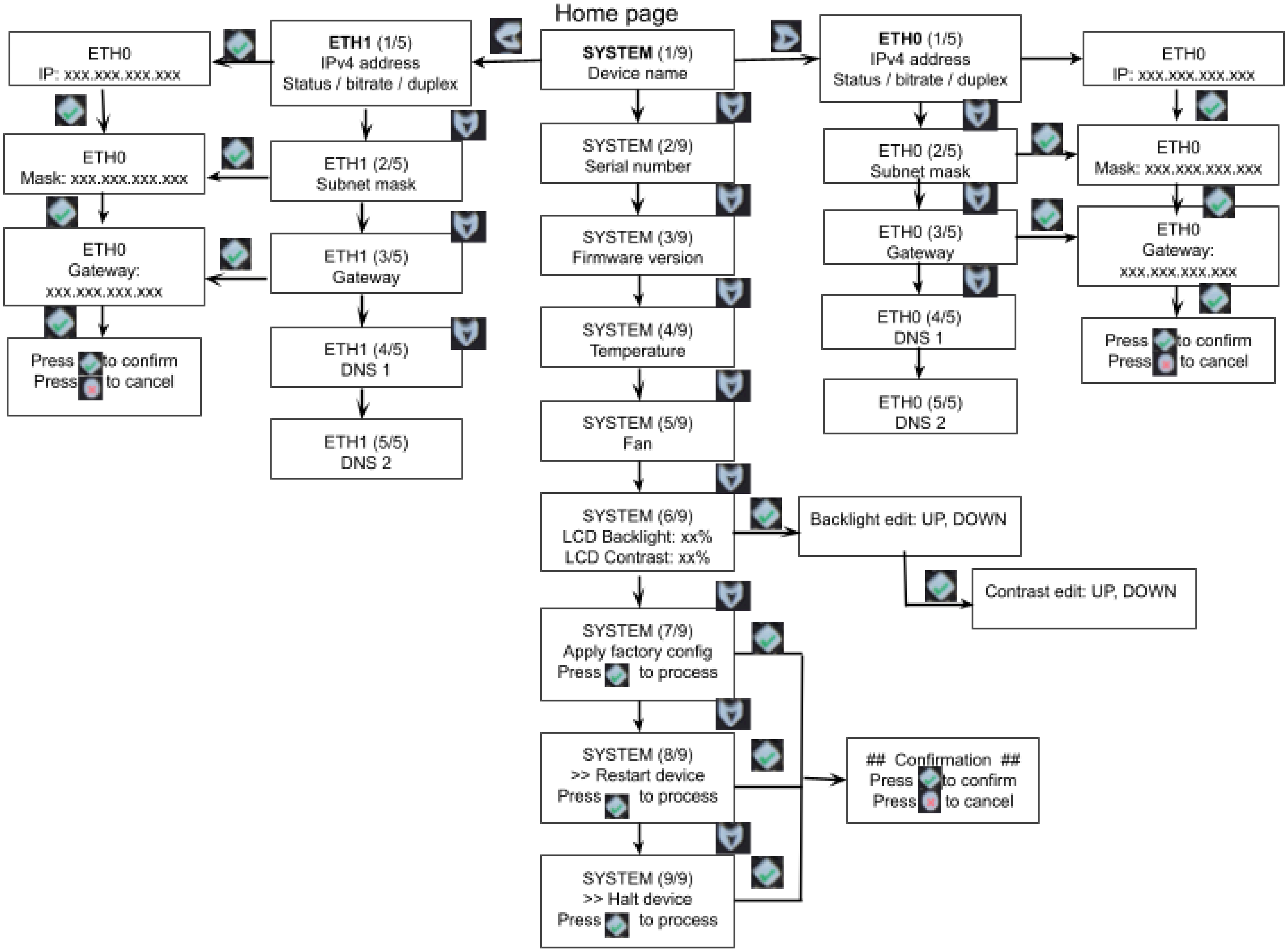
 Go to previous menu

 Go to next menu

 Go to previous item of current menu

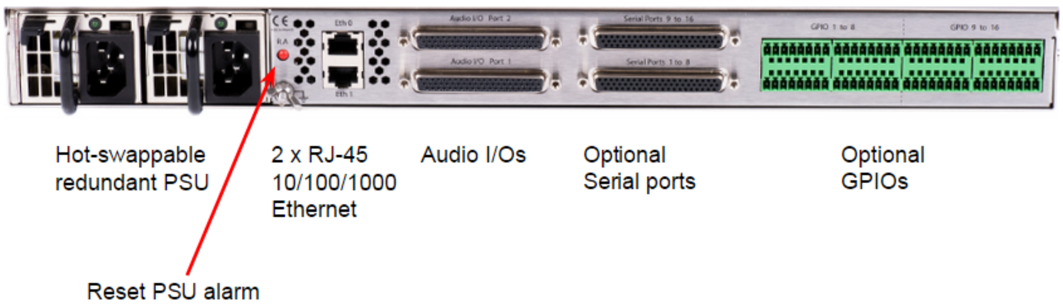
 Go to next item of current menu

5.1.3 Navigating menus on LCD display



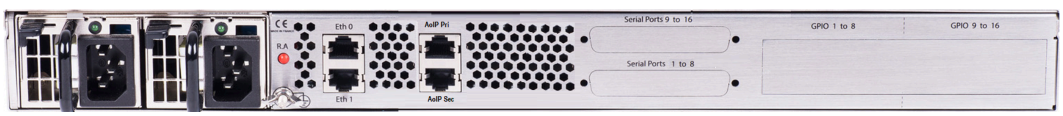
5.2 Back Panel

5.2.1 Analog & AES/EBU versions, AES/EBU versions



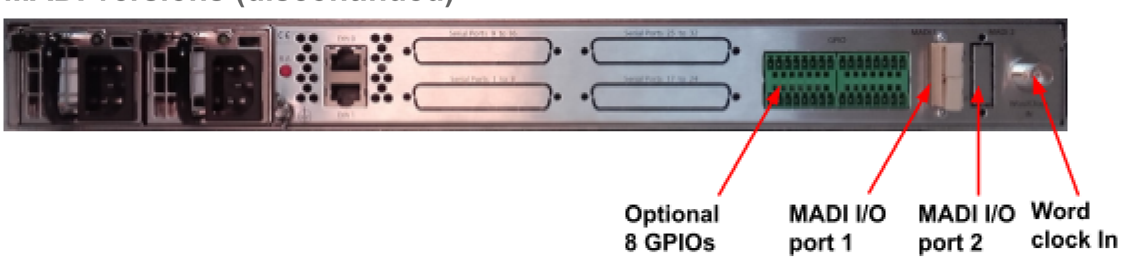
Note: IQOYA *SERV/LINK 881, 882, and 16161 do not feature connector “Audio I/O Port 2”.

5.2.2 DANTE versions and AES67 versions

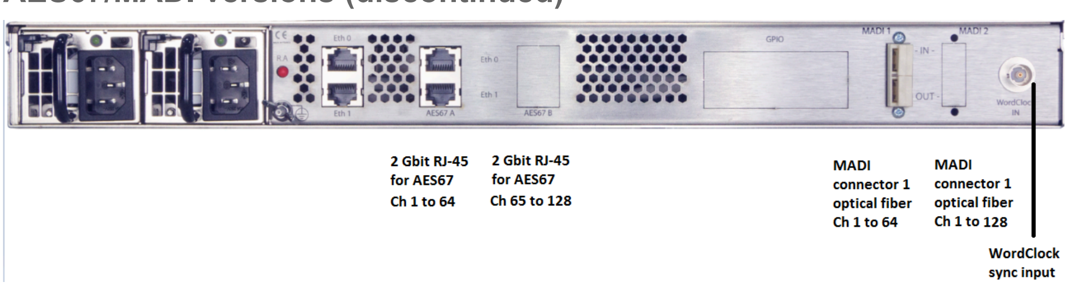


5.2.3 Transcoder (TC) version

5.2.4 MADI versions (discontinued)



5.2.5 AES67/MADI versions (discontinued)



6 INSTALLATION

6.1 Grounding IQOYA SERV/LINK

Attach a ground wire to the chassis (ideally the ground wire has a ring terminal).
Connect the other end of the ground wire to a good electrical ground point.
Once IQOYA is installed and properly grounded, you can connect the Eth ports and audio I/Os as required for your installation.

6.2 Connecting the audio I/Os and synchronization clocks

In case your IQOYA features analog and/or AES/EBU I/Os, audio connections are available from connectors “Audio Port 1” for SERV/LINK882 and SERV/LINK881, and from both connectors “Audio Port 1” and “Audio port 2” for SERV/LINK16162, 24241 and 32321. Please connect the audio I/Os as follows:

- Mono analog input signals 1 to 8 must be connected to inputs 1 to 8 of the breakout cable connected to “Audio I/O port 1”.
- Stereo AES/EBU input signals 1 to 4 must be connected to AES/EBU inputs 1 to 4 of the breakout cable connected to“Audio I/O port 1”
- Mono analog input signals 9 to 16 must be connected to inputs 1 to 8 of the breakout cable connected to “Audio I/O port 2”
- Stereo AES/EBU input signals 5 to 8 must be connected to AES/EBU inputs 1 to 4 of the breakout cable connected to“Audio I/O port 2”
- External synchro signal (AES clock, WordClock, Blackburst video reference) must be connected to the corresponding synchro connector of the cable connected to “Audio I/O port 1”.

Warning: In case you use an IQOYA SERV/LINK 24241 or 32321 (which are AES/EBU only versions), the Word Clock output from “Audio I/O Port 1” must be connected to the Word Clock input of “Audio I/O Port 2”.

In case your IQOYA SERV/LINK is for MADI connectivity, connect the MADI input fibre to the MADI IN connector, and connect the MADI output fibre to the MADI OUT connector.
Connect the WordClock signal to the WordClock input BNC connector if you have to synchronise to a Word Clock signal.

In case of an IQOYA SERV/LINK with AES67 connectivity that supports from 72 to 128 channels, it is necessary to synchronise the two on-board AES67 interfaces (each interface supports 64 I/O channels). They can be both synchronised on PTP (from their WEB GUI).

For a SERV/LINK with AES67 & MADI connectivity which supports from 72 to 128 channels, it is necessary to synchronise the two on-board AES67/MADI interfaces (each interface supports 64 I/O channels). They can be both synchronised on PTP, oir MADI, or one PTP and one MADI . In this last case the PTP clock and the MADI clock have to be at the same frequency (derived from the same master clock, such as GPS for instance).

6.3 Connecting IQOYA SERV/LINK to the network

We recommend that the first connection to the IQOYA codec is done on a LAN. The default IP addresses of IQOYA SERV/LINK WAN Eth ports are:

- Eth0: 192.168.0.100
- Eth1: 192.168.1.100

In case you do not know the IP addresses of your IQOYA SERV/LINK unit you want to connect to, you can read the IP addresses of the WAN Eth interfaces from the front panel (see paragraph “LCD display and keypad”).

Make sure all other devices connected to this LAN are in the same subnet and have different IP addresses (this includes the PC from which you will connect to the IQOYA codec to configure).

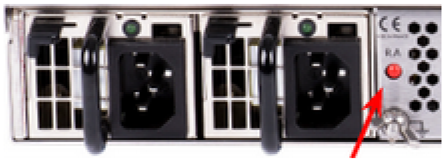
WARNING: Eth0 and Eth1 must not be configured on the same subnet!

For IQOYA SERV/LINK models with AES67 or Dante connectivity, the connection to the AES67 or Dante network is done via the Eth ports “AoIP Pri”, and “AoIP Sec” for the redundancy.

6.4 Powering up IQOYA SERV/LINK

It is recommended to establish all connections before powering the device up.
IQOYA SERV/LINK features two hot swappable redundant power supply units. It is recommended to connect the two power cords. However, only one cord may be used.
IQOYA SERV/LINK starts as soon as it is connected to the mains.

Using only one power supply unit triggers the buzzer of the power supply units.
Connecting the second PSU stops the buzzer.
In case the second PSU is not used, stop the buzzer by pressing the red button labelled “R.A” on the back panel. Note that this does not turn off the “Error” LED on the front panel and the PSU alarm on the LCD and WEB GUI.



7 ACCESSING IQOYA SERV/LINK WEB PAGES

IQOYA SERV/LINK can be configured through its WEB pages.
The codec configuration is done through one of the two WAN network ports (ETH0 and ETH1)
For versions with AES67 connectivity, the codec configuration is done from the two ports ETH0 and ETH1. The AES67 parameters are configurable by connecting through the AES67 dedicated ports.
For the Dante version, the codec configuration is done from the two ports ETH0 and ETH1. The Dante configuration is done via the Audinate's Dante Controller software application, downloadable from Audinate's WEB site.

7.1 WEB pages for setting the AES67 parameters

For IQOYA SERV/LINK with AES67 I/Os, the AES67 parameters can be configured from the Eth interfaces dedicated to AoIP AES67 connectivity which are set in factor to DHCP mode.
See section "AES67 parameters configuration" in this document.

7.2 WEB pages for setting the Dante parameters

For IQOYA SERV/LINK with Dante I/Os, the Dante parameters can be configured from the Dante controller application, through the Eth ports dedicated to AoIP Dante connectivity which are set in the factory to DHCP mode.
See section "Dante parameters configuration" in this document.

7.3 WEB pages for the configuration of the IP audio codec instances

The management of the IP audio codec instances running on IQOYA SERV/LINK can be done from any of the two WAN Eth ports (Eth 0 and Eth1)..
Eth0 default IP address: 192.168.0.100
Eth1 default IP address: 192.168.1.100
Once connected to a network, the IP addresses can be read from the LCD display, from the Eth0 or Eth1 menu.

Enter the appropriate IP address in a WEB browser. The HTTPS protocol is used for WEB access. If the WEB browser displays a message about security certificates. Select the option that allows you to continue with this WEB server.
Enter the requested username and password. The default administrator login is:
username = iqoya
password = iqoya

IQOYA SERV/LINK supports three categories of users: Administrator, User, Read only

- **"Administrator" category**
  A user from the "Administrator" category has all the access rights on the WEB pages.
  The login to the embedded WEB server as "Administrator" is:
  - username: iqoya
  - default password: iqoya
  Username and password can only be modified when logged as Administrator. See Preferences –> System –> Password.
- **"User" category**
  A user from the "User" category has limited access rights. "Write" access is limited to the audio parameters (audio format, source/target IP address and UDP port).
  The login to the embedded WEB server as "User" is:

- username: user
- default password: user
Username and password can only be modified when logged as Administrator. See Preferences –> System –> Password.

- **“Read-only” category**
A user from the Read-only category only has “Read” access rights. He cannot modify a single parameter of the codec.
The login to the embedded WEB server as “Read-only” is:
- username: guest
- default password: guest
Username and password can only be modified when logged as Administrator. See Preferences –> System –> Password.

8 CONFIGURE IQOYA SERV/LINK IN DISTRIBUTION MODE

IQOYA SERV/LINK features two modes of use :

- The 'Program Distribution' mode of use: In this mode, the available functions and the graphical user interfaces are suitable for the needs of fixed installations like STL and SSL links, delivery of WEB radios to CDNs, program delivery to DVB/cable operators, IP audio transcoding, etc ...
- The 'Remote Broadcasting' mode of use: In this mode, the available functions and the graphical user interfaces are suitable for the needs of temporary audio over IP connections like live remote broadcasts, intercom, etc ...

This section is dedicated to the Distribution mode.

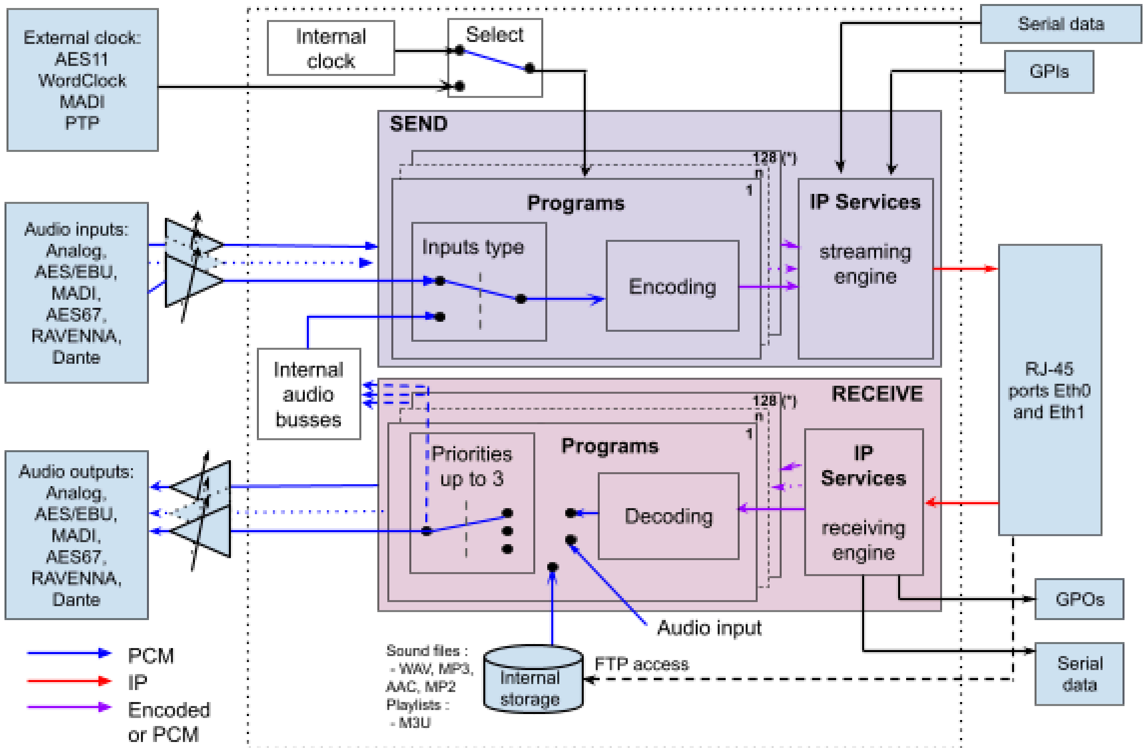
At first power up, the user is prompted to choose the mode of use from the configuration web interface. Later it is also possible to switch from one mode to another from the configuration web pages.

8.1 Working principles in "Program Distribution" mode of use

IQOYA SERV/LINK is a multi-channel IP audio codec. It allows at the same time:

- Encoding the incoming signals and streaming them over IP,
- Decoding IP audio streams to output channels
- Transcoding incoming IP audio streams

The number of audio inputs and outputs of the codec is defined by the licence (for instance SERV/LINK 16161 features 16 mono inputs and outputs).
The number of audio channels to be transcoded from the input streams is also defined by the licence (audio buses).



(*): up to 128 instances of encoding + decoding are supported (128 encodings, or 128 decodings, or 64 encodings and 64 decodings, etc... Note that this max number may be lower depending on the audio compression formats used.

Schematic diagram of IQOYA SERV/LINK

8.1.1 Audio inputs and outputs

The audio inputs receive the audio signals to be encoded. They can be analog, AES/EBU, MADI, or RAVENNA / AES6, or Dante. Each source can be encoded several times at different formats, and streamed to several destinations.
Audio samples decoded by SERV/LINK are played to the selected audio outputs. Audio outputs can be analog, AES/EBU, MADI, or RAVENNA / AES67, or Dante . In case of an IQOYA SERV/LINK that features both analog and digital outputs (for instance SERV/LINK 882), the decoded audio signal is available on both the analog and AES/EBU outputs.
Note that decoded audio samples can also be sent to internal audio buses, and audio buses can also be sources to be encoded. See paragraph 4.4 below.

8.1.2 Programs

In the encoding section of IQOYA SERV/LINK, a program is the encoding of one or several audio inputs. A program is then defined by the following parameters:

- the audio mode: mono, stereo, multi-channel.
- the audio input(s) that receive the signal to be encoded. The number of inputs is given by the audio mode. They are consecutive inputs for stereo and multi-channel modes.
- the encoding format: algorithm, bitrate, sampling frequency.

On the decoding part of IQOYA SERV/LINK, a program is the decoding of an audio source to the audio outputs. A program can be composed of three decoding priorities, with automatic switching from a priority to another in case the audio source is lost. Audio sources of the decoding priorities can be an RTP stream (raw RTP), an HTTP stream (Icecast/Shoutcast), audio inputs, sound files or playlists stored locally.

8.1.3 IP services

IP services are the way programs are streamed over the IP network. An IP service can include one audio program, or several multiplexed audio programs (case of MPEG-TS MPTS encapsulation).
When IQOYA streams, an IP service can be sent to one IP destination (unicast or multicast), or several IP destinations (multi-unicast or multi-multicast). The main parameters that define an IP service are:

- the streaming protocol: RTP, UDP, HTTP
- the encapsulation: raw (no encapsulation), MPEG-TS
- the audio program(s) included in the service: one program in case of raw or MPEG-TS SPTS transport; several programs in case of MPEG-TS MPTS transport
- the FEC scheme (IP data redundancy)
- the destination IP address and port. Several destination IP addresses and ports can be declared (multi-unicast / multi-multicast).

When IQOYA decodes, it listens to IP services and unpacks the IP frames in order to extract and decode the audio contents.
Optionally, the transport of the IP streams (UDP, RTP) can be done via RIST tunnels.

8.1.4 Audio buses

The audio buses serve for transcoding IP audio streams. An audio bus can be selected as the output of one or several output programs. In case several output programs are connected to the same internal bus, the bus mixes the audio from the different programs. Note that an output program can be simultaneously connected to an audio output and an internal bus.
An audio bus can also be selected as the audio source of an input program (like an audio input), so that it can be streamed as an IP Service.

8.2 Steps to follow to configure IQOYA SERV/LINK

Set the global parameters of your IQOYA SERV/LINK
Configuring encoding instances:

- Adjust the parameters of the audio inputs
- Declare the encoding programs
- Declare the IP services to be streamed over IP

Configuring decoding instances:

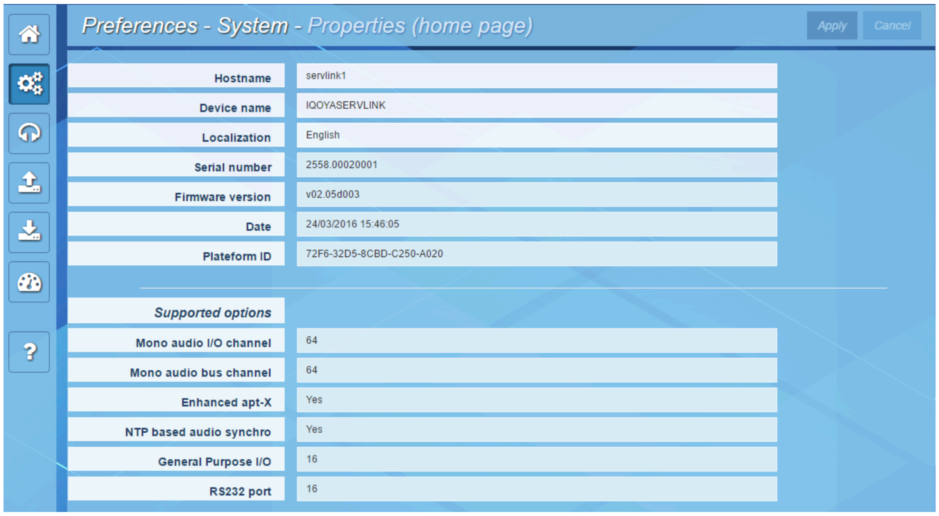
- Adjust the parameters of the audio outputs
- Declare the IP services to be received from the network
- Declare the programs to be decoded from the received IP services
- Check the statuses and metrics on the output programs.

Configuring decoding and encoding instances for transcoding:

- Declare the IP services to be received from the network: RTP, Icecast/Shoutcast, MPEG-TS (optional licence)
- Declare the programs to be decoded from the received IP services, and assign them to internal audio buses.
- Declare the input programs (select audio buses as the sources of these input programs)
- Declare the IP services to be streamed over IP (RTP, Icecast/Shoutcast, HLS, MPEG-TS)
- Check the statuses and metrics on the output programs.

8.3 WEB pages organization

Once logged in, the SERV/LINK home page is displayed.

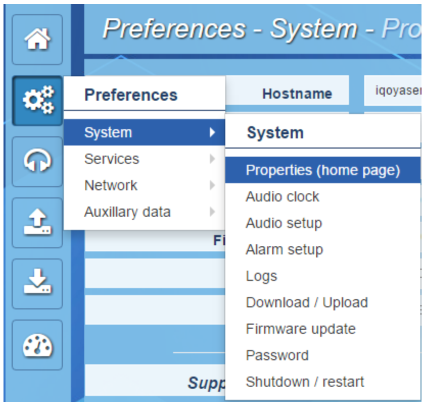


Click on the “value” field of any writeable parameter to enter the edit mode. The background colour of all the parameters that can be modified becomes white.
Select/enter the appropriate values for the parameters of the page, and click on “Apply” to confirm the settings, or “Cancel to ignore the changes.

The WEB pages are organized in categories which are always accessible from the left side of the WEB pages.

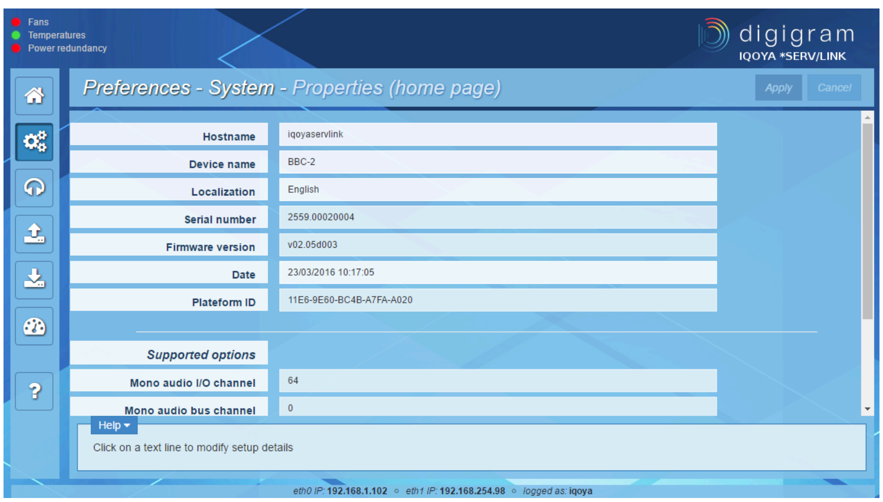
Icon	Category	Description
	Home page	Displays the properties of the unit as well as its software options
	Preferences	Global parameters of the unit.
	Audio I/Os	Audio inputs and outputs parameter settings: name, type selection, audio level adjustment, vu-meters)
	Encoders	Settings of programs (encodings of audio inputs) and IP services (streaming of programs).
	Decoders	Settings of IP services to be received, and associated audio programs to be decoded to the outputs.
	Status	Display the status of all the encoders and decoders, as well as the alarms.
	Help	Allows displaying this user manual.

8.3.1 "Preferences"

<table>
<tr>
<td>
</td>
<td>Click on  to display all the available menus.
Move the mouse pointer above the menus to display the submenus. Click on a sub-menu to display the corresponding page.</td>
</tr>
</table>

8.3.2 Preferences –> System

8.3.2.1 Preferences –> System –> Properties

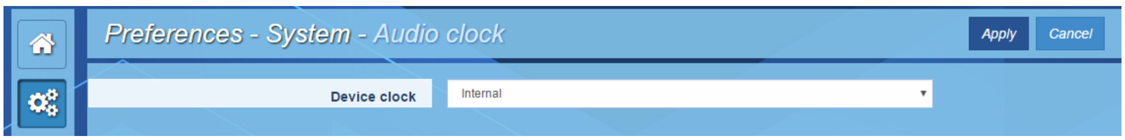


Parameter	Read/Write	Meaning
Hostname	R / W	Logical name given to the device on the network.
Device Name	R / W	Name given to the equipment
Localization	R / W	Language
Serial number	R	Serial number of the unit. This number is set in the factory and cannot be changed.
Firmware version	R	Version of the firmware running on the unit. The firmware can be updated.
Date	R / W	Date and time of the unit.

Platform ID	R	Identifier of the unit. This number is required for applying firmware options.
Mono audio I/O channels	R	Number of mono audio inputs and outputs allowed by the license.
Mono audio bus channels	R	Number of mono channels allocated for the buses and allowed by the license. Buses are used for transcoding.

8.3.2.2 Preferences –> System –> Audio Clock

This page allows defining the SERV/LINK sampling clock source .



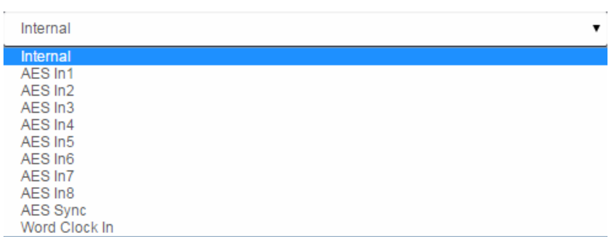
For a SERV/LINK with AES/EBU I/Os, or both analog and AES/EBU I/Os, clock source can be:

- internal: on-board clock
- extracted from an AES/EBU input (AES In x - where x is the number of the AES/EBU input)
- AES Sync: AES11 synchro input
- Word clock input

For a SERV/LINK with MADI I/Os clock source can be:

- internal: on-board clock
- MADI
- Word clock input

Click on the “Device clock” field to select the clock source.



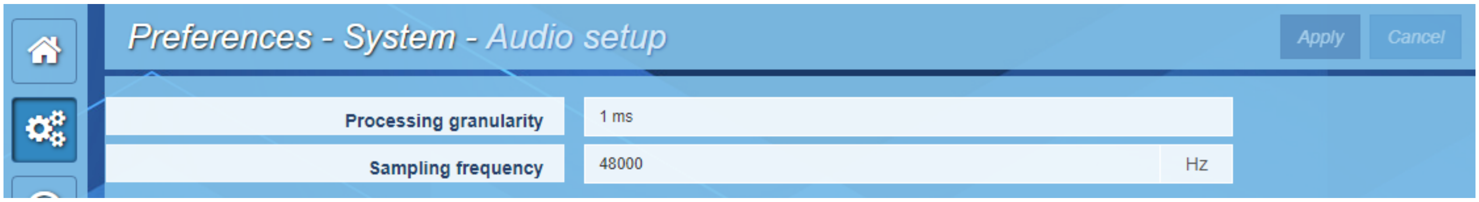
For a SERV/LINK with AES67, or “AES67 & MADI” I/O’s, or Dante I/O’s, the clock source is “Internal”.
The real clock source can be selected through the WEB pages accessible from:

- the AES67 network interfaces. See section “WEB pages for the configuration of the AES67 parameters”.
- The Dante controller application. See section “WEB pages for the configuration of the Dante parameters”.

Click on “Apply” to confirm your choice.

7.1.1.3 Preferences –> System –> Audio setup

This page allows setting the processing granularity and the working sampling frequency value IQOYA SERV/LINK



Click on a parameter field to be able to change the values.

Parameter	Description
Processing granularity	This is the smallest amount of data processed at a time by IQOYA. The lower the processing granularity, the lower the latency. Possible values are 1ms, 2ms, 3 ms, 4 ms. However, a value of 1ms may lead to audio underruns, depending on the features enabled on IQOYA. In case this happens, it is necessary to increase the processing granularity value. Not: the payload size of an IP frame is adjustable via parameter Payload size, from the Send page (see paragraph Encoder parameters configuration).
Sampling frequency	It defines the working sampling frequency of IQOYA. Note that received and generated IP streams can carry audio at different sampling frequencies (in which case a high quality frequency change is applied). When sampling frequency is set to 48 kHz, IP streams can be at 48 kHz, 32 kHz, 16 kHz (G722), and 8 kHz (G711). Note that 44.1 kHz is allowed for a HTTP stream. When sampling frequency is set to 44.1 kHz, IP streams must be at 44.1 kHz.

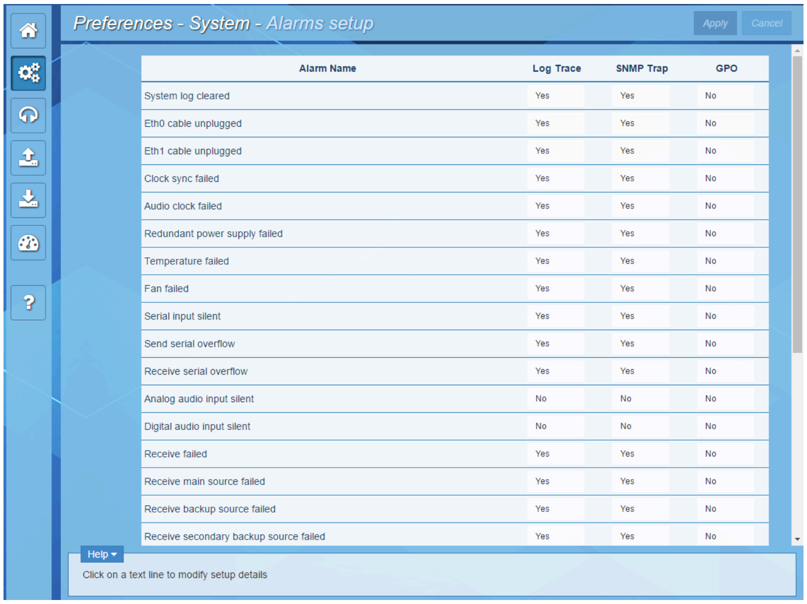
Note:

- From firmware 2.13, parameters for silence detection are configurable for each input program, and for each decoding priority of an output program.
- From firmware 2.13, backup switching criteria can be configured for each output program (Receive/Program). These criteria were global with previous firmware versions.

Click on “Apply” to confirm your changes.

8.3.2.3 Preferences –> System –> Alarms setup

Each alarm occurring on IQOYA can be written in a log file, or/and sent to a GPO, or/and signalled as an SNMP trap (not available in the first firmware version).
The “Alarms setup” page allows enabling/disabling each alarm notifications



Click on a parameter field to be able to change the values.
Click on "Apply" to confirm your changes.

Available alarms

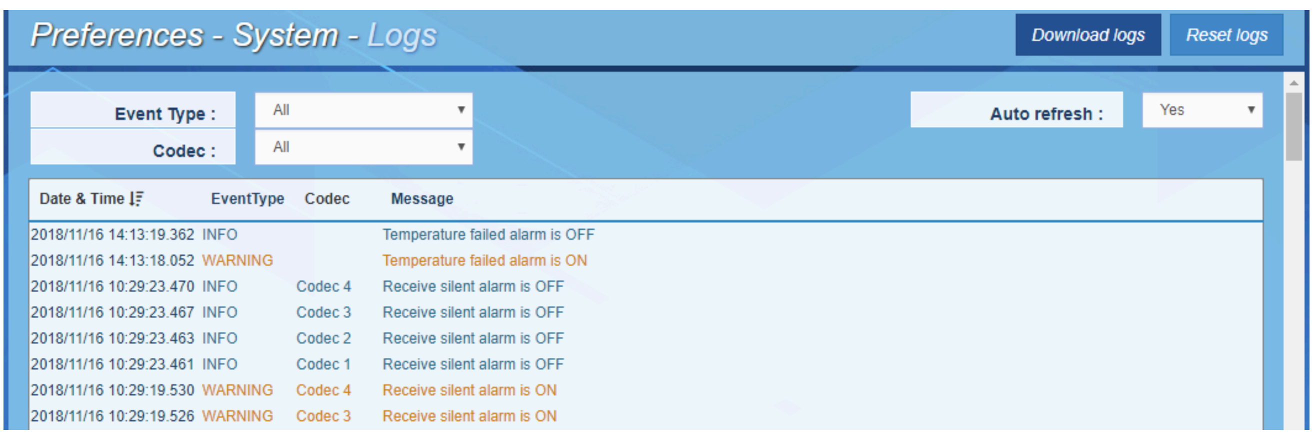
System log cleared	Log file has been cleared
Eth0 cable unplugged	No connection of Eth0
Eth1 cable unplugged	No connection of Eth1
Clock sync failed	External synchro failure (PTP, NTP)
Audio clock failed	Audio sampling clock failure
Redundant power supply failed	PSU failure
Temperature failed	Temperature too high
Fan failed	Internal fan failure
Serial input silent	
Send serial overflow	

Receive failed	No available defined IP stream on the output program
Receive main source failed	Priority 1 of the output program is not available
Receive backup source failed	Priority 2 of the output program is not available
Receive secondary backup source failed	Priority 3 of the output program is not available
Receive sync failed	
Receive main source disabled	Priority 1 on the output program is disabled
Receive backup source disabled	Priority 2 on the output program is disabled
Receive secondary backup source disabled	Priority 3 on the output program is disabled

Receive serial overflow	
Analog audio input silent	Silence detected on the analog input according the criteria of silence
Digital audio input silent	Silence detected on the analog input according the criteria of silence

Receive main source primary stream failed	In case of streaming with FEC on priority 1, this means that the primary stream is lost on priority 1
Receive backup source primary stream failed	In case of streaming with FEC on priority 2, this means that the primary stream is lost on priority 2
Receive secondary backup source primary stream failed	In case of streaming with FEC on priority 3, this means that the primary stream is lost.
Receive main source redundancy stream failed	In case of streaming with FEC on priority 1, this means that the FEC is lost.
Receive backup source redundancy stream failed	In case of streaming with FEC on priority 2, this means that the FEC is lost.

8.3.2.4 Preferences –> System –> Logs



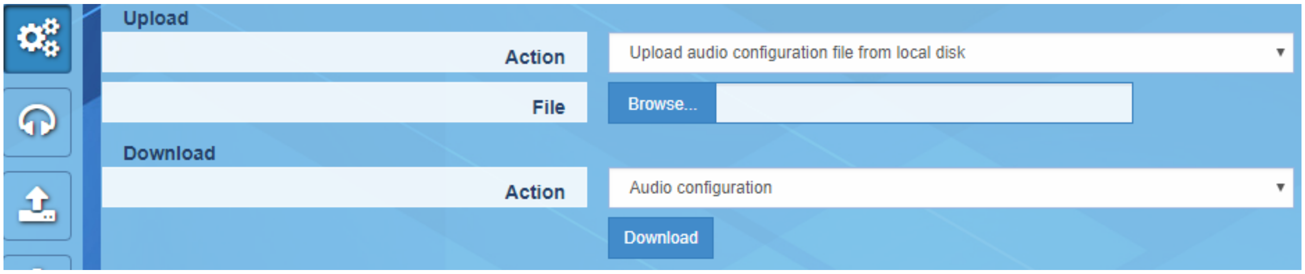
This page allows viewing and downloading the log file of IQOYA SERV/LINK. This log file gives information about the internal behaviour of IQOYA, and is useful for advanced diagnostics. Traces of enabled alarms are written into this log file (alarm ON, alarm OFF). This log file is stored internally and is persistent to a power cycle, a restart or reboot.

Event Type: allows selecting the category of traces to be displayed: Infos, Warnings, Errors, Errors & Warnings.
Codec: allows selecting one of the codecs so that only log traces related to this codec are displayed. The number of the codec can be seen from the Send/IP Services page, and from the Receive/ Programs page.
Auto refresh: Yes/No. When set to Yes, the log file display is refreshed every few seconds
Date & Time: clicking on this icon allows you to sort out the traces by date and time, starting by most recent traces or starting by oldest traces.
Reset logs: resets all the traces.

Download logs: allows for remotely downloading the traces log file.

8.3.2.5 Preferences –> System –> Download / Upload

This page allows downloading the IQOYA configuration to a remote PC, or uploading a configuration from a remote PC to IQOYA.



To save the current configuration of IQOYA to a remote PC, click on “ Download”.
To apply a configuration to IQOYA, click on “ Browse” to select the configuration file, and click on “Apply”.

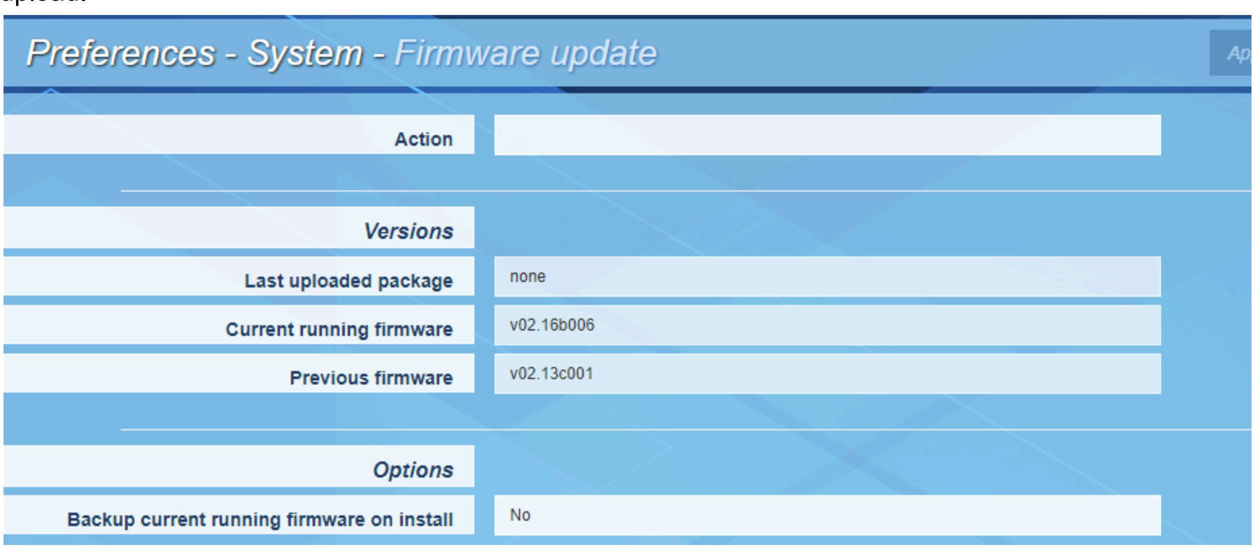
The configuration that can be uploaded/downloaded can be:

- The audio configuration only (includes the programs and IP services)
- The full codec configuration

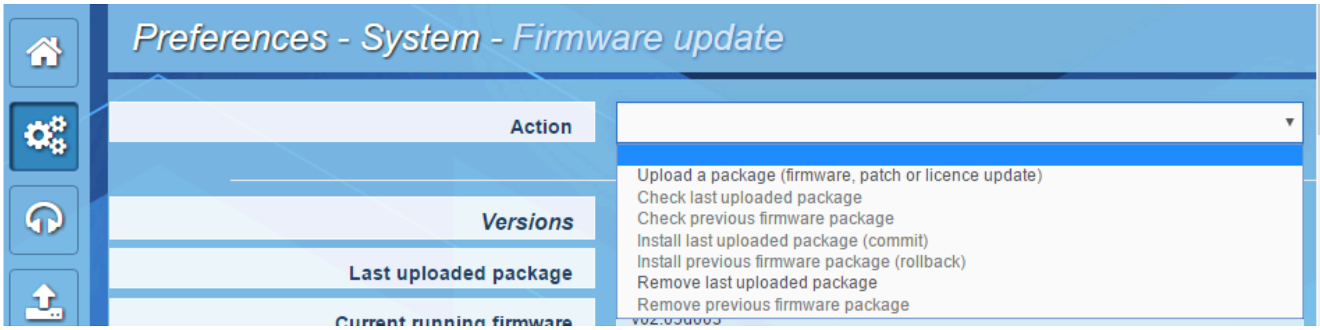
In addition, the html file which allows you to view all the parameters of the codec can be downloaded. From the download section, select “ Device Information”, and download

8.3.2.6 Preferences –> System –> Firmware update

IQOYA can be updated with a new firmware, a patch, or an optional license. The first phase of the update consists in uploading and checking the software package; during this phase, the audio activity is not stopped. The second phase consists in applying the uploaded package; audio activity is stopped during this phase.
Two firmware versions are stored locally: the currently running version, and the previous version. This allows to go back to the previous firmware version if an issue is experienced with the more recent version, without having to go through an upload.



Click on the “Action” field, and click on the arrow to display the list of possible actions.



Select the appropriate action through the list.

For a firmware update, select “Upload a package”, and click on “Browse” to select the file to be uploaded.
Click on “Apply” to start the upload. Audio activity is not stopped during the upload.
Once the package upload is completed, select the action “Install last uploaded firmware”,and click on “Apply”. Applying the firmware stops the audio activity. The equipment restarts automatically.

The following operations are also possible from the “Action” drop-down menu:

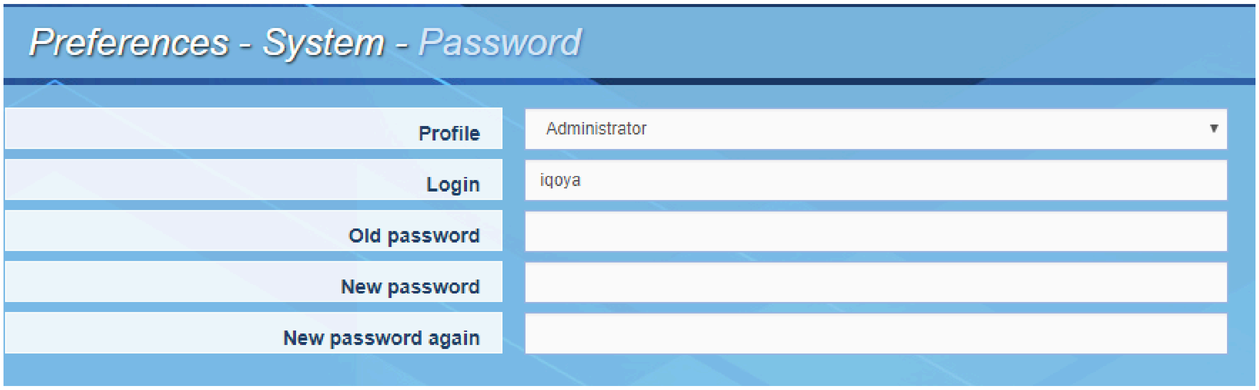
- **Check previous firmware package**: this allows checking that the previous firmware version that is stored locally is correct.
- **Check last uploaded package**: this allows checking that the last uploaded firmware version is correct. This operation is done automatically during the uploading phase.
- **Install previous firmware package** (rollback): this allows installing a previous version of the firmware that is stored locally. This is a firmware downgrade.
- **Remove last uploaded package**: this allows deleting the last uploaded package. This means that this package will not be installed.
- **Remove previous uploaded package**: this allows deleting the previous uploaded package. This means that an upload is necessary for a firmware downgrade.

Backup the current firmware when installing a new one firmware
One may want to save the current firmware when installing a new one. This allows easy firmware rollback if necessary.
Select “Yes” from the field **Backup current running firmware on install.**
It is recommended to set this option to “Yes”, otherwise the firmware version seen as “previous firmware” may not be the expected version, see table below).

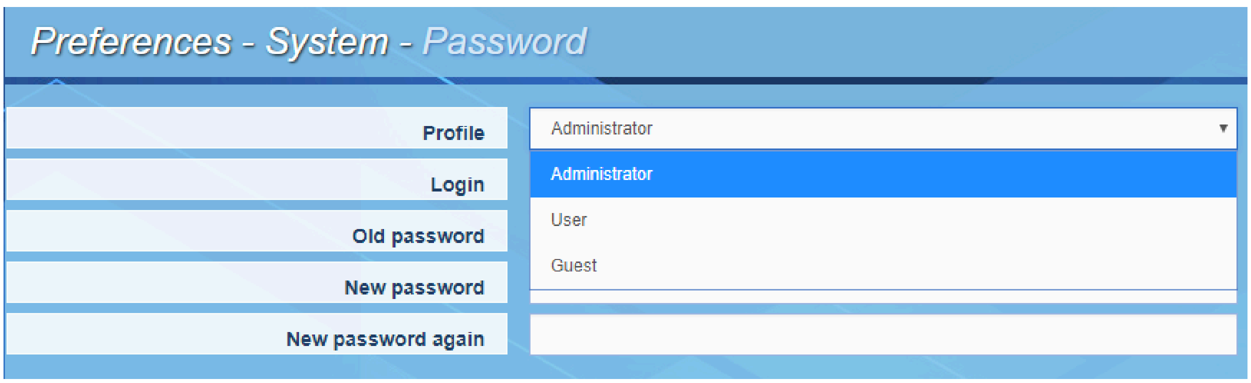
	Backup current running firmware on install	**Current firmware**	**Previous firmware**
Original firmware		1	
Update with Firmware 2	Yes	2	1
Update with Firmware 3	Yes	3	2
Update with Firmware 4	**No**	4	**2 (!)**

8.3.2.7 Preferences –> System –> Password

This page allows changing the login username password for a given user category.
This can be done when logged into the IQOYA as Administrator.



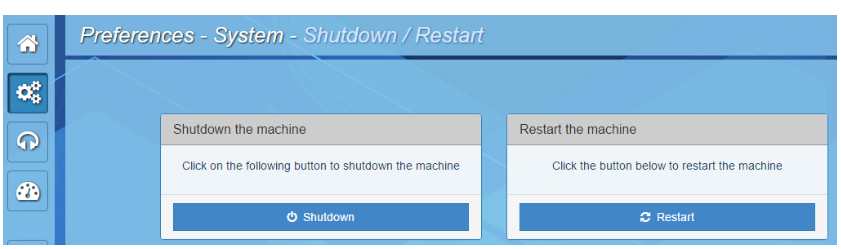
First select the profile for which credentials have to be changed.



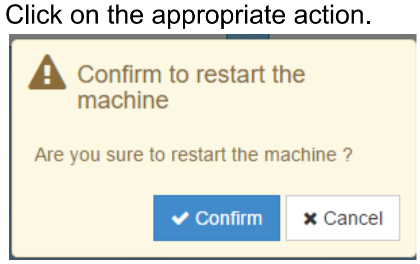
Login: allows configuring the username to be used in order to log to the WEB GUI with the selected profile.
Old password: Type the current password
New password: Type the new password
New password again: confirm the new password
Click on “Apply” to confirm the changes.

8.3.2.8 Preferences –> System –> Shutdown / Restart

This page allows you to restart or shutdown IQOYA.



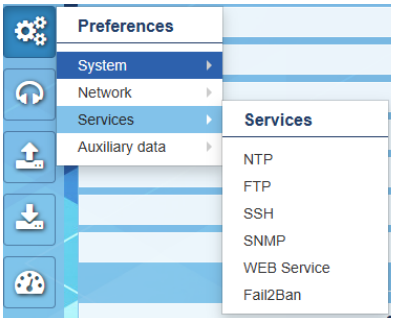
Click on the appropriate action.



Confirm or cancel your choice through the displayed confirmation window.

8.3.3 Preferences –> Services

This menu allows configuring the “network” services of IQOYA.

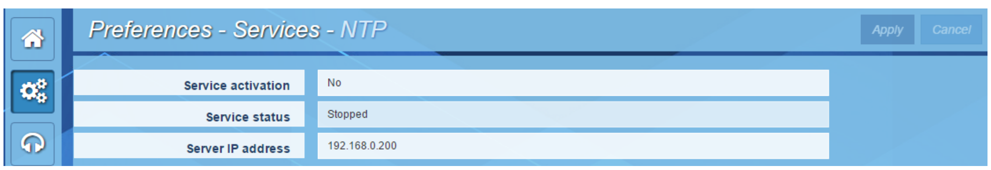


8.3.3.1 Preferences –> Services –> NTP

This page allows:

- configuring the date and time synchronization to an NTP server.
- enabling the optional feature “audio synchronization on NTP clock”.

NTP service is disabled by default.



Click on the **“service activation”** field to activate/deactivate the NTP service. Select “Yes” to activate it.
Enter the IP address of the NTP server.
In case you just need to activate the date and time NTP synchronization, click on “Apply”. The status of the service is displayed in the field “Service status”.

For activation of the NTP based audio synchronization, select “Yes” for parameter “**Sync audio on NTP cloc**k”. This requires that the software option is installed on the IQOYA SERV/LINK, as well as on the associated IQOYA decoders.

8.3.3.2 Preferences –> Services –> FTP

FTP is useful typically for managing the backup playlists and sound files on IQOYA’s internal storage (uploading/deleting).
FTP service is disabled by default.

Click on the “Service activation” field. Select “Yes” to enable the FTP service, “No” to disable it.
If necessary, you may change the port used for FTP (default value is 21).
Parameter “Bandwidth limitation” allows limiting the network bandwidth of the FTP traffic.
Click on “Apply” to confirm the changes.
Note that a username and password are required to establish an FTP

connection to IQOYA SERV/LINK. Username is: ftp. Password is the administrator password, by default: iqoya.

Playlist and sound files have to be stored in folder DEVICE_STORAGE.

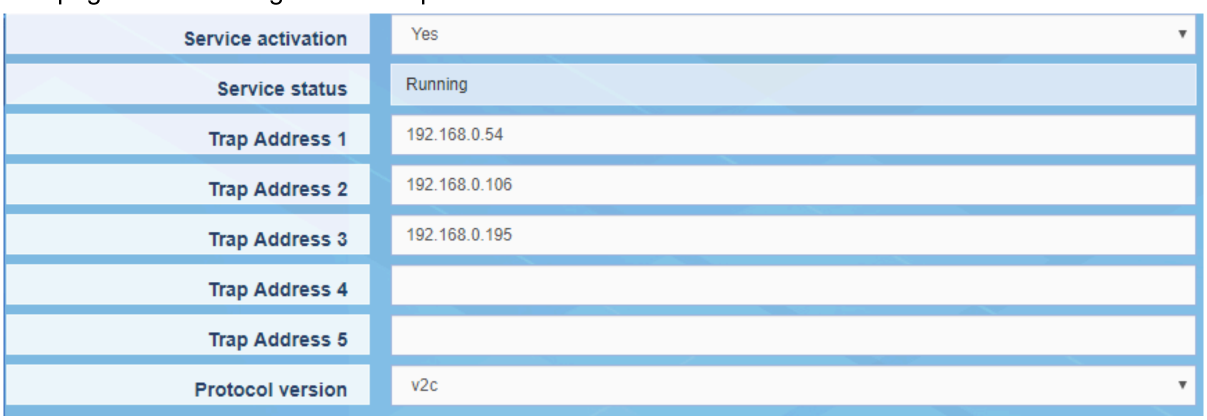
8.3.3.3 Preferences –> Services –> SSH

This page allows enabling/disabling the SSH service on IQOYA.
SSH is mainly to be used for advanced diagnostics.

SSH access to the IQOYA requires a password.
Up to firmware version 3.x, the login/password to open an SSH session is: root, iqoya.
As of firmware version 4.11, each IQOYA SERV/LINK unit has its own default password for SSH access: iqoyaxxxxx, where xxxxx are the last 5 numbers of the IQOYA serial number.

8.3.3.4 Preferences –> Services –> SNMP

This page allows setting the SNMP parameters.



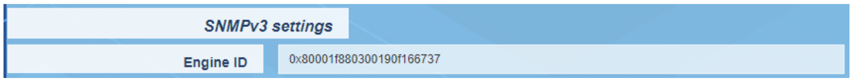
IQOYA can be controlled and monitored via SNMP (SET, GET, Traps) provided that the SNMP service is activated (“Service activation” field).
IQOYA can send the SNMP traps to up to 5 SNMP supervisors (Trap addresses 1 to 5).

Select the version of the used SNMP protocol from the field “Protocol version”: v2c or v3.

SNMPv3 brings the following features compared to SNMPv2c:

- Message integrity: Ensures that a packet has not been tampered with during transit.
- Authentication: Determines that the message is from a valid source.

- Encryption: Scrambles the content of a packet to prevent it from being understood by an unauthorised source.When SNMP v3 is selected the following additional parameters are displayed:



"User 1" allows the user to set the parameters for a user account that has the read-only rights.
"User 2" allows the user to set the parameters for a user account that has the read and write rights.
"TRAP user" allows the user to set the parameters for a user account that is authorised for receiving traps for the SERV/LINK.

SNMPv3 Parameter	Type	Description
Engine ID	Read	ID delivered by the IQOYA SNMPv3 agent. May be necessary for a supervisor to receive the SNMP traps issued by the IQOY.
Access Type	Read Only	Read Only: A user of this category can access the SNMP variables only in read mode Read / Write: A user of this category can access the SNMP variables in read or write mode.
Name	Read / Write	Name of the user
Authentication encryption	Read / Write	Specifies the authentication protocol (HMAC-MD5 or

		HMAC-SHA algorithms) for this SNMPv3 user.
Authentication pass-phrase	Read / Write	Lets you assign an authentication passphrase for this SNMPv3 user; 8 to 64 characters; no spaces or double quotes allowed.
Frame encryption	Read / Write	Specifies the privacy protocol (DES or AES-128) for this SNMPv3 user.
Encryption pass-phrase	Read / Write	Lets you assign a privacy passphrase for this SNMPv3 user; 8 to 64 characters; no spaces or double quotes allowed.

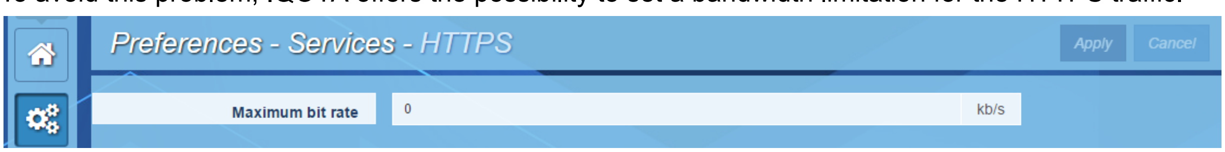
It also displays the System group MIB-II information.

Note that the IQOYA MIB can be downloaded by clicking on .

Click on “Apply” to confirm the settings.

8.3.3.5 Preferences –> Services –> HTTPS

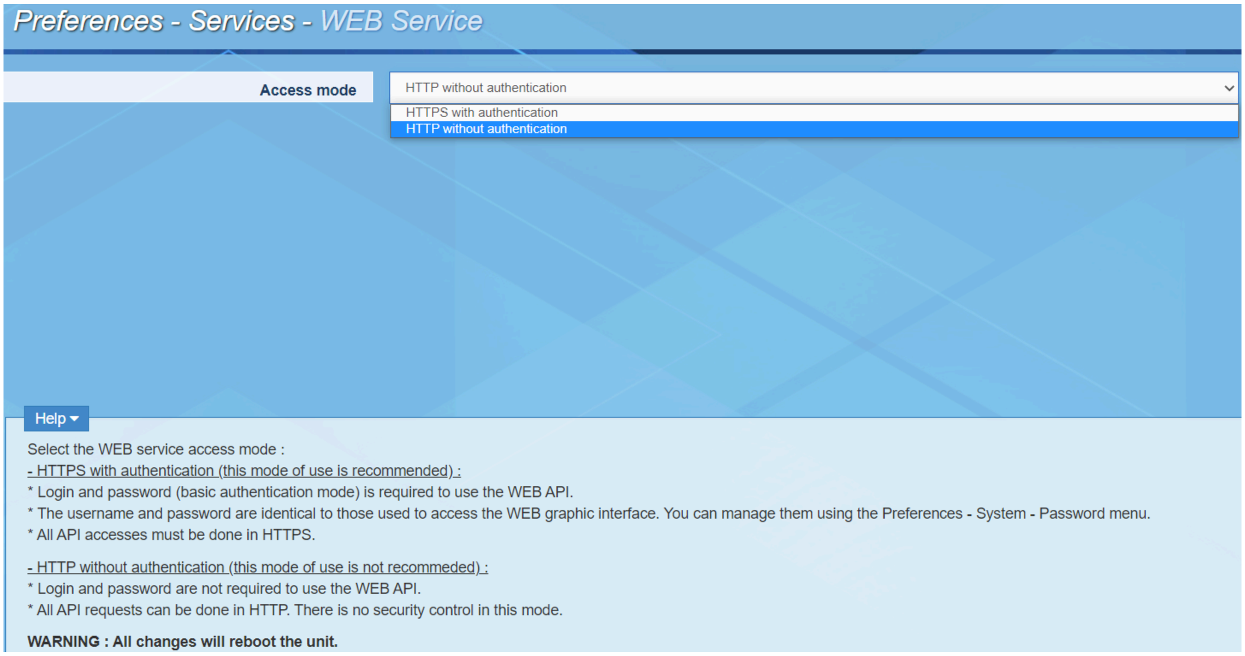
This page allows setting a bandwidth limitation to the HTTP traffic.
In case the IP audio stream takes almost all the available network bandwidth, the HTTP traffic generated when accessing the WEB pages may disturb the IP audio frames transmission, because the total bandwidth necessary for the IP audio stream plus HTTP traffic may exceed the available network bandwidth.
To avoid this problem, IQOYA offers the possibility to set a bandwidth limitation for the HTTPS traffic.



Click on the “Maximum bit rate” field, and enter the maximum bit rate allowed for HTTPS traffic.
Default value is 0, which means no limitation on HTTPS traffic.
Click on “Apply” to confirm the settings.

8.3.3.6 Preferences –> Services –> WEB Service

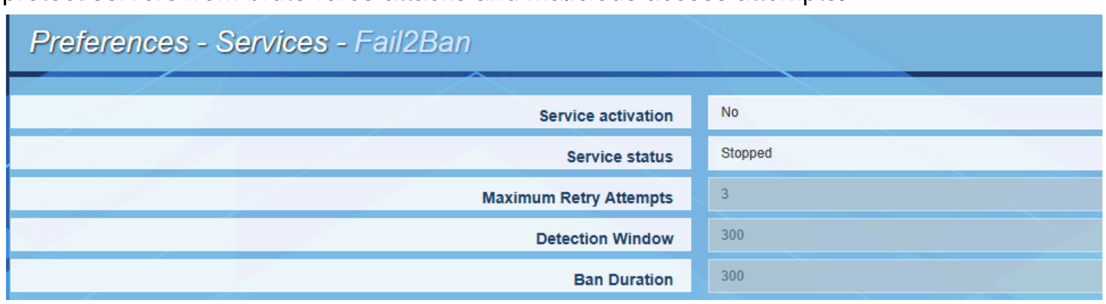
This page allows for selecting the protocol used for the WEB Service API access.



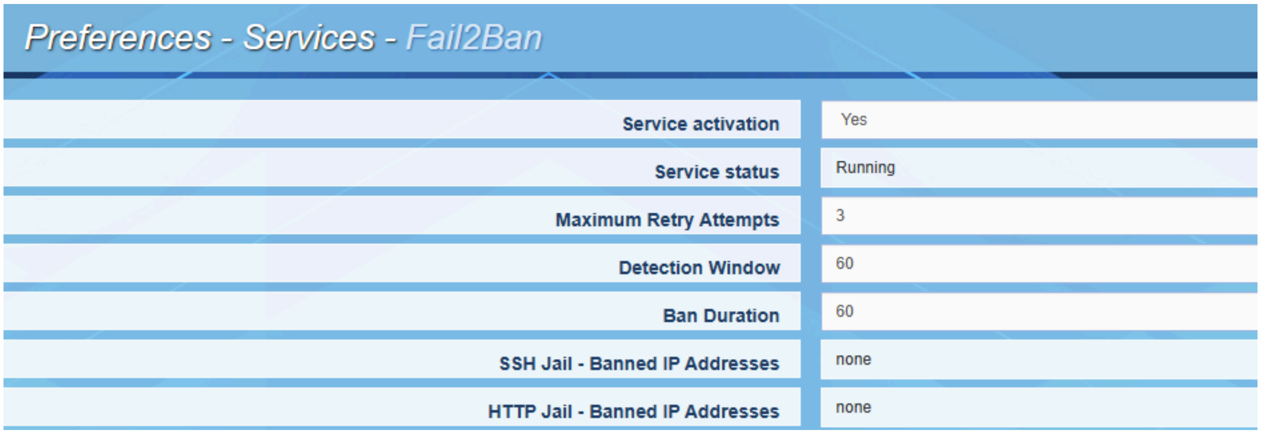
The default protocol is HTTPS (authentication through the username and password used to access the WEB pages).
HTTP can be selected, in which case no authentication is required; this last mode is less secure than the HTTPS mode, and is not recommended.
Changing the WEB Service API access protocol restarts the codec.

8.3.3.7 Preferences –> Services –> Fail2Ban

This page allows you to enable / disable the Fail2Ban service on the IQOYA. Fail2ban is an intrusion prevention tool used to protect servers from brute-force attacks and malicious access attempts.



The principle of this service is that if a source (a given IP address) makes a certain number of attempts to connect to the IQOYA during a period of time, it is banned for a certain duration.This applies to HTTP/HTTPS and SSH connection attempts.

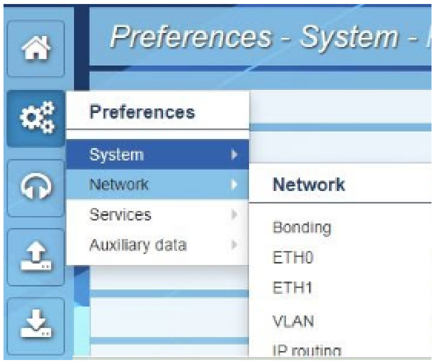


Parameter	Type	Description
Service activation	Read / Write	Yes: activation of the service. No: service disabled
Service status	Read Only	Values: Stopped or Running
Maximum retry attempts	Read / Write	Number of consecutive connection attempts from the same source. When this number is reached, the IP address of the source is banned.
Detection Windows	Read / Write	Time window (in seconds) duration for consecutive attempts.
Ban duration	Read / Write	Duration (in seconds) of the address ban.
SSH Jail - Banned IP addresses	Read / Write	IP addresses which are banned for SSH connection attempts
HTTPJail - Banned IP addresses	Read / Write	IP addresses which are banned for HTTP/HTTPS connection attempts

When a source IP address is banned, it appears as in the screen capture below.

A banned IP address can be unbanned by clicking on the “Unban” button.

8.3.4 Preferences –> Network



This menu allows accessing the network configuration of IQOYA.

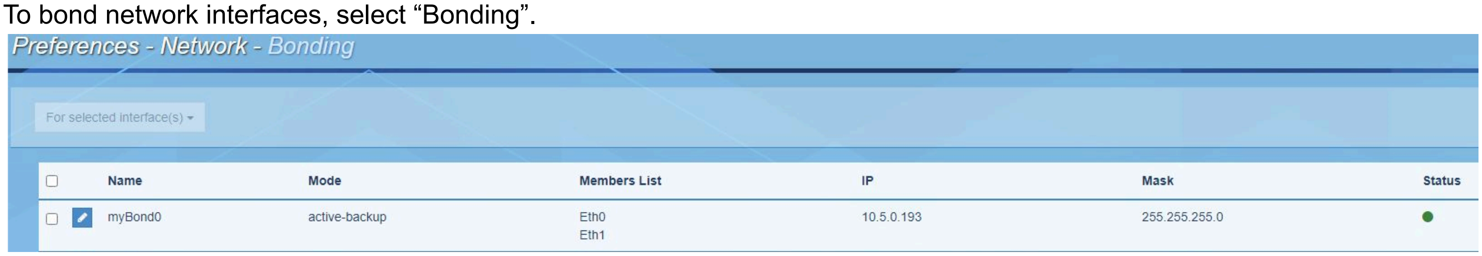
8.3.4.1 Preferences –> Network –> Bonding

This page allows configuring the bonding of network interfaces.

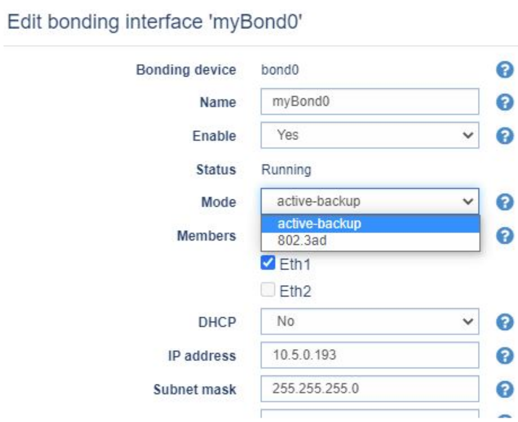
Two bonding modes are supported:

- Active-backup mode:
  This mode uses only one active slave interface to transmit packets. An additional slave interface only becomes active if the primary one fails. Active-backup is the best choice in high availability setups with multiple switches that are interconnected.
- 802.3ad (or LACP: Link Aggregation Control Protocol)
  This mode provides a method to control the bundling of several physical ports together to form a single logical channel

SERV/LINK features 2 physical WAN Eth ports, and 1 possible bond: bond0.
To bond network interfaces, select “Bonding”.



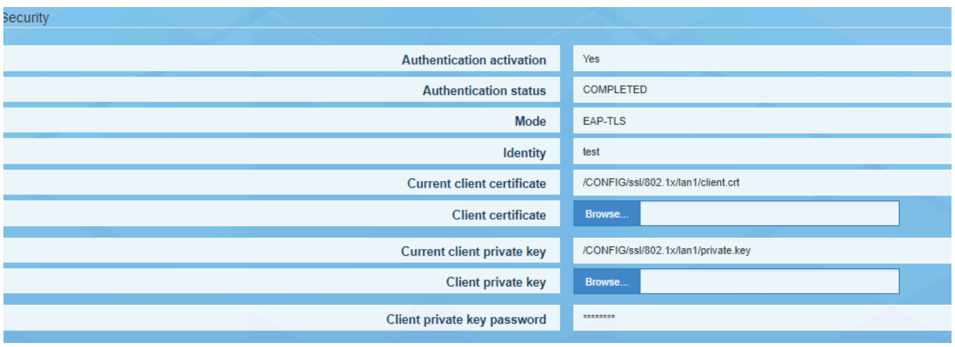
Click on the icon of the bond to configure it.
The configured bond can be enabled/disabled by selecting it (thanks to the check box on the left end), and then by selecting Enable or Disable from the menu “For selected interfaces”. Status of the enabled bond is displayed in green on the right end.



Parameter	Type	Description
Name	Read/Write	Give a name to the bonding interface. This name will be displayed for selecting this interface from other WEB pages (like Send->IP Service->, Receive->IP Service).
Enable	Read/Write	Select “Yes” for activating the bonding interface. Select “No” to disable it. Note that the physical Eth ports included in a bond cannot be directly used as long as the bonding interface is enabled and running. Note that the bond can also be enabled/disabled from the bonding page by selecting it and by selecting Enable or Disable from the menu “For selected interfaces”.
Status	Read	Status of the Bond.
Mode	Read/Write	Allows selecting the working mode of the bond: Active-backup: 802.3ad
Members	Read/Write	Select here the Eth ports that are part of the bond.
DHCP	Read/Write	Allows enabling/disabling DHCP (Dynamic Host Configuration Protocol). Default value is OFF (disabled). Click on “On” to enable DHCP. This mode disables the following parameters.
IPv4 address	Read if DHCP is On Write if DHCP is Off	DHCP Off: Enter the IP address of this bond logical interface. DHCP On: Displays the IP address automatically set through DHCP.
Subnet mask	Read if DHCP is On Write if DHCP is Off	DHCP Off: Enter the mask of the subnet this bond interface belongs to. DHCP On: Displays the subnet mask automatically set through DHCP.
Gateway	Read if DHCP is On Write if DHCP is Off	DHCP Off: Enter the gateway IP address. Streams sent through this bond interface will go through this gateway. DHCP On Displays the gateway IP address automatically set through DHCP.
Primary DNS	Read if DHCP is On	DHCP Off: Enter the IP address of the used DNS (if any).

	Write if DHCP is Off	DHCP On: Displays the IP address of the DNS set through DHCP.
Secondary DNS	Read if DHCP is On Write if DHCP is Off	DHCP Off: Enter the IP address of the secondary DNS (if any). DHCP On: Displays the IP address of the secondary DNS set through DHCP.
Authentication activation	Read/write	Set to Yes, this parameter allows configuring the 802.1x authentication parameters (see parameters description below). Set to No, 802.1x authentication is disabled.

802.1x authentication parameters

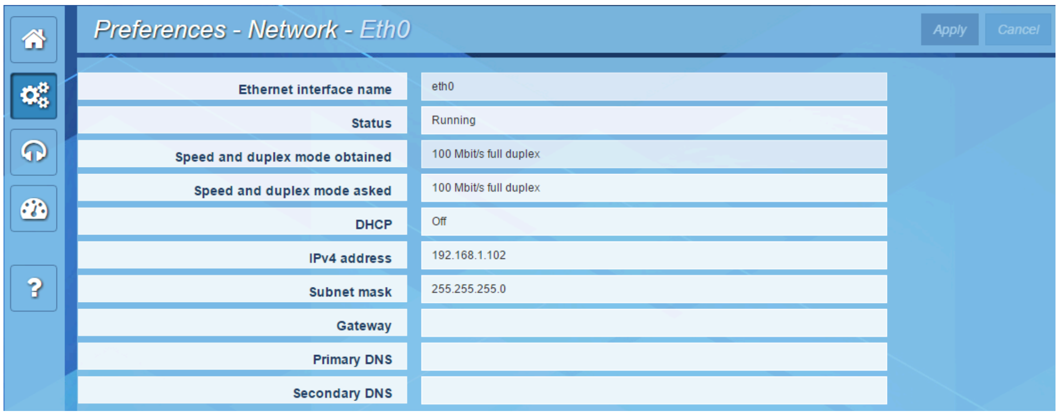


Authentication status	Read	Reports the status of the authentication process: • COMPLETED-SUCCESS-Authorized : Authentication is successful. IQOYA is authorised on the network. • COMPLETED-FAILURE-Unauthorized : The connection to the authentication service has been lost. IQOYA is not authorised on the network. • ASSOCIATED-CONNECTING-Unauthorized-IDLE : connection in progress - IQOYA is not yet authorised on the network. • ASSOCIATED-AUTHENTICATING -Unauthorized-IDLE : authentication in progress - IQOYA is not yet authorised on the network. • ASSOCIATED-HELD-Unauthorized-FAILURE : Authentication failed - IQOYA is not yet authorised on the network.
Mode	Read/Write	One standard is currently supported: EAP LTS (Extensible Authentication Protocol - Transport Layer Security)
Identity	Read/Write	Identity string for EAP
Current client certificate	Read	Displays the client certificate filename currently in use.
Client certificate	Write	Allows for the selection of the certificate file to be used (.crt file)
Current client private key	Read	Displays the client private key filename currently in use.
Client private key	Read	Allows for the selection of the private key file to be used (.key file)

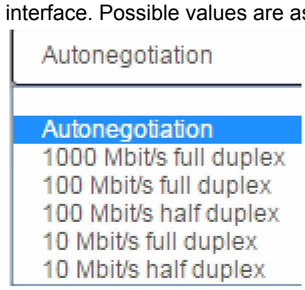
Client private key password	Write	A password must be entered to save the authentication settings. Enter the password for the client key. Once the password is saved, it is no longer displayed on the WEB page and is replaced by stars.

8.3.4.2 Preferences –> Network –> Eth0 / Eth1

These two pages allow configuring the two network ports of IQOYA SERV/LINK.



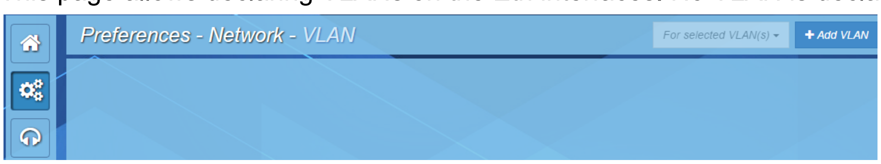
Click on a parameter field (“Status” for instance) to enter the editing mode.

Parameter	Type	Description
Internet interface name	Read	Displays the name of the Eth port
Status	Read/Write	This parameter allows enabling/disabling the interface Default value=Running Possible values: Running: Eth port is enabled. Stopped; Eth port is disabled
Speed and duplex mode obtained	Read	Displays the current speed and mode of the Eth interface.
Speed and duplex mode asked	Read/Write	Allows selecting the working mode of the Eth interface. Possible values are as follows:  We recommended avoiding the “Auto-negotiation” mode. Select the mode supported by the network node connected to the IQOYA.
DHCP	Read/Write	Allows enabling/disabling DHCP (Dynamic Host Configuration Protocol). Default value is OFF (disabled).

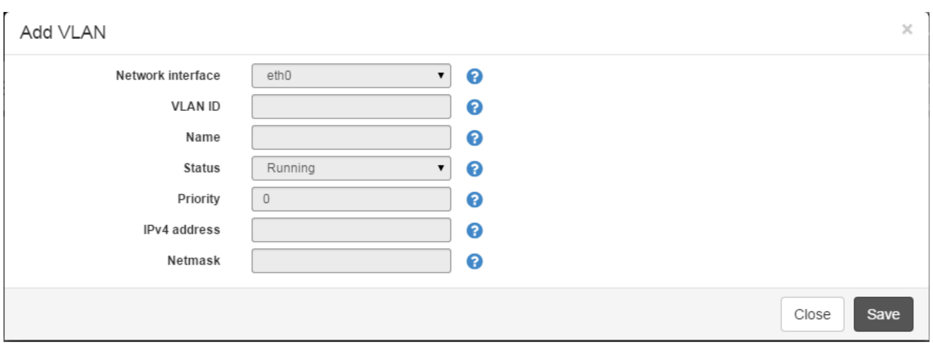
		Click on “On” to enable DHCP. This mode disables the following parameters.
IPv4 address	Read if DHCP is On Write if DHCP is Off	**DHCP Off:** Default value is 192.168.0.100 for Eth0, and 192.168.1.100 for Eth1. Enter the IP address of this Eth interface. **DHCP On:** Displays the IP address automatically set through DHCP.
Subnet mask	Read if DHCP is On Write if DHCP is Off	**DHCP Off:** Enter the mask of the subnet this Eth port belongs to. **DHCP On:** Displays the subnet mask automatically set through DHCP.
Gateway	Read if DHCP is On Write if DHCP is Off	**DHCP Off:** Enter the gateway IP address. Streams sent through this interface will go through this gateway. **DHCP On:** Displays the gateway IP address automatically set through DHCP.
Primary DNS	Read if DHCP is On Write if DHCP is Off	**DHCP Off:** Enter the IP address of the used DNS (if any). **DHCP On:** Displays the IP address of the DNS set through DHCP.
Secondary DNS	Read if DHCP is On Write if DHCP is Off	**DHCP Off:** Enter the IP address of the secondary DNS (if any). **DHCP On:** Displays the IP address of the secondary DNS set through DHCP.

8.3.4.3 Preferences –> Network –> VLAN

This page allows declaring VLANs on the Eth interfaces. No VLAN is declared by default.



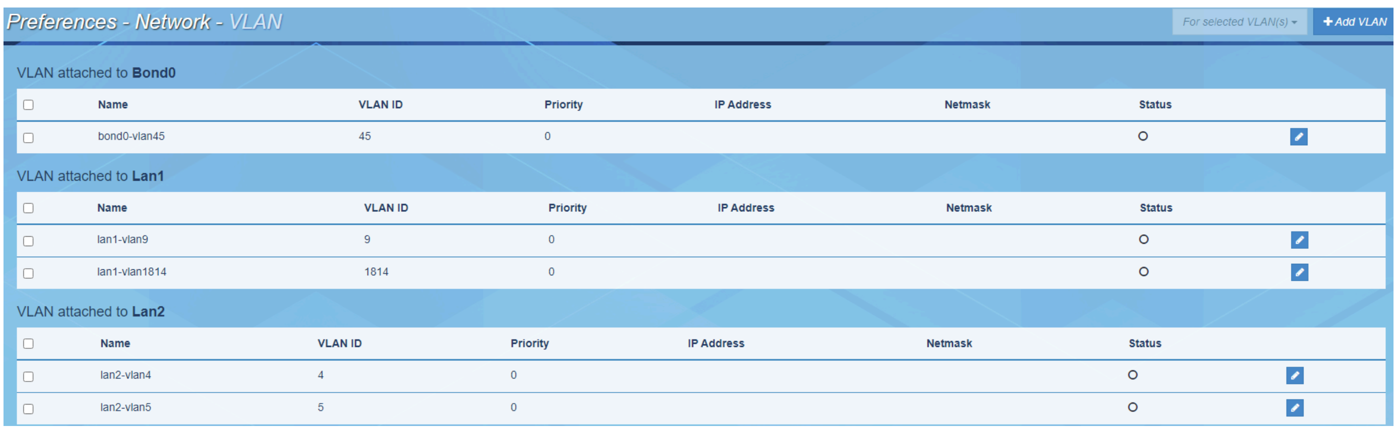
Click on “+Add VLAN” to declare a new VLAN.



Parameter	Type	Description
Network interface	Read/Write	Select the network interface that will support the VLAN (Eth0, or Eth1, or bond)

VLAN ID	Read/Write	Enter the VLAN ID in the ranges [1-1001] [1006-4095]
Name	Read/Write	Enter a logical name for this VLAN
Status	Read/Write	Allows enabling/disabling this VLAN. Select “Running” to enable this VLAN. Select “Stopped” to disable this vLAN.
Priority	Read/Write	Enter the VLAN priority in the range [0-7].
IPv4 address	Read/Write	Enter the IP address of the selected Eth port in this VLAN. If no value is entered, the IP address is the IP address of the selected Eth port.
Netmask	Read/Write	Enter the netmask for this VLAN interface. If no value is entered, the netmask is the same as the selected Eth port netmask.

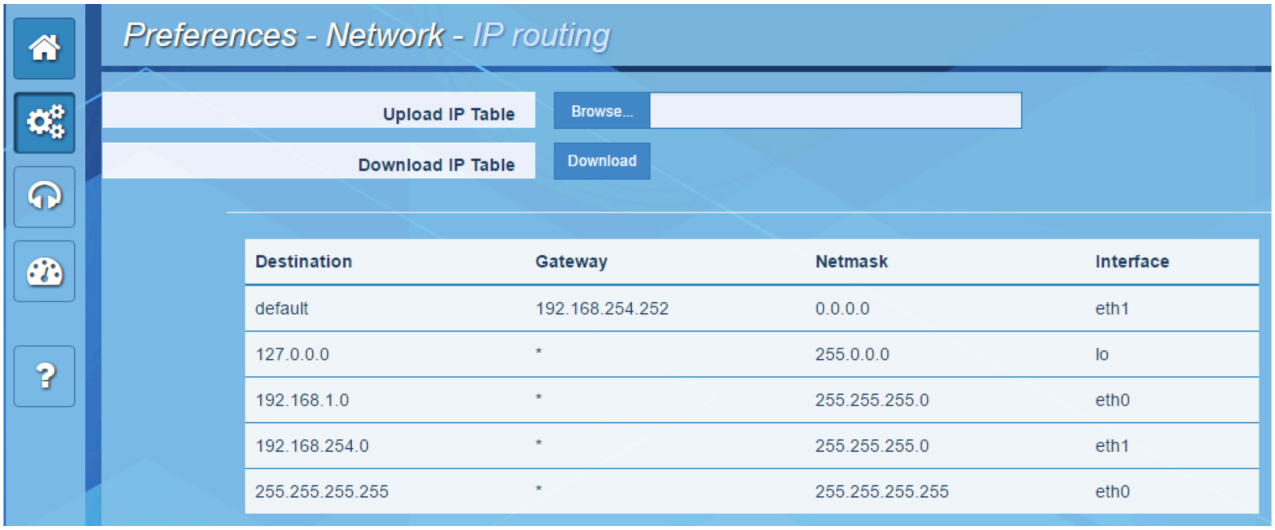
Click on “Save” to save your modifications.



VLANs can be started, stopped or deleted from the VLAN WEB page. Select the VLANs thanks to the check boxes on the left-end, and go to the menu “For selected LAN’s”, and select the appropriate action.
Click on the pencil icon on the right-end to modify the settings LAN.

8.3.4.4 Preferences –> Network –> IP routing

This page allows viewing the current IP routing table, downloading it, and uploading a modified IP routing table.



In case the routing table has to be modified, click on “Download”.

The routing table can be edited with a standard text editor such (as notepad). You may add IP routes, as described in the downloaded file.

Note: Only one default gateway must be declared. Routes may be added on the other interfaces that do not have a gateway. .

Example:

We want to stream in dual streaming, with one stream going through a network via Eth0, and the redundant stream going through a separate network via Eth1.

- Eth0 is set to IP@ 192.168.0.100, with the gateway 192.168.0.254 declared from the WEB GUI (default gateway).
- Eth1 is set to IP@ 192.168.1.100 , with the gateway 192.168.1.254 that is not declared on the SERV/LINK.

Let’s suppose dual streaming is as follows:

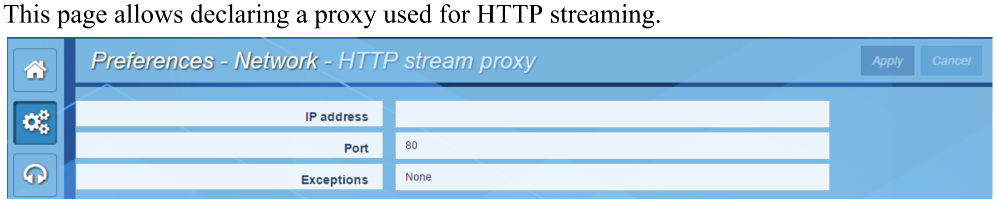
- first stream sent to IP@ 10.0.0.140
- redundant stream sent to 193.0.0.13

If the routing table is not modified, the two streams will by default flow via Eth0 and the default gateway 192.168.0.254.
The following rule must be added via the file IpRoutingTable.cfg so that the redundant stream flows via Eth1:
-net 193.0.0.13 netmask 255.255.255.255 gw 192.168.1.254

If a range of IP addresses have to be routed through Eth1, a rule like the following has to be added:
-net 193.0.0.0 netmask 255.255.255.240 gw 192.168.1.254
In this example, any stream which destination IP@ is in the range 193.0.0.1 - 193.0.0.14 will flow through Eth1.

8.3.4.5 Preferences –> Network –> HTTP stream proxy

This page allows declaring a proxy used for HTTP streaming.

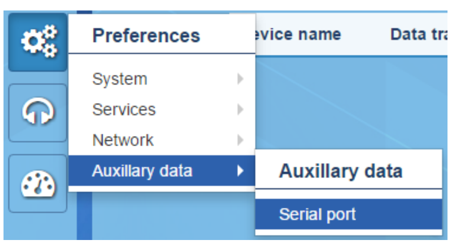


Parameter	Type	Description
IP address	Read/Write	IP address (or domain name) of the HTTP proxy.
Port	Read/Write	TCP Port for the HTTP proxy (80 by default)
Exceptions	Read/Write	Default is None. Select “Locals” to bypass the HTTP stream proxy for local IP addresses.

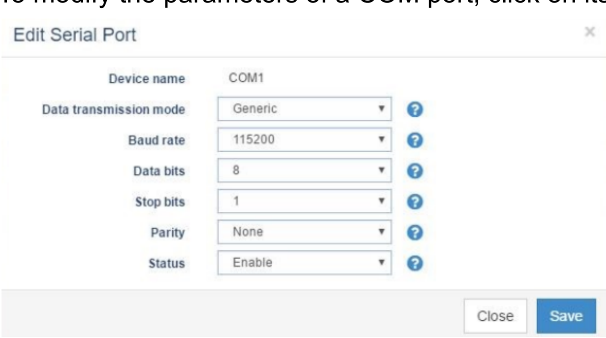
8.3.5 Preferences –> Auxiliary data

8.3.5.1 Preferences –> Auxiliary data –> Serial port

This page allows enabling/disabling the available RS232 ports (optional hardware), and setting their configuration.



To modify the parameters of a COM port, click on its icon on the left column.



Parameter	Type	Description

Device name	Read	Name of the RS232 port
Data transmission mode	Read/Write	Defines the way serial data are inserted into the IP audio stream. Generic: serial data are inserted as they arrive. UECP: serial data are inserted each time a complete RDS UECP frame is fully received from the RS232 port.
Baud rate	Read/Write	Serial port baud rate in bits/s, from 1200 bps to 40 Kbits/s
Data bits	Read/Write	Select the number of bits for each character (6, 7 or 7)
Stop bits	Read/Write	Enter the number of bits used to signal the end of a character: 1 or 2.
Parity	Read/Write	Select the method used for detecting errors on the RS232 port transmission: ● None: No ● Odd: number of bits of each character (including the parity bit) is always odd. ● Even: number of bits of each character (including the parity bit) is always even.
Status	Read/Write	Enable: the COM port is enabled. Disable: the COM port is disabled.

Click on “Save” to confirm the changes.

8.3.5.2 Preferences –> Auxiliary data –> GPIO

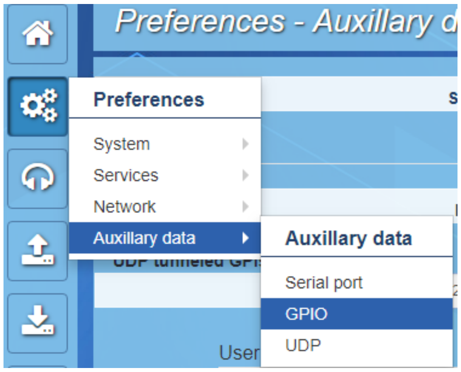
SERV/LINK offers the possibility to use physical GPIOs, and/or virtual GPIOs received through a UDP port. The status of the physical or virtual GPI’s is tunneled in-band so that the decoder can output the status information on physical or virtual GPOs. Note that physical GPIOs are a hardware option of IQOYA SERV/LINK.
Virtual GPIOs allow third party applications to send/receive status information via IP to/from IQOYA. IQOYA SERV/LINK can tunnel 32 virtual GPI status among all the status it receives.

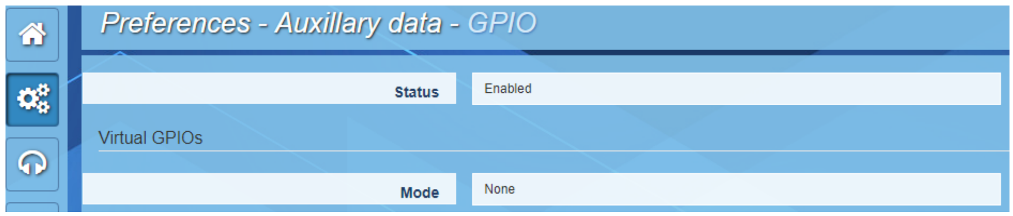
Structure of a virtual status information frame over UDP
Here is the structure of a UDP frame with virtual GPI triggers, that has to be sent to IQOYA SERV/LINK .

32-bit word 1:	Version number (4 bits) = 0000	User ID1 (24 bits)
32-bit word 2:	32 bits. Bit 0 = Status GPI0 –> Bit 31 = Status GPI31	
32-bit word 3	Validation mask (32 bits)	

The frame contains 32 status bits (triggers). The validation mask allows defining what are the relevant status bits delivered by the application.
The User ID allows identifying the application that generates the frame.
Several applications may send UDP GPI frames to the same SERV/LINK (on the same UDP port).

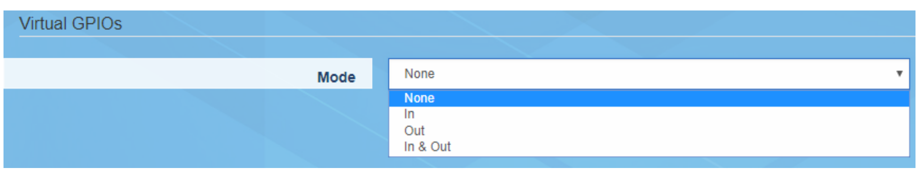
The page Preferences –> Auxiliary data –>GPIO allows enabling/disabling the in-band tunnelling of GPI status information to GPO.





Status: Select enable to activate the status tunnelling.

To declare virtual GPIs to be tunneled, select “In” from the parameter “Mode”.
To declare virtual GPO’s, select “Out” from the parameter “Mode”.
To declare both virtual GPI’s and GPO’s, select “In & Out”.

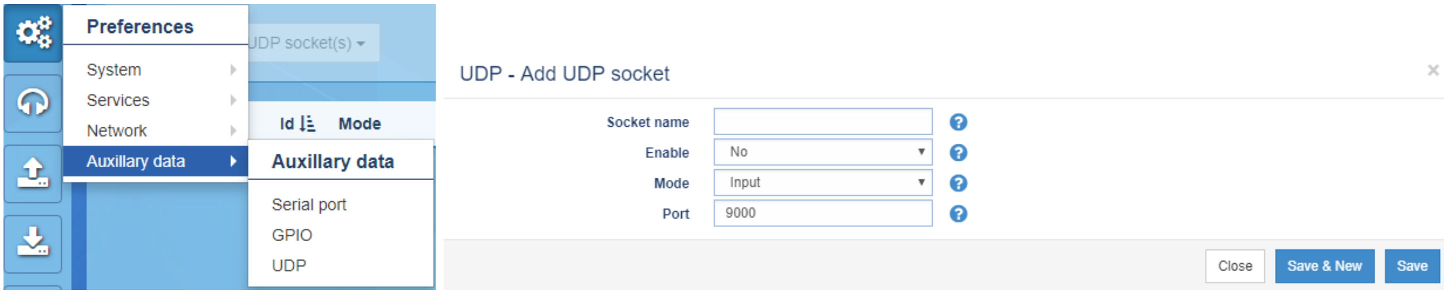


The following screen capture corresponds to the mode “In & Out”.

Parameter	Type	Description
UDP Tunneled GPIs		
User ID	Read/Write	Allows defining the application application. A group contains 32 input status bits.
UDP GPI1	Read/Write	Click on [+] to declare an additional input status. You may enter a maximum of 32 virtual status, if the SERV/LINK does not have physical GPIs. Enter for each input status (UDP GPIn) its rank among the 32 transported status.
UDP Tunneled GPOs		
User ID	Read/Write	Identifies the IQOYA that sends the Virtual GPOs frame.
Repetition frequency	Read/Write	Defines how often the GPO values have to be repeated
IP Destination:Port	Read/Write	IP@and UDP port the UDP frames of virtual GPOs are sent to.
DSCP	Read/Write	Quality of service giver to the virtual GPOs UDP frames.

8.3.5.3 Preferences –> Auxiliary data –> UDP

This page allows defining the UDP ports used for receiving and /or sending data over IP (text, characters, commands).

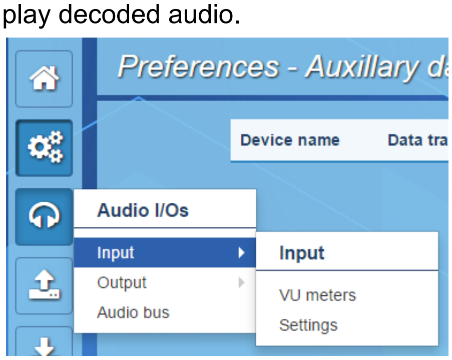


Parameter	Type	Description
Socket name	Read/Write	Name given to the UDP socket. This name allows selecting the socket for tunnelling data, in the Send–>IPService and Receive–>Program pages.
Enable	Read/Write	Yes: socket is enabled. No, socket is disabled.
Mode	Read/Write	Input: IQOYA reads the data to be tunneled from the socket. Output: IQOYA sends data through this socket.
Port	Read/Write	UDP port of the socket

Data received via a UDP port is inserted in the IP audio stream, provided that this UDP port has been selected as the source of auxiliary data to be tunneled.
For an Icecast/Shoutcast, data has to conform to the standard ICY-metadata syntax.

8.4 Audio I/Os

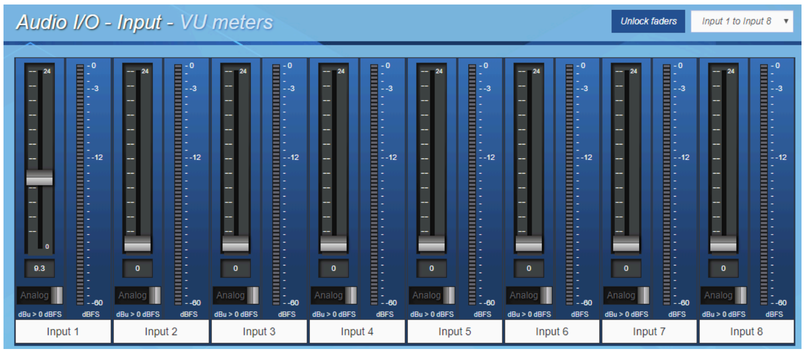
This category gathers all the menus allowing for the configuration of the inputs that can be encoded, and the outputs that play decoded audio.



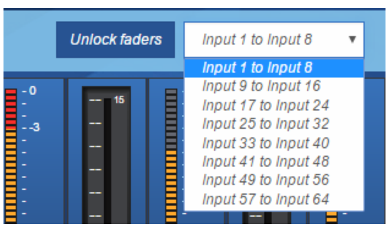
8.4.1 Audio I/O –> Input

8.4.1.1 Audio I/O –> Input –> VU meters

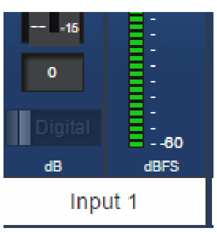
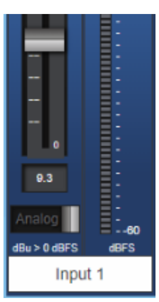
This page displays the level of the signals incoming on the inputs (Line analog, AES/EBU, or MADI depending on the SERV/LINK configuration).



Displayed VU-meters unit is dBfs.
VU-meters are displayed in groups of 8 channels. For a SERV/LINK with more than 8 mono channels, the group of channels to be displayed is selectable from the top right menu.



Select "Unlock faders" to change the input gains.
If the SERV/LINK features analog inputs, it is possible to adjust both the analog input gain and the digital input gain.
Selection of analog or digital gain is done thanks to the selector below the fader.

	When Digital is selected, a digital gain/attenuation is applied to the input signal.
	When Analog is selected (only possible on SERV/LINK versions with analog inputs) an analog gain/attenuation is applied to the input signal. The value displayed below the fader corresponds to the input signal level which gives 0 dBfs after analog to digital conversion

8.4.1.2 Audio I/O -> Input -> settings

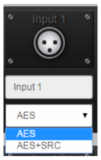
This page allows the following:

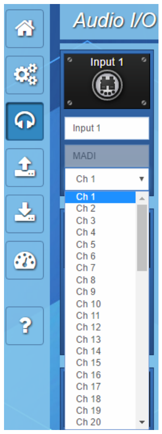
- select the input signals to be allocated to the encoder inputs
- naming of the encoder inputs

For analog & AES/EBU SERV/LINK versions, input type can be “Line”, “AES” or “AES+SRC” (choice is valid for two consecutive channels).
For AES/EBU only SERV/LINK versions, input type can be “AES” or “AES+SRC” (choice is valid for two consecutive channels).
For MADI versions, it is possible to select the MADI channel associated with the input.

AES+SRC means that the hardware sample rate converter of the AES/EBU input is enabled. This is useful when the AES/EBU audio signal is not synchronous of the reference sampling clock

Click on the name of an input (below an XLR icon) to be able to select the audio input and change the input name.



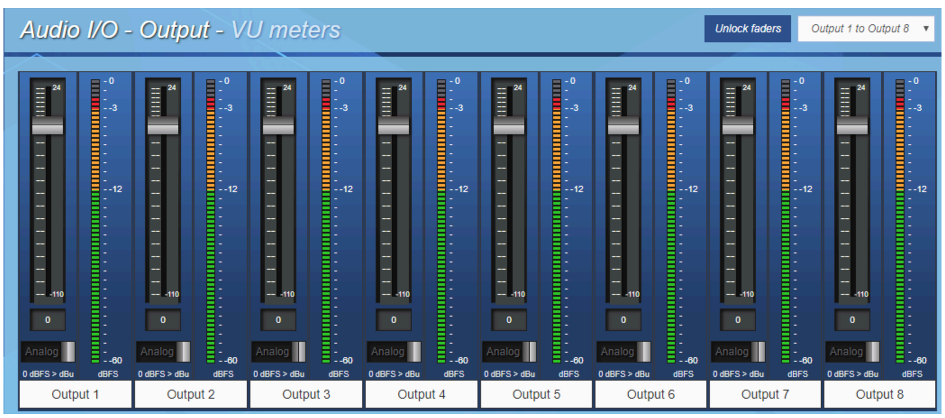


Click on “Apply” to confirm the changes.

8.4.2 Audio I/O –> Output

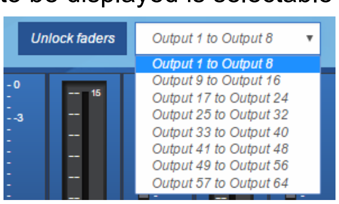
8.4.2.1 Audio I/O –> Output –> VU meters

This page displays the level of the output signals.



Displayed VU-meters unit is dBfs.

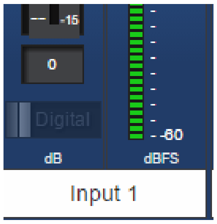
VU-meters are displayed in groups of 8 channels. For a SERV/LINK with more than 8 mono channels, the group of channels to be displayed is selectable from the top right menu.



Select “Unlock faders” to change the output gains.

If the SERV/LINK features analog outputs, it is possible to adjust both the analog output gain and the digital output gain.

Selection of analog or digital gain is done thanks to the selector below the fader.

	When Digital is selected, a digital gain/attenuation is applied to the output signal.
	When Analog is selected (only possible on SERV/LINK versions with analog outputs) an analog gain/attenuation is applied to the output signal. The value displayed below the fader corresponds to the level of the output signal for a 0 dBfs digital signal.

8.4.2.2 Audio I/O -> Output -> settings

This page allows the following:

- assign a physical output to a decoder output signal
- naming of the encoder outputs

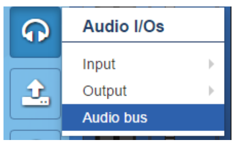
For analog & AES/EBU SERV/LINK versions, output type is “Line+AES”: output signal is sent on both analog and digital outputs..
For AES/EBU only SERV/LINK versions, output type is “AES”.
For MADI versions, it is possible to select the MADI channel associated with the output signal.

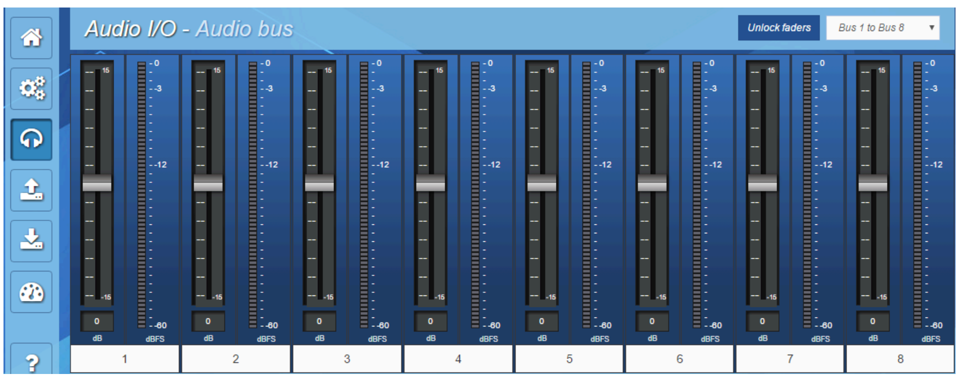
Click on the name of an output (below an XLR icon) to be able to select the audio output and change the output name.

Click on “Apply” to confirm the changes.

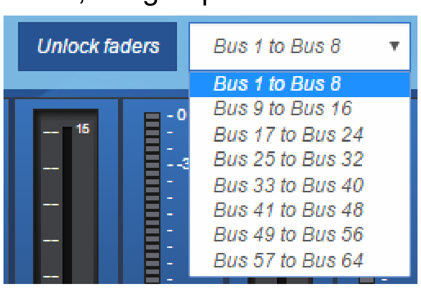
8.4.2.3 Audio I/O -> Audio Bus

Audio buses are useful for the transcoding of IP streams. An audio bus can be the destination of one or several output programs, and the source of input programs. The number of available audio buses is defined by the license.





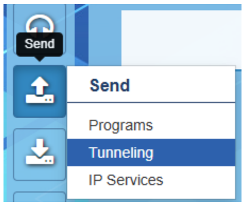
Audio buses are displayed in groups of 8 channels. For a SERV/LINK offering more than 8 mono channels for the buses, the group of channels to be displayed is selectable from the top right menu.



To change the name of a bus channel, click on its name below its fader.

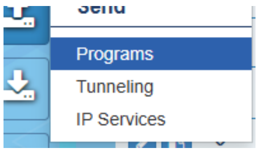
8.5 “Send”

This category allows defining the programs and the IP services to be streamed. The principle consists in first declaring the programs, and then declaring the IP services that carry the programs.



8.5.1 Send –> Programs

This page allows viewing and declaring the audio encoding instances: the programs.



It can be accessed either from the left column, or from the icon “Go to programs” on the top right of the IP Services page 

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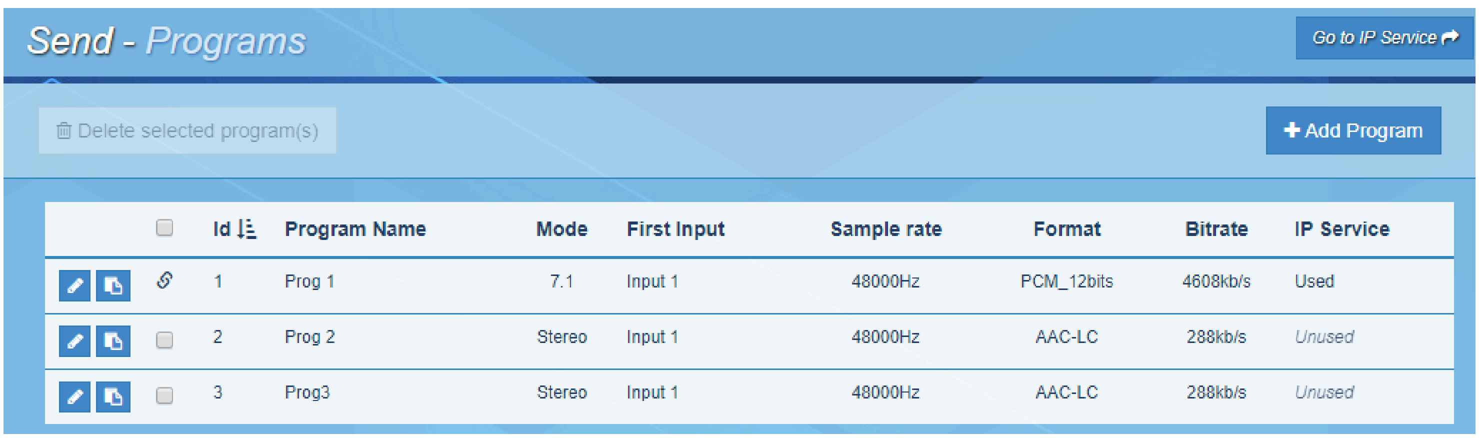
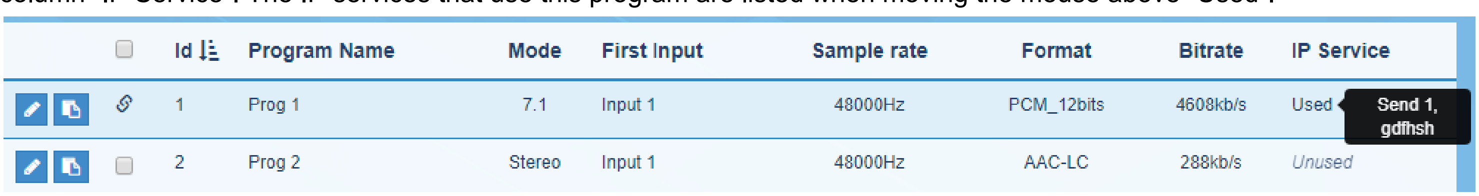


	Id	Program Name	Mode	First Input	Sample rate	Format	Bitrate	IP Service
	1	Prog 1	7.1	Input 1	48000Hz	PCM_12bits	4608kb/s	Used
	2	Prog 2	Stereo	Input 1	48000Hz	AAC-LC	288kb/s	*Unused*
	3	Prog3	Stereo	Input 1	48000Hz	AAC-LC	288kb/s	*Unused*

In case some programs are already created, they are listed in the Programs page, with their characteristics: name, mode, first audio input, sample rate, audio format, bitrate, IP Service using this Program.

If a Program is used in at least one IP Service, the icon  is displayed on the left of its name, and “Used” appears in the column “IP Service”. The IP services that use this program are listed when moving the mouse above “Used”.



	Id	Program Name	Mode	First Input	Sample rate	Format	Bitrate	IP Service
	1	Prog 1	7.1	Input 1	48000Hz	PCM_12bits	4608kb/s	Used
	2	Prog 2	Stereo	Input 1	48000Hz	AAC-LC	288kb/s	*Unused*

If a Program is not used by any IP Service, the selection button  is displayed on the left of its name.
A Program can be associated with one IP Service. Only unused Programs can be selected in an IP Service.

To declare a new Program, click on the icon 

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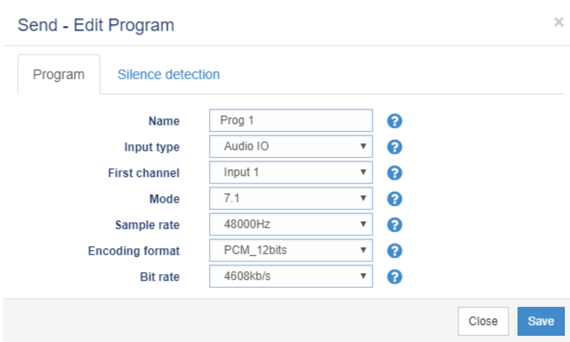


Give a unique name to the program.
Click on “Save” to confirm the parameters.
Click on “Close” to discard the changes.
Click on “Save & New” to confirm the settings, and duplicate them so that to create a new program with similar settings, except the name.

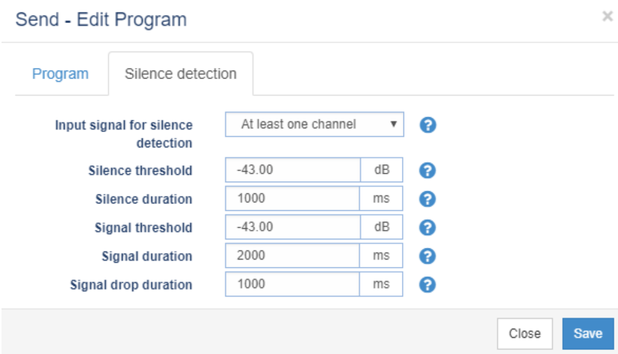
To edit an existing Program, click on the icon in the left side of the Program line.

A new program can be created by duplicating one of the displayed programs; click on the icon in front of the program to be duplicated.

Parameter	**Type**	**Description**
Name	Read/Write	Name given to the encoding instance. This name will be selected when declaring an IP service.
Input type	Read/Write	Audio source of the program: it can be an audio input, or an audio bus.
First channel	Read/Write	First input channel of the audio signal to be encoded, to be selected among the list of input channels.
Mode	Read/Write	Mono, Stereo, Multi-channel 5.1
Sample rate	Read/Write	Frequency of the encoded audio, to be selected from the list box. It may be different from the IQOYA sampling frequency.
Encoding format	Read/Write	Audio format of the encoding, to be selected from the list box.
Bit rate	Read/Write	Bit rate of the encoded audio.

Silence detection parameters

Click on the “Silence detection” tab to set the criteria for silence detection on this program. An alarm is signalled when silence is detected, and it is reset when signal is detected again.
It is also possible to automatically stop/start the streaming upon silence/signal detection. This can be configured from the IP Service page (see next paragraph **Send –> IP services**).



Input signal for silence detection	In case IQOYA is used as an encoder, it can generate an alarm when silent audio is detected on the audio inputs, and sets this alarm off when the audio signal is detected again. (Note that all the alarms handled by IQOYA can be enabled/disabled from the “Alarms setup” menu). The parameter “Input signal for silence detection” allows defining on which input signal the silence detection is applied. Possible choices are: **- Mean of left + right channels**: compares the mean value of a left and right sample to the threshold. In case the calculated values are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- Left channel only**: compares the left channel samples to the silence threshold. In case the sample values are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- Right channel only**: compares the right channel samples to the silence threshold. In case the sample values are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- Left and right channels**: compares both the left and right channel samples to the silence threshold. In case the sample values on both channels are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- At least one channel:** compares both the left and right channel samples to the silence threshold. In case the sample values on at least one of the two channels are always lower to the silence threshold during the defined silence duration, the silence condition is reached.
Silence threshold & *Silence duration*	Silent audio is defined through these two parameters, expressed in dBfs. When the audio level is below the threshold value during at least the defined duration, the alarm “Analog audio in silent” or “Digital audio in silent” is set (if it is enabled from the “Alarms setup” menu).
Signal threshold *Signal duration* *Signal drop duration*	Audio signal is defined through the three parameters. Audio signal is considered as recovered if all the following conditions are true: ● Audio level exceeds the Signal threshold (dBfs) within the “Signal duration” analysis window (ms). ● Audio level does not stay below the Signal threshold during the “Signal drop duration”, within the “Signal duration” analysis window. Note the following rule: Signal drop duration <= (Signal duration / 2). Once the signal is recovered, the alarm “Analog audio in silent” or “Digital audio in silent” is reset (if it is enabled from the “Alarms setup” menu).

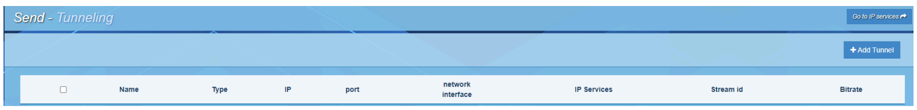
Click on Save button to confirm the new Program.

To delete one or several unused programs, select them by clicking on the icon on the left of their names, and click on the button Delete selected program(s) on the top of the Programs list.

If all Programs are unused and you want to delete them all, click on the icon on the left of the column title "Program Name" (this selects all the Programs), and click on the button Delete selected program(s) on the top of the Programs list. Confirm or cancel your choice in the displayed confirmation window.

8.5.2 Send –> Tunneling

This page allows creating tunnels that support unidirectional and bidirectional transport of IP audio streams.



To declare a tunnel, click on the icon "+Add Tunnel".

RIST tunnel

RIST (Reliable Internal Stream Transport) is an open specification transport protocol designed for reliable transmission of video and audio over lossy networks (including the internet) with low latency and high quality. RIST uses retransmission bandwidth throttling which ensures the link keeps going and that it retains as much quality as possible even when bandwidth is limited.

RIST specifies several profiles, each adding more capabilities. The following describes the most significant features of each profile.

Simple Profile

- Interoperable Automatic Repeat reQuest (ARQ) with configurable behaviour for recovery of packet loss, packet reordering and link failure
- Removal of network introduced jitter
- Transport of point-to-point dual stream services
- Bonding of several links using link aggregation (multipath)

Main Profile

It is based on the simple profile, and adds the following features:

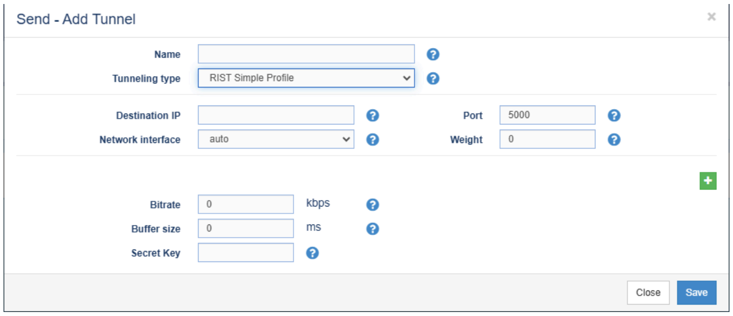
- Transport of point-to-multipoint services
- Stream encryption for secure content
- VPN tunnelling for secure sender/receiver communication
- NAT traversal for improved interworking with consumer-style internet connectivity
- Null packet suppression for saving bandwidth

Enhanced Profile

It adds the following improvements to the main profile.

- Enhanced tunnelling capabilities
- Enhanced PSK Security
- Reduce size of packet headers

The parameters of a tunnel are the following



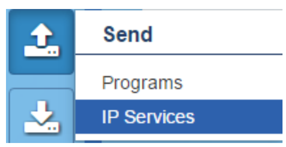
Parameter	Type	Profile	Description
Name	Read/Write	All	Name given to the tunnel
Tunneling type	Read/Write	All	Selection of profile: RIST simple, main, or Advanced profile
Destination IP	Read/Write	All	Destination IP address or domain name You can declare several destinations for multipath streaming, by clicking on the icon
Port	Read/Write	All	Destination UDP port of the destination
Network interface	Read/Write	All	“Auto” for a unicast destination. Eth selection for multicast (not supported in the RIST simple profile).
Weight	Read/Write	All	Sets the relative share (load balancing) for multipath connections. Exemple: in a setup with two declared paths (two destination “IP@:UDP port”, if a paths are given weights of 5 and 10 respectively, the former would receive 1/3 of packets sent (5 / 15), and the latter would receive 2/3 (10/15). A weight of 0 means that all packets flow through the path.
Bitrate	Read/Write	All	Sets the maximum bandwidth in Kbps. It is necessary to configure the bandwidth to be higher than the max bandwidth of your stream(s). This is in order to allow room for messaging headroom, plus the re-requested packets.

			When tuning a connection for the first time, analyze your stream statistics locally at first, then start at 10% higher for a constant bitrate, 100% higher for variable bitrate
Buffer size	Read/Write	All	Sets the buffer size in milliseconds. The buffer size will work best at four to seven times the ping time. This allows time for requests for the retransmission of a lost or corrupted packet, and the subsequent retransmission of its replacement.
Secret key	Read/Write	Main, Advanced	Sets the specified passphrase for Main or Advanced profile encryption.

Click “Save” to save the tunnel parameters.

8.5.3 Send –> IP services

This page allows viewing and declaring the IP Services to be streamed over IP.

It can be accessed either from the left column 

, or directly from the icon “Go to IP Service” on the top right of the Programs page 

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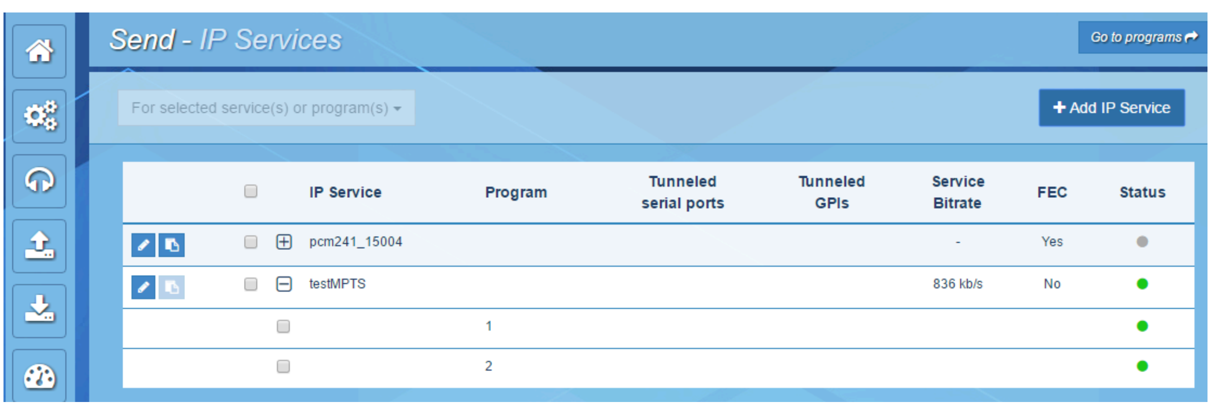

In case some IP Services are already created, they are listed in the IP Services page, with their characteristics: Name, Program, tunneled ports, tunneled GPIs, bitrate, FEC, Status.

The program(s) carried by an IP service can be displayed by clicking on the icon  right on the left of the IP service name (an IP service can contain several programs in case of MPEG-TS MPTS encapsulation).

To declare a new IP Service, click on the icon 

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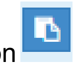
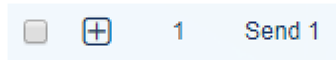
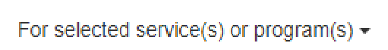
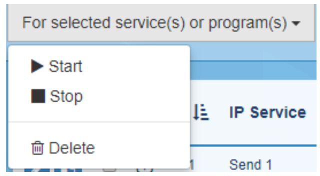
A new IP service can also be created by duplicating an existing one. Click on the icon  on the left of the IP service to be duplicated.

To start, stop, or delete an IP service, check the box on the left of its name 

, and select the appropriate action:





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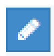
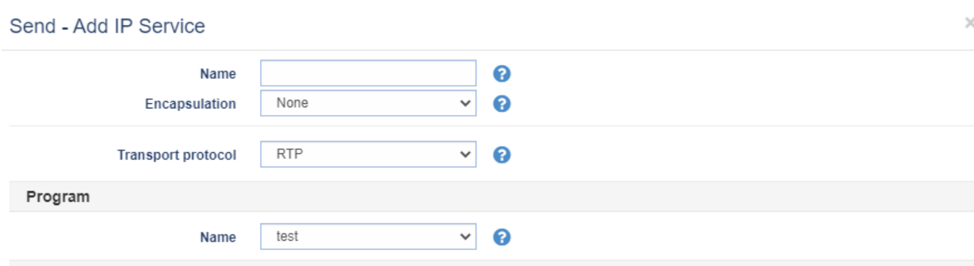
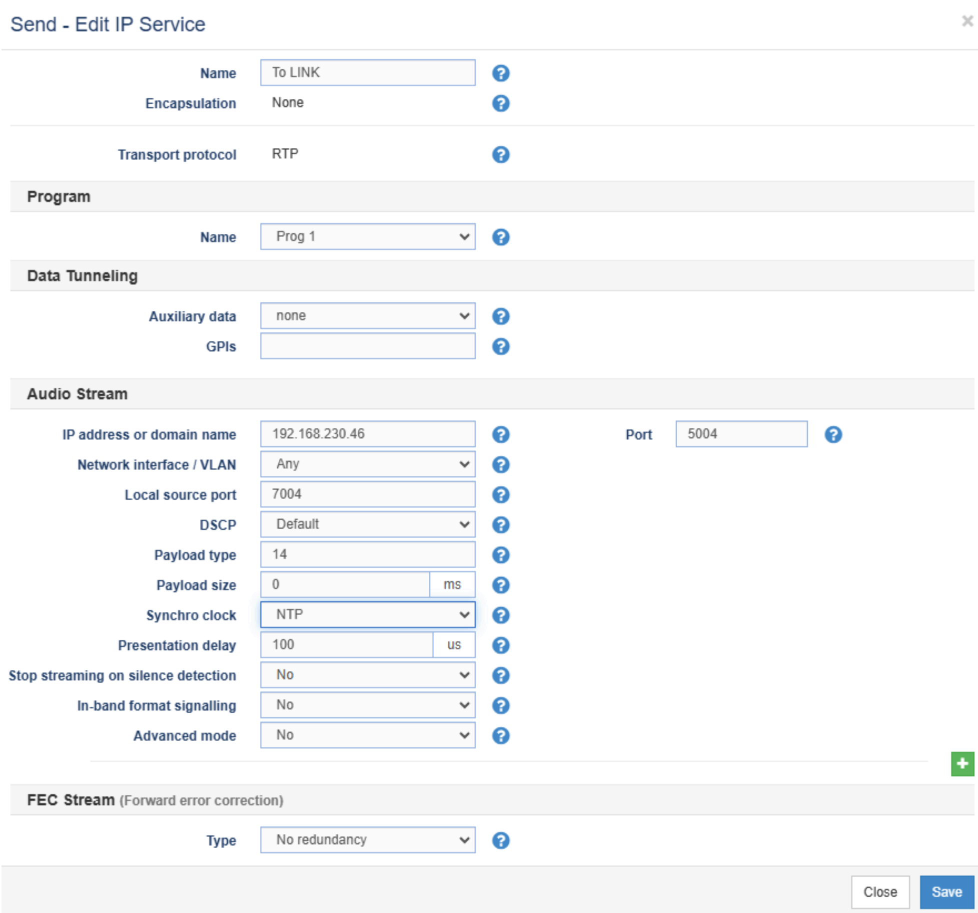
Note that a list of consecutive services can be selected by clicking on the first service check box, and shift clicking on the check box of the last service of the list.
Non consecutive services can be selected by CTRL clicking on their check boxes.

To edit an existing IP Service, click on the icon  on the right end of the IP Service line.
The following window is displayed.

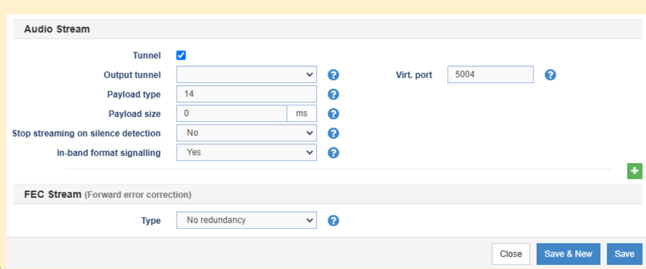


Parameter	Type	Description
Name	Read/Write	Name given to this IP service
Encapsulation	Read/Write	None: The IP Service includes one Program, and audio data is not encapsulated (raw mode). MPEG-TS SPTS: The IP Service includes one audio Program with Transport Stream encapsulation. MPEG-TS MPTS: The IP Service includes several audio Programs which are multiplexed in a single MPEG-TS stream
Transport protocol	Read/Write	Available protocols are: RTP, UDP, HTTP, HLS. HTTP is to be used for streaming to an Icecast/Shoutcast server. HLS is to be used to stream to an HLS streaming server. For low latency real time streaming, we recommend RTP.. UDP is to be used only if the equipment that receives the stream does not support standard ACIP RTP streams.

Encapsulation = None and Transport protocol = RTP

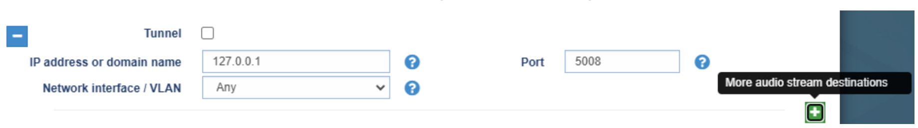


Parameter	Type	Description
Program		
Name	Read/Write	Select the Program to be streamed from the list of Programs. A Program can be used by several IP services.
Data tunnelling (This section is displayed only if tunnelling has been activated from Preferences->Auxiliary data->GPIO)		
Auxiliary data	Read/Write	If there is a serial port hardware option installed, select the serial port that provides the serial data to be tunneled in-band.
GPIs	Read/Write	If there is a GPIO hardware option installed, enter a list of GPI numbers whose status is to be tunneled in-band. Numbers start from 1 and must be separated by commas
Audio stream		
Tunnel	Read/Write	Check this box if the stream has to be included in a tunnel (RIST protocol)
Case “Tunnel” unchecked (the stream is not included in a RIST tunnel).		
IP address or Domain name	Read/Write	Destination IP address (unicast or multicast) or domain name.
Port	Read/Write	Enter the destination UDP port.

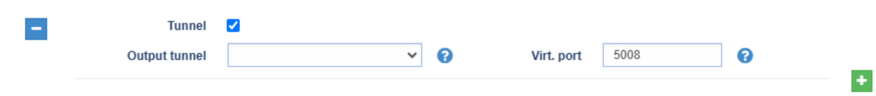
Network interface/VLAN	Read/Write	Select the network interface, bond or VLAN for this stream. In case the target address is unicast, select “Any” so that the Eth interface is determined automatically according to this IP address, or select a bond or a VLAN. In case the target IP address is multicast, select the Eth interface, the bond or VLAN.
Audio stream: Local source port	Read/Write	Local UDP port number of IQOYA SERV/LINK
Audio stream: DSCP	Read/Write	Select the quality of service (QoS) class of the stream.
Audio stream: Payload type	Read/Write	RTP payload value that defines the audio profile. Standard values are: ● 0 for G711 ● 9 for G722; 14 for MPEG ● 96 for AAC, apt-X, Opus
Audio stream: Payload size	Read/Write	Size (in ms) of the audio transported by an RTP frame. For unframed formats (like PCM, G7xx, apt-X), payload size value is rounded to the nearest multiple value that is equal or higher than the processing granularity value. For framed formats (like MPEG, AAC), payload size value is rounded to the nearest multiple value equal or higher than the frame size.
Audio stream: Stop streaming on silence detection	Read/Write	IQOYA can automatically stop streaming and restart streaming upon silence/signal detection on the audio source. This feature can be enabled by setting this option to “Yes”. As a consequence, a decoder receiving the stream will switch to a backup when silence is detected on the input of the encoder that generates the stream. Set this option to “No” if you want the encoder to stream even when the audio source is silent.
Audio stream:Synchro Clock	Read/Write	None, NTP, or NTP for *LINK. “NTP” and “NTP for *LINK” can be selected if the option “NTP based audio synchro” is installed. When streaming to IQOYA SERV/LINK and IQOYA X/LINK, “NTP” must be selected. When streaming to old IQOYA *LINK, “NTP for *LINK” must be selected.
Audio stream:Presentation delay	Read/Write	Valid if Synchro Clock is set to NTP. Offset of time added to the current NTP time for time-stamping the IP packets so that several decoders play the packets at the same time. This value, expressed in microseconds, must be at least equal to the maximum network transport time for an IP packet to reach the target decoders. Once the encoder and the decoders are configured, this value can be tuned by checking the IP metrics. The maximum value is 2 000 000 microseconds (2 seconds) for unframed audio formats (PCM, apt-X, G7xx), and 256 frames for framed audio formats (MPEG, AAC). In MPEG Layer 2 48 kHz, this corresponds to 6 seconds (6 000 000 microseconds).
In-band format signalling	Read/Write	**Yes**: the description of the audio format is inserted in the IP audio stream so that the decoder can automatically adapt to the received format. This works only with IQOYA encoders and decoders. In this mode, FEC stream is sent to the same destination IP address as the IP audio stream, on UDP port +2. **No**: the decoder must be configured to receive the appropriate audio format. In this mode, FEC stream destination IP address and UDP port can be configured.
Case “Tunnel” checked (the stream is included in a RIST tunnel).		

Output tunnel	Read/Write	Select the RIST tunnel from the list of tunnels that have been created.
Virtual port	Read/Write	Enter a virtual port number which identifies this stream in the tunnel.
Payload type	Read/Write	RTP payload value that defines the audio profile. Standard values are: ● 0 for G711 ● 9 for G722; 14 for MPEG ● 96 for AAC, apt-X, Opus
Payload size	Read/Write	Size (in ms) of the audio transported by an RTP frame. For unframed formats (like PCM, G7xx, apt-X), payload size value is rounded to the nearest multiple value that is equal or higher than the processing granularity value. For framed formats (like MPEG, AAC), payload size value is rounded to the nearest multiple value equal or higher than the frame size.
Stop streaming on silence detection	Read/Write	IQOYA can automatically stop streaming and restart streaming upon silence/signal detection on the audio source. This feature can be enabled by setting this option to "Yes". As a consequence, a decoder receiving the stream will switch to a backup when silence is detected on the input of the encoder that generates the stream. Set this option to "No" if you want the encoder to stream even when the audio source is silent.
Inband format signalling	Read/Write	**Yes**: the description of the audio format is inserted in the IP audio stream so that the decoder can automatically adapt to the received format. This works only with IQOYA encoders and decoders. In this mode, FEC stream is sent to the same destination IP address as the IP audio stream, on UDP port +2. **No**: the decoder must be configured to receive the appropriate audio format. In this mode, FEC stream destination IP address and UDP port can be configured.

Click on the icon on the bottom right of the page to add an IP destination.

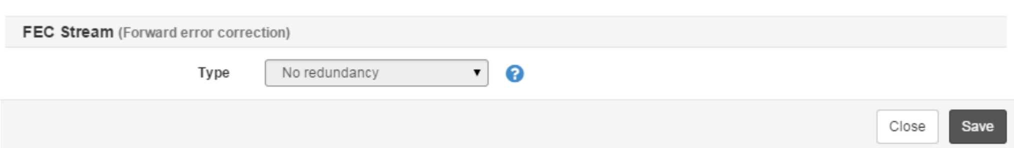


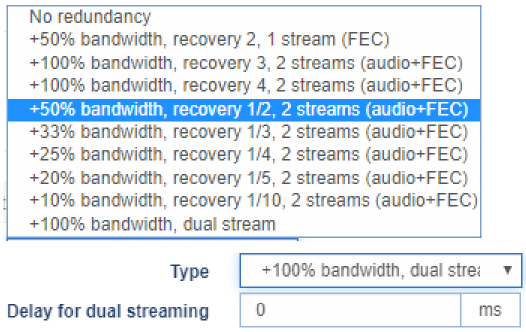
If no tunnel is to be used for this destination, enter the new target IP address, UDP port, and the network interface through which the stream is sent.
If a tunnel is to be used, select the tunnel and a virtual port to identify this stream in the tunnel.



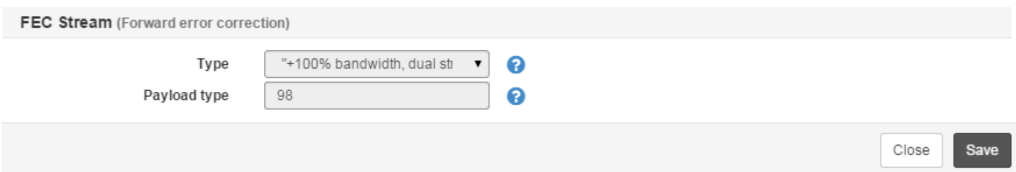
Click on the icon to remove a destination.

A FEC can be selected. FEC consists in sending additional data so that the decoder can recover lost packets.
The amount of additional frames defines the recovery performance.
Note that if a RIST tunnel is used, FEC should not be used, as RIST aims to get a reliable transport with optimised bitrate and latency

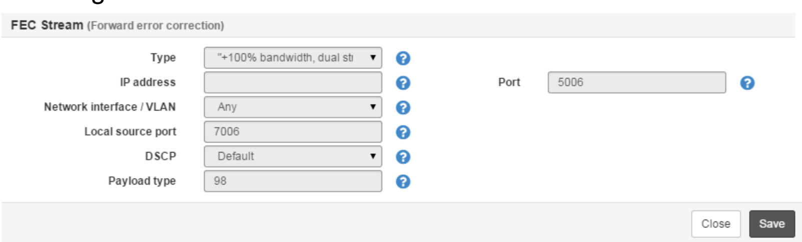


	FEC on 1 stream means that additional data is sent in the IP audio stream (in-band). FEC on 2 streams means additional data is sent as a second IP stream. Dual stream FEC means that the IP stream is duplicated. When no delay is selected, primary stream and redundant stream are sent at the same time. When a delay is selected, the redundant stream is delayed with regards to the primary stream.

In case parameter “In-band format signalling” is set to “Yes”, the destination IP address and UDP port of the FEC stream cannot be configured. The destination IP address is the same as for the primary stream, and the destination UDP port is equal to “primary stream destination UDP port + 2”.



In case parameter “In-band format signalling” is set to “No”, the destination IP address and UDP port of the FEC stream can be configured.

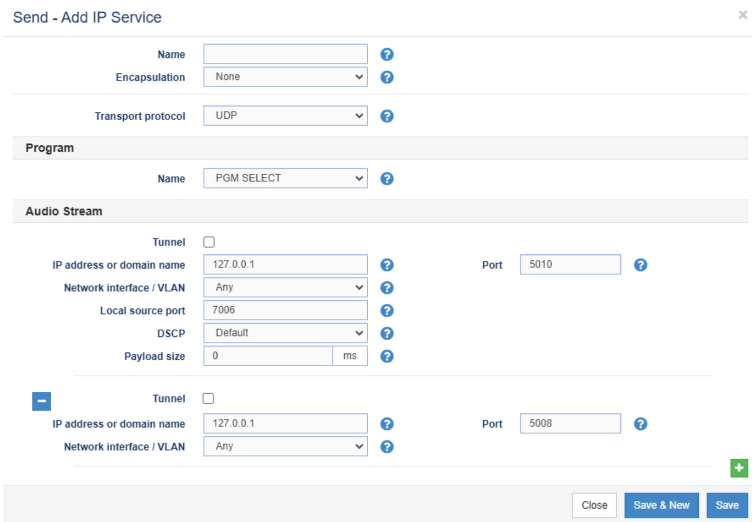


Parameter	Type	Description
FEC stream (Forward error Correction)		
IP address	Read/Write	Enter the destination IP address (unicast or multicast) of the FEC stream.
Port	Read/Write	Enter the destination UDP port of the FEC stream.
Network interface/VLAN	Read/Write	Select the network interface or VLAN or bond for this FEC stream. In case the target address is unicast, select “Any” so that the Eth interface is determined automatically according to this IP address, or select a VLAN or a bond. In case the target IP address is multicast, select the Eth interface or the VLAN, or the bond.
Local source port	Read/Write	Local UDP port number of IQOYA SERV/LINK
DSCP	Read/Write	Select the quality of service (QoS) class of the FEC stream.
Payload type	Read/Write	RTP payload of the FEC stream. Value 98 is recommended.

Click on “Save” to confirm the settings. Click on “Save & New” to confirm the settings and create a new IP service with the same parameters.

Click on “Close” to discard the settings.

<u>Encapsulation = None and Transport protocol = UDP</u>



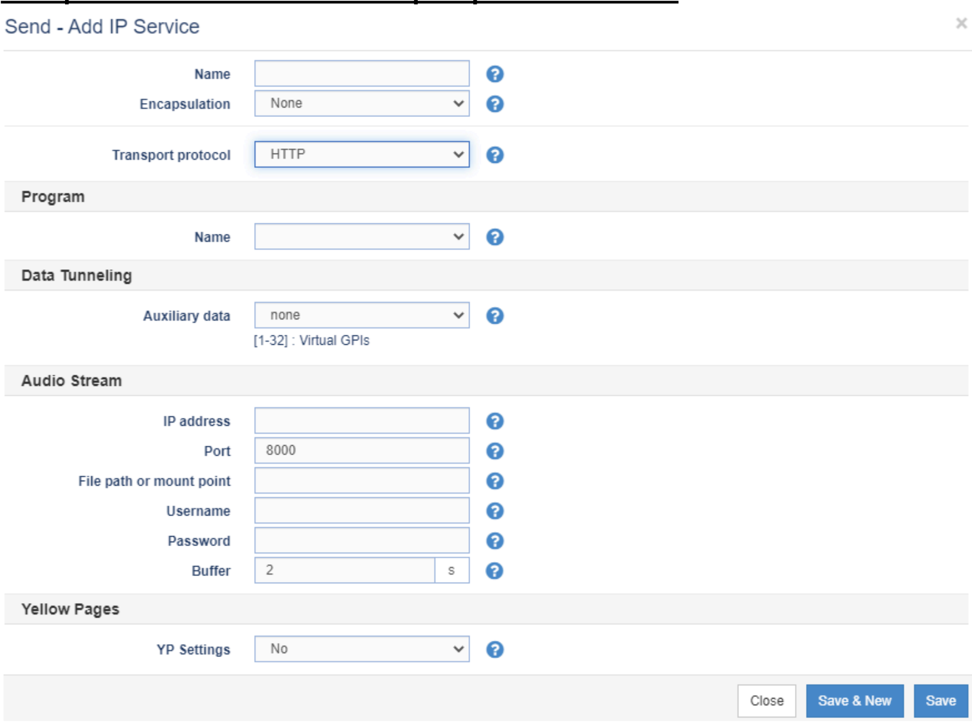
Parameter	**Type**	**Description**
Program		
Name	Read/Write	Select the Program to be streamed from the list of Programs. A Program can be used by several IP services.
Audio stream		
Tunnel	Read/Write	Check this box if the stream has to be included in a tunnel (RIST protocol)
Case “Tunnel” unchecked (the stream is not included in a RIST tunnel).		
IP address or Domain name	Read/Write	Destination IP address (unicast or multicast) or domain name.
Port	Read/Write	Enter the destination UDP port.
Network interface / VLAN	Read/Write	Displayed if a multicast destination IP address has been entered. Select the network interface, or VLAN or bond that is to be used for streaming.
Audio stream: Local source port	Read/Write	Local UDP port number of IQOYA SERV/LINK
Audio stream: DSCP	Read/Write	Select the quality of service (QoS) class of the stream.
Audio stream: Payload size	Read/Write	Size (in ms) of the audio transported by an RTP frame. For unframed formats (like PCM, G7xx, apt-X), payload size value is rounded to the nearest multiple value that is equal or higher than the processing granularity value. For framed formats (like MPEG, AAC), payload size value is rounded to the nearest multiple value equal or higher than the frame size.
Case “Tunnel” checked (the stream is to be included in a RIST tunnel).		
Output tunnel	Read/Write	Select the tunnel name from the list of tunnels that have been created.
Virtual port	Read/Write	Enter a virtual port number which identifies this stream in the tunnel.

Payload size	Read/Write	Size (in ms) of the audio transported by an RTP frame. For unframed formats (like PCM, G7xx, apt-X), payload size value is rounded to the nearest multiple value that is equal or higher than the processing granularity value. For framed formats (like MPEG, AAC), payload size value is rounded to the nearest multiple value equal to or higher than the frame size.

Click on the icon to steam the program to an additional IP destination.

Click on “Save” to confirm the settings. Click on “Save & New” to confirm the settings and create a new IP service with the same parameters.
Click on “Close” to discard the settings

Encapsulation = None and Transport protocol = HTTP



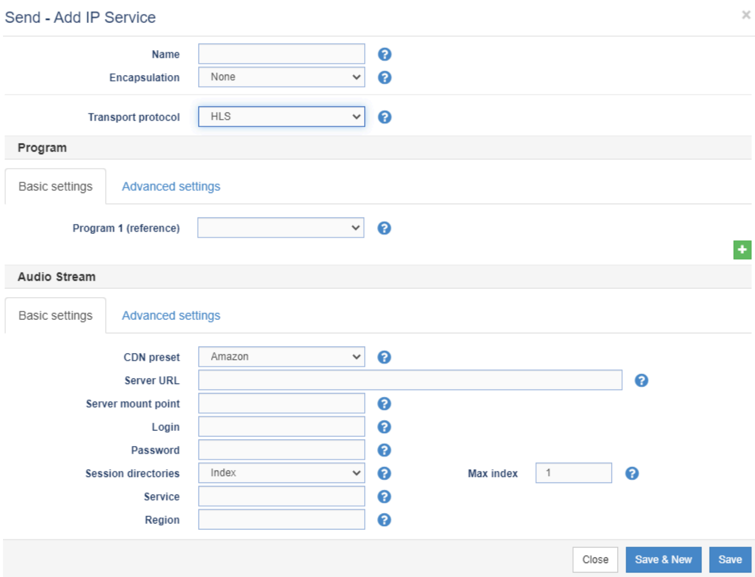
Parameter	Type	Description
Program		
Name	Read/Write	Select the Program to be streamed from the list of Programs. A Program can be used by several IP services.
Data tunnelling (This section is displayed only if tunnelling has been activated from Preferences->Auxiliary data->GPIO)		
Auxiliary data	Read/Write	This is the input port to be selected for collecting the **dynamic metadata** to be inserted in the icecast/shoutcast stream. It can be a serial port (if the hardware option for RS232 ports is installed), or an input UDP port declared from Preferences->Auxiliary data->UDP.
Audio stream		

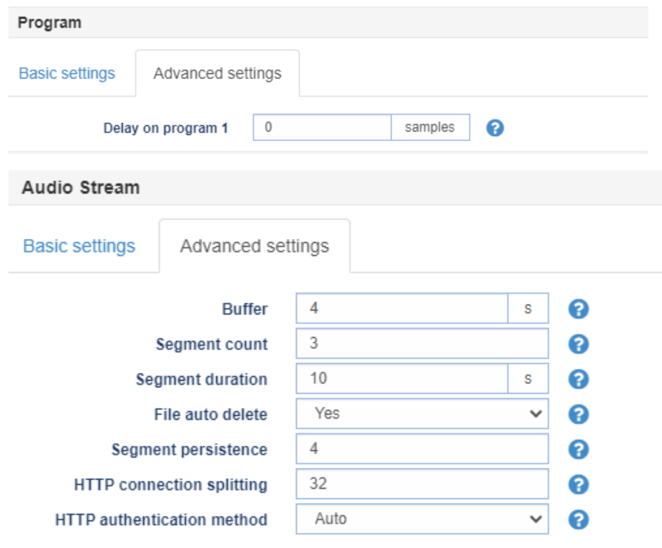
IP address or Domain name	Read/Write	Destination IP address or domain name of the Icecast/Shoutcast streaming server
Port	Read/Write	Destination TCP port.
File path or mount point	Read/Write	File path of the source on a Shoutcast streaming server. File path of the source or mount point of the source on an Icecast streaming server. Example: server URL= http://streamer.myorgnization.com:6400/M1 Mount point is: /M1
Username	Read/Write	Username to access the streaming server.
Password	Read/Write	Password to access the streaming server
Buffer	Read/Write	Size of the buffer used for the streaming to the icecast/Shoutcast server (from 2 to 60 seconds). In case the log file contains 'codec sender underrun" traces, this buffer size has to be increased.
Yellow pages (static metadata for Icecast/Shoutcast streaming)		
YP settings	Read/Write	If static metadata are to be inserted, select Yes. If static metadata is not needed, select No. Each new metadata field to be added can be displayed by clicking on .
Public server	Read/Write	If you would like to make your radio station (server) public.
Stream name	Read/Write	Generally used to specify the name of the radio station or broadcast
Stream description	Read/Write	Generally used to specify the description (or title) of the radio station or broadcast
Stream URL	Read/Write	Generally used to specify the internet address of the radio Web site
Stream genre	Read/Write	Generally used to specify the genre of music or content streamed by the radio station
ICQ#	Read/Write	These labels allow your listeners to make instant music requests or leave feedback on your streams.If you do not have an AIM or ICQ username, or do not wish to include it along with your stream, you should leave these fields blank.
Nt to	Read/Write	
IRC	Read/Write	Useful for those who wish to link their stream to an Internet Relay Chat server. If you do not have a chat room on an IRC server, or do not wish to include it with your stream,you should leave this field blank.
Other	Read/Write	Use this field to send specific metadata to your server. Your data will be inserted without processing.

Click on "Save" to confirm the settings. Click on "Save & New" to confirm the settings and create a new IP service with the same parameters.
Click on "Close" to discard the settings

Encapsulation = None and Transport protocol = HLS

Note: Since version 3.11, it is possible to enable the descriptor “Maximum_Bitrate_Descriptor” in the PMT, and the flag “ES_Rate_flag" in the PES. See appendix D.





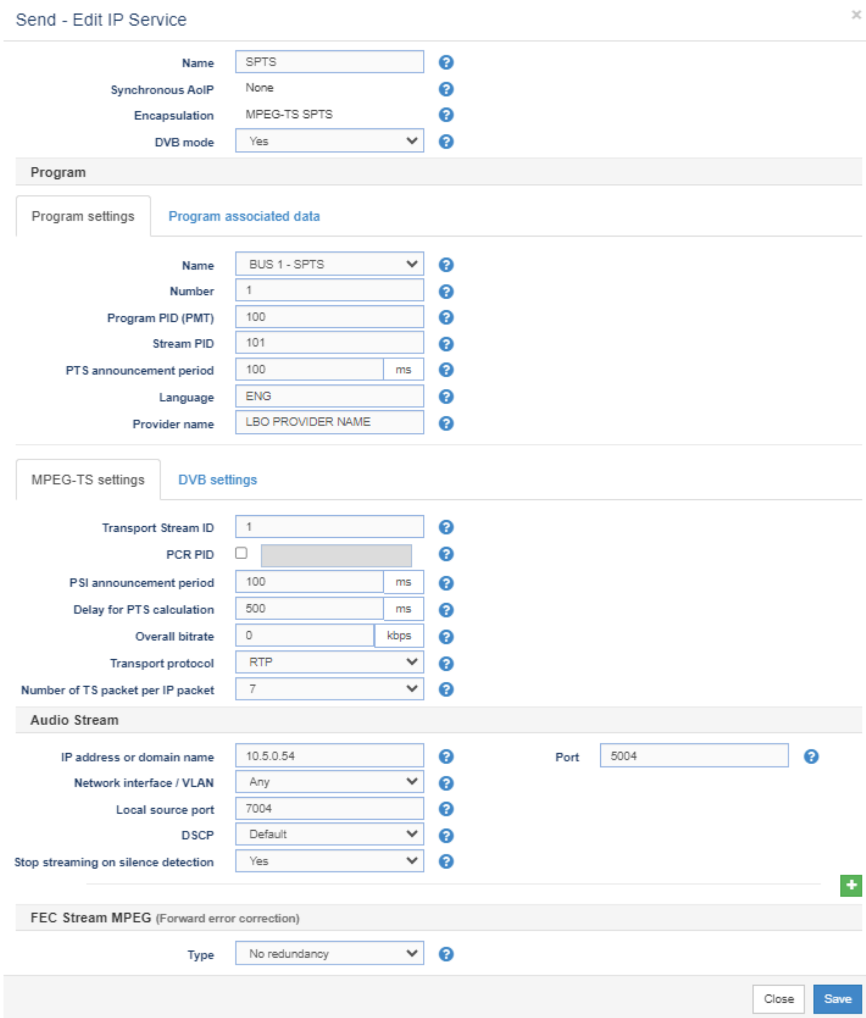
Parameter	Type	Description
Program: Basic settings		
Program 1(reference)	Read/Write	Select the Program to be streamed from the list of Programs. Program can be in MPEG Layer 2, MPEG Layer 3, and AAC-ADTS formats. For multi bitrate HLS, up to 5 additional audio Programs can be selected. They all must be the same audio compression format (but of course with different bitrates). Click on the icon to add a Program.
Program: Advanced settings		
Delay on Program x	Read/Write	Delay (expressed in audio samples) to be applied to the Program number X. This can be used to adjust the synchronization between Programs..
Audio stream: Basic settings		
CDN Preset	Read/Write	Select the preset that corresponds to the CDN you want to use; this will only display the parameters required by this CDN. Presets are: - AWS: for Amazon Web Services (with AWS authentication) - Akamaï (MSL4): for Akamaï CDN Select “None” to access other CDN’s.
Server URL	Read/Write	Address of the origin server to connect to.
Server mount point	Read/Write	Mount point on the origin server the HLS stream has to be pushed to.
Login	Read/Write	For authentication to the origin server.

Password	Read/Write	For authentication to the origin server.
Protocol		Transport protocol used to push HLS data to the origin server. Select HTTP or HTTPS.
Network port		Network port used to push HLS data to the origin server.
Session directories	Read/Write	For AWS. Selectable values are None, date time, Index. - Date time: each time the IP service is started, a subdirectory under the server mountpath will be created with the current date and time. - Index: Date time: each time the IP service is started, a subdirectory"SessionX" under the server mountpath is created. "X" is the index of the directory and is defined with the session index parameter. -None: no subdirectory under the server mountpath is created. When using "Index" or "None" values, previous files can be overwritten each time the IP service is started;
Max index	Read/Write	For AWS CDN only. This parameter is displayed if "Session directories" is set to "Index". This is the number of subdirectories allowed.
Service	Read/Write	For AWS. Name of your amazon media storage service used on the origin server.
Region	Read/Write	For AWS. Name of the region used on the origin server.
Audio stream: Advanced settings		
Buffer	Read/Write	Internal buffer used for the connection to the origin server (from 2 to 60 seconds).
Segment count	Read/Write	Number of segments contained in the HLS playlist (from 3 to 20)
Segment duration	Read/Write	Duration of a segment (from 1 to 30 seconds).
File auto delete	Read/Write	Allows keeping or removing the segment stored on the origin server. The number of segments to keep is given by parameter "Segment persistence".
Segment persistence		Number of segments to store on the origin server. The entered value must be as follows: 4 <= Value, and Value > ("Segment count" + 1)
HTTP connection splitting	Read/Write	Some CDNs may unexpectedly break the connection after some chunks have been pushed to them. The connection is then lost. This parameter allows to avoid these breaks, forcing the connection to stop and immediately restart before an unexpected break. The value reflects the number of chunks that are pushed to the server during an HTTP connection. Default value is 32.
HTTP authentication method	Read/Write	This parameter is only displayed when "CDN Preset" is set to "None". It defines the method used to negotiate credentials - such as username or password Values: - None: No authentication is required. - Auto: basic method - AWS Signature v4 - Akamaï edge grid v1 - Digest

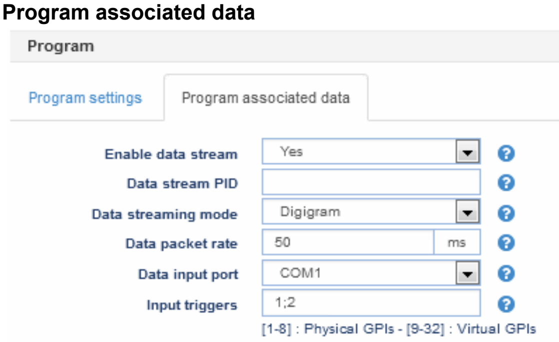
Click on "Save" to confirm the settings. Click on "Save & New" to confirm the settings and create a new IP service with the same parameters.
Click on "Close" to discard the settings

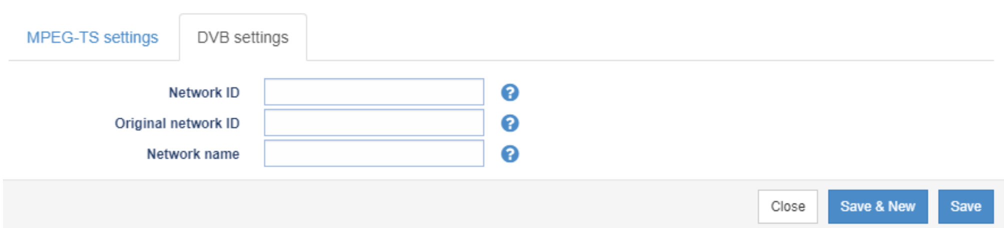
Encapsulation = MPEG-TS SPTS / MPEG-TS MPTS

Note: Since version 3.11, the MPEG-TS streams generated by IQOYA by default contain the descriptor "Maximum_Bitrate_Descriptor" in the PMT, and the flag "ES_Rate_flag" in the PES. They can be disabled by modifying the IQOYA configuration file. See appendix D.



Parameter	Type	Description
Name	Read/Write	Give a name to this TS IP service.
Synchronous AoIP	Read/Write	Set to "None" in MPEG-TS
Encapsulation	Read/Write	Select MPEG-TS SPTS if only one audio program is to be declared in the TS stream. Select MPEG-TS MPTS if several audio programs are to be declared in the TS stream.
DVB Mode	Read/Write	Set this parameter to "Yes" to include the following DVB information tables into the stream: CAT, EIT, NIT, SDT, TDT, TSDT. Note that CAT and EIT are dummy tables inserted for compatibility with DVB. Set DVB Mode to "No" if you do not need the TS to carry the DVB information tables.

Programs If MPEG-TS MPTS is selected, several programs can be inserted in the TS stream.		
Name	Read/Write	Select the Program to be streamed from the list. Only the Programs encoded in the following audio formats are supported in the MPEG-TS streams: MPEG Layer2, MPEG Layer 3, AAC-ADTS (LC, HEv1, HEv2). Programs encoded in other audio formats are not displayed. . The name of the selected program is the name of the Service in the TS stream.
Number	Read/Write	Enter the program number (1 to 65535)
PID (PMT)	Read/Write	Enter the Program Map Table PID (16 to 8190)
Stream PID	Read/Write	Enter the PID of the elementary stream (16 to 8190)
PTS announcement period	Read/Write	Enter the Program Time Stamps announcement period (from 50ms to 10000 ms)
Language	Read/Write	Enter the language descriptor, according to ISO 639-2
Provider Name	Read/Write	Only available if DVB mode is set to “Yes”. This is the service provider name which is set in the SDT table.
Program associated data 		This tab allows the insertion of data associated with the audio program. They are transported as a component of the program. They are marked as private stream (stream type 0x06), they have a specific PID to be set, and are inserted in PES packets (this allows inserting a presentation time stamp (PTS) so that the decoder keeps the synchronization between data and audio). Data can come from COM ports or UDP ports, and triggers come from physical GPI’s or virtual GPI’s (UDP ports). See Appendix B for the structure of the data packet. See also Appendix C for information about the time-stamping of private data packets.
Enable data stream	Read/Write	Select “Yes” if data has to be transported along with the audio program. Select “No” if no data transport is required.
Data stream PID	Read/Write	Enter a unique PID value for the data packets. Values can be from 32 to 8186, and from 8188 to 8190.
Data stream mode	Read/Write	This parameter allows selecting the format of the data packets. Two formats are available: “Digigram” and “Raw”. -” Digigram”: Data and triggers are inserted in the same packet. Data is preceded by the identifier "COM", and GPI triggers are preceded by the identifier "GPI". This allows for easy extraction of the appropriate data on the receiver side. -” Raw”: Data is inserted without a specific identifier. GPI triggers are not supported in this mode.
Data packet rate	Read/Write	This is the elapsed time between two data packets, in ms. Possible values are from 10ms to 5000ms.The maximum amount of data that can be inserted in a packet is 128 bytes.
Data input port	Read/Write	Select the input port to be used to receive the data to be inserted in the data packets. This port can be a COM port, or a UDP port declared from Preferences-Auxiliary data-UDP
Input triggers	Read/Write	Enter a list of GPI indexes separated by semicolons (ex: 1;3;5). The status of the listed GPI's are inserted in the data packets. This is supported only in the Digigram data streaming mode. GPI's must be configured from Preferences - Auxiliary data-GPIO

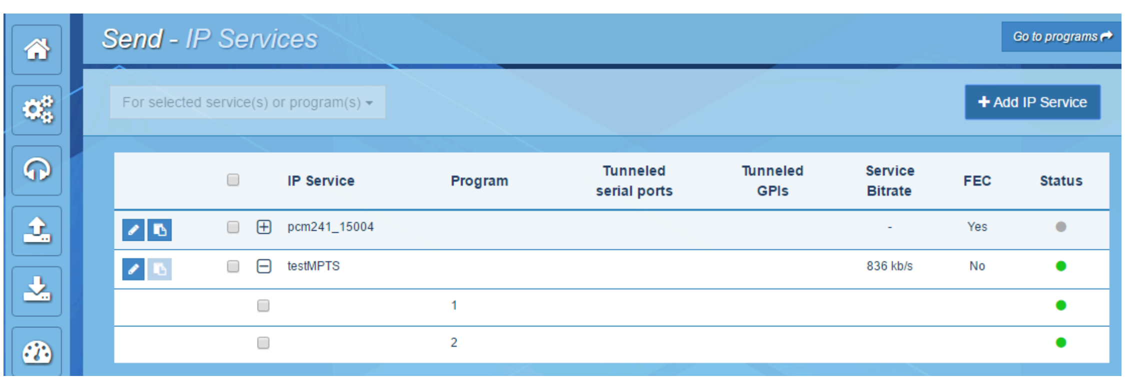
For adding another audio program in the MPTS stream, click on the icon .		
MPEG-TS Settings		
Transport Stream ID	Read/Write	Unique identifier of a TS within an original network (from 0 to 65535)
PCR ID	Read/Write	Program Clock Reference Select this option in case the PCR is sent as an elementary stream, and enter its packet ID (16 to 8190)
PSI announcement period	Read/Write	Program Specific Information Enter the announcement period (from 100 to 5000 ms)
Delay for PTS calculation	Read/Write	Enter the relative delay to be used to calculate the Presentation Time Stamp (100 to 2000ms)
Overall bit rate	Read/Write	Enter the overall bit rate of the MPEG-TS stream. When set to 0, the bit rate is set automatically.
Transport Protocol	Read/Write	Streaming protocol of the MPEG-TS stream: RTP or UDP.
Number of TS packet per IP packet	Read/Write	Sets the number of TS packets per IP packet. Default value is 7.
DVB Settings (if DVB Mode is set to "yes"). 		
Network ID	Read/Write	Network ID is used to identify the delivery system.
Original network ID	Read/Write	Network_id of the originating delivery system.
Network name:	Read/Write	Name of the network.
Audio Stream When "DVB Settings" has been selected, select again "MPEG-TS settings" to access the Audio stream parameters.		
IP address	Read/Write	Enter the destination IP address (unicast or multicast)
Port	Read/Write	Enter the destination UDP port.
Network interface/VLAN	Read/Write	Select the network interface or bond or VLAN for this stream. In case the target address is unicast, select "Any" so that the Eth interface is determined automatically according to this IP address, or select a VLAN. In case the target IP address is multicast, select the Eth interface or the VLAN.
Local source port	Read/Write	Local UDP port number of IQOYA X/LINK
DSCP	Read/Write	Select the quality of service (QoS) class of the stream.
Stop streaming on silence detection	Read/Write	This parameter is available for MPEG-TS SPTS encapsulation. Select "Yes" so that the SPTS stream is automatically stopped upon silence condition,

		and started upon signal detection. Silence/signal detection criteria can be set from the parameters of the audio input Program. When this parameter is set to “No”, the SPTS stream is continuously active.
FEC stream MPEG This section allows configuring a Pro MPEG COP#3.2 FEC for the MPEG-TS stream.		
Type	Read/Write	**No Redundancy**: No FEC is generated. **Column**: 1 dimension FEC scheme. Only FEC frames generated from columns are streamed. Number of columns can be set from 1 to 20. This FEC is ideal for correcting packet burst errors and random errors. The column FEC frames are sent to UDP port = MPEG-TS stream UDP port + 2. **Column and row**: 2 dimensions FEC scheme. Provides correction for non-consecutive lost frames, and can correct any single packet loss within a row of media packets. • 4 <= Number of Columns (L) <= 20. • 4 <= Number of rows (D) <= 20 • L x D <= 100 The column FEC frames are sent to UDP port = MPEG-TS stream UDP port + 2. The row FEC frames are sent to UDP port = MPEG-TS stream UDP port + 4.
Number of columns (L)	Read/Write	Column depth Column scheme: value from 1 to 20 Column and row scheme: 4 <= L <= 20
Number of rows (D)	Read/Write	Row depth: 4 <= Number of rows (D) <= 20

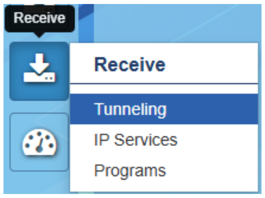
Click on the icon on the bottom right of the page to add IP destinations.

Click on “Save” to confirm the settings. Click on “Save & New” to confirm the settings and create a new IP service with the same parameters.
Click on “Close” to discard the settings.

Log traces associated to an IP service
To view the log traces associated with an IP service, click on its status LED on the right, as shown on the screen capture below.

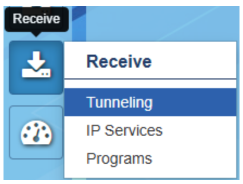


8.6 “Receive”



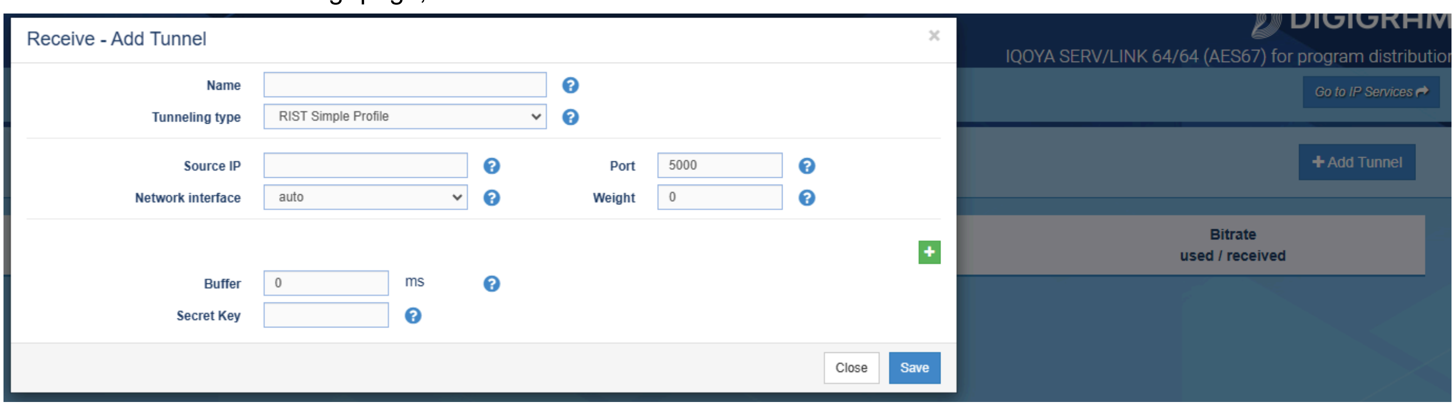
This category allows defining the IP services to be received by IQOYA, and the audio programs to be played to the outputs of IQOYA; three decoding priorities can be defined per audio program.

8.6.1 Receive –> IP Tunnel



This page allows declaring the RIST tunnels used to transport the IP stream(s) to be received. The RIST tunneling feature requires that the “RIST protocol” firmware licence is applied to the SERV/LINK.
A RIST tunnel supports RTP, UDP streams with or without MPEG-TS encapsulation.
Note that MPEG-TS decoding is allowed when the firmware licence “MPEG-TS decoding” is applied to the SERV/LINK.

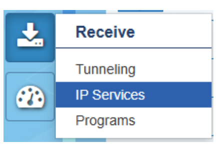
From the “Receive-Tunneling” page, click on “Add Tunnel” to declare a tunnel.



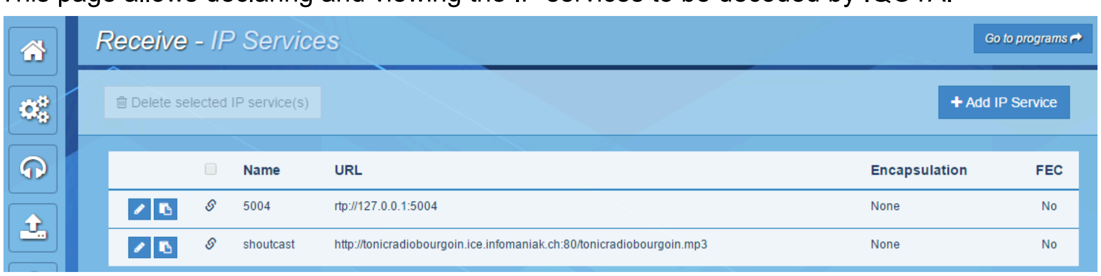
Parameter	Type	Description
Name	Read/Write	Logical name for the tunnel.
Tunneling type	Read/Write	RIST profile: Simple, Main, Advanced
Source IP	Read/Write	Enter 127.0.0.1 or left blank for a point to point tunnel (unicast). Enter the multicast IP address otherwise.
Port	Read/Write	UDP port for the reception of the tunnel. It must match the destination port configured on the encoder.

Network interface	Read/Write	Select “Auto” in unicast. Select the Eth port in multicast.
Weight	Read/Write	Sets the relative share (load balancing) for multipath connections. Exemple: in a setup with two declared paths (two destination “IP@:UDP port”, if a paths are given weights of 5 and 10 respectively, the former would receive 1/3 of packets sent (5 / 15), and the latter would receive 2/3 (10/15). A weight of 0 means that all packets flow through the path.
If the RIST tunnel is multipath, click on the icon + to declare a new reception IP address/port.		
Buffers	Read/Write	Sets the buffer size in milliseconds. The buffer size will work best at four to seven times the ping time. This allows time for requests for the retransmission of a lost or corrupted packet, and the subsequent retransmission of its replacement.
Secret key	Read/Write	Sets the specified passphrase for Main or Advanced profile encryption. This must match the secret key set on the encoder.

8.6.2 Receive –> IP services



This page allows declaring and viewing the IP services to be decoded by IQOYA.



In case some IP Services are already created, they are listed in the IP Services page, with their characteristics: : name, URL, encapsulation (MPEG-TS or not), FEC.

To edit an existing IP service, click on the icon on the right end of its line.

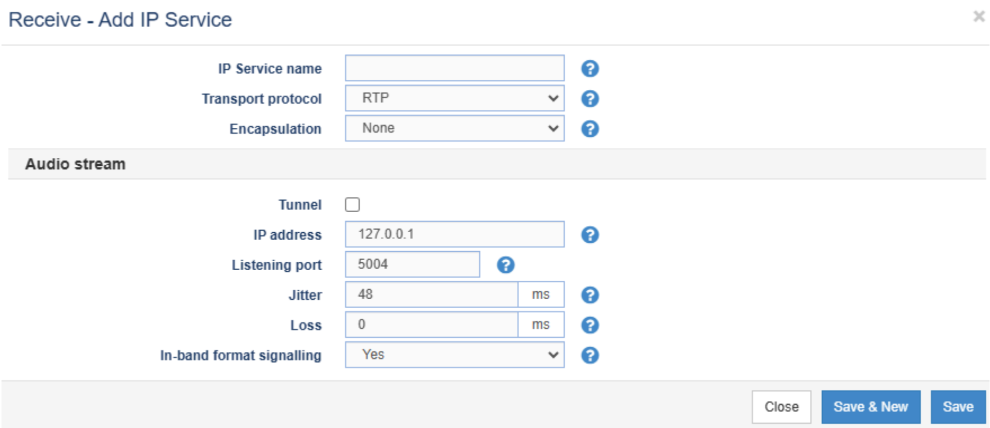
To remove an IP service, click on the icon on the left end of its line, and select Delete selected IP service(s) .

To delete all the IP services, click on the icon on the left of “Name”, and select Delete selected IP service(s) .

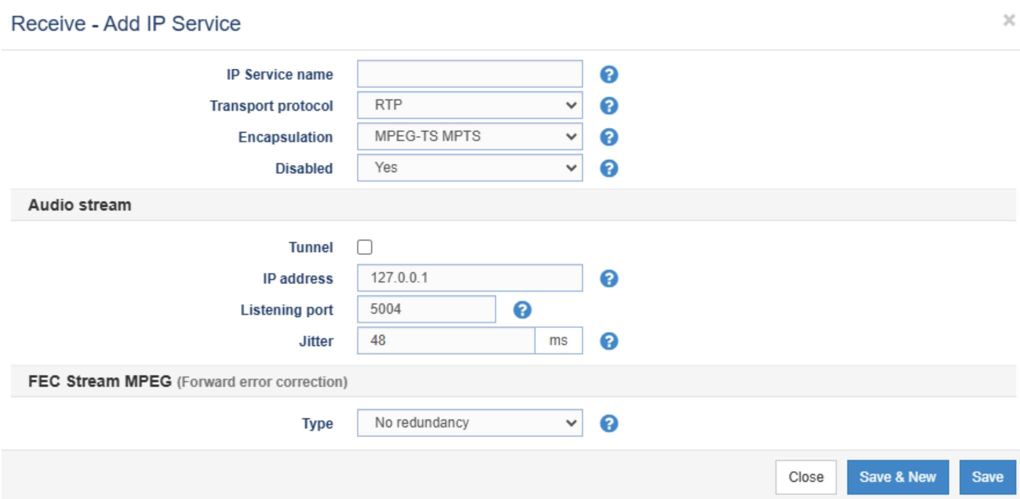
To declare a new IP service, click on the button + Add IP Service .

An IP service can also be created by duplicating an existing one. Click on the icon on the left on the IP service to be duplicated.

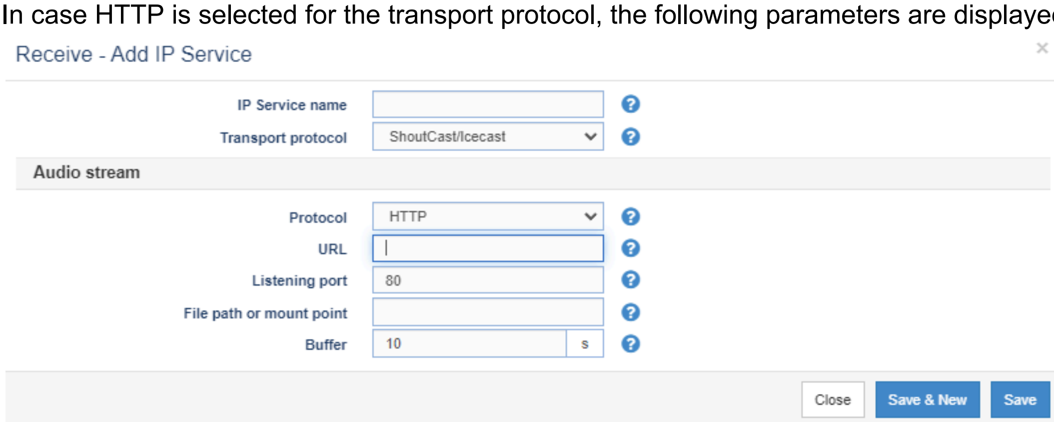
When declaring or editing an IP service the following page is displayed.



In case MPEG-TS encapsulation is selected for a UDP or RTP stream, the following parameters are displayed.



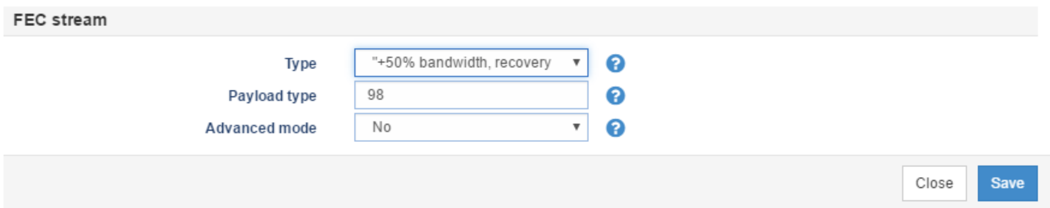
In case HTTP is selected for the transport protocol, the following parameters are displayed.



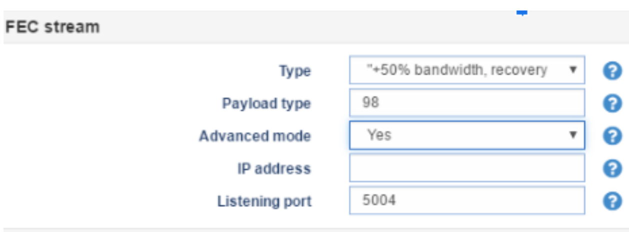
Parameter	Type	Description
IP service name	Read/Write	Name given to this IP service. This is the name that can be selected in the source of a decoding priority of an output program.
Transport protocol	Read/Write	Values: RTP, UDP, Icecast/Shoutcast
Encapsulation	Read/Write	Only displayed if the selected transport protocol is different from Icecast/Shoutcast. Values are: None, MPEG-TS SPTS or MPEG-TS MPTS.
Audio stream		
Tunnel	Read/Write	Select this option if the stream is to be received through a RIST tunnel.
IP address (for RTP and UDP, and if “Tunnel” is not checked)	Read/Write	Displayed if “Tunnel” is not selected. In unicast, set this parameter to 127.0.0.1, otherwise enter the multicast IP address to listen to.
Listening port (for RTP and UDP, and if “Tunnel” is not checked)	Read/Write	Displayed if “Tunnel” is not selected. For RTP and UDP protocols, value of the UDP port to listen to. For Icecast/Shoutcast, value of the TCP port to listen to.
Input tunnel (for RTP and UDP, and if “Tunnel” is checked)	Read/Write	Select the RIST tunnel from the list of declared tunnels.
Stream virtual port (for RTP and UDP, and if “Tunnel” is checked)	Read/Write	Select the virtual port of the appropriate stream (this virtual port is set on the encoder of the stream).
Network interface	Read/Write	Displayed if the IP address is multicast. Select the Eth interface, bond or VLAN from the list.
Jitter (for RTP and UDP)	Read/Write	Enter the input buffering size to compensate for the jitter of the network. This value, expressed in ms, must be at least equal to the measured jitter. In case there is FEC, it is necessary to consider the measured jitter for “primary and FEC stream”.
Loss (for RTP)	Read/Write	Defines the duration of consecutive lost packets until which IQOYA replaces lost frames by silence, without flushing the buffer of jitter. If the absence of received consecutive packets exceeds this duration, the jitter buffer is then flushed, and filled again with received packets; this allows resynchronization on the incoming IP audio stream, but this may generate a silence longer than the packet's loss. To avoid long audio silences when only a few consecutive packets are lost (especially for high jitter values), it is recommended to set the Loss value to approximately 3/4 of the jitter buffer.
Synchro clock (for RTP, without encapsulation)	Read/Write	Select NTP in case the audio synchronization on NTP is used for decoding this stream (optional feature).
In-band format signalling (for RTP without encapsulation)	Read/Write	Set this parameter to “Yes” if it is also set to “Yes” on the IQOYA encoder. Set this parameter to “No” if it is not configured on the IQOYA encoder, or if the encoder is another brand.
payload type (for RTP)	Read/Write	Only displayed if “In-band format signalling” is set to “No”. Enter the payload value of the audio stream (same payload value as configured on the stream encoder).
Protocol	Read/Write	Only displayed when the Transport Protocol is set to Icecast/Shoutcast.

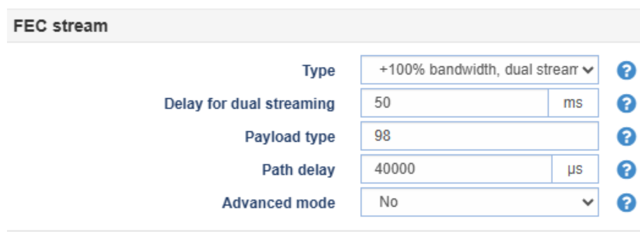
(for HTTP)		Possible values: HTTP, HTTPS.
URL (for HTTP)	Read/Write	Only displayed when Transport Protocol is set to Icecast/Shoutcast URL of the Icecast/Shoutcast server. Example: streamer.mysite.com.
Listening port (for HTTP)	Read/Write	Listening port of the Icecast/Shoutcast server (usually port 80). For a URL like: http://streamer.digigram.com:6200/servlink1, listening port is 6200.
Buffer (for HTTP)	Read/Write	Buffer value in seconds necessary to decode correctly the HTTP stream. This value may depend on the HTTP server. In case the decoding is producing audio breaks, this value has to be increased.

In case the audio format of the IP stream is not signalled in-band, it is necessary to declare if the received IP service includes an FEC.
Select the appropriate FEC for the Type field as shown below. The payload type is set automatically.

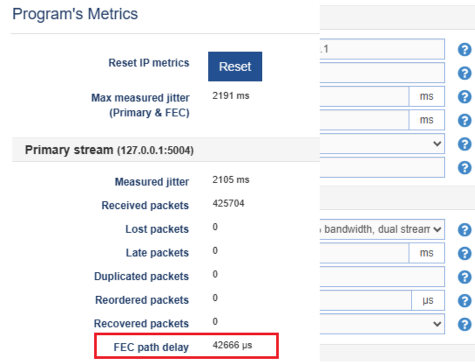


In case FEC is not sent on the default UDP port and IP address, select “Yes” in the “ Advanced mode” field, to be able to enter the IP address and UDP port.





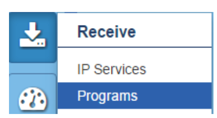
FEC stream parameters	Type	Description
Type	Read/Write	Select the FEC that is configured on the encoder of the received IP stream.
Delay for dual streaming	Read/Write	Only displayed when the FEC Type is “+100% bandwidth, dual stream”. Enter the delay between the main stream and the redundant stream that has been set on the encoder side. From0 to 3000 seconds.
Payload type	Read/Write	Enter the same FEC payload type that is configured on the encoder of the received IP stream.
Path delay	Read/Write	Only displayed when the FEC Type is “+100% bandwidth, dual stream”. The value to be put here is the “Path delay” value that is measured and displayed in the

		“Metrics” page. 
Advanced mode	Read/Write	Select “Yes” if the FEC destination IP address is not the same as the IP stream destination IP address, or of it is to be received on a UDP port different from “IP stream UDP port +2”
IP address	Read/Write	Displayed if Advanced mode is set to “Yes”. In unicast, set the IP address to 127.0.0.1. In multicast, enter the multicast IP address.
Listening port	Read/Write	Displayed if Advanced mode is set to “Yes”. Enter the UDP port for receiving the FEC.

Click on “Save” to confirm the settings.

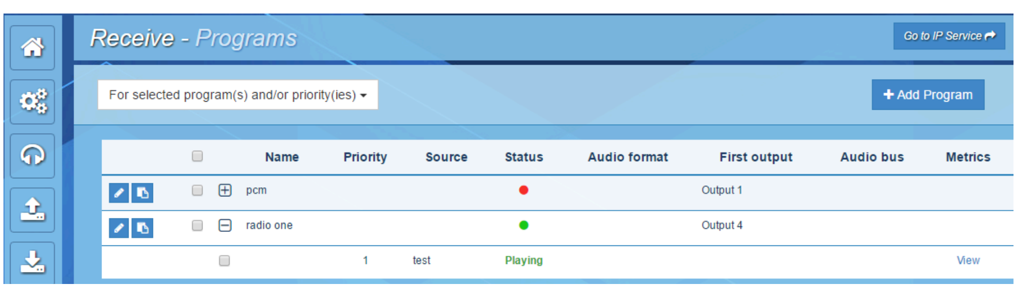
8.6.3 Receive –> Programs

Output programs are composed of a list of audio sources organised in priorities. Up to 3 decoding priorities can be defined. The highest priority is priority 1. If the audio source of priority 1 is lost, IQOYA switches to priority 2 if the corresponding audio source is available, or to priority 3 if the corresponding audio source is available. If no declared audio source is available, the program output is silent.

Output programs configuration is accessible either from the left column 

, or directly from the icon “Go to IP Service” on the top right of the IP Services 

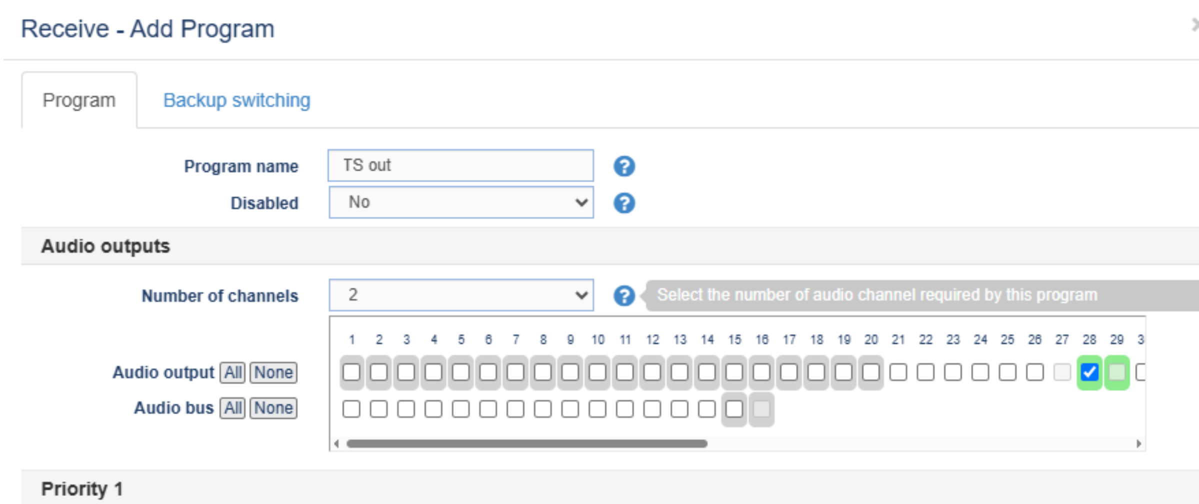
.



The “Programs” page displays the declared output programs.

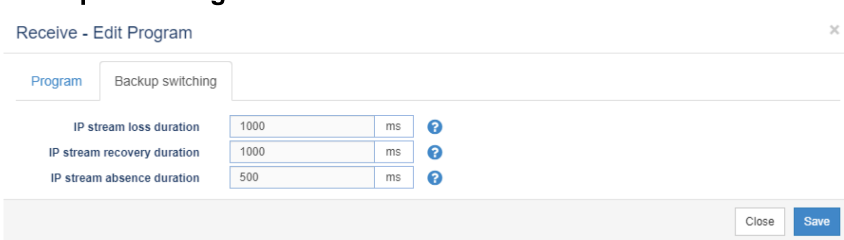
To declare a new output program, click on 

, or create it from an existing one by selecting the icon  on its left.



Program Parameter	Type	Description
Program		
Program name	Read/Write	Name given to this output Program.
Disabled	Read/Write	Set this parameter to “Yes” if you want to disable the program. This means that IQOYA does not process it. Set this parameter to “No” so that IQOYA processes this program and decodes audio.
Audio outputs		
Number of channels	Read/Write	Select the numof channels of the audio program to be decoded: 1 (mono), 2 (stéréo)
First audio bus	Read/Write	Select the first audio bus channel associated with this program. Audio bus is useful for IP stream transcoding.
Number of outputs	Read/Write	Set this parameter to 1 for mono, 2 for stereo, 6 for 5.1, 8 for 7.1.

Backup switching criteria



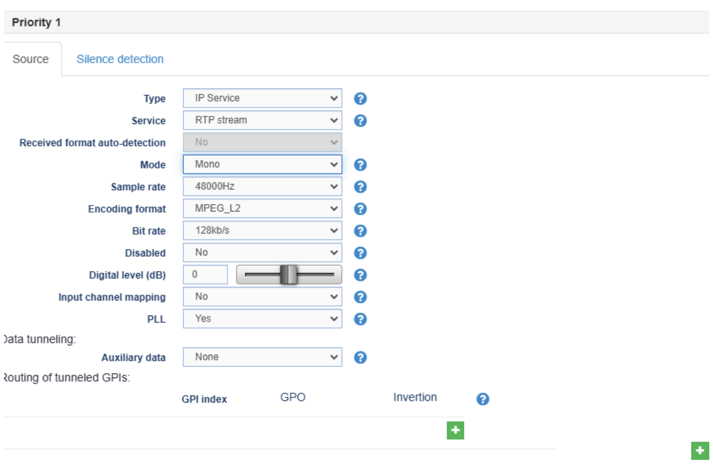
Program Parameter	Description
IP stream loss duration	In case the codec is configured to decode an IP audio stream and at least one backup is defined, you can configure the backup switching criteria. IP stream loss duration , expressed in ms, is the duration of absence of the stream. When this condition is encountered on priority 1 or priority 2, IQOYA automatically switches to the lower priority. The minimum value for this duration is the jitter value set from the Receive page.

IP stream recovery duration	This value, expressed in ms, is the duration of presence of the stream after it has been lost. When this condition is encountered, IQOYA automatically switches to the higher priority where the stream is recovered. These two criteria apply to the main received IP stream as well as to the backup IP stream.
IP stream absence duration	During the stream recovery process, if a received packet is followed by a packet's absence duration larger than this "IP stream absence duration", the stream is considered as absent. This value (in ms) should be lower than half the "IP stream recovery duration". If the value is set to 0, this parameter is ignored..

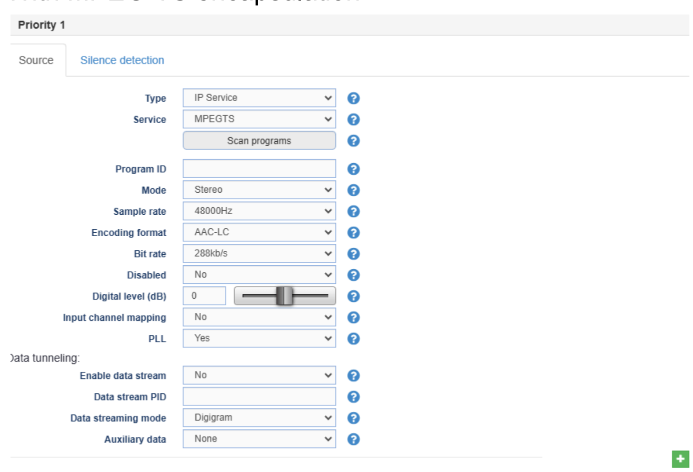
The following parameters define the audio source associated with priority 1. It is possible to declare two additional priorities by clicking on the button on the right below the decoding priority.
The parameters listed for a decoding priority depend on the selected source "Type": IP service, File, Playlist, Audio input.

Source Type = IP service, and IP service is RTP

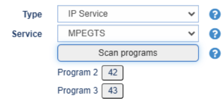
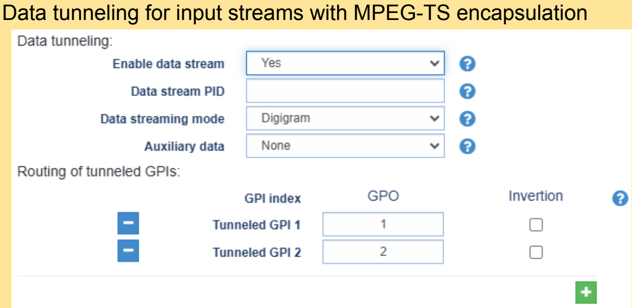
Without encapsulation

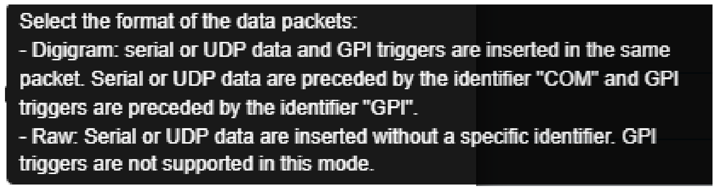


With MPEG-TS encapsulation

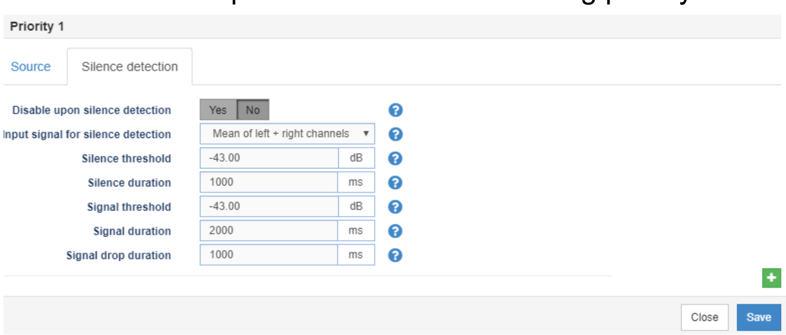


Priority Source Parameter	Type	Description
Type	Read/Write	Select the audio source for this priority. **IP Service**: audio will be extracted from a declared IP service. File: audio source is a local file Playlist: audio source is a local "m3u" playlist Audio input: audio source is an audio input.
Service	Read/Write	Select the IP service from the list of declared IP services. (IP services must have been declared first from the IP Services page).
For MPEG-TS only: Scan programs	Click on this button so that the receiver detects the program IDs in the TS stream. The TS stream must be started on the encoder, otherwise a message "MPEG MPTS IP stream is not started" or "Could not find input stream" for an SPTS stream. The detected program IDs are then displayed	

Program ID	 Enter the program ID to be decoded in the field “Program ID”.	
Received format auto-detection	Read	This parameter is set automatically according to the selected IP service. If the IP service has been declared with in-band format signalling, auto-detection is set to “Yes”.
Disabled	Read/Write	Set this parameter to “Yes” to disable this decoding priority. Disabling a defined priority is useful when some servicing is in progress on it (network servicing, servicing on the source of the IP stream). The priority can then be enabled when servicing operations are finished.
Digital Level (dB)	Read/Write	Digital gain applied to the audio samples on this priority.
Input channel mapping	Read/Write	Select how the channels of the selected source are processed: • No: each input channel is assigned to an output channel. • Mix: the input channels are mixed to a single output channel. An attenuation of -6 dB is applied to each channel before they are mixed. The gain/attenuation set through “Digital level” comes in addition to this attenuation. • First channel only: only the first channel is processed. • Second channel only: only the second channel is processed.
PLL	Read/Write	Set to Yes in most of the cases. It allows synchronization of the incoming IP audio to the sampling clock, thus guaranteeing a constant delay. It has to be set to No when samples must ne be modified between the encoder and the decoder (this requires that the encoder and the decoder have clock sources having the exact same sampling frequency)
Data tunneling for input streams without MPEG-TS encapsulation		
Auxiliary data	Read/Write	Select “None” if there is no auxiliary data to extract. Otherwise, select the serial port or UDP socket to be used to output the serial data extracted from the decoded IP stream.
Routing of tunneled status data: Status data indexes	Read/Write	 For each tunnelled GPI, enter the GPO number (starting from 1) that will reflect the tunneled GPI status. Click on the “+” icon to add a new GPO.
Routing of tunneled status data: GPO inversion mask	Read/Write	Check the box under a GPO so that it reflects the inverted status of the tunneled GPI.
Data tunneling for input streams with MPEG-TS encapsulation 		

Enable data stream	Read/Write	Select “Yes” if program associated data has to be extracted from the incoming MPEG-TS stream.
Data stream PID	Read/Write	Enter the packet ID of the Data stream. Authorised values:from 32 to 8186, and from 8188 to 8190.
Data stream mode	Read/Write	
Auxiliary data	Read/Write	Select “None” if there is no auxiliary data to extract. Otherwise, select the serial port or UDP socket to be used to output the serial data extracted from the decoded IP stream.
Routing of tunneled status data: Status data indexes	Read/Write	 For each tunnelled GPI, enter the GPO number that will reflect its status.. Click on the “+” icon to add a new GPO.

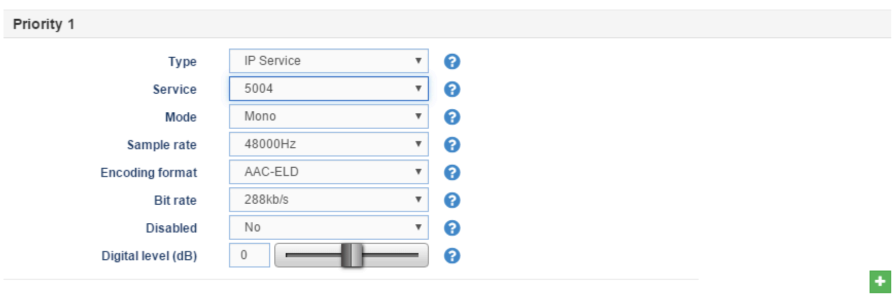
Silence detection parameters for the decoding priority



Priority Source Parameter	Type	Description
Disable upon silence detection	Read/Write	IQOYA can also automatically disable the decoding priority in case of silence detection in the audio source. The priority can then be enabled again via the WEB site, or via SNMP.
Input signal for silence detection	Read/Write	The parameter “Input signal for silence detection” allows defining on which source signal the silence detection is applied. Possible choices are: **- Mean of left + right channels**: compares the mean value of a left and right sample to the threshold. In case the calculated values are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- Left channel only**: compares the left channel samples to the silence threshold. In case the sample values are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- Right channel only**: compares the right channel samples to the silence threshold. In case the sample values are always lower to the silence threshold during the defined silence duration, silence condition is reached.

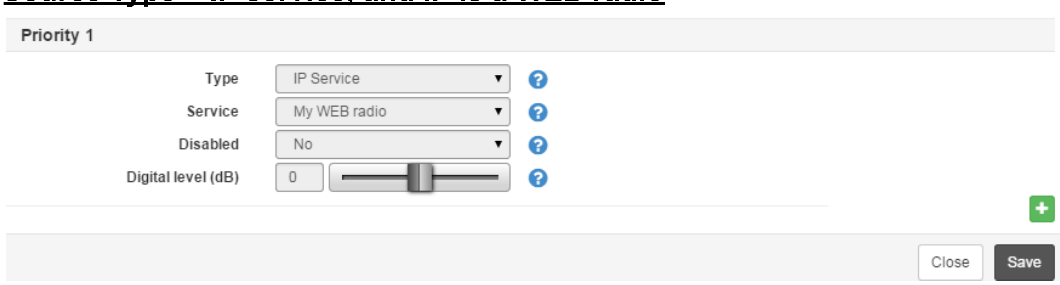
		- Left and right channels: compares both the left and right channel samples to the silence threshold. In case the sample values on both channels are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- At least one channel:** compares both the left and right channel samples to the silence threshold. In case the sample values on at least one of the two channels are always lower to the silence threshold during the defined silence duration, silence condition is reached.
Silence threshold & *Silence duration*	Read/Write	Silent audio is defined through these two parameters, expressed in dBfs. When the audio level is below the threshold value during at least the defined duration, the alarm “Analog audio in silent” or “Digital audio in silent” is set (if it is enabled from the “Alarms setup” menu).
Signal threshold *Signal duration* *Signal drop duration*	Read/Write	Audio signal is defined through the three parameters. Audio signal is considered as recovered if all the following conditions are true: ● Audio level exceeds the Signal threshold (dBfs) within the “Signal duration” analysis window (ms). ● Audio level does not stay below the Signal threshold during the “Signal drop duration”, within the “Signal duration” analysis window. Note the following rule: Signal drop duration <= (Signal duration / 2). Once the signal is recovered, the alarm “Analog audio in silent” or “Digital audio in silent” is reset (if it is enabled from the “Alarms setup” menu).

Source Type = IP service, and IP service is UDP



In UDP mode, the audio format has to be declared.

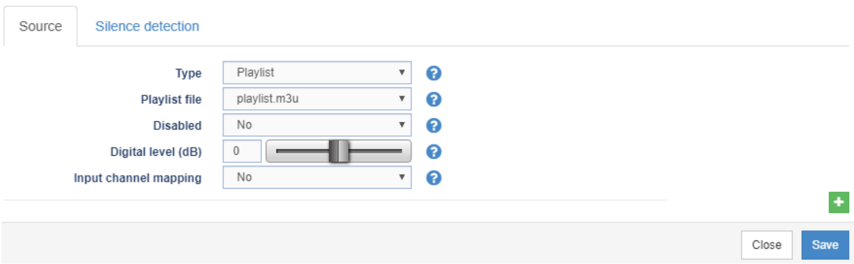
Source Type = IP service, and IP is a WEB radio



Parameter	Type	Description

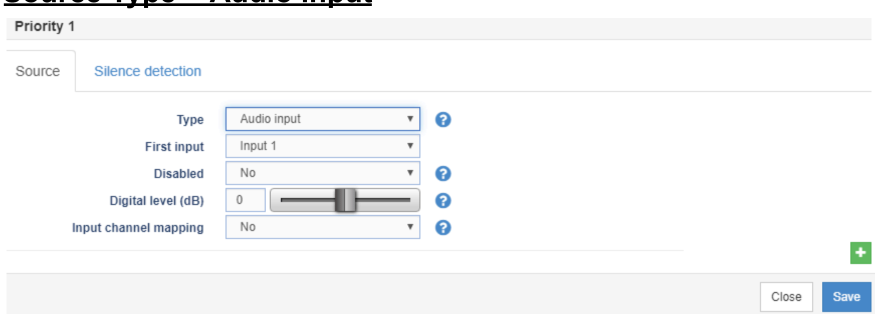
Type	Read/Write	IP Service
Service	Read/Write	Select the IP service from the list of declared IP services. (IP services must have been declared first from the IP Services page).
Disabled	Read/Write	Set this parameter to “Yes” to disable this decoding priority.
Digital Level (dB)	Read/Write	Digital gain applied to the audio samples on this priority.

Source Type = File or Playlist



Parameter	**Type**	**Description**
Type	Read/Write	File or Playlist. These files are stored locally on the internal DOM (disk on module).
Audio File	Read/Write	Select the audio file or playlist from the list.
Disabled	Read/Write	Set this parameter to “Yes” to disable this decoding priority.
Digital Level (dB)	Read/Write	Digital gain applied to the audio samples on this priority.
Input channel mapping	Read/Write	Select how the channels of the selected source are processed: ● No: each input channel is assigned to an output channel. ● Mix: the input channels are mixed to a single output channel. An attenuation of -6 dB is applied to each channel before they are mixed. The gain/attenuation set through “Digital level” comes in addition to this attenuation. ● First channel only: only the first channel is processed. ● Second channel only: only the second channel is processed.

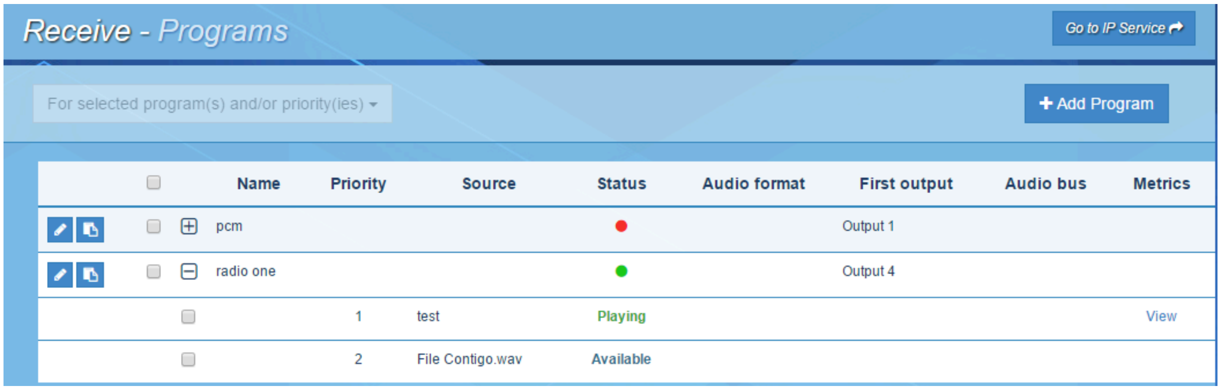
Source Type = Audio input



Parameters are the same as above, except the audio input that must be selected instead of a sound file or playlist.

Once output programs have been defined, they are listed in the “Programs” page.

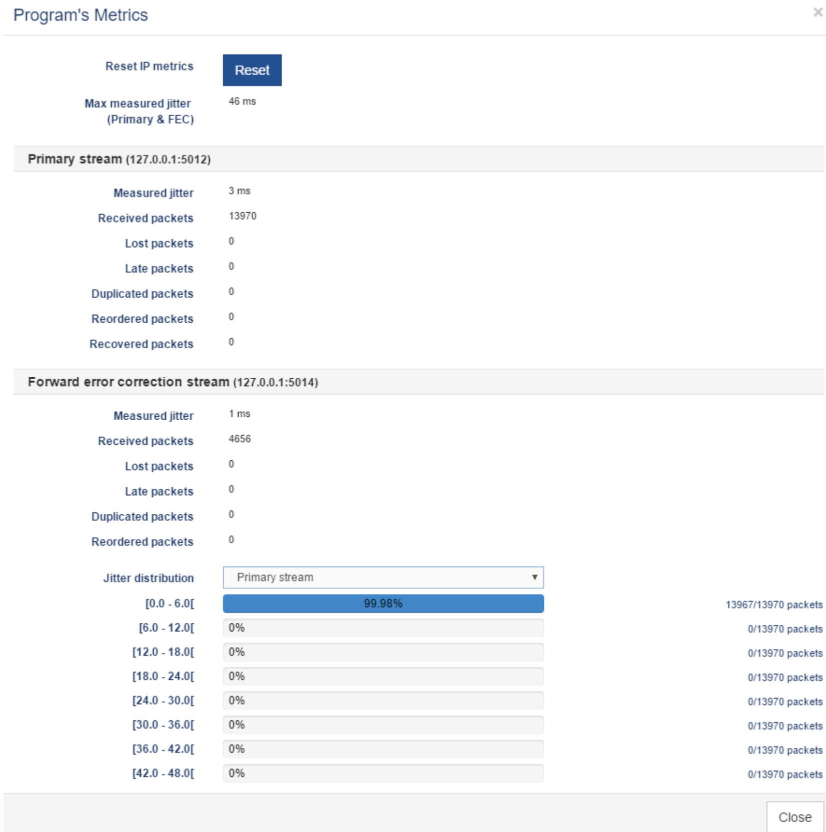
To view the content of a program, click on ⊞ on the left of its name.



The decoding priorities of the program are displayed as well as some associated information:

- Name: program name
- Priority: 1, 2, or 3 priorities are displayed, depending on what has been defined.
- Source: displays the name of the audio source defined for the priority
- Program status: Displays the status of the program, and the status of each priority.
  Possible program statuses are:
  - Green LED => the first enabled priority is decoded.
  - Orange LED => the source of a priority is missing
  - Red LED => all the defined sources are missing.
  Possible priority statuses are:
  - Playing: IQOYA is playing this priority
  - Missing: the audio source of the priority is missing
  - Disable: the decoding priority is disabled
  - Available: means that the source of this priority is detected, but a higher priority source is being played.
- Audio format: display the audio format of the decoded IP service.
- First output: displays the first output used for the program
- Serial: displays the serial port that outputs tunneled serial data.
- GPOs: displays the GPO that reflect the tunneled statuses

- Metrics/view: Click on view to display the metrics of the IP service.



These metrics are important characteristics of the network path. In case an FEC is used, metrics are available for both the primary stream and the FEC stream.
Note that the measured jitter (Primary + FEC streams)

Variable	**Meaning**
Max Measured jitter	Displayed only if an FEC stream is received. Defines the minimum jitter to be configured in Receive−>IP Services (it includes the primary stream and the FEC stream). On unmanaged networks, we recommend configuring a higher value as the jitter may evolve and reach higher values.
Measured jitter	Jitter measured for the considered stream (primary or FEC). If no FEC stream is received, this value defines the minimum jitter to be configured in Receive−>IP Services (it includes the primary stream and the FEC stream). On unmanaged networks, we recommend configuring a higher value as the jitter may evolve and reach higher values.
Receive packets	Number of IP frames received for the considered stream (primary or FEC). If this value does not increase regularly, the IP stream is not received.
Lost packets	Number of IP frames that have not been received.
Late packets	Number of IP frames that have been received late.

Duplicated packets	Number of IP frames that are received more than once. IQOYA automatically removes duplicated frames.
Reordered packets	Number of IP frames that have been reordered after being received disordered.
Recovered packets	Number of IP frames that are recovered thanks to the FEC. If “Lost packets - Recovered packets” equals 0, the FEC is adapted to the network path. If “Lost packets > Recovered packets”, the selected FEC does not allow to recover all the lost packets. It is then necessary to select another FEC. Make sure that the jitter value set in Receive–> IP services is higher than the max measured jitter.

8.6.4 Managing sound files and playlists via FTP

Available in “Program Distribution” mode of use only.
Local sound files and playlists can be uploaded and removed via FTP.
Connect to IQOYA SERV/LINK via an FTP software application. Login is as follows:

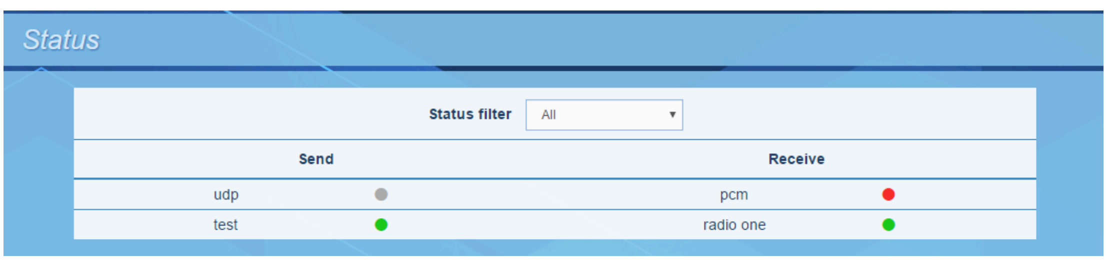
- username: ftp
- password: iqoya

Playlists (.m3u) and sound files must be stored in the folder DEVICE_STORAGE.

8.7 Send / Receive status monitoring

The status page displays a synthesis of the statuses of sent IP Services and output programs, and gives access to all the alarms of each IP service and output program.

This page is accessible by clicking on the icon .



All sent IP services are listed on the left under “Send”.
All output programs are listed on the right under “Receive”.

The parameter “Status filter” allows filtering on the type of alarms to take into account for the display. Possible values are:

- All: all alarms are taken into account.
  Green LED means no alarm is ON.
  Orange LED means there are warnings ON.
  Red LED means there are alarms and the stream is stopped.
- Warnings: only the warnings are taken into account. They concern the receivers.
- Failures: only failures are taken into account; this is typically when there is a streaming failure (no stream received, no stream sent).

A list of all the alarms can be displayed by clicking on the IP Service name (Send), and on the output program name (Receive).

9 CONFIGURE IQOYA SERV/LINK FOR REMOTES

IQOYA SERV/LINK features two modes of use :

- The ‘Program Distribution’ mode of use: In this mode, the available functions and the graphical user interfaces are suitable for the needs of fixed installations like STL and SSL links, delivery of WEB radios to CDNs, program delivery to DVB/cable operators, IP audio transcoding, etc ...
- The ‘Remote Broadcasting’ mode of use: In this mode, the available functions and the graphical user interfaces are suitable for the needs of temporary audio over IP connections like live remote broadcasts, intercom, etc ...

This section is dedicated to the RemoteBroadcasting mode.

At first power up, the user is prompted to choose the mode of use from the configuration web interface. Later it is also possible to switch from one mode to another from the configuration web pages.

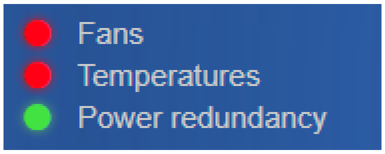
9.1 WEB pages organization

The WEB pages are organized in categories which are always accessible from the left side of the WEB pages.

Icon	Category	Description
	Operations (Home page)	Displays the mosaic of call pages of the different active codec instances.
	Connections	Connection parameters of the unit and of the codec instances: - at network level - ethernet and IP - at audio and SIP level - at user level - contacts and call profiles.
	Advanced Settings	System parameters (System properties, clock settings, audio advanced settings, alarm settings, logs, configuration up- and download, firmware and license update, password change, shutdown/restart, mode of use switch). Secondary network service settings (NTP, FTP, SSH). Auxiliary data settings (from/to serial ports, GPIO or UDP sockets).
	Audio I/Os	Audio input and audio output settings: name, type selection, audio level adjustment, vu-meters
	Help	About IQOYA SERV/LINK and this user manual.

All the web pages have the same header showing the following information:

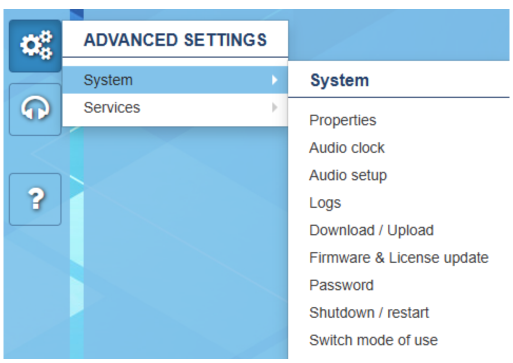
- On the left, the status of the fans, the status of the temperatures and the status of the redundant power supply unit:



The led is green when the status is ok,
The led blinks red when the status is not good.
Note that the redundant power supplies are hot swappable.

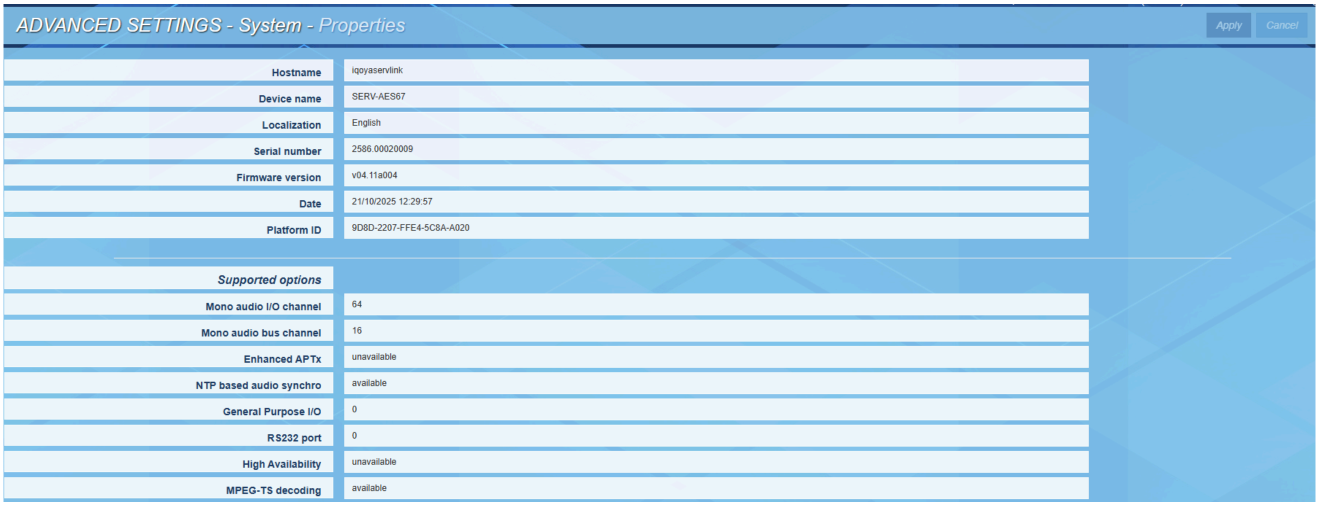
- On the right, the device model and the current mode of use:

9.1.1 "Advanced settings"

	Click on  to display all the available menus. Move the mouse pointer above the menus to display the submenus. Click on a sub-menu to display the corresponding page.

9.1.1.1 Advanced settings –> System

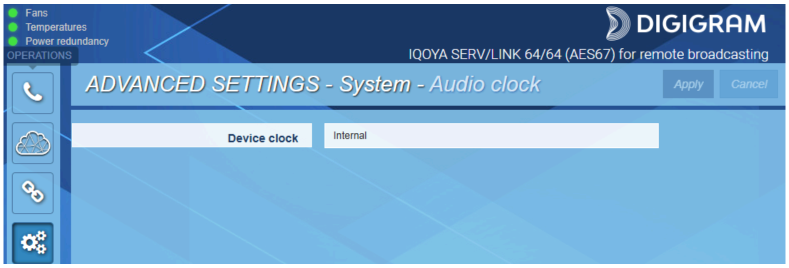
7.1.2.1 Advanced settings –> System –> Properties



Parameter	Read/Write	Meaning
Hostname	R / W	Logical name given to the device on the network.
Device Name	R / W	Name given to the equipment
Localization	R / W	Language
Serial number	R	Serial number of the unit. This number is set in the factory and cannot be changed.
Firmware version	R	Version of the firmware running on the unit. The firmware can be updated.
Date	R / W	Date and time of the unit.
Platform ID	R	Identifier of the unit. This number is required for applying firmware options.
Mono audio I/O channels	R	Number of mono audio inputs and outputs allowed by the license.
Mono audio bus channels	R	Number of mono channels allocated for the buses and allowed by the license. Buses are used for transcoding.

9.1.1.2 Advanced settings –> System –> Audio Clock

This page allows defining the SERV/LINK sampling clock source .



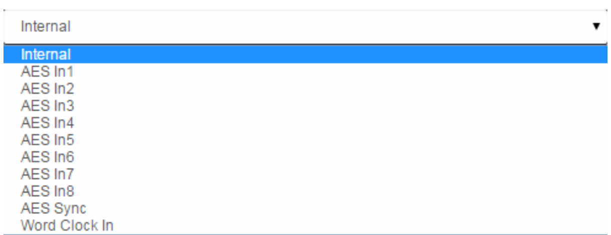
For a SERV/LINK with AES/EBU I/Os, or both analog and AES/EBU I/Os, clock source can be:

- internal: on-board clock
- extracted from an AES/EBU input (AES In x - where x is the number of the AES/EBU input)
- AES Sync: AES11 synchro input
- Word clock input

For a SERV/LINK with MADI I/Os clock source can be:

- internal: on-board clock
- MADI
- Word clock input

Click on the “Device clock” field to select the clock source.



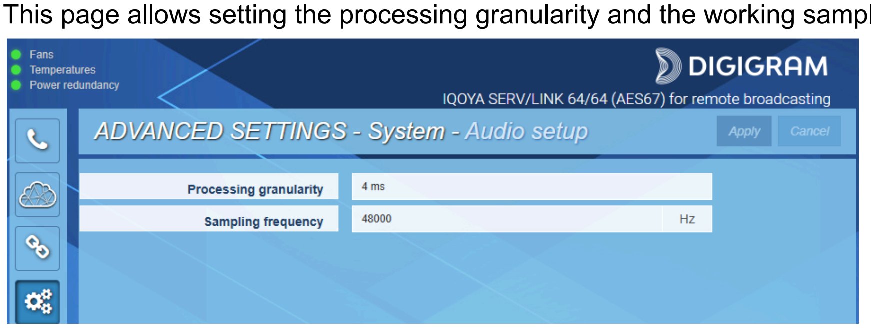
For a SERV/LINK with AES67, or “AES67 & MADI” I/O’s, or Dante I/O’s, the clock source is “Internal”.
The real clock source can be selected through the WEB pages accessible from:

- the AES67 network interfaces. See section “WEB pages for the configuration of the AES67 parameters”.
- The Dante controller application. See section “WEB pages for the configuration of the Dante parameters”.

Click on “Apply” to confirm your choice.

7.1.1.3 Advanced settings–> System –> Audio setup

This page allows setting the processing granularity and the working sampling frequency value IQOYA SERV/LINK



Click on a parameter field to be able to change the values.

Parameter	Description
Processing granularity	This is the smallest amount of data processed at a time by IQOYA. The lower the processing granularity, the lower the latency. Possible values are 1ms, 2ms, 3 ms, 4 ms. However, a value of 1ms may lead to audio underruns, depending on the features enabled on IQOYA. In case this happens, it is necessary to increase the processing granularity value. Not: the payload size of an IP frame is adjustable via parameter Payload size, from the Send page (see paragraph Encoder parameters configuration).
Sampling frequency	It defines the working sampling frequency of IQOYA. Note that received and generated IP streams can carry audio at different sampling frequencies (in which case a high quality frequency change is applied). When sampling frequency is set to 48 kHz, IP streams can be at 48 kHz, 32 kHz, 16 kHz (G722), and 8 kHz (G711). Note that 44.1 kHz is allowed for a HTTP stream. When sampling frequency is set to 44.1 kHz, IP streams must be at 44.1 kHz.

Note:

- From firmware 2.13, parameters for silence detection are configurable for each input program, and for each decoding priority of an output program.
- From firmware 2.13, backup switching criteria can be configured for each output program (Receive/Program). These criteria were global with previous firmware versions.

Click on “Apply” to confirm your changes.

9.1.1.3 Advanced settings–> System –> Logs

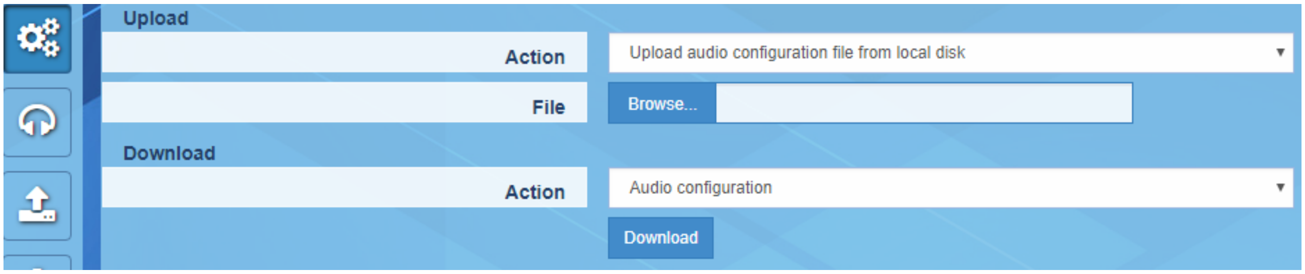
This page allows viewing and downloading the log file of IQOYA SERV/LINK. This log file gives information about the internal behaviour of IQOYA, and is useful for advanced diagnostics. Traces of enabled alarms are written into this log file (alarm ON, alarm OFF). This log file is stored internally and is persistent to a power cycle, a restart or reboot.

Event Type: allows selecting the category of traces to be displayed: Infos, Warnings, Errors, Errors & Warnings.
Codec: allows selecting one of the codecs so that only log traces related to this codec are displayed. The number of the codec can be seen from the Send/IP Services page, and from the Receive/ Programs page.
Auto refresh: Yes/No. When set to Yes, the log file display is refreshed every few seconds
Date & Time: clicking on this icon allows you to sort out the traces by date and time, starting by most recent traces or starting by oldest traces.
Reset logs: resets all the traces.

Download logs: allows for remotely downloading the traces log file.

9.1.1.4 Advanced settings–> System –> Download / Upload

This page allows downloading the IQOYA configuration to a remote PC, or uploading a configuration from a remote PC to IQOYA.



To save the current configuration of IQOYA to a remote PC, click on " Download".
To apply a configuration to IQOYA, click on " Browse" to select the configuration file, and click on "Apply".

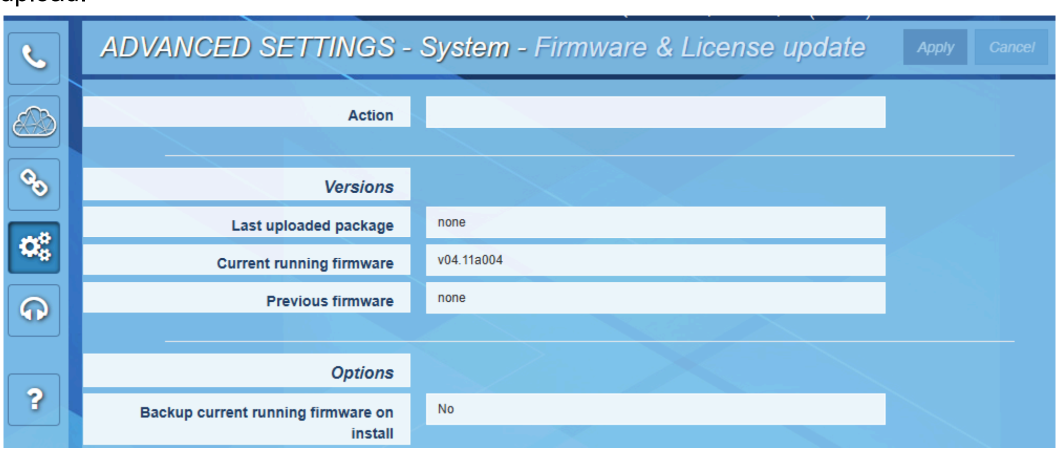
The configuration that can be uploaded/downloaded can be:

- The audio configuration only (includes the programs and IP services)
- The full codec configuration

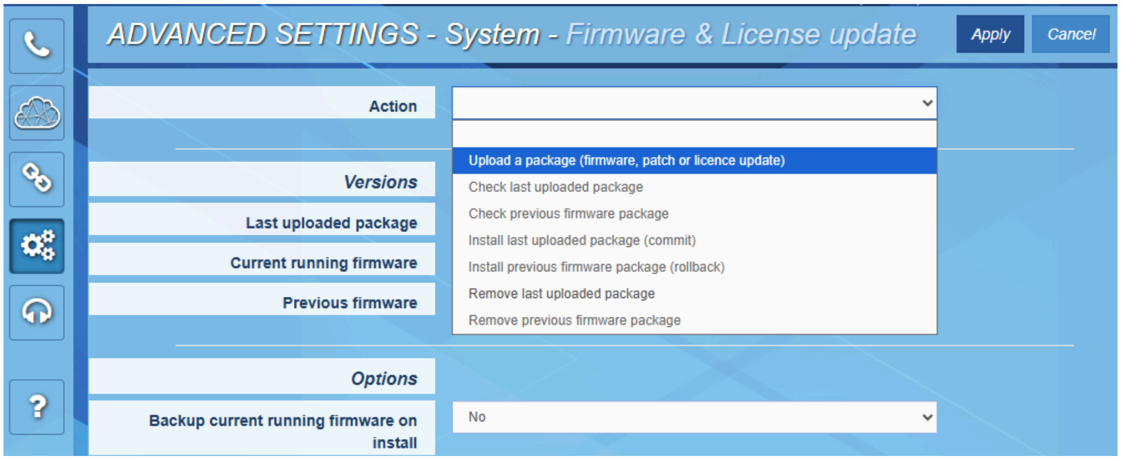
In addition, the html file which allows you to view all the parameters of the codec can be downloaded. From the download section, select " Device Information", and download

9.1.1.5 Advanced settings–> System –> Firmware update

IQOYA can be updated with a new firmware, a patch, or an optional license. The first phase of the update consists in uploading and checking the software package; during this phase, the audio activity is not stopped. The second phase consists in applying the uploaded package; audio activity is stopped during this phase.
Two firmware versions are stored locally: the currently running version, and the previous version. This allows to go back to the previous firmware version if an issue is experienced with the more recent version, without having to go through an upload.



Click on the "Action" field, and click on the arrow to display the list of possible actions.



Select the appropriate action through the list.

For a firmware update, select “Upload a package”, and click on “Browse” to select the file to be uploaded.
Click on “Apply” to start the upload. Audio activity is not stopped during the upload.
Once the package upload is completed, select the action “Install last uploaded firmware”,and click on “Apply”. Applying the firmware stops the audio activity. The equipment restarts automatically.

The following operations are also possible from the “Action” drop-down menu:

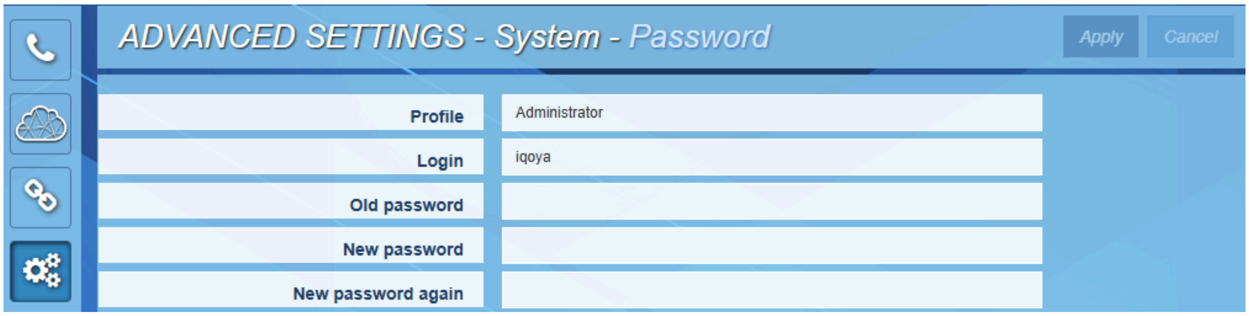
- **Check previous firmware package**: this allows checking that the previous firmware version that is stored locally is correct.
- **Check last uploaded package**: this allows checking that the last uploaded firmware version is correct. This operation is done automatically during the uploading phase.
- **Install previous firmware package** (rollback): this allows installing a previous version of the firmware that is stored locally. This is a firmware downgrade.
- **Remove last uploaded package**: this allows deleting the last uploaded package. This means that this package will not be installed.
- **Remove previous uploaded package**: this allows deleting the previous uploaded package. This means that an upload is necessary for a firmware downgrade.

Backup the current firmware when installing a new one firmware
One may want to save the current firmware when installing a new one. This allows easy firmware rollback if necessary.
Select “Yes” from the field **Backup current running firmware on install.**
It is recommended to set this option to “Yes”, otherwise the firmware version seen as “previous firmware” may not be the expected version, see table below).

	Backup current running firmware on install	**Current firmware**	**Previous firmware**
Original firmware		1	
Update with Firmware 2	Yes	2	1
Update with Firmware 3	Yes	3	2
Update with Firmware 4	**No**	4	**2 (!)**

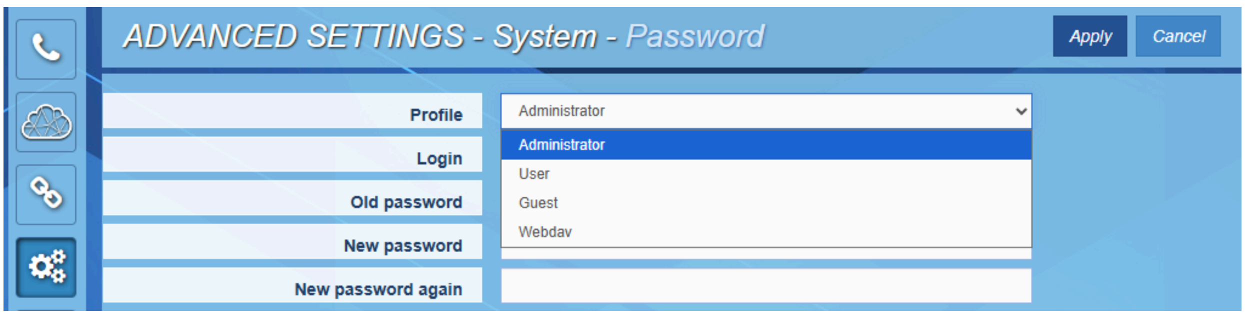
9.1.1.6 Advanced settings–> System –> Password

This page allows changing the login username password for a given user category.
This can be done when logged into the IQOYA as Administrator.



First select the profile for which credentials have to be changed.

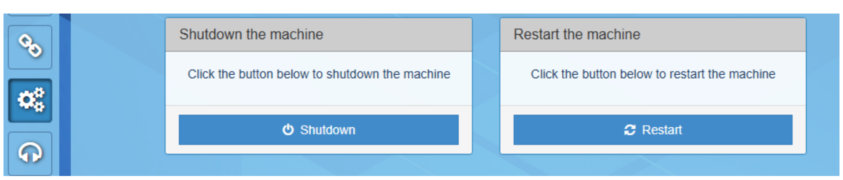
Login: allows configuring the username to be use



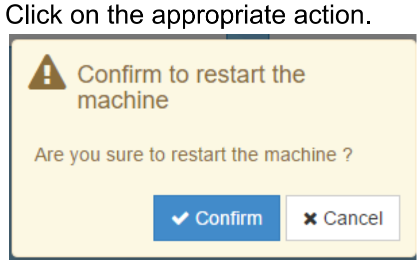
d in order to log to the WEB GUI with the selected profile.
Old password: Type the current password
New password: Type the new password
New password again: confirm the new password
Click on “Apply” to confirm the changes.

9.1.1.7 Advanced settings–> System –> Shutdown / Restart

This page allows you to restart or shutdown IQOYA.



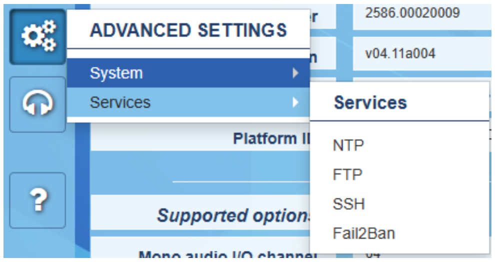
Click on the appropriate action.



Confirm or cancel your choice through the displayed confirmation window.

9.1.2 Advanced settings–> Services

This menu allows configuring the “network” services of IQOYA.

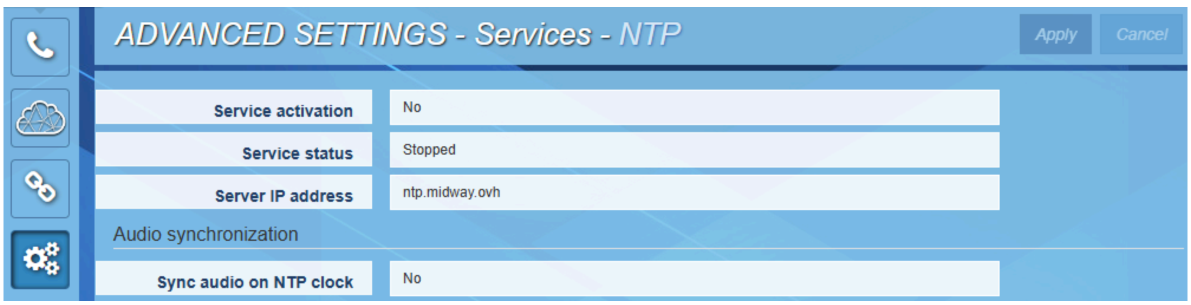


9.1.2.1 Advanced settings–> Services –> NTP

This page allows:

- configuring the date and time synchronization to an NTP server.
- enabling the optional feature “audio synchronization on NTP clock”.

NTP service is disabled by default.

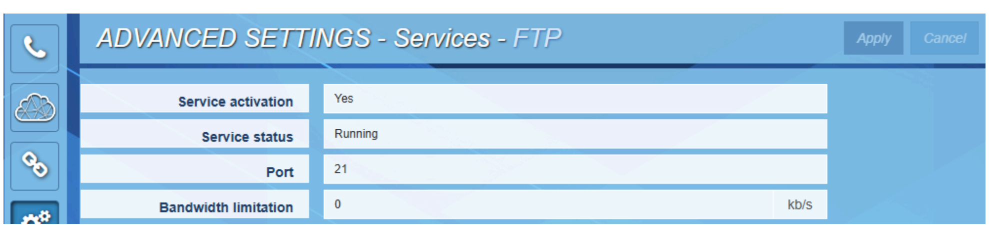


Click on the **“service activation”** field to activate/deactivate the NTP service. Select “Yes” to activate it.
Enter the IP address of the NTP server.
In case you just need to activate the date and time NTP synchronization, click on “Apply”. The status of the service is displayed in the field “Service status”.

For activation of the NTP based audio synchronization, select “Yes” for parameter “**Sync audio on NTP clock**”. This requires that the software option is installed on the IQOYA SERV/LINK, as well as on the associated IQOYA decoders.

9.1.2.2 Advanced settings–> Services –> FTP

FTP is mainly to be used for advanced support purpose. FTP service is disabled by default.

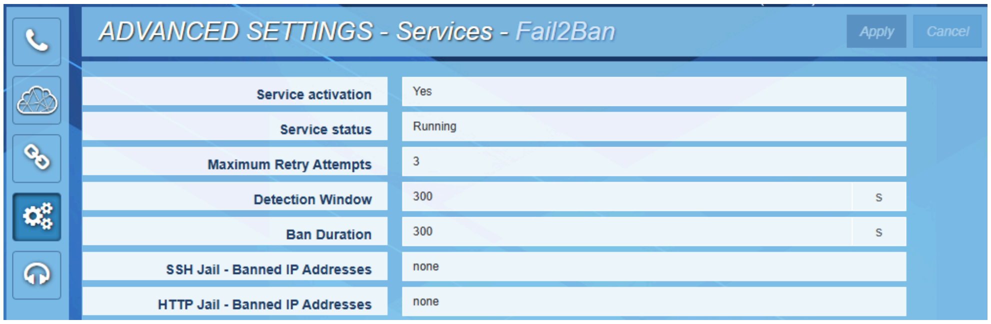


Click on the “Service activation” field. Select “Yes” to enable the FTP service, “No” to disable it.
If necessary, you may change the port used for FTP (default value is 21).
Parameter “Bandwidth limitation” allows limiting the network bandwidth of the FTP traffic.
Click on “Apply” to confirm the changes.

Note that a username and password are required to establish an FTP connection to IQOYA SERV/LINK. Username is: ftp. Password is the administrator password.

9.1.2.3 Advanced settings–> Services –> Fail2Ban

This page allows you to enable / disable the Fail2Ban service on the IQOYA. Fail2ban is an intrusion prevention tool used to protect servers from brute-force attacks and malicious access attempts.



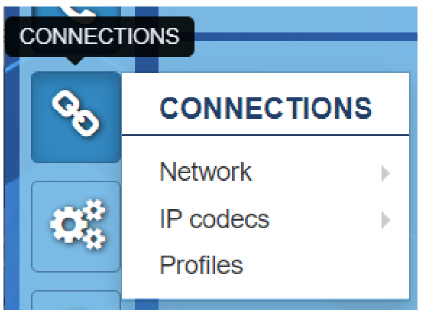
The principle of this service is that if a source (a given IP address) makes a certain number of attempts to connect to the IQOYA during a period of time, it is banned for a certain duration.This applies to HTTP/HTTPS and SSH connection attempts.

Parameter	Type	Description
Service activation	Read / Write	Yes: activation of the service. No: service disabled
Service status	Read Only	Values: Stopped or Running
Maximum retry attempts	Read / Write	Number of consecutive connection attempts from the same source. When this number is reached, the IP address of the source is banned.
Detection Windows	Read / Write	Time window (in seconds) duration for consecutive attempts.
Ban duration	Read / Write	Duration (in seconds) of the address ban.
SSH Jail - Banned IP addresses	Read / Write	IP addresses which are banned for SSH connection attempts
HTTPJail - Banned IP addresses	Read / Write	IP addresses which are banned for HTTP/HTTPS connection attempts

When a source IP address is banned, it appears as in the screen capture below.

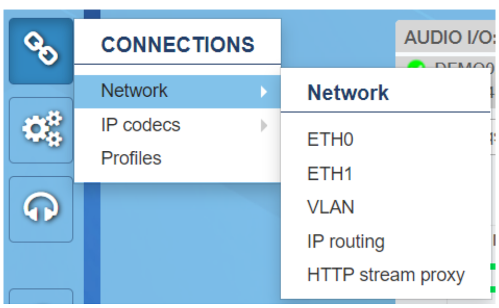
A banned IP address can be unbanned by clicking on the “Unban” button.

9.1.3 “Connections”



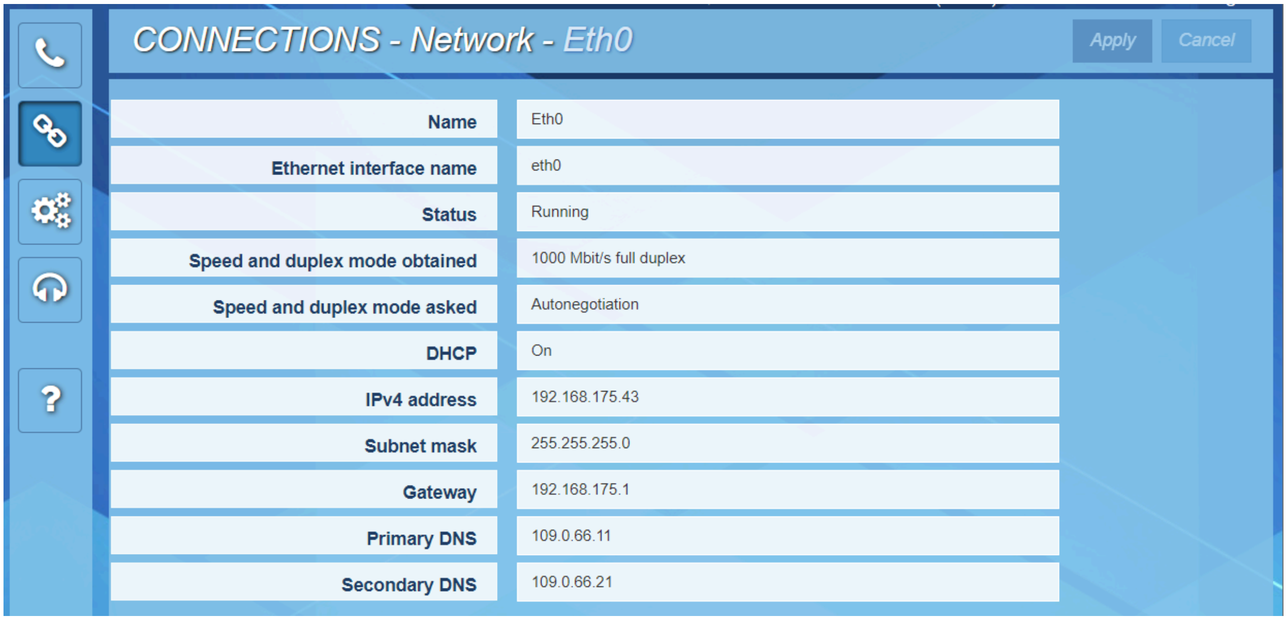
Click on  to display all the available menus.
Move the mouse pointer above the menus to display the submenus. Click on a sub-menu to display the corresponding page.

9.1.3.1 Connections –> Network

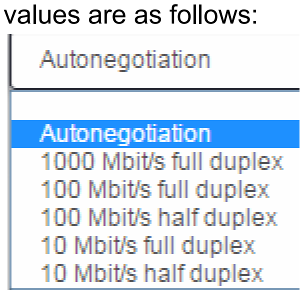
	This menu allows accessing the network configuration of IQOYA SERV/LINK.

9.1.3.2 Connections –> Network –> Eth0 / Eth1

These two pages allow configuring the two network ports of IQOYA SERV/LINK.

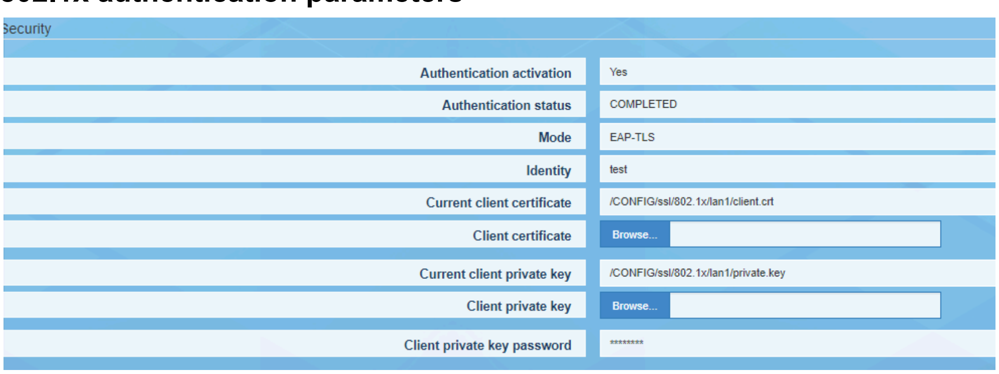


Click on a parameter field (“Status” for instance) to enter the editing mode.

Parameter	**Type**	**Description**
Internet interface name	Read	Displays the name of the Eth port: Eth0 or Eth1
Status	Read/Write	This parameter allows enabling/disabling the interface Default value=Running Possible values: Running: Eth port is enabled. Stopped; Eth port is disabled
Speed and duplex mode obtained	Read	Displays the current speed and mode of the Eth interface.
Speed and duplex mode asked	Read/Write	Allows selecting the working mode of the Eth interface. Possible values are as follows:  We recommended to avoid the “Auto-negotiation” mode. Select the mode supported by the network node connected the IQOYA.
DHCP	Read/Write	Allows enabling/disabling DHCP (Dynamic Host Configuration

		Protocol). Default value is OFF (disabled). Click on “On” to enable DHCP. This mode disables the following parameters.
IPv4 address	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Default value is 192.168.0.100 for Eth0, and 192.168.1.100 for Eth1. Enter the IP address of this Eth interface. **DHCP On** Displays the IP address automatically set through DHCP.
Subnet mask	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Enter the mask of the subnet this Eth port belongs to. **DHCP On** Displays the subnet mask automatically set through DHCP.
Gateway	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Enter the gateway IP address. Streams sent through this interface will go through this gateway. **DHCP On** Displays the gateway IP address automatically set through DHCP.
Primary DNS	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Enter the IP address of the used DNS (if any). **DHCP On** Displays the IP address of the DNS set through DHCP.
Secondary DNS	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Enter the IP address of the secondary DNS (if any). **DHCP On** Displays the IP address of the secondary DNS set through DHCP.
Authentication activation	Read/write	Set to Yes, this parameter allows configuring the 802.1x authentication parameters (see parameters description below). Set to No, 802.1x authentication is disabled.

802.1x authentication parameters

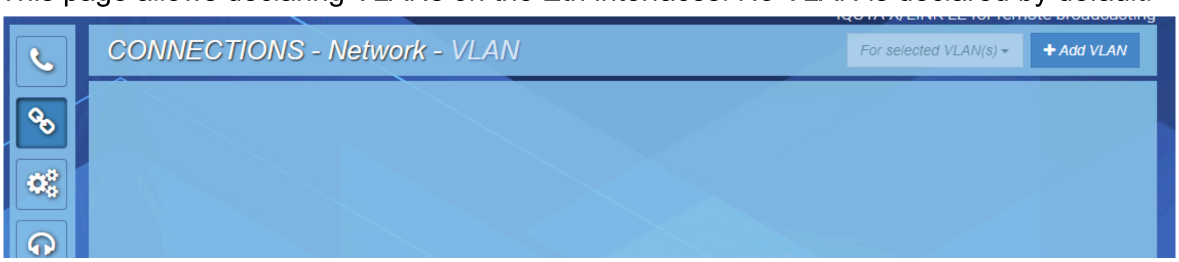


Authentication status	Read	Reports the status of the authentication process: ● COMPLETED-SUCCESS-Authorized : Authentication is successful. IQOYA

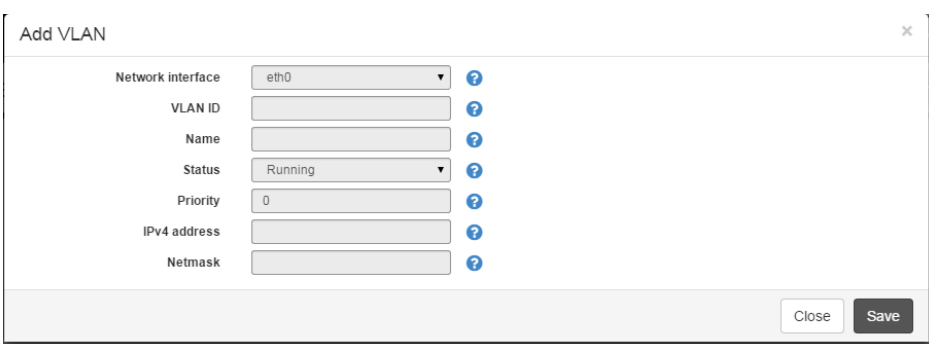
		is authorised on the network. ● COMPLETED-FAILURE-Unauthorized : The connection to the authentication service has been lost. IQOYA is not authorised on the network. ● ASSOCIATED-CONNECTING-Unauthorized-IDLE : connection in progress - IQOYA is not yet authorised on the network. ● ASSOCIATED-AUTHENTICATING -Unauthorized-IDLE : authentication in progress - IQOYA is not yet authorised on the network. ● ASSOCIATED-HELD-Unauthorized-FAILURE : Authentication failed - IQOYA is not yet authorised on the network.
Mode	Read/Write	One standard is currently supported: EAP LTS (Extensible Authentication Protocol - Transport Layer Security)
Identity	Read/Write	Identity string for EAP
Current client certificate	Read	Displays the client certificate filename currently in use.
Client certificate	Write	Allows for the selection of the certificate file to be used (.crt file)
Current client private key	Read	Displays the client private key filename currently in use.
Client private key	Read	Allows for the selection of the private key file to be used (.key file)
Client private key password	Write	A password must be entered to save the authentication settings. Enter the password for the client key. Once the password is saved, it is no longer displayed on the WEB page and is replaced by stars.

9.1.3.3 Connections –> Network –> VLAN

This page allows declaring VLANs on the Eth interfaces. No VLAN is declared by default.



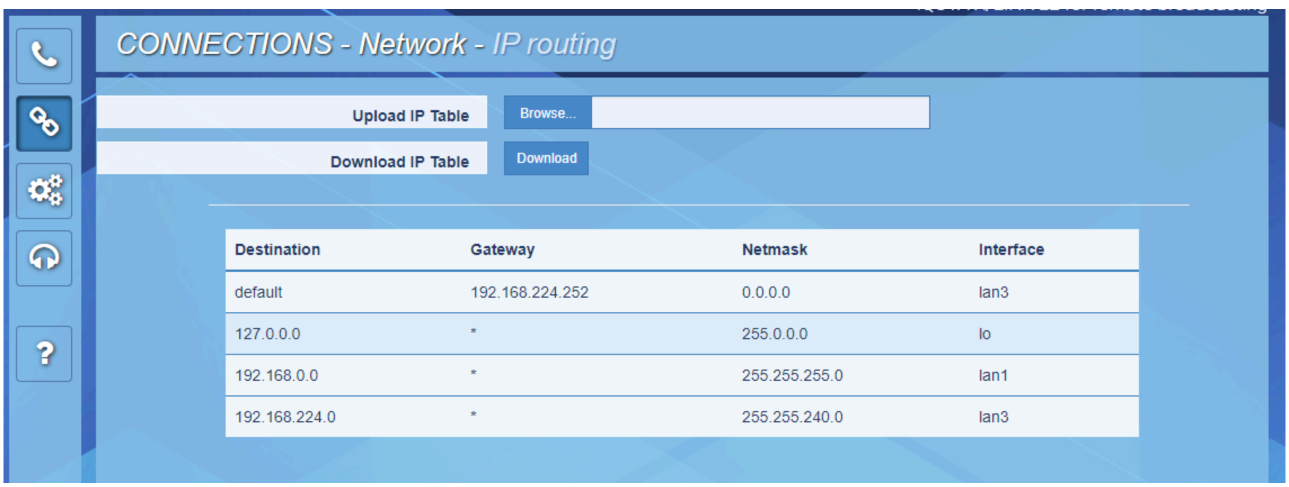
Click on “+Add VLAN” to declare a new VLAN.



Parameter	Type	Description
Network interface	Read/Write	Select the network interface that will support the VLAN (Eth0, or Eth1)
VLAN ID	Read/Write	Enter the VLAN ID in the ranges [1-1001] [1006-4095]
Name	Read/Write	Enter a logical name for this VLAN
Status	Read/Write	Allows enabling/disabling this VLAN. Select “Running” to enable this VLAN. Select “Stopped” to disable this vLAN.
Priority	Read/Write	Enter the VLAN priority in the range [0-7].
IPv4 address	Read/Write	Enter the IP address of the selected Eth port in this VLAN. If no value is entered, the IP address is the IP address of the selected Eth port.
Netmask	Read/Write	Enter the netmask for this VLAN interface. If no value is entered, the netmask is the same as the selected Eth port netmask.

9.1.3.4 Connections –> Network –> IP routing

This page allows viewing the current IP routing table, downloading it, and uploading a modified IP routing table.



In case the routing table has to be modified, click on “Download”.

The routing table can be edited with a standard text editor such (as notepad). You may add IP routes, as described in the downloaded file.

Note: In case you use both Eth0 and Eth1, do not declare two default gateways. Declare instead one default gateway for instance on Eth0, and routes on Eth1 (or vice versa).

Example:

We want to stream in dual streaming, with one stream going through a network via Eth0, and the redundant stream going through a separate network via Eth1.

- Eth0 is set to IP@ 192.168.0.100, with the gateway 192.168.0.254 declared from the WEB GUI (default gateway).
- Eth1 is set to IP@ 192.168.1.100 , with the gateway 192.168.1.254 that is not declared on the SERV/LINK.

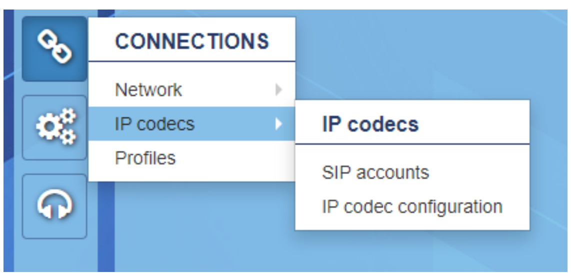
Let's suppose dual streaming is as follows:

- first stream sent to IP@ 10.0.0.140
- redundant stream sent to 193.0.0.13

If the routing table is not modified, the two streams will by default flow via Eth0 and the default gateway 192.168.0.254.
The following rule must be added via the file IpRoutingTable.cfg so that the redundant stream flows via Eth1:
-net 193.0.0.13 netmask 255.255.255.255 gw 192.168.1.254

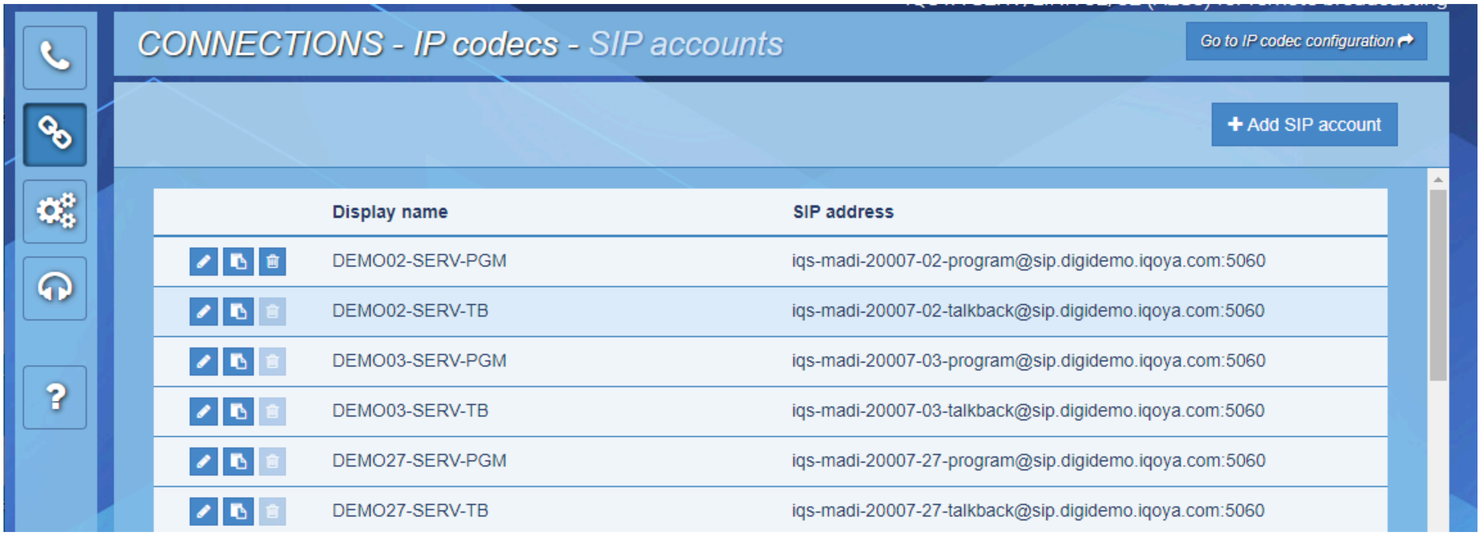
If a range of IP addresses have to be routed through Eth1, a rule like the following has to be added:
-net 193.0.0.0 netmask 255.255.255.240 gw 192.168.1.254
In this example, any stream whose destination IP@ is in the range 193.0.0.1 - 193.0.0.14 will flow through Eth1.

9.1.3.5 Connections –> IP codecs

	This menu allows accessing the configuration of the IP codec instances.

9.1.3.6 Connections –> IP codecs –> SIP accounts

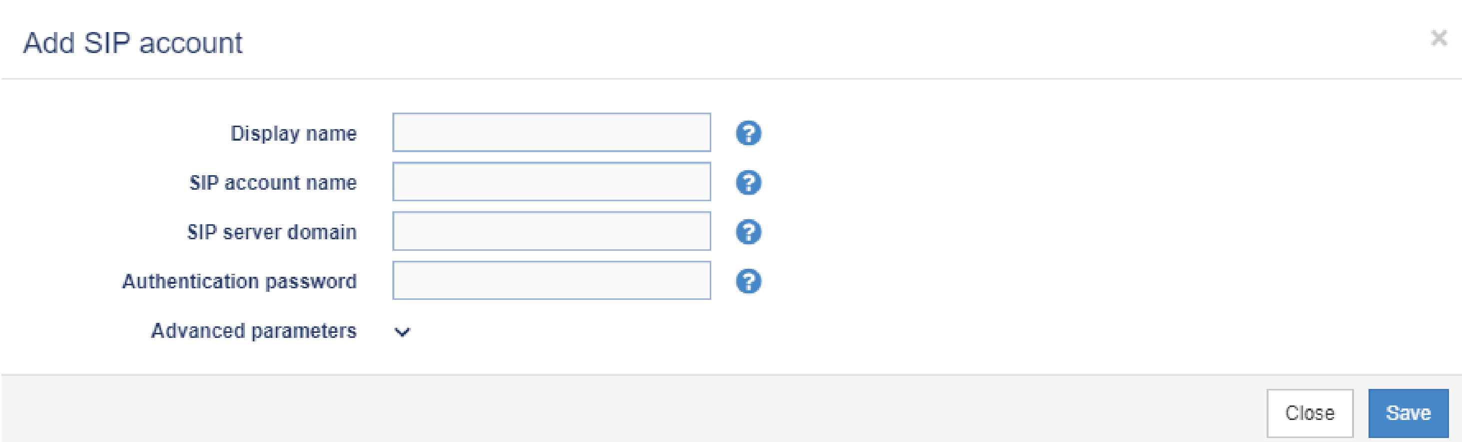
This page shows the declared SIP accounts and allows declaring new SIP accounts or editing/deleting existing ones. The SIP accounts declared in this page can be used by IP codec instances to register on SIP servers.



The shortcut *Go to IP codec configuration* allows you to quickly jump to the IP codec configuration page described below.

8.2.2.1.1 Declare a new SIP account

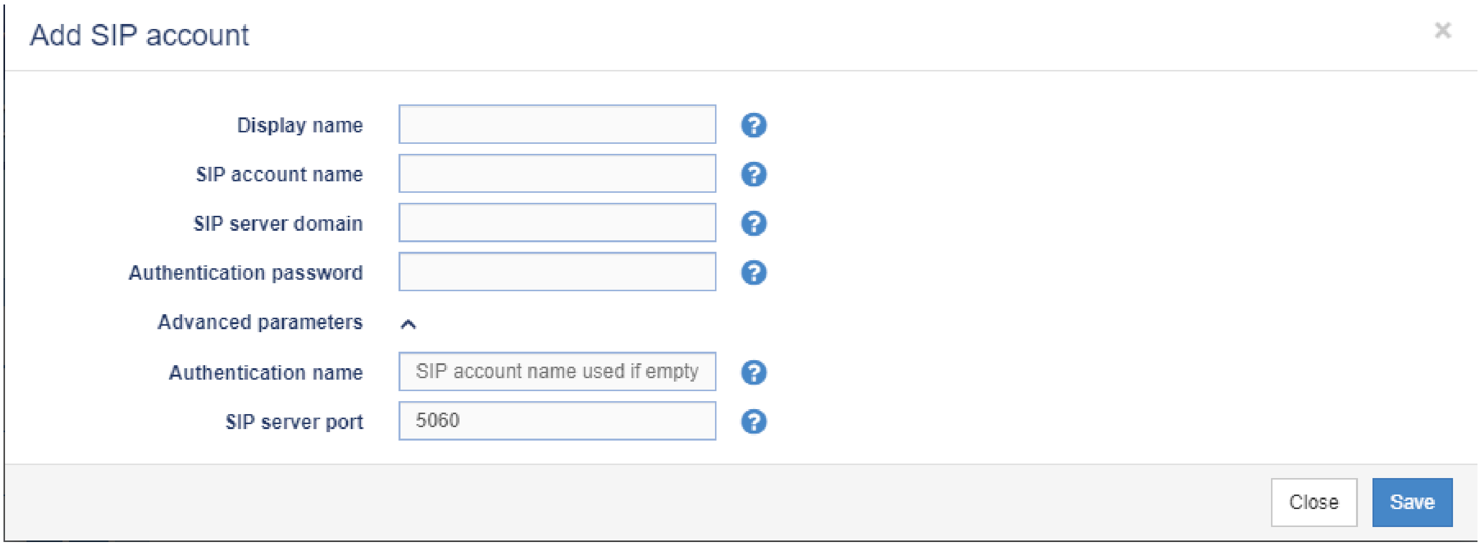
To declare a new SIP account, click on **+ Add SIP account**, or create it from an existing one by clicking the icon on the left of this latter. Then provide the requested parameters and click on the "Save" button. To cancel the declaration of a new SIP account, you can click on the "Close" button at any time. The requested parameters are described below:



SIP account parameter	Type	Description
Display name	Read/Write	Name given to this SIP account. This name will be presented to the remote party at call time by the codec instance registered with this SIP account.

SIP account name	Read/Write	Name that will be used to register with the SIP server (also called SIP registrar).
SIP server domain	Read/Write	Domain name or the IP address of the SIP server (also called SIP registrar) providing the SIP account.
Authentication password	Read/Write	The access to the SIP server is usually protected by an authentication name and password. This is the password of the SIP account on the SIP server.

With some SIP infrastructures you might have to adjust advanced parameters. Click on the chevron to access to the advanced parameters:



SIP account advanced parameter	Type	Description
Authentication name	Read/Write	The access to the SIP server is usually protected by an authentication name and password. This is the authentication name of the SIP account on the SIP server. This parameter is optional, if no authentication name is provided, the SIP account name will be used.
SIP server port	Read/Write	Listening port of the SIP server providing the SIP account. This parameter is optional, if no listening port is provided 5060 the default SIP listening port is used.

9.2.2.1.2 Edit a SIP account

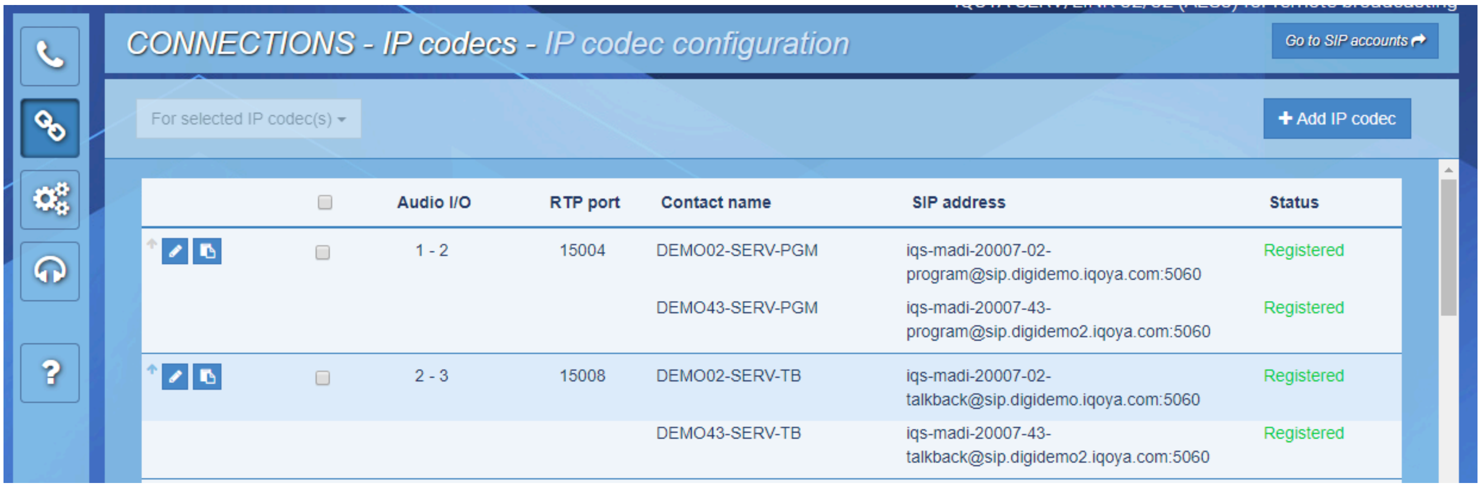
To edit an existing SIP account, click the icon on the left of this latter.
The edit page is identical to the add page described in the previous paragraph.

9.2.2.1.3 Delete a SIP account

To delete a SIP account, click the icon on the left of this latter.
Only SIP accounts that are not currently used to register IP codec instances can be deleted.

9.1.3.7 Connections –> IP codecs –> IP codec configuration

This page shows the IP codec instances and allows creating new IP codec instances or editing/deleting existing ones. The IP codec instances created on this page must be activated to be operational and to appear in the codec mosaic of the "Operations" page.



This page shows the following parameters for each IP codec instance:

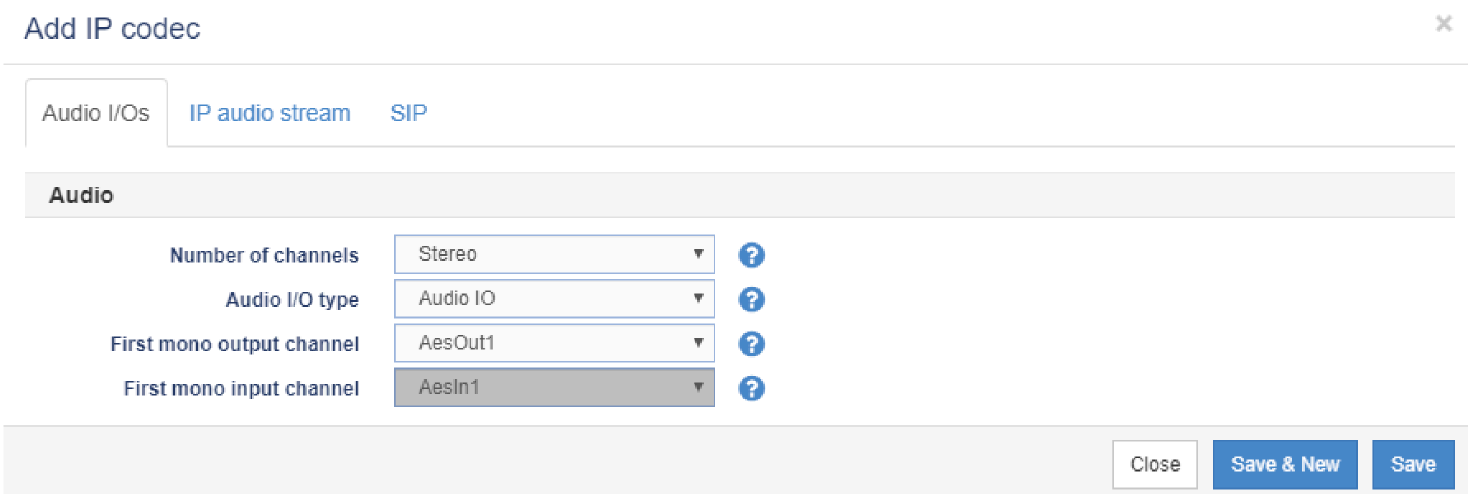
- **Audio I/O**: The audio I/Os associated with the IP codec instance.
- **RTP port**: The port used by the the IP codec instance to listen to the IP audio stream coming from the remote party.
- **Contact name**: The display names of the SIP accounts used to register the IP codec instance with SIP servers (only SIP is activated for this instance). There can be up to 2 registrations per IP codec instance.
- **SIP address**: The SIP addresses of the IP codec instance, one per registration.
- **Status**: The status of the IP codec instance is empty when the IP codec instance is disabled else the possible statuses are:
  - "Registered": SIP is activated and the IP codec instance is successfully registered with the SIP server.
  - An error message in red: SIP is activated and the IP codec instance fails to register with the SIP server. The possible error messages are:
    - "Invalid address, check DNS": the SIP domain is wrong,
    - "Unknown name or user": the SIP account name, the SIP authentication name or the SIP authentication password is wrong.
    - "No remote response": The SIP server is unreachable.
  - "Not registered": The user has manually unregistered the IP codec instance.
  - "Ready": SIP is not activated and the IP codec instance is ready for a symmetric RTP connection.
  - "Failed": SIP is not activated and the IP codec instance is not ready for a symmetric RTP connection probably because the audio listening port is not available.

9.2.2.2.1 Create a new IP codec instance

To create a new IP codec instance, click on **+ Add IP codec** button, or create it from an existing one by clicking the icon on the left of this latter. Then provide the requested parameters and click on the **Save** button. To create several instances successively, click on **Save & New** rather than on **Save**. To cancel the creation of a new IP codec instance, you can click on the **Close** button at any time.

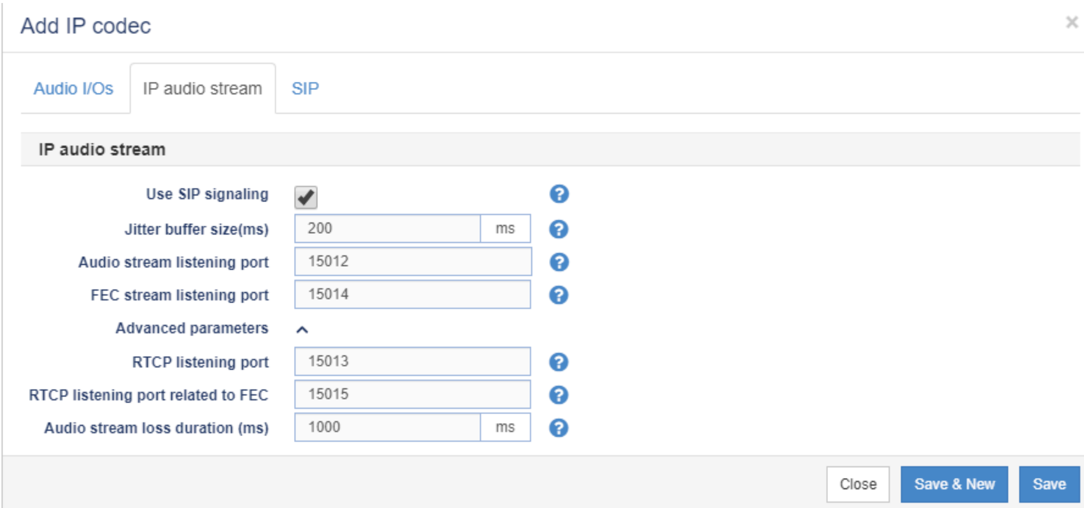
The requested parameters are described below:

- Parameters related to the audio I/Os



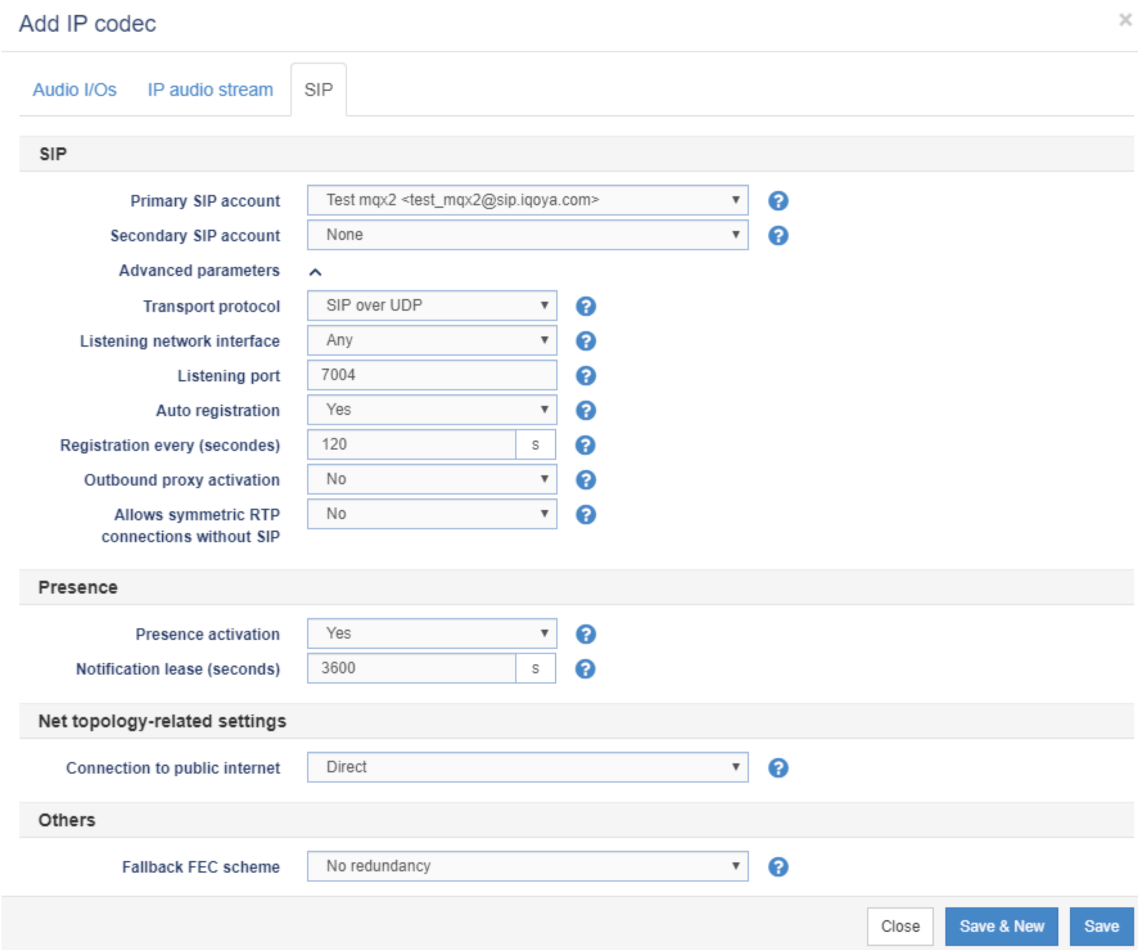
IP codec parameter	Type	Description
Number of channels	Read/Write	Number of audio channels managed by the IP codec instance. It can be Mono or Stereo.
Audio I/O type	Read/Write	Type of the audio I/Os allocated to the IP codec instance. It can be “Audio IO” for Analog or AES/EBU I/Os or “AoIP” for AES67 audio channels.
First mono output channel	Read/Write	First mono audio output allocated to the IP codec instance. If the IP codec instance is stereo, the next mono audio output is also allocated to the instance. Audio outputs already allocated are greyed out in the drop-down menu.
First mono input channel	Read/Write	First mono audio input allocated to the IP codec instance. If the IP codec instance is stereo, the next mono audio input is also allocated to the instance. By default, the input with the same number as the output is allocated.

- Parameters related to the IP audio stream received from the remote party



IP codec parameter	Type	Description
Use SIP signaling	Read/Write	Check this box if you want to establish connections via SIP (through a SIP infrastructure or directly). Checking this box brings up the SIP configuration tab.
Jitter buffer size(ms)	Read/Write	Size of the jitter buffer for the IP audio stream received from the remote party in milliseconds. The larger the buffer, the more the IP codec instance is immune to the network jitter but the higher the latency.
Audio stream listening port	Read/Write	Number of the UDP port used by the IP codec instance to listen to the IP audio stream coming from the remote party.
FEC stream listening port	Read/Write	Number of the UDP port used by the IP codec instance to listen to the FEC stream coming from the remote party if there is one.
Click on the chevron to access to these advanced parameters:		
RTCP listening port	Read/Write	Number of the UDP port used to listen to the RTCP traffic related to the audio stream coming from the remote party.
RTCP listening port related to FEC	Read/Write	Number of the UDP port used to listen to the RTCP traffic related to the FEC stream coming from the remote party.
Audio stream loss duration (ms)	Read/Write	When the IP codec instance no longer receives the IP audio stream from the remote party for a duration equal to this parameter value, a hang-up is triggered as if the hang-up button has been pressed. The value is expressed in milliseconds and must be greater than 100ms.

- Parameters related to SIP



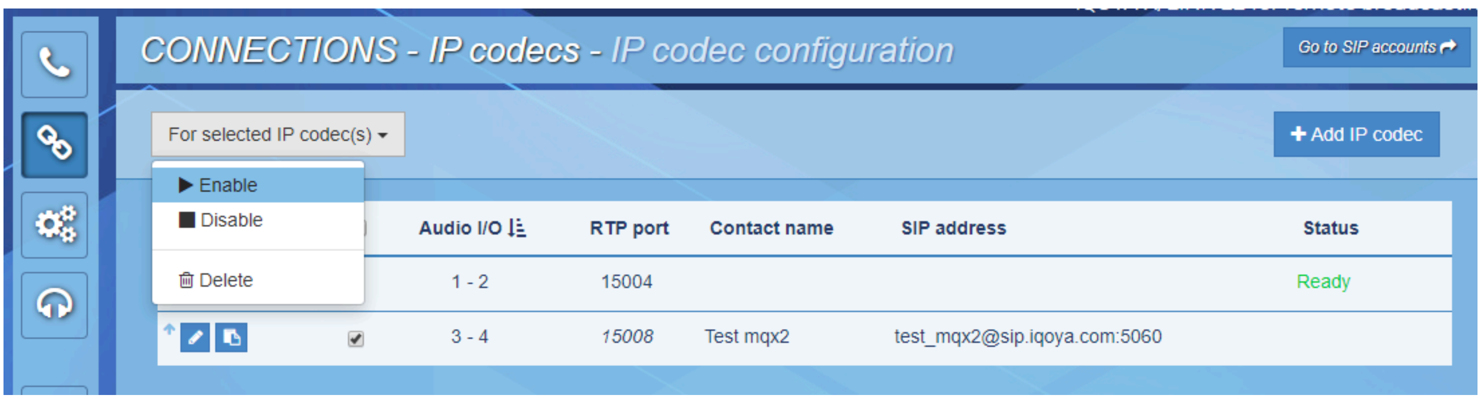
IP codec parameter	Type	Description
SIP section		
Primary SIP account	Read/Write	Primary SIP account to be used by the codec instance to register with a SIP server
Secondary SIP account	Read/Write	The codec instance can register on 2 SIP servers at the same time. So if one SIP infrastructure breaks down, the codec remains accessible through the other infrastructure.This is useful for example to implement a disaster recovery plan. This is the SIP account to be used by the codec instance to register with a secondary SIP server
Click on the chevron to access to these advanced parameters:		
Transport protocol	Read/Write	The protocol to be used to transport SIP signaling. It can be UDP or TCP. The choice depends on your SIP infrastructure. IQOYA CONNECT, Digigram's SIP infrastructure, supports both but UDP is preferable.

Listening network interface	Read/Write	The network interface to be used by the IP codec instance to listen to the SIP signaling. Use “Any” if you do not have instructions from your IT team on this.
Listening port	Read/Write	Port to be used by the IP codec instance to listen to the SIP signaling. The web interface proposes you a free port. Keep the proposed value to avoid port conflicts.
Auto registration	Read/Write	'Yes' enables automatic and periodic SIP registration(s) of the IP codec instance with the SIP server(s). The refresh period of the SIP registration is defined below. 'No' disables the automatic and periodic SIP registration(s) of the IP codec instance with the SIP server(s). Note that manual registration is possible in the call window.
Registration every (secondes)	Read/Write	This is the refresh period of the SIP registration in seconds. It is not recommended to enter a value below 30s. The default value is 120.
Outbound proxy activation	Read/Write	Enable/disable the use of an outbound SIP proxy.
Outbound proxy domain	Read/Write	Visible only if “Outbound proxy activation” is yes. This is the IP address or the domain name of the outbound SIP proxy.
Outbound proxy port	Read/Write	Visible only if “Outbound proxy activation” is yes. This is the listening port of the outbound SIP proxy.
Allows symmetric RTP connections without SIP	Read/Write	Enables/disables the possibility of also establishing or accepting symmetric RTP connections.
Presence		
Presence activation	Read/Write	Enable/disable the SIP presence service. Do not disable the SIP presence service if you use Digigram’s SIP infrastructure IQOYA CONNECT.
Notification lease (seconds)	Read/Write	This is the refresh period of the subscription to the presence service. The lease value must be greater than the field 'Registration every (seconds)’. The default value is 3600.
Net topology-related setings		
Connection to public internet	Read/Write	Select the proposition that best matches with your internet connection topology. Ask you IT team if you don’t know. If you are using IQOYA CONNECT, Digigram’s SIP infrastructure, choose “Direct” because IQOYA CONNECT integrates a NAT traversal solution.
Public IP address	Read/Write	Visible only if “Connection to public internet” is “From behind NAT specifying public address”. Enter the public IP address or domain name of the device.
STUN server address	Read/Write	Visible only if “Connection to public internet” is “From behind NAT using STUN”. This is the IP address or domain name of a STUN server.

STUN server port	Read/Write	Visible only if “Connection to public internet” is “From behind NAT using STUN”. This is the listening port of the STUN server.
Others		
Fallback FEC scheme	Read/Write	The IP codec instance enables the FEC scheme given here when the SIP signaling coming from a third party codec requires a FEC stream without specifying any FEC scheme. In this case, the FEC scheme used by the third party codec needs to match this fallback FEC scheme. Note that this field is only relevant with SIP and has no use for a communication between two Digigram’s codecs.

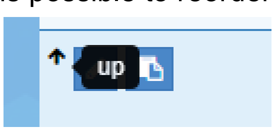
9.2.2.2.2 Enable IP codec instances

After creation, the IP codec instances must be enabled to appear in the “Operations” page.

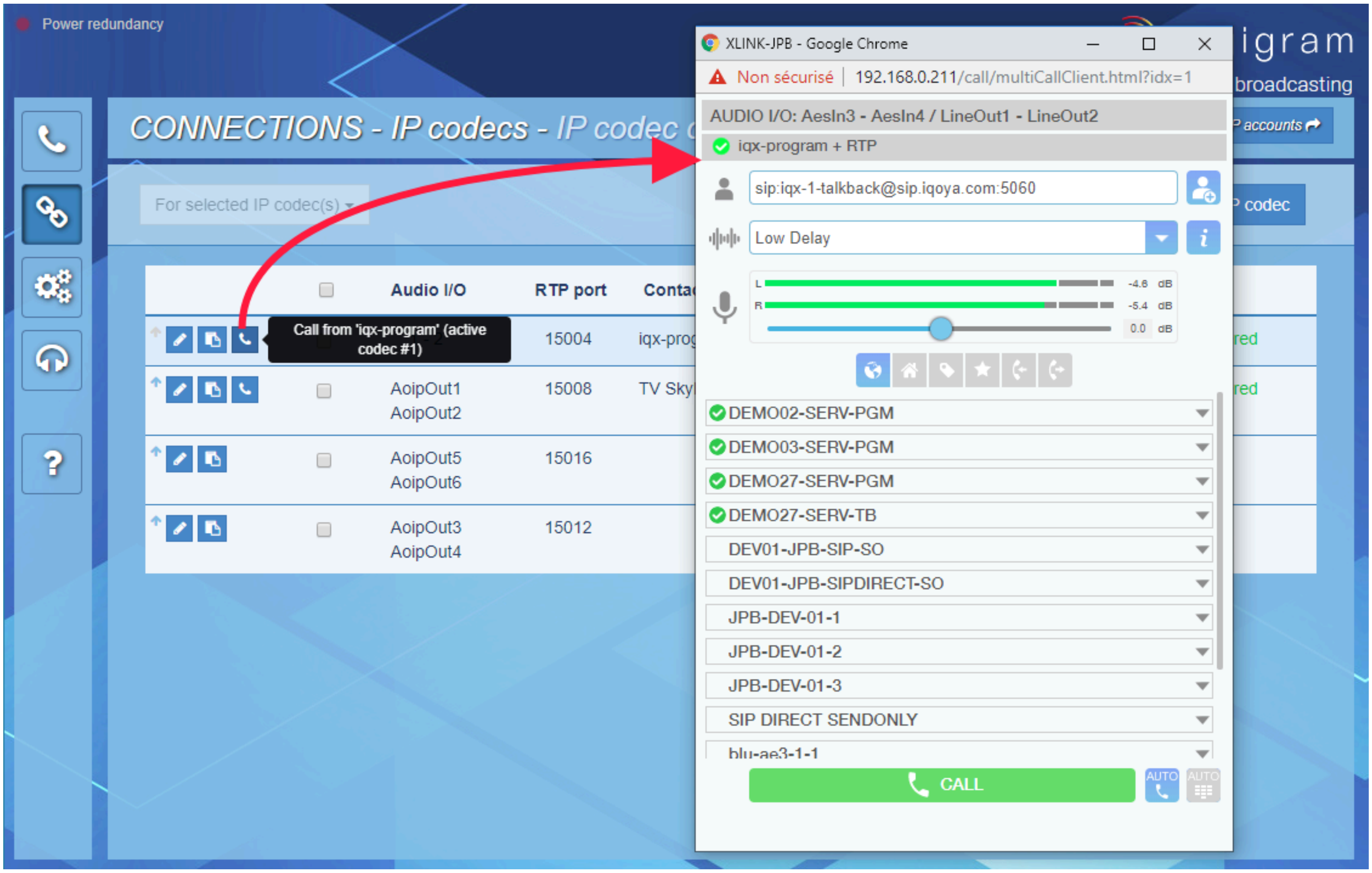


To enable IP codec instances, select them checking the box on the corresponding lines of the list, then open “For selected IP codec(s)” menu at the top of the list and click “Enable” item.
Once enabled, the codec is added to the “Operations” page.

The enabled IP codec instances appear on the “Operations” page in the same order as they appear in the list. It is possible to reorder the list clicking the “up” icon present at the beginning of each line:



To test an enabled IP codec instance, it is possible to access its call page clicking the call icon at the beginning of the line:

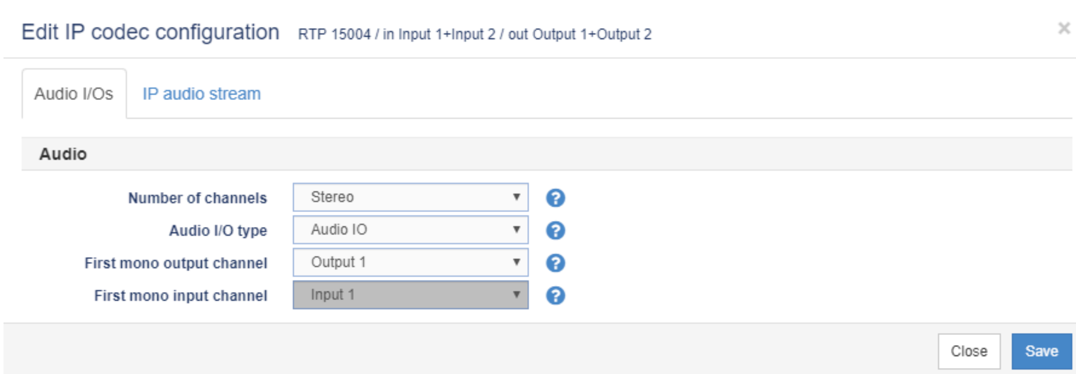


9.2.2.2.3 Edit an IP codec instance

It is possible to edit for modification an existing IP codec instance.

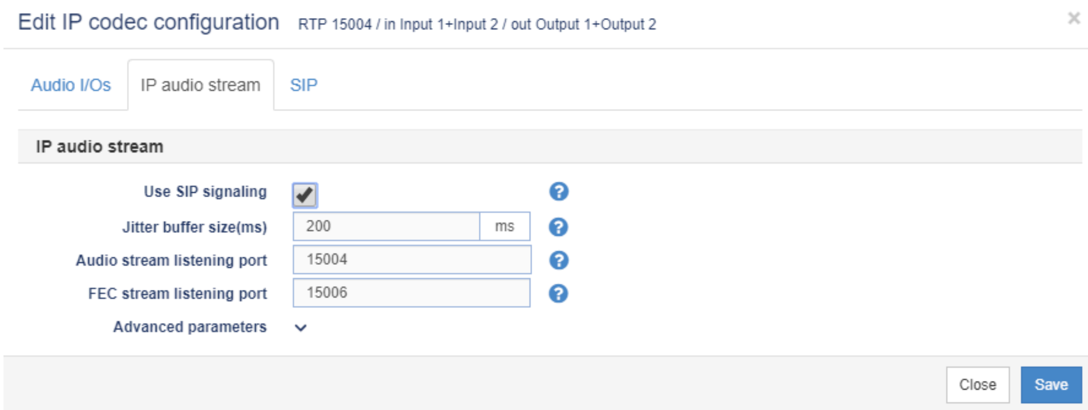
Click the pencil icon on the line of the IP codec instance you want to modify to start editing. The edition gives access to the same settings pages as the creation:

- Parameters related to the audio I/Os



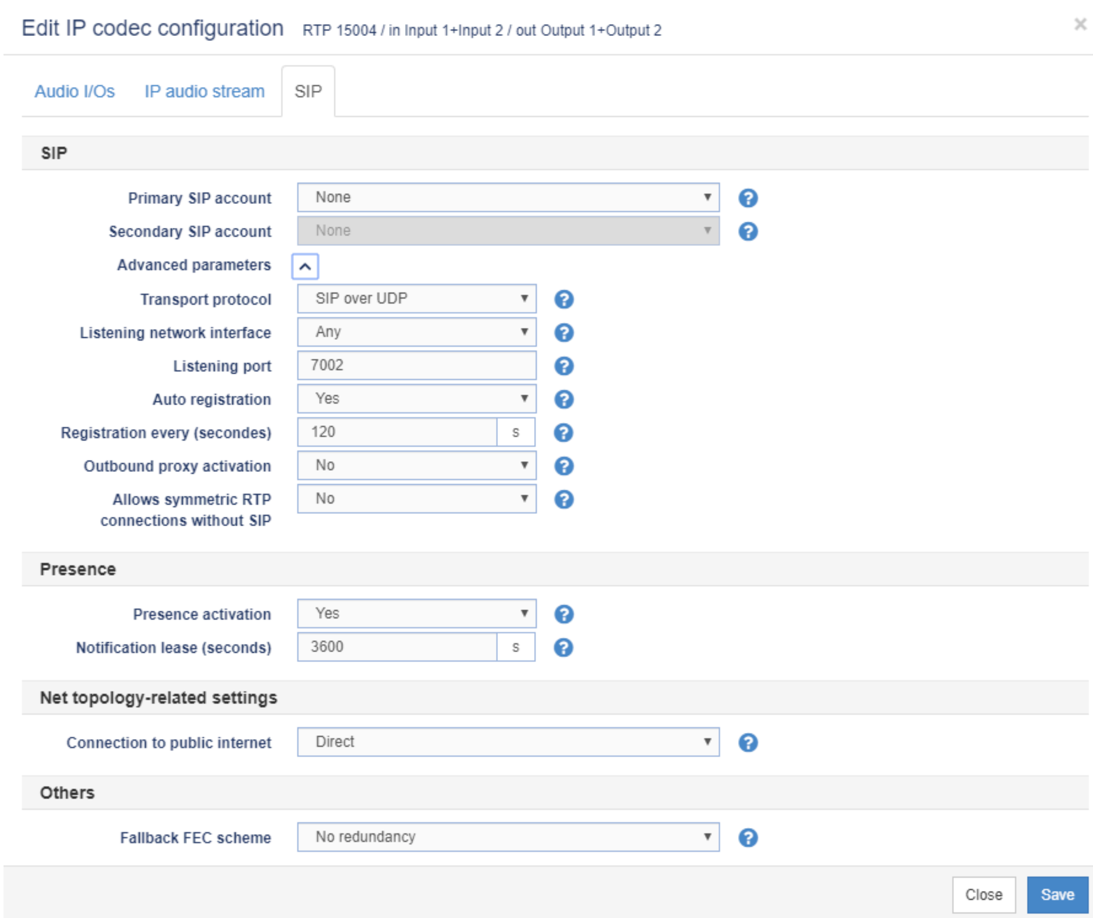
Refer to the paragraph "Create a new IP codec instance" to know the meaning of the each parameter.

- Parameters related to the IP audio stream received from the remote party



Refer to the paragraph "Create a new IP codec instance" to know the meaning of the each parameter.

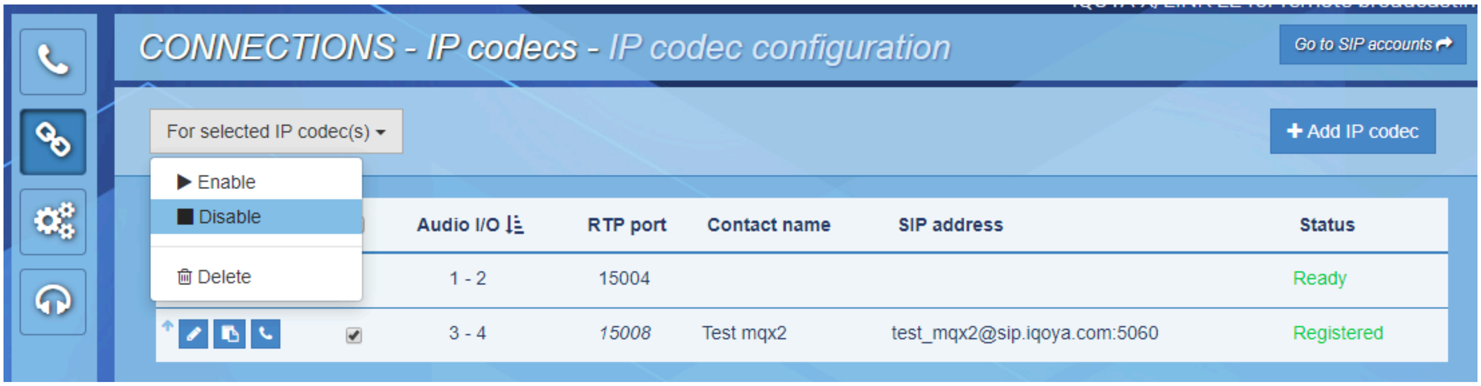
- Parameters related to SIP



Refer to the paragraph "Create a new IP codec instance" to know the meaning of the each parameter.

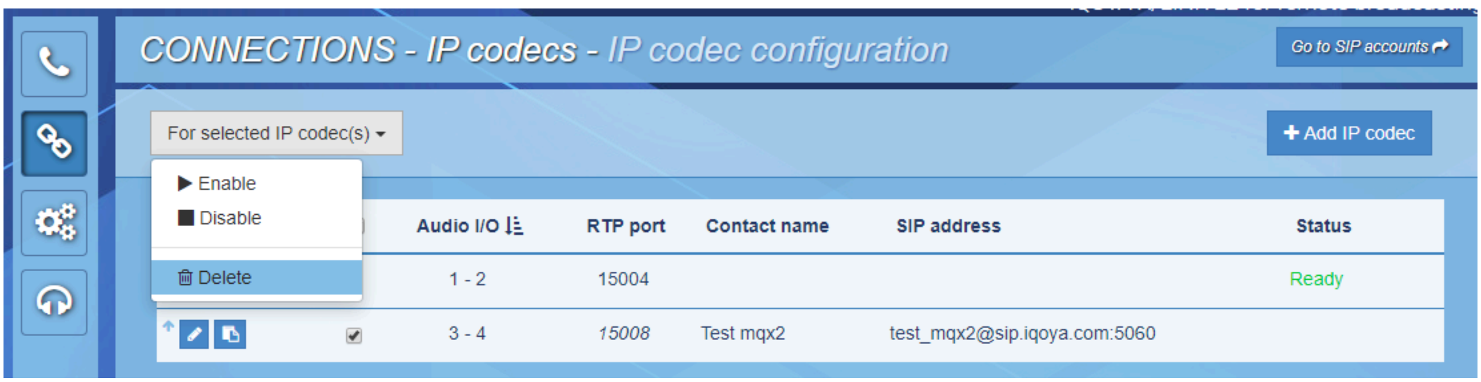
9.2.2.2.4 Disable IP codec instances

To disable IP codec instances, select them checking the box on the corresponding lines of the list, then open “For selected IP codec(s)” menu at the top of the list and click “Disable” item.
Once disabled, the IP codec instance disappeared from the “Operations” page.



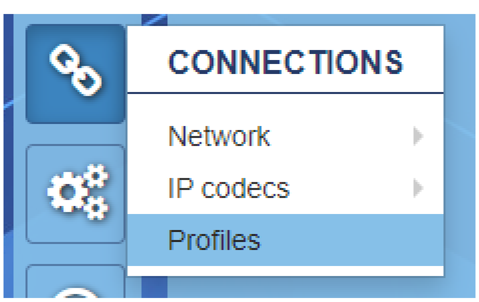
9.2.2.2.5 Delete IP codec instances

To delete IP codec instances, select them checking the box on the corresponding lines of the list, then open “For selected IP codec(s)” menu at the top of the list and click “Delete” item.

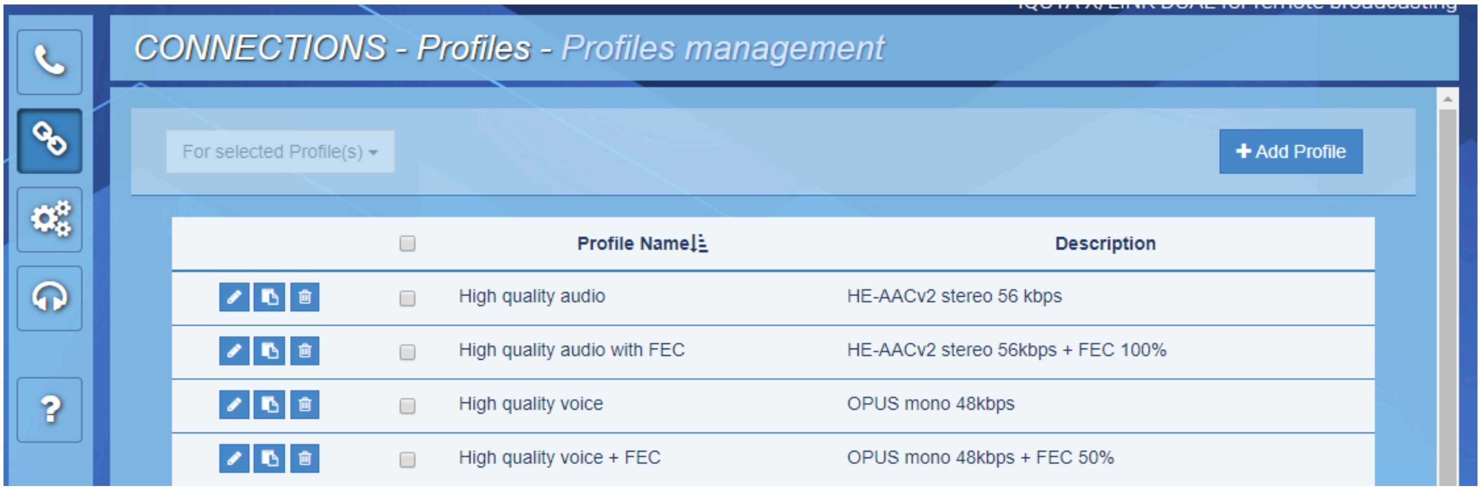


⚠ An IP codec instance must be disabled before it can be deleted.

9.1.3.8 Connections –> Profiles

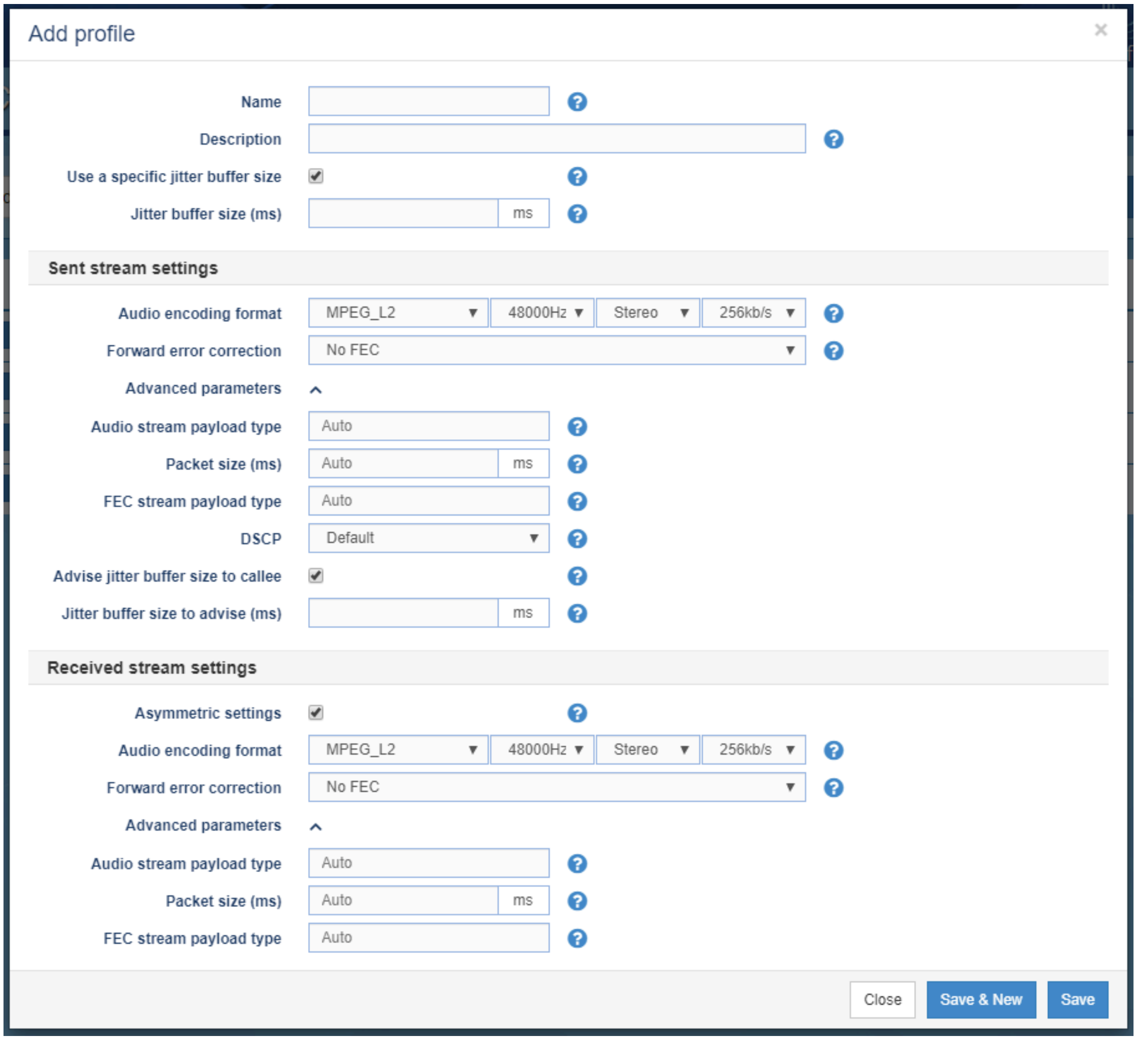
	This menu gives access to the call profile management page. This page allows you to add, modify or delete call profiles stored only locally on the device.

The profile management page shows the list of local call profiles currently defined:



9.1.3.9 Add a call profile

To create a new call profile, click on **+ Add Profile**, or create it from an existing one by clicking the icon on the left of this latter. Then provide the requested parameters and click on the "Save" button. To cancel the creation of a new call profile, you can click on the "Close" button at any time.



The parameters requested at creation are described below:

SIP account parameter	**Type**	**Description**
Name	Read/Write	Name of the call profile
Description	Read/Write	Call profile description
Use a specific jitter buffer size	Read/Write	This parameter defines the jitter buffer size to be used when the user selects the profile at call time: - Checked: the specific jitter buffer size specified below will be used, - Unchecked: the default jitter buffer size defined at IP codec instance level will be used.
Jitter buffer size (ms)	Read/Write	Visible only if “Use a specific jitter buffer size” is checked. Size of the jitter buffer to be allocated by the IP codec instance when the user selects this profile at call time.

Sent stream settings section		
Audio encoding format	Read/Write	Audio encoding format of the stream sent to the remote party.
Forward error correction	Read/Write	Forward Error Correction (FEC) is a technique used to reduce data transmission errors on unreliable networks by sending additional information allowing to correct them. This parameter allows to select the FEC scheme for the FEC stream sent to the remote party. Possible values are: ● No FEC stream ● +50% bandwidth, recovery 2, 1 stream (audio) ● +100% bandwidth, recovery 3, 2 streams (audio + FEC) ● +100% bandwidth, recovery 4, 2 streams (audio + FEC) ● +50% bandwidth, recovery 1/2, 2 streams (audio + FEC) ● +33% bandwidth, recovery 1/3, 2 streams (audio + FEC) ● +25% bandwidth, recovery 1/4, 2 streams (audio + FEC) ● +20% bandwidth, recovery 1/5, 2 streams (audio + FEC) ● +10% bandwidth, recovery 1/10, 2 streams (audio + FEC) 'recovery N' means that up to N consecutive lost IP packets can be reconstructed thanks to the FEC scheme, 'recovery 1/N' means one lost IP packet out of N consecutive packets can be reconstructed thanks to the FEC scheme.
Advanced parameters:		
Audio stream payload type	Read/Write	Payload type of the audio stream sent to the remote party. It's an integer between 0 and 127.
Packet size (ms)	Read/Write	Defines the size in ms of the audio packets sent to the remote party or 0 to use the default value. The packet size is the amount of audio data to be put in the network packets, expressed in ms. For PCM, G711, G722, and aptX formats: The entered value is adjusted to the nearest greater or equal multiple of the processing granularity. It is the amount of audio samples processed by the audio engine at each cycle. For MPEG formats: The entered value is adjusted to the nearest greater or equal multiple of the MPEG frame. For AAC formats: The entered value is adjusted to the nearest greater or equal multiple of the AAC frame.
FEC stream payload type	Read/Write	Payload type of the FEC stream sent to the remote party. It's an integer between 0 and 127.
DSCP	Read/Write	Defines the Quality of Service (QoS) class for the audio stream as defined in the Differentiated Services Code Point (DSCP) standard. Possible values are: ● Default ● Class 1 ● Class 2 ● Class 3 ● Class 4 ● Class 5

		• Class 6 • Class 7 • Assured Forwarding 11 (AF 11) • Assured Forwarding 12 (AF 12) • Assured Forwarding 13 (AF 13) • Assured Forwarding 21 (AF 21) • Assured Forwarding 22 (AF 22) • Assured Forwarding 23 (AF 23) • Assured Forwarding 31 (AF 31) • Assured Forwarding 32 (AF 32) • Assured Forwarding 33 (AF 33) • Assured Forwarding 41 (AF 41) • Assured Forwarding 42 (AF 42) • Assured Forwarding 43 (AF 43) • Expedited Forwarding (EF)
Advise jitter buffer size to callee	Read/Write	This parameter can be checked to recommend a size for the receiving jitter buffer of the remote party's device.
Jitter buffer size to advise (ms)	Read/Write	Visible only if "Advise jitter buffer size to callee" is checked. This parameter is the recommended size for the receiving jitter buffer of the remote party's device. The size is in milliseconds.
Receive stream settings section		
Asymmetric settings	Read/Write	This parameter allows to negotiate different settings for the audio stream sent by the remote party than for the audio stream sent to the remote party. - Checked: The settings for the audio streams sent by the remote party and to the remote party are different. - Unchecked: The settings for the audio streams sent by the remote party and to the remote party are the same.
Audio encoding format	Read/Write	Visible only if "Asymmetric settings" is checked. Audio encoding format of the stream sent by the remote party.
Forward error correction	Read/Write	Visible only if "Asymmetric settings" is checked. Forward Error Correction (FEC) is a technique used to reduce data transmission errors on unreliable networks by sending additional information to correct these errors. This parameter allows the user to select the FEC scheme for the FEC stream sent by the remote party. Possible values are: • No FEC stream • +50% bandwidth, recovery 2, 1 stream (audio) • +100% bandwidth, recovery 3, 2 streams (audio + FEC) • +100% bandwidth, recovery 4, 2 streams (audio + FEC) • +50% bandwidth, recovery 1/2, 2 streams (audio + FEC) • +33% bandwidth, recovery 1/3, 2 streams (audio + FEC) • +25% bandwidth, recovery 1/4, 2 streams (audio + FEC) • +20% bandwidth, recovery 1/5, 2 streams (audio + FEC) • +10% bandwidth, recovery 1/10, 2 streams (audio + FEC) 'recovery N' means that up to N consecutive lost IP packets can be reconstructed thanks to the FEC scheme, 'recovery 1/N' means one

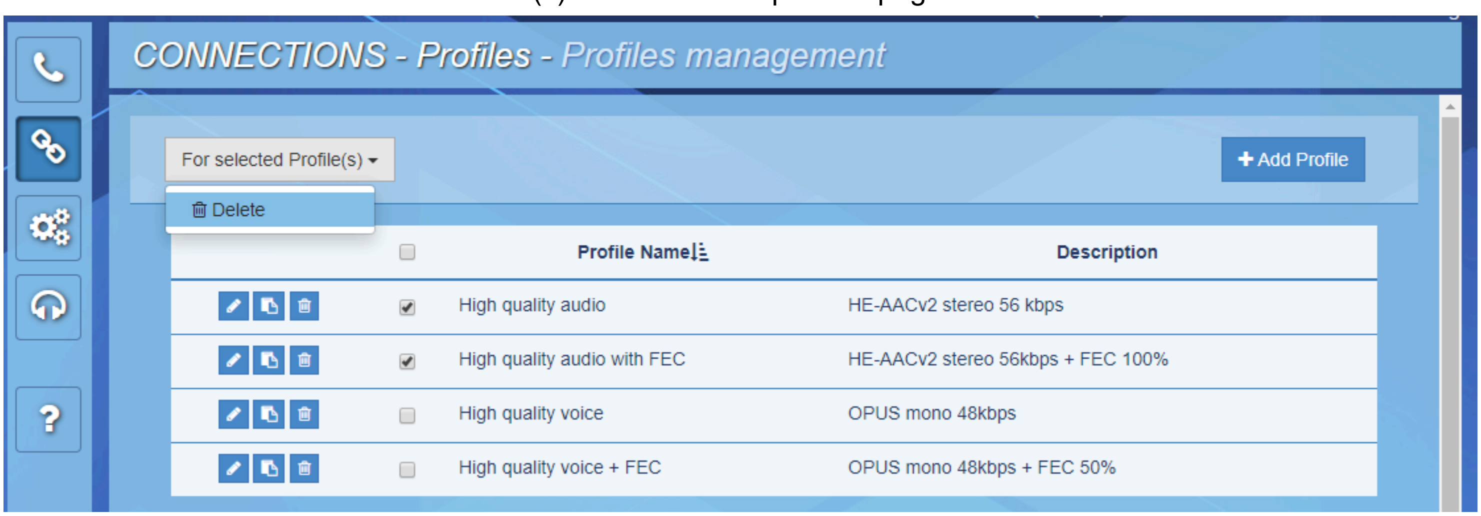
		lost IP packet out of N consecutive packets can be reconstructed thanks to the FEC scheme.
Advanced parameters:		
Audio stream payload type	Read/Write	Visible only if “Asymmetric settings” is checked. Payload type of the audio stream sent by the remote party. It’s an integer between 0 and 127.
Packet size (ms)	Read/Write	Visible only if “Asymmetric settings” is checked. Defines the size in ms of the audio packets sent by the remote party or 0 not to negotiate the packet size.
FEC stream payload type	Read/Write	Visible only if “Asymmetric settings” is checked. Payload type of the FEC stream sent by the remote party. It’s an integer between 0 and 127.

9.1.3.10 Edit a call profile

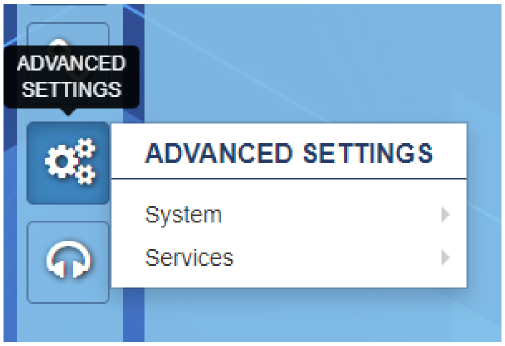
To edit an existing call profile, click the icon on the left of this latter.
The edit page is identical to the add page described in the previous paragraph.

9.1.3.11 Delete call profiles

To delete a call profile, click the icon on the left of this latter.
To delete several call profiles at the same time, check the box of the call profiles you want to delete then click “Delete” item in the “For selected Profiles(s)” menu at the top of the page:



9.1.4 “Advanced settings”



9.1.4.1 Advanced settings –> System

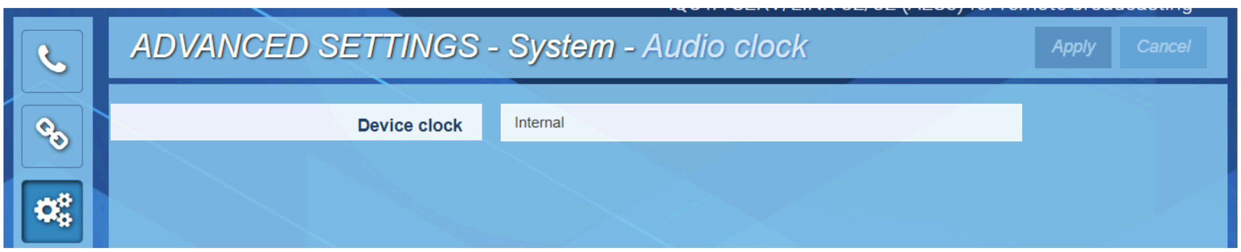
9.3.1.1 Advanced settings –> System –> Properties

Parameter	Read/Write	Meaning
Hostname	R / W	Logical name given to the device on the network.
Device Name	R / W	Name given to the equipment
Localization	R / W	Language

Serial number	R	Serial number of the unit. This number is set in the factory and cannot be changed.
Firmware version	R	Version of the firmware running on the unit. The firmware can be update.
Date	R / W	Date and time of the unit.
Platform ID	R	Identifier of the unit. this number is required for applying firmware options.
Mono audio I/O channels	R	Number of mono audio inputs and outputs allowed by the license.
Mono audio bus channels	R	Number of mono channels allocated for the buses and allowed by the license. Buses are used for transcoding.

9.3.1.2 Advanced settings –> System –> Audio Clock

This page allows defining the SERV/LINK sampling clock source:



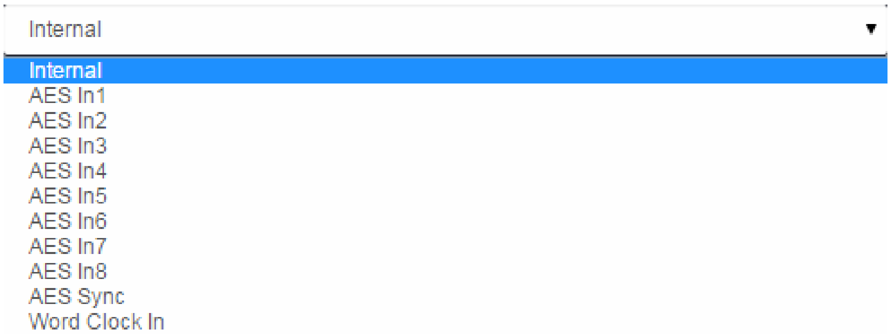
For a SERV/LINK with AES/EBU I/Os, or both analog and AES/EBU I/Os, clock source can be:

- internal: on-board clock
- extracted from an AES/EBU input (AES In x - where x is the number of the AES/EBU input)
- AES Sync: AES11 synchro input
- Word clock input

For a SERV/LINK with MADI I/Os clock source can be:

- internal: on-board clock
- MADI
- Word clock input

Click on the “Device clock” field to selected the clock source.



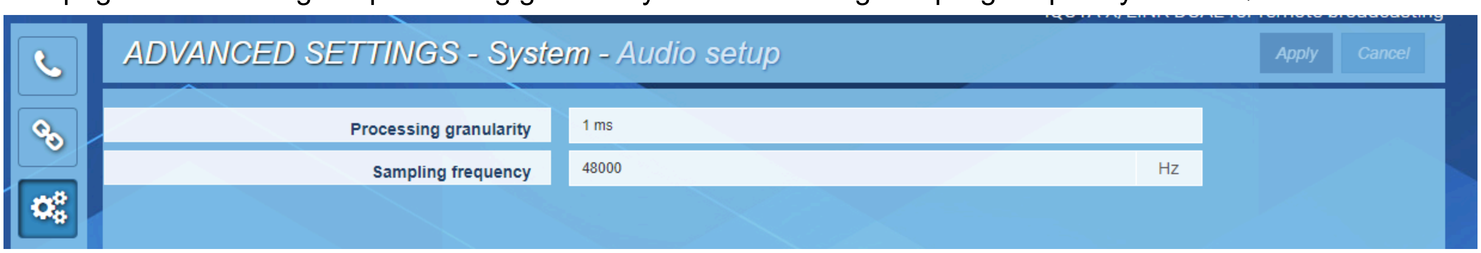
For a SERV/LINK with AES67, or “AES67 & MADI” I/O’s, or Dante I/O’s,, the clock source is “Internal”.
The real clock source can be selected through the WEB pages accessible from:

- the AES67 network interfaces. See section “WEB pages for the configuration of the AES67 parameters”.
- The Dante controller application. See section “WEB pages for the configuration of the Dante parameters”.

Click on “Apply” to confirm your choice.

9.3.1.3 Advanced settings –> System –> Audio setup

This page allows setting the processing granularity and the working sampling frequency value IQOYA SERV/LINK



Click on a parameter field to be able to change the values.

Parameter	Description
Processing granularity	This is the smallest amount of data processed at a time by IQOYA. The lower the processing granularity, the lower the latency. Possible values are 1ms, 2ms, 3 ms, 4 ms. However, a value of 1ms may lead to audio underruns, depending on the features enabled on IQOYA. In case this happens, it is necessary to increase the processing granularity value. Not: the payload size of an IP frame is adjustable via parameter Payload size, from the Send page (see paragraph Encoder parameters configuration).
Sampling frequency	It defines the working sampling frequency of IQOYA. Note that received and generated IP streams can carry audio at a different sampling frequencies (in which case a high quality frequency change is applied).

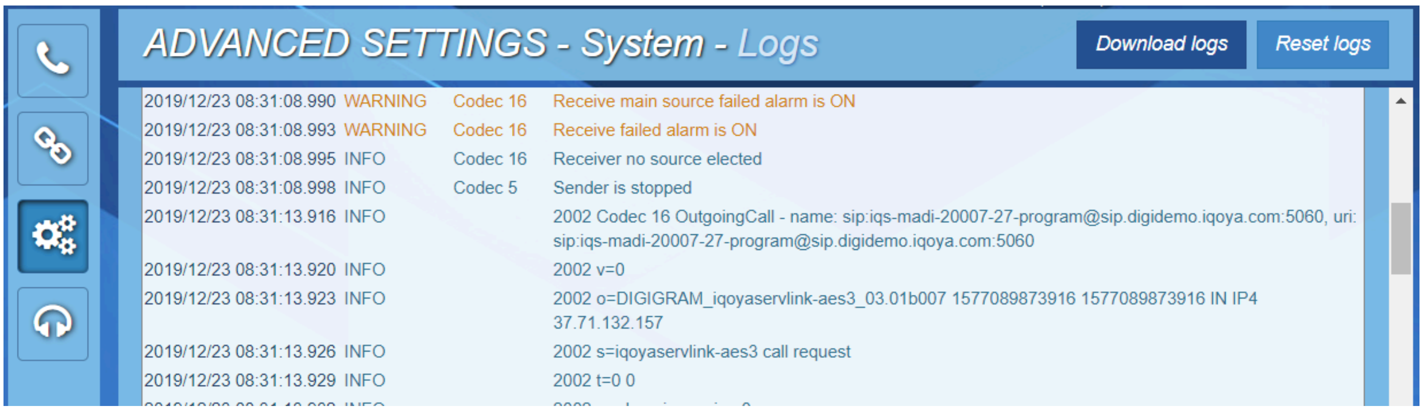
	When sampling frequency is set to 48 kHz, IP streams can be at 48 kHz, 32 kHz, 16 kHz (G722), and 8 kHz (G711). Note that 44.1 kHz is allowed for a HTTP stream. When sampling frequency is set to 44.1 kHz, IP streams must be at 44.1 kHz.

Note:

- From firmware 2.13, parameters for silence detection are configurable for each input program, and for each decoding priority of an output program.
- From firmware 2.13, backup switching criteria can be configured for each output program (Receive/Program). These criteria were global with previous firmware versions.

Click on “Apply” to confirm your changes.

9.3.1.4 Advanced settings –> System –> Logs

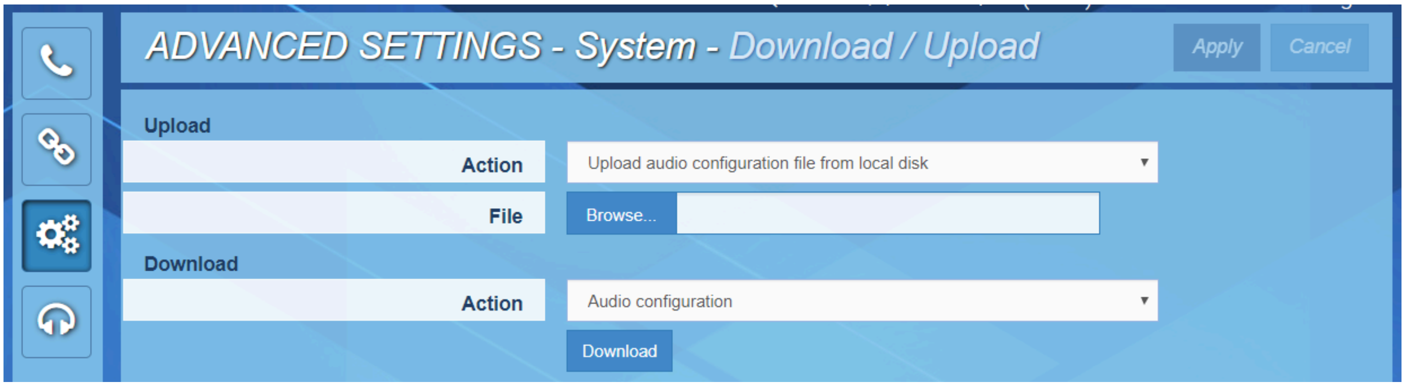


This page allows viewing and downloading the log file of IQOYA SERV/LINK. This log file gives information about the internal behaviour of IQOYA, and is useful for advanced diagnostics. Traces of enabled alarms are written into this log file (alarm ON, alarm OFF). This log file is stored internally and is persistent to a power cycle, a restart or reboot.

Event Type: allows selecting the category of traces to be displayed: Infos, Warnings, Errors, Errors & Warnings.
Codec: allows selecting one of the coedcs so that only log traces related to this codec are displayed. The number of the codec can be seen from the Send/IP Services page, and from the Receive/ Programs page.
Auto refresh: Yes/No. When set to Yes, the log file display is refreshed every few seconds
Date & Time: clicking on this icon allows to sort out the traces by date and time, starting by most recent traces or starting by oldest traces.
Reset logs: resets all the traces.
Download logs: allows for remotely downloading the traces log file.

9.1.4.2 Advanced settings –> System –> Download / Upload

This page allows downloading the IQOYA configuration to a remote PC, or uploading a configuration from a remote PC to IQOYA.



To save the current configuration of IQOYA to a remote PC, click on “ Download”.
To apply a configuration to IQOYA, click on “ Browse” to select the configuration file, and click on “Apply”.

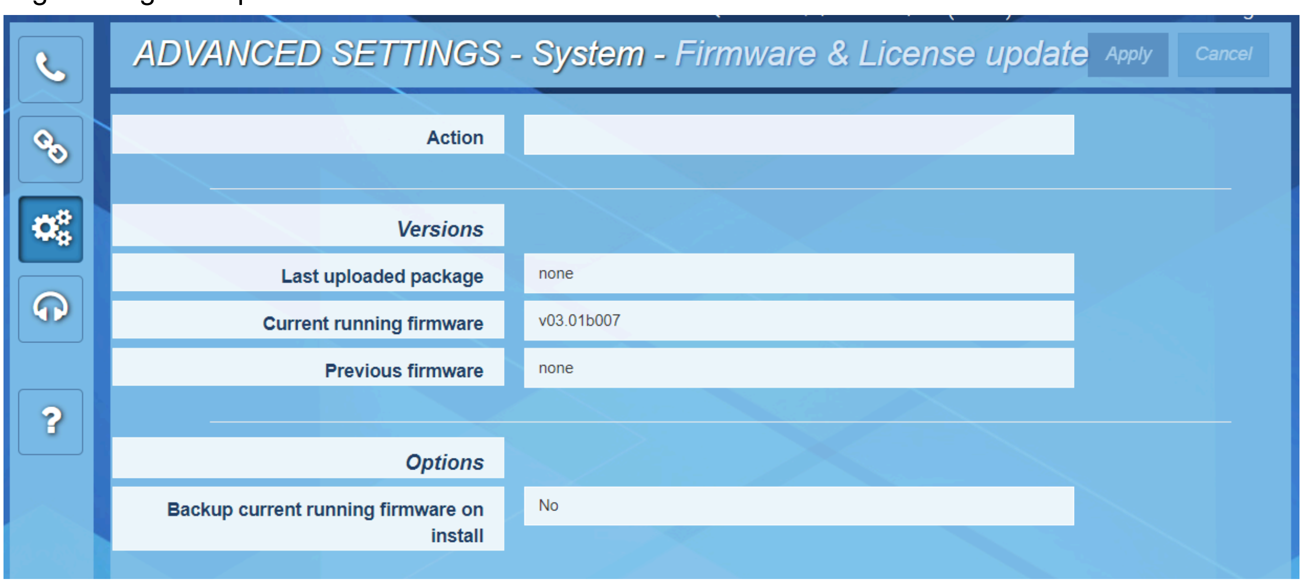
The configuration that can be uploaded/downloaded can be:

- The audio configuration only (includes the programs and IP services)
- The full codec configuration
- The connection book: The connection book is the concatenation of the contact list and the call profile list.

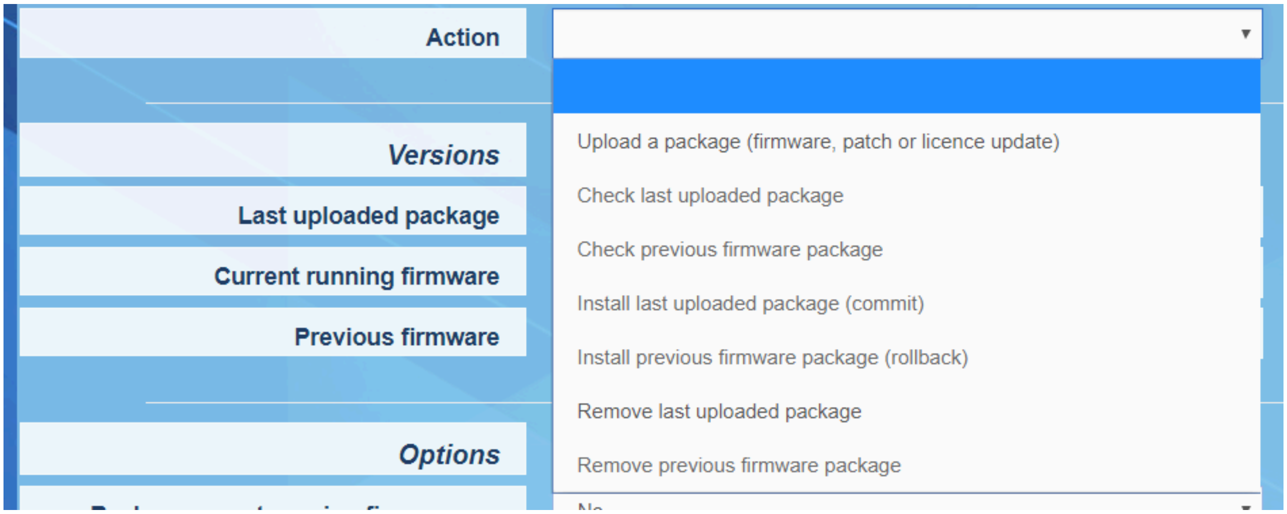
In addition, the html file which allows to view all the parameters of the codec can be downloaded. From the download section, select “ Device Information”, and download.

9.3.1.6 Advanced settings –> System –> Firmware update

IQOYA can be updated with a new firmware, a patch, or an optional license. The first phase of the update consists in uploading and checking the software package; during this phase, the audio activity is not stopped. The second phase consists in applying the uploaded package; audio activity is stopped during this phase.
Two firmware versions are stored locally: the currently running version, and the previous version. This allows to go back to the previous firmware version if an issue is experienced with the more recent version, without having to go through an upload.



Click on the “Action” field, and click on the arrow to display the list of possible actions.



Select the appropriate action through the list.

For a firmware update, select “Upload a package”, and click on “Browse” to select the file to be uploaded.
Click on “Apply” to start the upload. Audio activity is not stopped during the upload.
Once the package upload is completed, select the action “Install last uploaded firmware”,and click on “Apply”.
Applying the firmware stops the audio activity. The equipment restarts automatically.

The following operations are also possible from the “Action” drop-down menu:

- **Check previous firmware package**: this allows checking that the previous firmware version that is stored locally is correct.
- **Check last uploaded package**: this allows checking that the last uploaded firmware version is correct. This operation is done automatically during the uploading phase.
- **Install previous firmware package** (rollback): this allows installing a previous version of the firmware that is stored locally. This is a firmware downgrade.
- **Remove last uploaded package**: this allows deleting the last uploaded package. This means that this package will not be installed.
- **Remove previous uploaded package**: this allows deleting the previous uploaded package. This means that an upload is necessary for a firmware downgrade.

Backup the current firmware when installing a new one firmware

One may want to save the current firmware when installing a new one. This allows easy firmware rollback if necessary. Select “Yes” from the field **Backup current running firmware on install.**

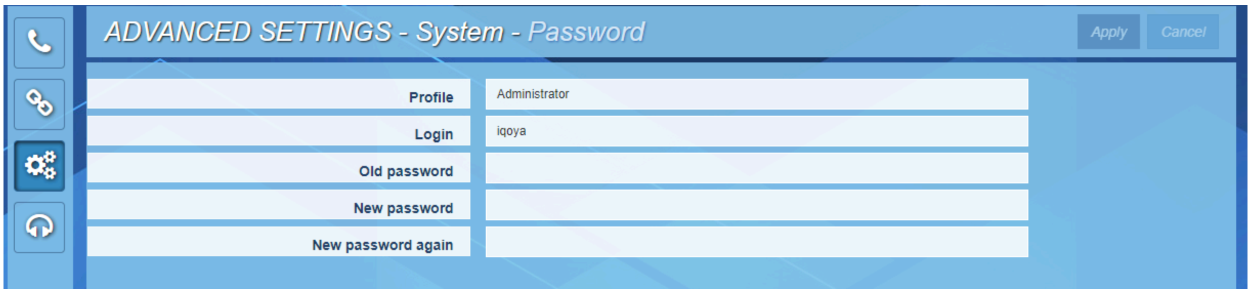
It is recommended to set this option to “Yes”, otherwise the forware version seen as “provious firmware” may not be the expected version, see table below).

	Backup current running firmware on install	**Current firmware**	**Previous firmware**

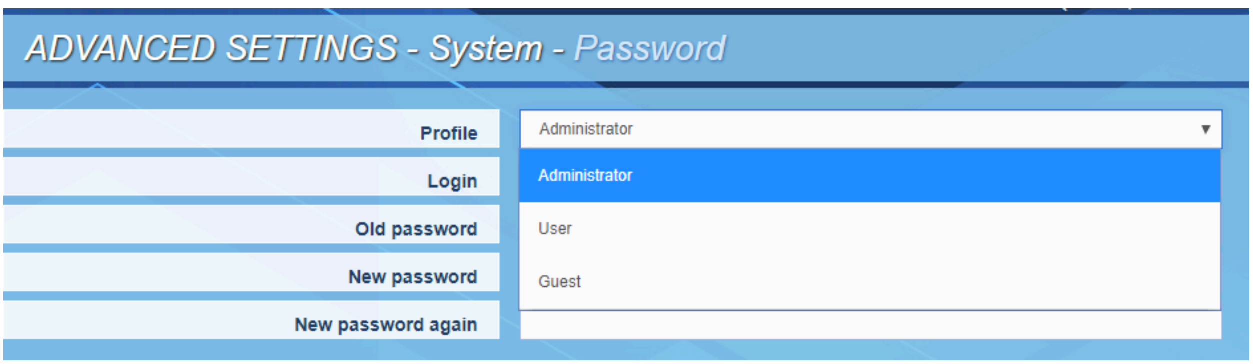
Original firmware		1	
Update with Firmware 2	Yes	2	1
Update with Firmware 3	Yes	3	2
Update with Firmware 4	**No**	4	**2 (!)**

9.3.1.7 Advanced settings –> System –> Password

This page allows changing the login username password for a given user category.
This can be done when logged into the IQOYA as Administrator.



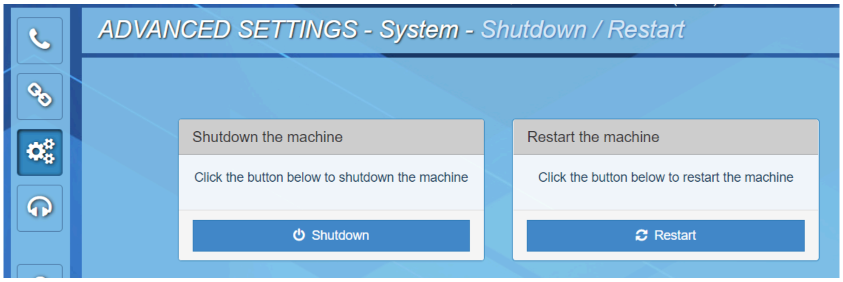
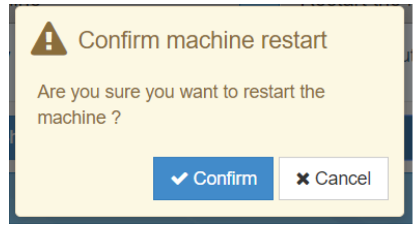
First select the profile for which credentials have to be changed.



Login: allows configuring the username to be used in order to log to the WEB GUI with the selected profile.
Old password: Type the current password
New password: Type the new password
New password again: confirm the new password
Click on “Apply” to confirm the changes.

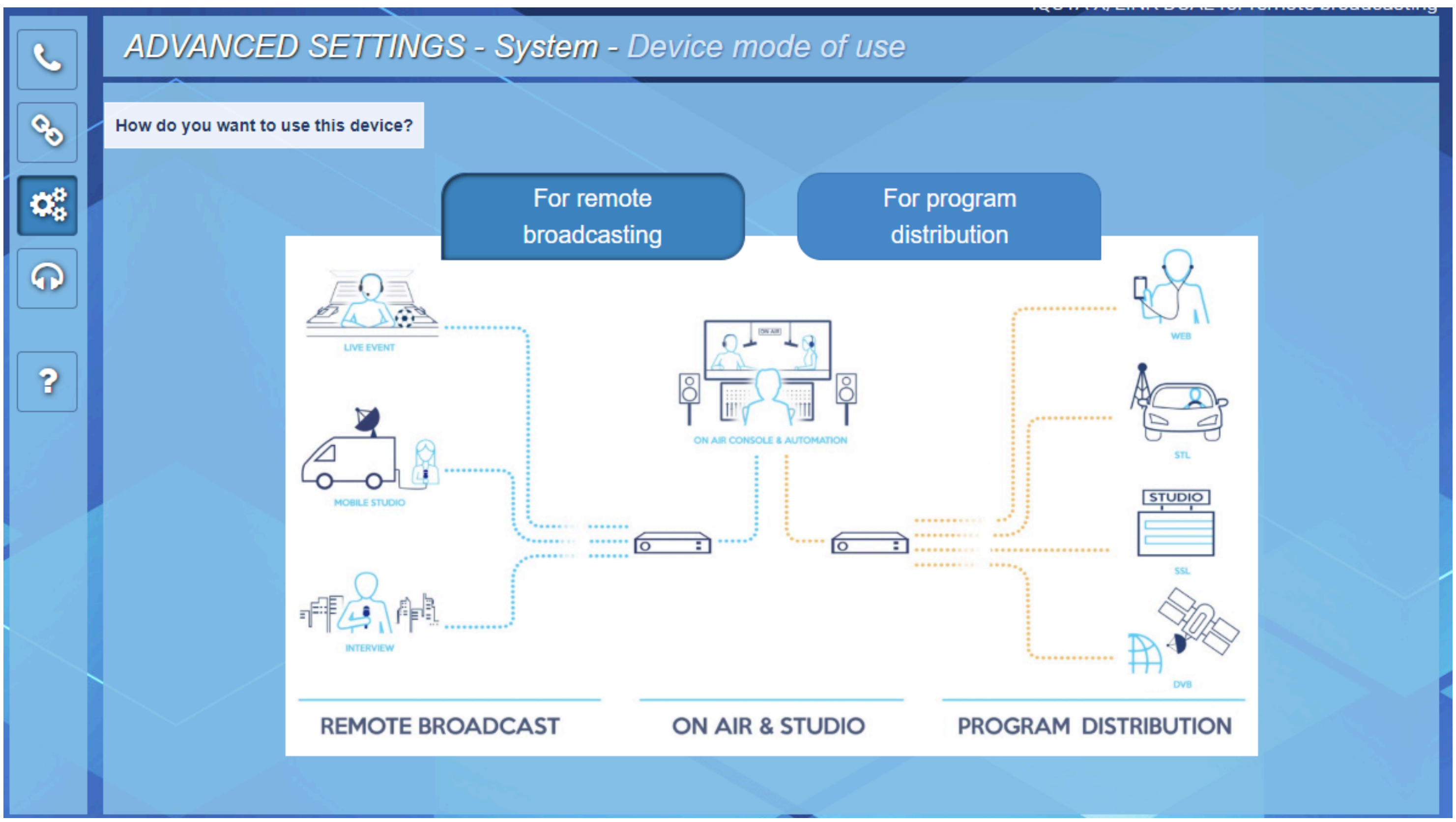
9.3.1.8 Advanced settings –> System –> Shutdown / Restart

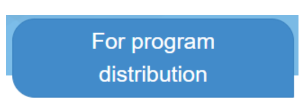
This page allows to restart or shutdown IQOYA.

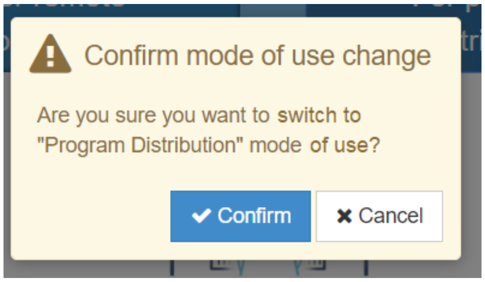
	Click on the appropriate action.  Confirm or cancel your choice through the displayed confirmation window.

9.3.1.9 Advanced settings –> System –> Switch mode of use

This page allows switching from “Remote Broadcasting” mode of use to “Program Distribution” mode of use and vise versa:



To switch to “Program Distribution” mode of use, click  button then confirm your choice through the displayed confirmation window:



9.1.4.3 Advanced settings –> Services

This menu allows configuring the “network” services of IQOYA.
Please see section 6.3.3.

9.1.5 Audio I/Os

Please refer to Audio I/Os.

9.2 Connection to IQOYA CONNECT

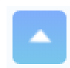
This page allows registering IQOYA X/LINK to an organisation in IQOYA CONNECT. This requires a subscription to the IQOYA CONNECT service.

	Once you are subscribed to the IQOYA CONNECT service, you have a login password or pincode that you have to enter in this window together with your email address. Press “Sign In”.

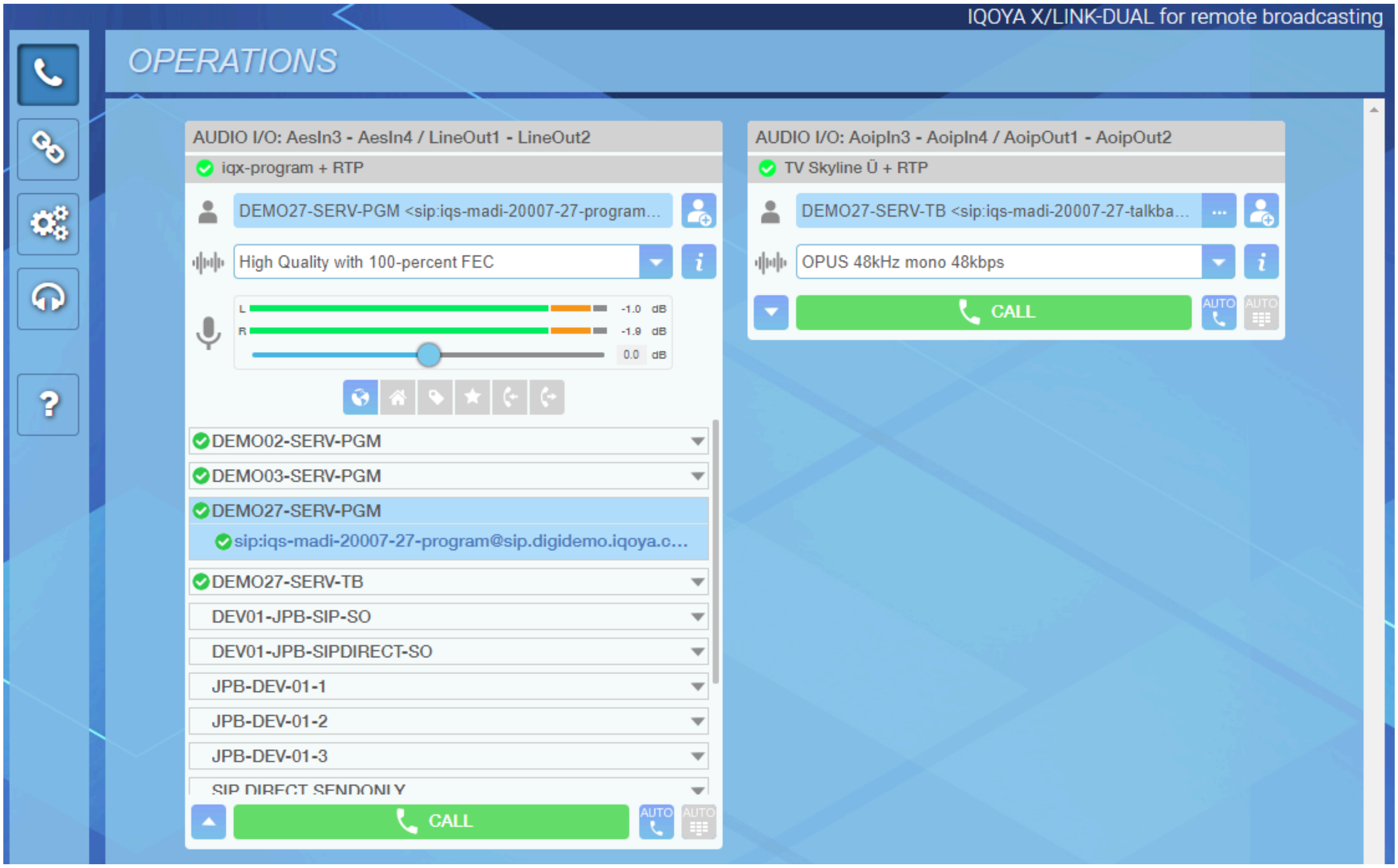
	If the login gives access to several organizations in IQOYA CONNECT (the user is subscribed to more than one organization), the list of possible organizations is displayed. Select the organization by clicking on its checkbox, and select “Attach to the selected organization".

When using IQOYA in the context of IQOYA CONNECT, all the configuration about contacts, audio profiles, SIP accounts and codec instances is done from IQOYA CONNECT.
The “Operations” page described here below can still be used, although all operations are supposed to be done from the IQOYA CONNECT GUI.

9.2.1 “Operations”

This page presents the call windows of the codec instances currently configured and enabled. Each call window can be expanded clicking on  or collapsed clicking on 

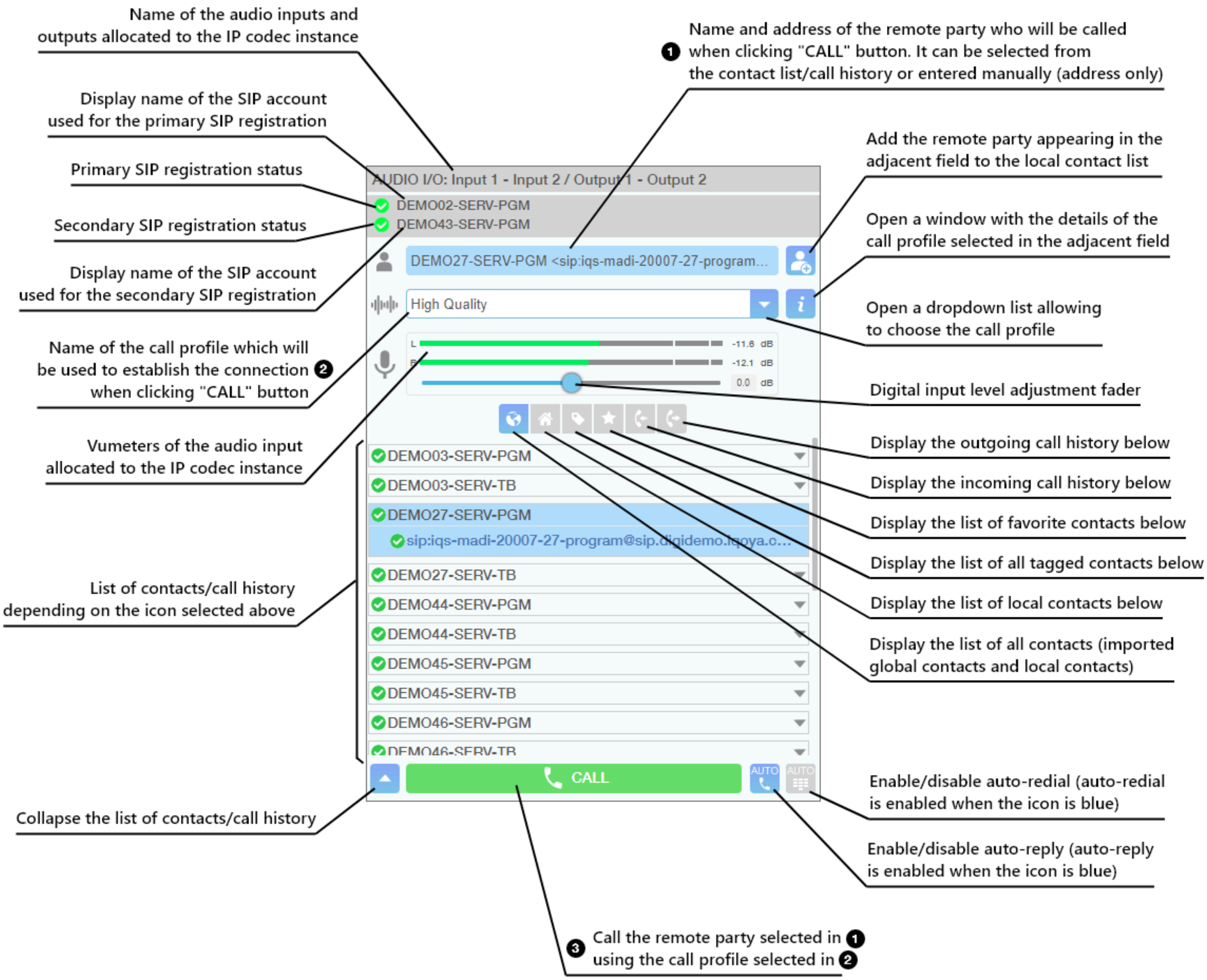
In the example below, the call window of the first IP codec instance is expanded while the call window of the second codec instance is collapsed:



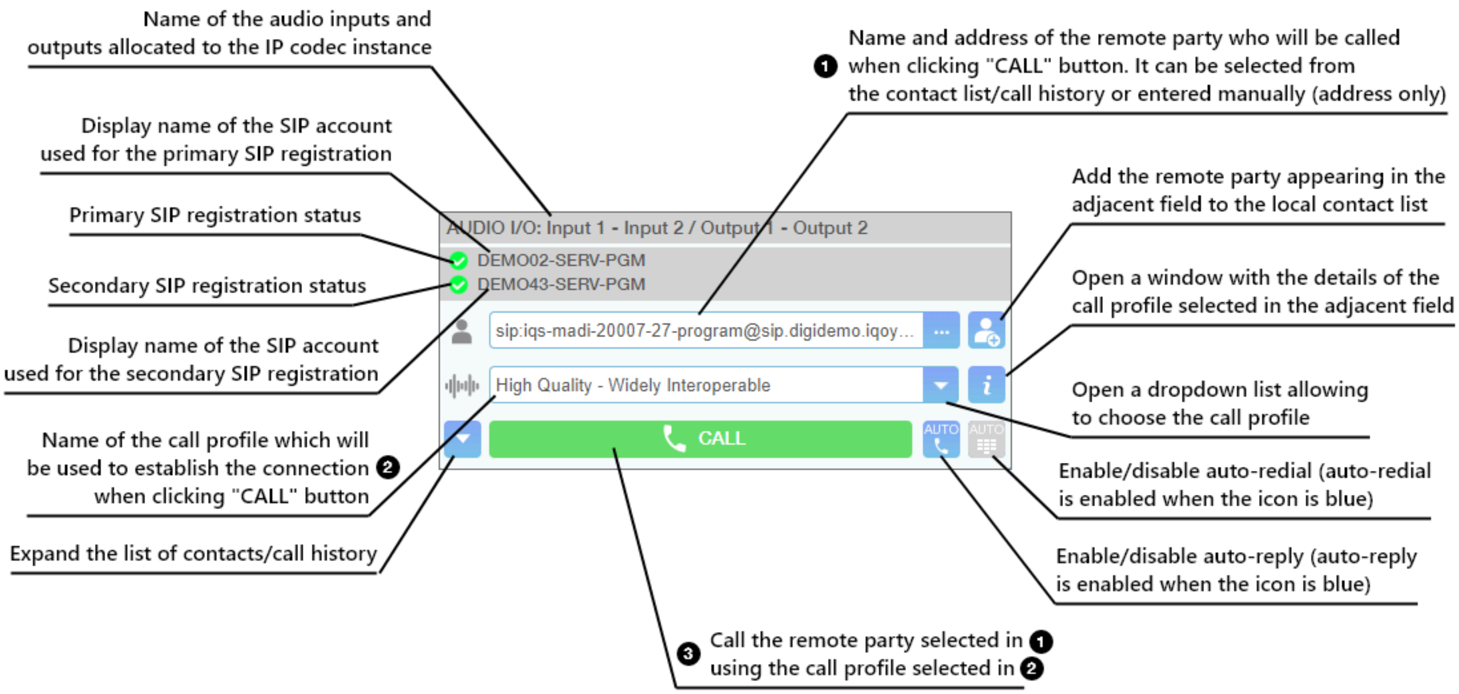
Each call window can be reopened in an independent window by double clicking on its title bar.

9.2.1.1 Call window when no communication is in progress

Call window (expanded version):



Call window (collapsed version):



9.2.1.2 Place a call

Please refer to the image of the previous paragraph for references to the graphical interface.

To place a call the user (1) select a remote party in the contact list/call history or enter the remote party address, (2) select a call profile and (3) press the CALL button. The CALL button is grayed out until the remote party and the profiles have been specified.

Format of the remote party address

- For a SIP connection, the address is:

  sip:sip_account_name@sip_server_domain:sip_server_port

  (the *sip_server_port* is optional, 5060 is used as default).

  The "*sip:*" prefix must not be forgotten when the SIP address is entered manually.
- For a direct SIP connection, the address is:

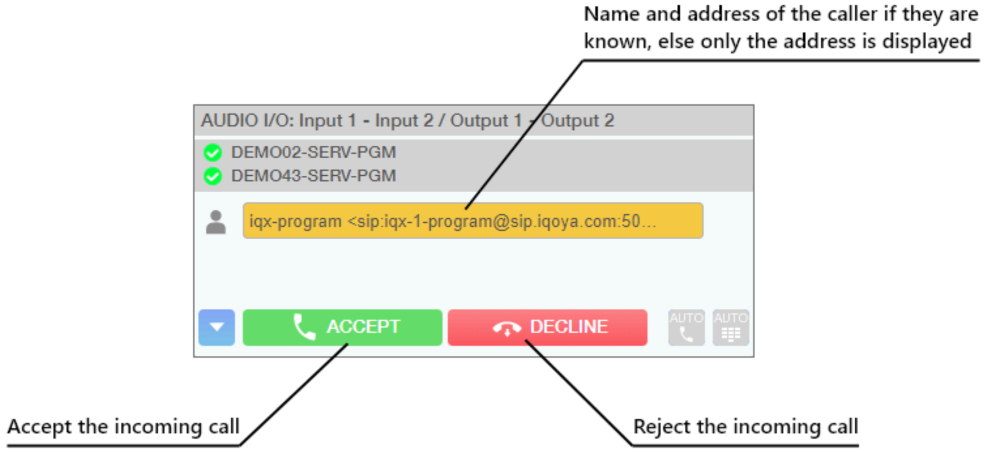
  sip:@remote_party_IP_address:remote_party_SIP_listening_port

  (the *remote_party_SIP_listening_port* is optional, 5060 is used as default)
- For a symmetric RTP connection, the address is:

  remote_party_IP_address:remote_party_audio_listening_port
-

9.2.1.3 Accept or reject a call

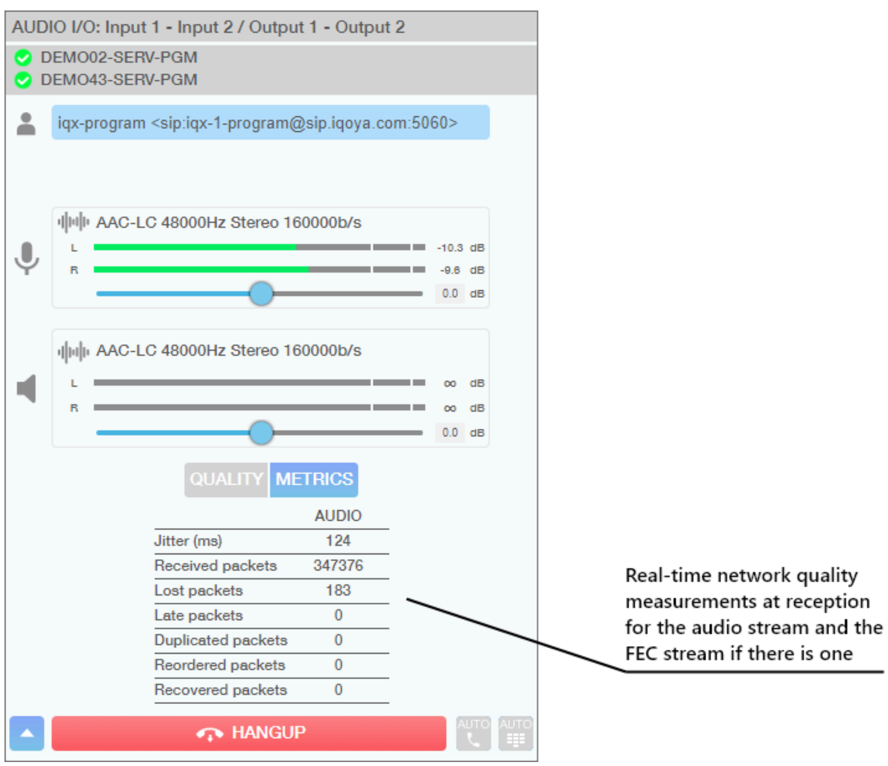
Click ACCEPT button to accept an incoming call or DECLINE button to reject it:



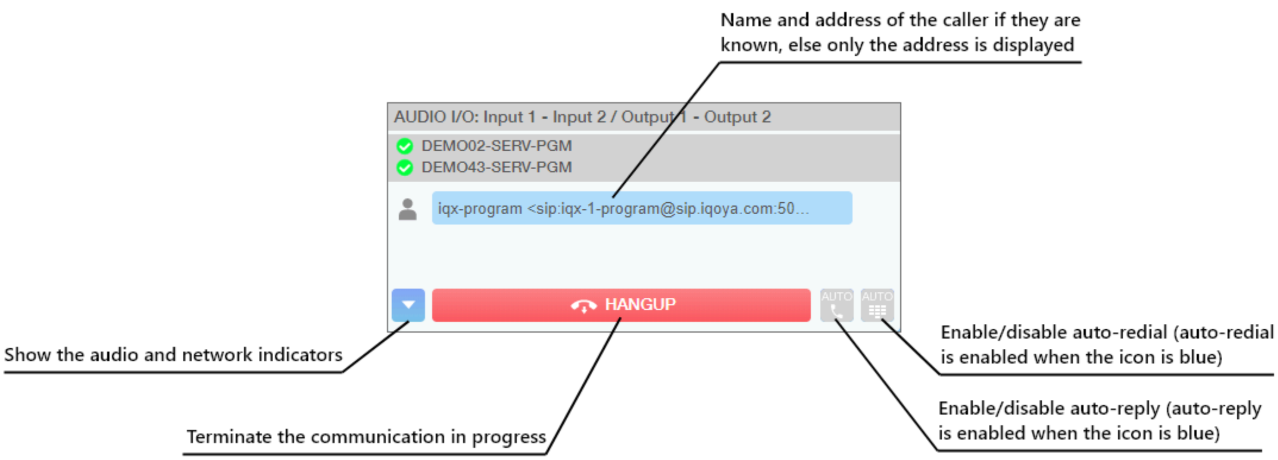
9.2.1.4 Call window when a communication is in progress

Call window (expanded version with network quality selector in QUALITY position):

Call window (expanded version with network quality selector in METRICS position):



Call window (collapsed version):



9.2.1.5 Hang up a call

Click HANGUP button to terminate the communication.

When auto-redial is activated on the caller's side, only the caller can terminate the communication. If the callee hangs up, the communication is automatically re-established by the caller device.

10 AES67 PARAMETERS CONFIGURATION

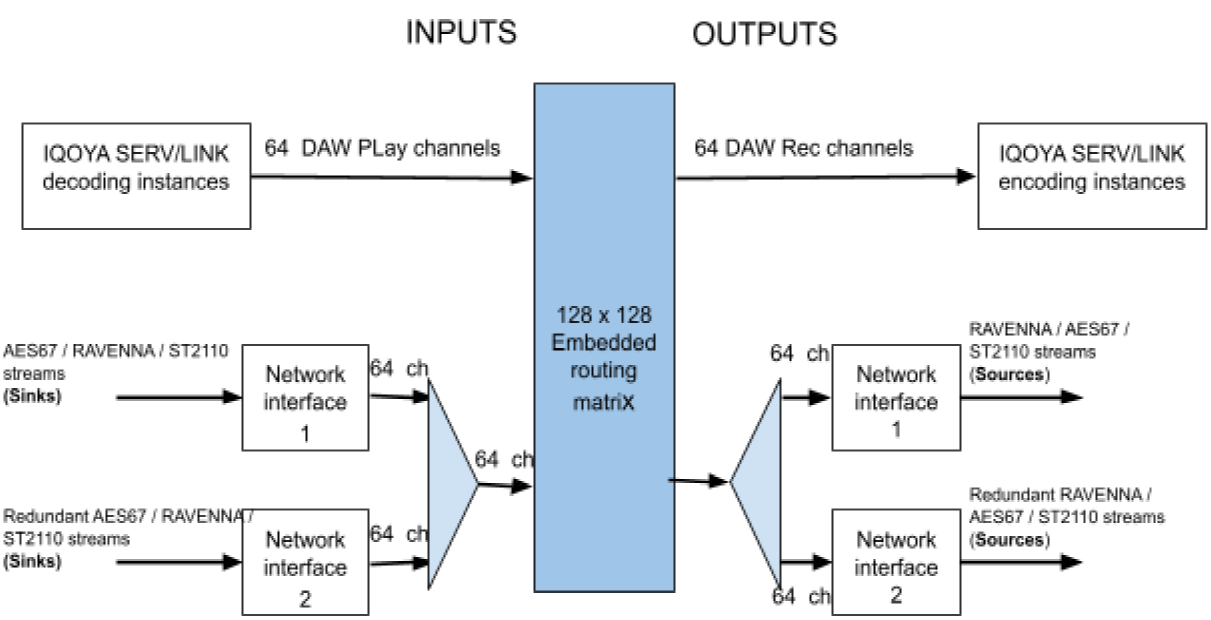
This section only applies to SERV/LINK-AES67.
This section refers to IQOYA SERV/LINK units sold as of 2025, and equipped with the following AES67 hardware modules:

- IQOYA SERV/LINK 88-AES67 - Product code=VB2274A0501
  One AES67 module.
  Supports up to 64 I/O channels
  Module name: SERV/LINK-XXXXX-AES67-1-64
  XXXXX being the serial number.
- IQOYA SERV/LINK 7272-AES67 - Product code= VB2274A0601
  Two AES67 modules.
  Supports up to 128 I/O channels
  Module names: SERV/LINK-XXXXX-AES67-1-64 and SERVLINK-XXXXX-AES67-65-128
  XXXXX being the serial number.

These modules are detected from the same AES67 network via the Eth ports “AoIP Pri” and “AoIP Sec” on the back panel. Eth port AoIP Sec is to be used for redundancy (ST-2022-7).

For older SERV/LINK AES67 versions, please refer to APPENDIX E.

10.1 Embedded routing matrix of an AES67 module



The inputs of the on-board routing matrix are:

- 64 playback channels coming from the software playback audio devices used by IQOYA SERV/LINK decoding instances.
- 64 channels coming from the network interface 1 (AoIP Pri) (or interfaces 1 & 2 - AoIP Pri & AoIP Sec - when they are configured in redundant mode). The audio channels are extracted from the incoming AES67 IP streams which are named “Sinks”.

The outputs of the routing matrix are:

- 64 channels assigned to the software recording audio devices used by IQOYA SERV/LINK encoding instances.
- 64 channels assigned to the network interface 1 (AoIP Pri) (or network interfaces 1 & 2 - AoIP Pri & AoIP Sec - when they are configured in redundant mode). These channels are the audio sources of the AoIP streams published to the network 0. These streams are named “Sources”.

10.2 Clock

An internal AES67 module can be configured as a PTP clock follower, or as a PTP clock leader on the network.
For a SERV/LINK that supports more than 64 channels, if a module is set as PTP leader, the second module must be set as PTP follower.

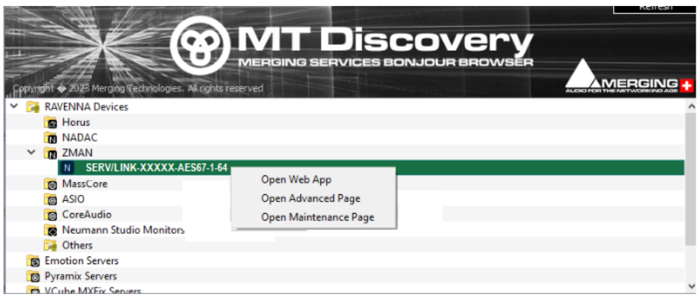
- PTP follower: the source clock is the PTP clock.
- PTP leader: : the clock source has to be internal, and the PTP clock priority has to be the highest on the network.

10.3 CONFIGURATION OF THE AoIP PARAMETERS

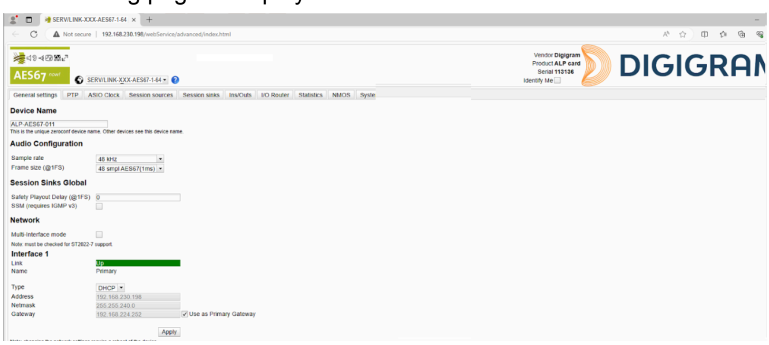
The network parameters of each AES67 module(s) can be configured through its embedded WEB server that is accessible from any of the two Eth ports.

To access the WEB interface of an AES67 module, connect the PC network interface to an AoIP Eth interface of IQOYA SERV/LINK (“AoIP Pri” for instance) which is by default set to DHCP. In case there is no DHCP service on your network, the ports are in APIPA mode (Automatic Private Internet Protocol Addressing), with the range of IP addresses 169.254.0.0/16

You can use an application such as MT Discovery to detect the module(s) and access their WEB pages:

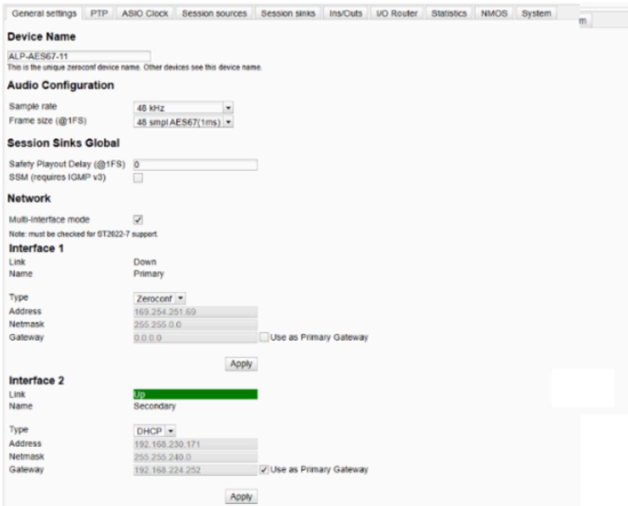


The following page is displayed.



The settings are accessible via different tabs:
General settings, PTP, ASIO clock, Session sources, Session sinks, Ins/Outs, I/O router, Statistics, NMOS, System

10.3.1 General settings



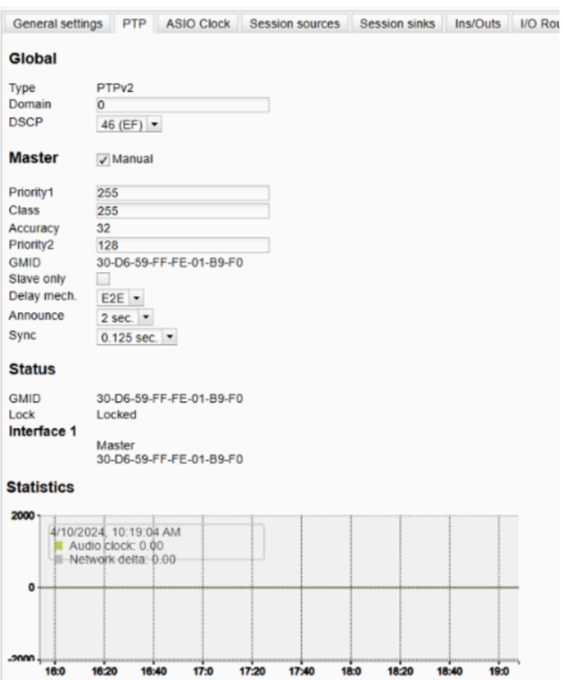
Parameter	Description
Device Name	This is a unique device name. Other devices on the network see this device name. We recommend not exceeding 32 characters. Default device name is SERV/LINK-XXXXX-AES67-1-64, and SERV/LINK-XXXXX-AES67-65-128.
Audio Configuration	
Sample Rate	Non modifiable. Reflects the current sample rate (44100-48000) that is set from IQOYA SERV/LINK.
Frame size (@1Fs)	Current frame size (64 smpl , 48 smpl AES67 (1ms) - 32 smpl -16 smpl - 12 smpl AES67 (0.25ms) - 6 smpl AES67 (0.125ms)). Note that available frame size values may not be available or differ according to the device and/or firmware. ⚠ When generating an AES67 stream to a Dante device configured in AES67 compatibility mode, the value “48 smpl AES67 (1ms)” must be selected.
Session Sinks Global	
Safety Playout Delay (@1Fs)	Additional playout delay. The value is described at 1Fs (44.1-48 kHz) in samples.
SSM (requires IGMP v3)	Source-Specific Multicast. If you activate this option, make sure your network switch supports and is configured for IGMP V3.
Network	
Multi-Interface mode	Activate / deactivate ST2022-7 mode
Interface 1-2 (Interface 2 is displayed if “Multi-interface mode” has previously been enabled and applied)	
Link	Status

Name	Information only - can't be modified
Type	Defines the type of IP V4 address used (Zeroconf - DHCP- Static)
Address	To be set only if Type is set to Static. Otherwise, these fields display the assigned values.
NetMask	
Gateway	

Reboot is required to apply the changes

.

10.3.2 PTP settings

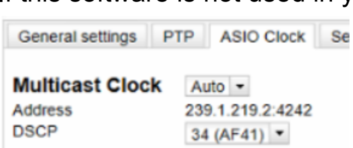


Parameter	Description
Global	
PTP Domain	Allows to define a specific PTP domain, usually when several PTP masters are required in the same network (default value is 0).
DSCP	PTP DSCP. Values: 46-EF : PTP AES67 48-CS6: PTP RAVENNA 56-CS7
Master	

Manual	This checkbox must be checked to modify the PTP settings
Priority 1	Main priority value (1= Highest priority; 255=lowest priority)
Class	Device class. This value should not be modified
Accuracy	Non modifiable
Priority 2	Only used if the other parameters do not allow for the election of a PTP master.
GMID	Non modifiable. Current GrandMasterID (PTP Master)
Slave only	Forces the AES67 module to always be PTP slave
Delay Mech	PTP Profile related - E2E or P2P
Announce	PTP announcement interval (1 - 2 - 4 - 8 -16 seconds)
Sync	0.0625 - 0.125 - 0.25 - 0.5 seconds
Status	
GMID	Non modifiable. Current GrandMasterID (PTP Master)
Lock	Shows if the device is locked to PTP (Locked -Locking - Unlocked)
Interfaces 1 & 2	
PTP status	Master : Device is Master Slave : Device is Slave Listening : The interface is not used for synchronization
Statistics	The graph is only active in slave mode. Green curve: shows the audio clock delay vs the current Master PTP clock. Grey curve: shows the network delay vs the current Master PTP clock, before the device synchronisation algorithm.

10.3.3 ASIO Clock

This tab allows setting the parameters of the clock that can be generated for MERGING AUDIO DEVICE software synchronization.
If this software is not used in your system, just disable this feature (Multicast clock = Off).

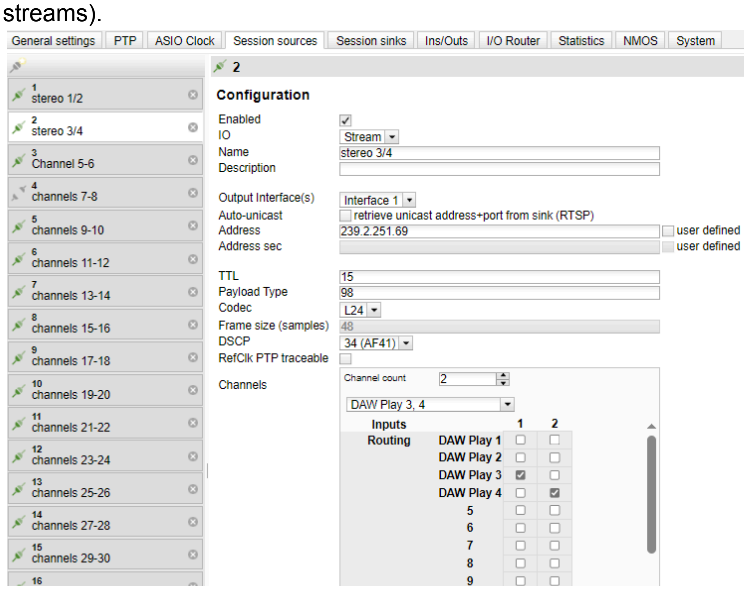


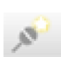
Parameter	Description

Global	
Multicast Clock	Off, On, Auto Allows to enable/disable the multicast ASIO clock. Auto activates the ASIO clock only on the master PTP device.
Address	Multicast address of the ASIO Multicast Clock
DSCP	Allows setting the ASIO Multicast Clock DSCP value: 34 (Default), 46, 48, 56. When using a switch, make sure that the ASIO Clock DSCP has a high value in the switch DSCP to Queue table

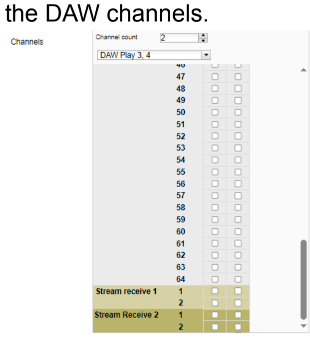
10.3.4 Session sources

This page allows for the declaration of session source instances on the SERV/LINK AES67 module (transmitters that generate the IP streams).



Click on the button  to create a new session.

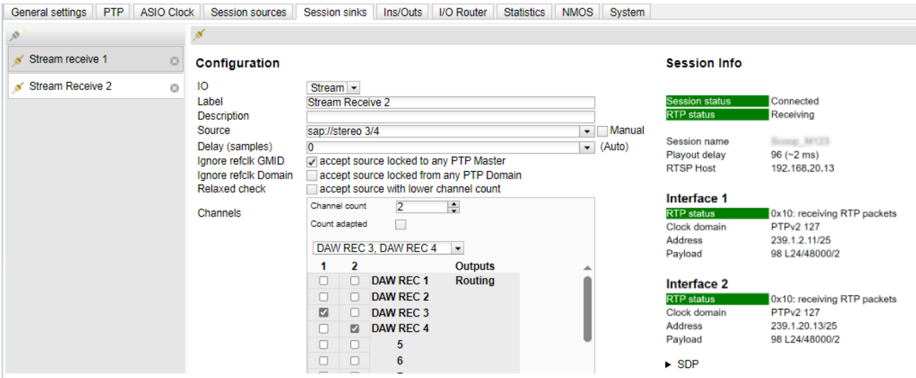
Parameter	Description
Enable	Enables the selected source (active by default)
IO	Non modifiable (Stream)
Name	Source name (63 characters max).
Output interface(s)	Select the network interface (1, 2, or 1&2 in case of ST_2022-7) used for streaming this session.
Auto-unicast	Automatically retrieves the IP address of the sink (listener) for a unicast connection

Address / User defined	Check “user defined” to manually enter or modify the multicast IP address of the generated stream. When generating AES67 streams to Dante devices configured in AES67 compatibility mode, the multicast addresses must be 236.69.xxx.xxx, otherwise routing this stream from Dante Controller is not possible.
Address sec / User defined	Not accessible if only one Eth interface is enabled. Check “user defined” to manually enter or modify the multicast IP address of the redundant stream (ST-2022-7).
TTL	Time to Live (also called Hop Limit) - this value should not be modified
Payload type	RTP Payload type - this value should not be modified
Codec	L24 - L16 - DSD64 - DSD64_32 - DSD128 - DSD128_32 - DSD256 (bit rate). Note that those values are sampling rate dependent;
Frame size (samples)	Frame size of the current source. Reflects the frame size set from the General settings section.
DSCP	Ip audio stream DSCP value (34: RTP AES67 / 46: RTP Ravenna / 26 / 0)
RefClk PTP traceable	This feature is useful when you want to connect a stream through the Internet (e.g. with two PTP Masters (GPS) at each location). It allows making connections with devices locked to different traceable PTP Masters. See also “Ignore refclk GMID” - accept source locked to any PTP master on the Session Sinks section below.
Channels	
Channels: Channel count	Number of channels in the stream. The “Channel count” drop down menu allows selecting the number of contiguous channels from the 64 available channels.
Routing table	The routing table allows for the selection of the content of each channel of the stream, among the 128 possible output channels of the AES67 module internal matrix. The first 64 displayed channels are the output channels of SERV/LINK decoding instances. The following channels are the channels received from declared sinks. In the picture above, the channels to be streamed are channels 3 & 4 (named DAW Play 3 & DAW Play 4) from IQOYA SERV/LINK Note that the channels can be selected from the sinks receivers, listed after the DAW channels. 

The URL of the SDP of this session is	Allows saving the Session Description into a file (useful for specific third party devices if manual SDP has to be provided). Note : Devices in 2022-7 mode will produce a 2022-7 SDP, containing both interfaces session parameters. If you load such SDP in a non-2022-7 device, only the session parameters from interface1 will be used.

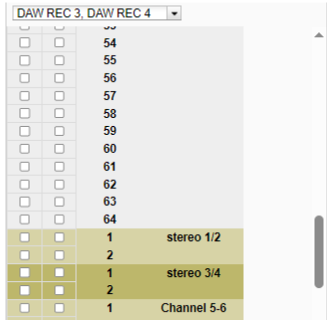
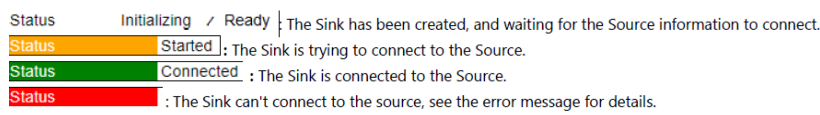
10.3.5 Session sinks

This page allows for the declaration of session sinks (receiver instances) on the selected SERV/LINK AES67 module.



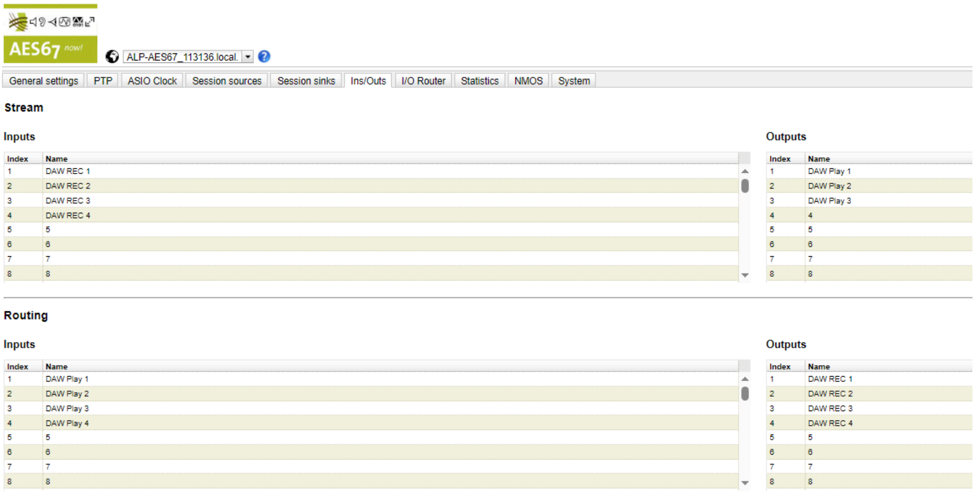
Click on the button  to create a new session.

Parameter	Description
IO	Non modifiable (Stream)
Label	Sink name (=receiver name)
Source	Drop down menu to select a source (both sap and bonjour advertised sources are listed).
Source: Manual	Allows to manually enter a SDP
Delay (samples)	Playback delay. 0 is an automatic delay (works with devices based on the AES67 technology from Merging Technologies). If set to 0, and when using devices that are not based on the Merging Technologies technology, the playout delay is calculated based on the value a=framecount in the SDP. In any case, the frames must be time aligned; all devices must run an integer number of frames from time zero (epoch).
Ignore refclk MGID: - accept source locked to any PTP master	This feature is useful when you want to connect a stream through the Internet (e.g.with two PTP Masters (GPS) at each location), it allows making connections with devices locked to different traceable PTP Masters. See also RefClk PTP traceable on the Session Sources page.
Ignore refclk MGID: - accept source locked from any PTP master	This feature is useful when you want to connect a stream through the Internet, it allows for connections with devices locked to different PTP domains.
Relaxed check	Allow connection to a source that has less channels
Channels: Channel count	Number of channels in the stream. The associated drop down menu allows selecting the "Channel count" contiguous channels from the 64 available channels. Received channels can be routed towards the DAW recording channels, or towards the declared source streams listed after the DAW rec devices:

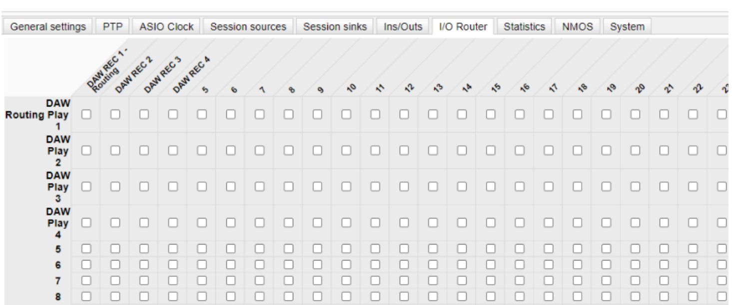
	
Channels: Count adapted	To be used if the number of channels does not match
Session Info	
Session status	
Interface 1-2	
RTP Status	Connection status. The displayed value is the sum of the following error value: 0x10 receiving RTP packets 0x01: wrong RTP sequence id 0x02: wrong RTP SSRC 0x04: wrong RTP payload type 0x08: wrong RTP SAC 0x20 stream has been muted 0x40: Horus implementation - an incoming stream is muted 0x80: ST2022-7 mode only - both interfaces are not receiving RTP packets properly → muted Exemple. : Stream muted (0x20) and Wrong payload (4) will be displayed as 0x24
Clock domain	PTP clock type and domain
Address	Multicast IP address of the selected sink
Payload	Payload / Codec / Sampling Rate / Number of channels
SDP	Displays the detailed SDP information on the current stream.

10.3.6 Ins / Outs

This page allows you to change the name of the Inputs and / or Outputs.



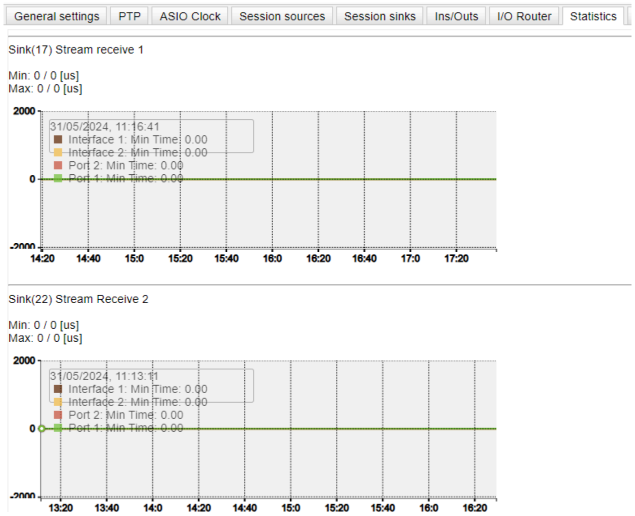
10.3.7 I/O Router



This page allows configuring the routing from the input channels of the internal matrix to its output channels.
An input channel can be routed to several output channels.
But multiple input channels **cannot** be routed to an output channel (no mixing on the output channel).
The input channels of the matrix are listed vertically on the left. The scroll bar on the right of the page allows scrolling down/up.
The output channels of the mixer are listed horizontally. The scroll bar at the bottom of the page allows scrolling right/left.
To access the input channels coming from the received streams, scroll first down until you see the horizontal scroll bar.

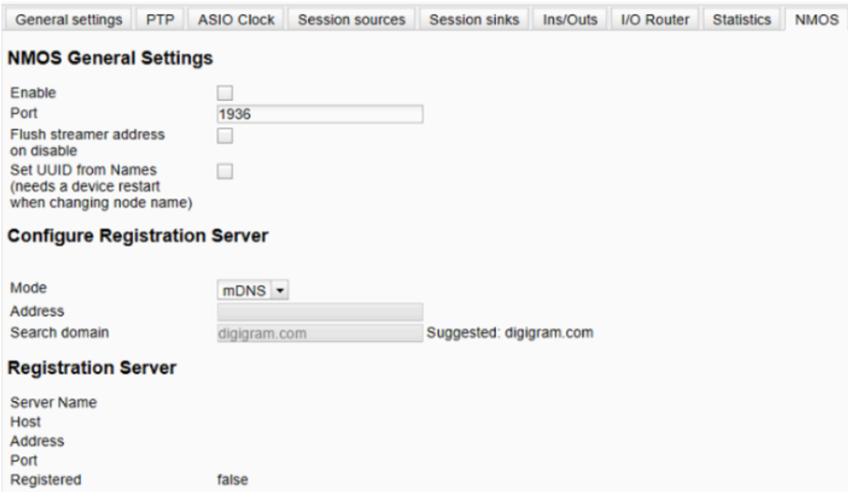
10.3.8 Statistics

This page gives statistics about the sinks (received IP streams).



10.3.9 NMOS

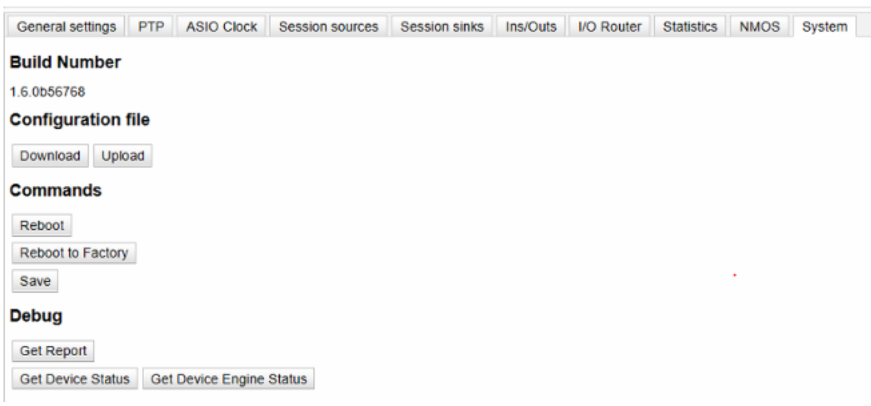
This page allows configuring the NMOS settings for discovery/registration (NMOS IS=04), and for device connection management (NMOS IS=05).



Parameter	Description
	NMOS General Settings
Enable	Enable or disable the NMOS client, following the Port and Registration server configuration. NMOS must be disabled to modify the registration server settings.
Port	ALP-AES67 NMOS communication port
Flush streamer address on disable	Clears the streamer address when the stream is disabled. This option might be useful with some NMOS servers
	Configure Registration Server
Mode	Registration Server communication modes : ● mDNS(Bonjour): Automatic communication to a Registration Server, using Bonjour service (The server needs to advertise with Bonjour Service) ● Search domain: A search domain needs to be supplied. The search domain associated with the primary network interface is shown as Suggested. ● Static Address: The registration server is found using the address given in the “Address” field. This is currently the preferred and most common way to establish communication with a NMOS Registration Server. (Sony NMOS CPP, RiedelNMOS,...)
Address	When Mode is set to Static Address, this allows the user to enter the server IP address and port (<ip address>:<port number>). In other modes, this field only displays information when the communication is established.
Search Domain	When Mode is set to Search Domain, this allows the user to enter a domain (Unicast DNS).

	In other modes, this field only displays information when the communication is established.
Registration Server (This section provides status information about the Registration Server)	
Server Name / Host	Provided by the Server (with DNS or mDNS). Note that this field may remain empty when Mode is set to static
Address / Port	Server IP address and Port
Registered	True if a registration server can be reached

10.3.10 System



Parameter	Description
Build number	Current firmware version.
Configuration file	
Download/upload	Save/Load a configuration file (preset)
Commands	
Reboot	Restart the device
Reboot to factory	Restore all settings to factory default, and restart the device
Save	save the current configuration.
Debug	
Get report	Generates a debug report, and saves it on the local computer. If the report is not saved automatically, make sure your web browser did not block the download
Get device status	Displays the device status (SysLog)
Get device engine status	Displays the device engine

11 DANTE PARAMETERS CONFIGURATION

This section only applies to SERV/LINK-DANTE.

11.1 Connecting to a Network

In order to use your SERV/LINK DANTE, you will need to set up an Ethernet network connecting:

- The SERV/LINK DANTE unit. Connection is made through the Eth port “AoIP Pri” on the rear panel.
- The computer running Dante Controller (if it’s not the same computer). Connection is made through the computer network adapter.
- Any other Dante-enabled audio device you may have

Gigabit Ethernet Support

Your SERV/LINK DANTE is designed to perform with Gigabit Ethernet networks. Connecting it to 100 Mbit/s Ethernet devices is not supported. Make sure to connect the Dante ports of SERV/LINK DANTEto a Gigabit Ethernet switch.

Choosing a Gigabit Ethernet Network Switch

You get the best performance out of your Dante network even if you use standard Gigabit Ethernet network switches. Dante uses standard Ethernet and IP Quality of Service (QoS) to ensure its high-quality synchronization is not affected, even on loaded networks. Make sure that you choose network switches which have the following features.

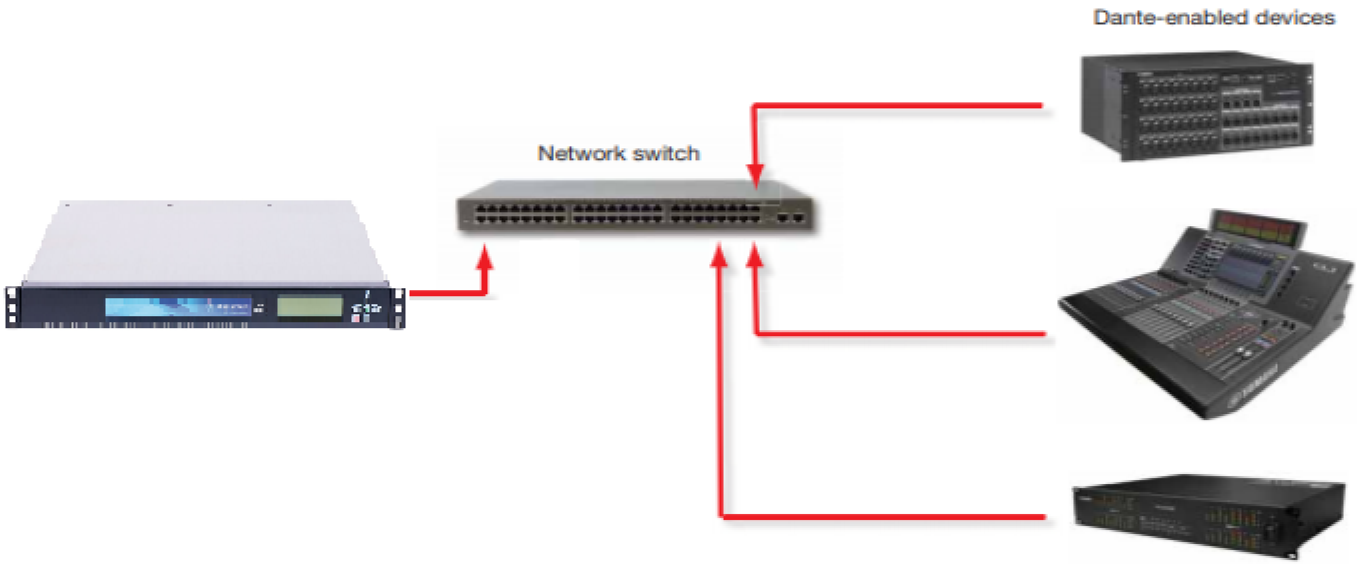
- DSCP-based QoS with four queues and strict priority queuing
- ACL filtering
- Fiber-optic cable support including SFP pluggable modules (if you need to run long cables)
- Managed network switches which allow you to manage the network switches and monitor your network If you have chosen a network switch that has been used before, you may need to check its settings again. For more information about choosing network switches, please visit the Support section of the Audinate website (www.audinate.com).

Choosing Ethernet Cabling

Dante uses completely standard Ethernet and IP, so it also uses standard Ethernet cabling (STP), including Cat5e or higher and fiber-optic. Make sure your Ethernet cables and ports are in good condition. Remember that Ethernet cables which are Cat5e or higher have a maximum length of 100 meters* at speeds of 1Gbps (the cable length limit depend on the cable type). If you require longer distances, you can use fiber-optic cables.

Network Configuration

The Dante Controller application from Audinate allows setting up the audio routing over a Dante network.
Please refer to the Audinate’s WEB site for information about the Dante Controller, and for downloading it.



Basic network configuration

If your network switch has a mix of Gigabit and 100Mbps ports, make sure you connect SERV/LINK Dante's port(s), and if possible all devices, to the Gigabit ports.

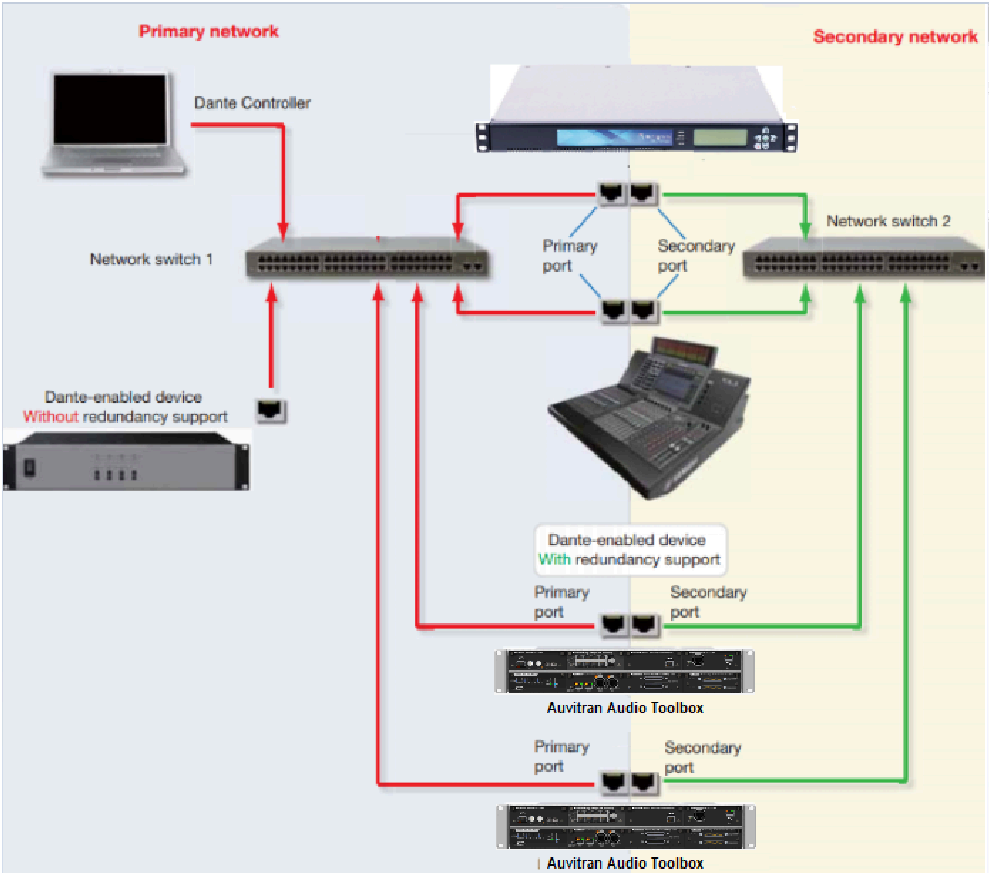
- Make sure all computers are set to automatically configure their IP address.
- Power on the network switch
- Connect your Ethernet cables from each device to the Gigabit ports of the network switch. You may need to reboot the computers if they have active previous IP network configurations. All devices will automatically be assigned IP network configurations.
- Use the Primary Ethernet port of the SRV/LINK (ETH0/PRI) to connect it to the Dante network.

Connection via a Redundant Network

The Dante redundancy works by using two completely independent and separate networks, the Primary Network and the Secondary Network. To set up and use Dante Redundancy, connect your redundant Dante-enabled device using duplicate network switches and Ethernet cables as shown below. Connect the following to the Primary Network only:

- Any computers running Dante Controller
- Any non-redundant Dante-enabled devices

All Dante-enabled devices that support redundancy should be connected to both the Primary and Secondary networks.
The primary and secondary networks MUST NOT be interconnected at any point.



1. Make sure all computers are set to automatically configure their IP address.
2. Power on the network switch.
3. Connect your Primary Ethernet cables from each device to the Gigabit ports of the Primary network switch.
4. Connect your Secondary Ethernet cables from each device that supports redundancy to the Gigabit ports of the Secondary network network switch.

You may need to reboot the computers if they have active previous IP network configurations. All devices will automatically be assigned IP network configurations.

Unsupported Dante Network

Connecting Primary and Secondary redundant networks to the same network switch

When using Dante redundancy with any Dante-enabled device, two separate networks must be used. You CANNOT connect any secondary network connections to a network switch used for the primary network, or any primary network connections to a network switch used for the secondary network.

Dante Audio Data over Wireless Networks

Wireless Ethernet networks should not be used to carry Dante audio data, and Dante Controller installed on a PC or Mac will not allow selection of Wireless Ethernet interface or any other non-standard wired Ethernet interface. You should NOT install any wireless components in your Dante network.

Use of 100Mbps Network interface

The use of 100 Mbps Ethernet device with the SERV/LINK-DANTE is **NOT SUPPORTED**. You must connect the SERV/LINK-DANTE to Gigabit interfaces: • Gigabit Ethernet network switch • Gigabit network interface on a PC or Mac

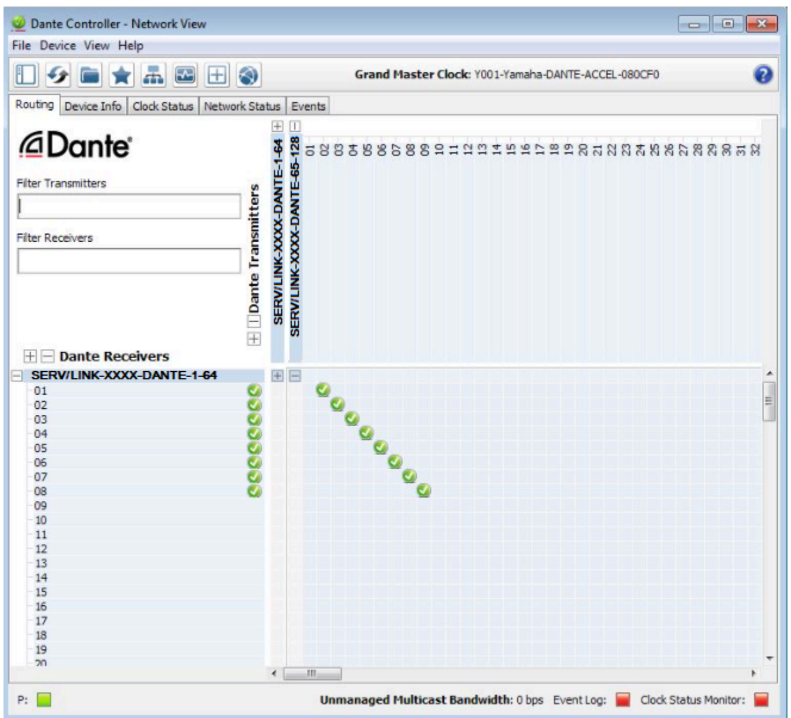
11.2 Using Dante Controller

Please download the Dante Controller application from Audinate's WEB site, and refer

When you open the Dante Controller it displays the Network View, which shows all the Dante-enabled devices on the network. Initially it will display devices, but not channels within devices. Devices with transmitter (Tx) channels will be displayed along the top row and devices with receiver (Rx) channels will be displayed in the left hand column.
Channels can be viewed by clicking on the + symbol next to Dante Transmitters or Dante Receivers, or the + symbol next to a particular Dante device.

The Dante Controller User Guide contains detailed information about all aspects of using the Dante Controller. This section is only a brief overview – you will need to refer to the Dante Controller User Guide to be able to use all features of the Dante Controller

Dante Controller Network View



Setting Up Audio Routing

The Dante Controller can be used to configure audio routing between Dante devices.
In the expanded view, wherever there is a blue cell at the intersection of a transmitting channel column and a receiving channel row, it is possible to establish an audio routing between them.
Using the Network View, click on the cell at the intersection of a transmit and receive channel to configure an audio routing from the transmitting channel to the receiving channel. A green icon will appear when the routing is established. Click again to remove the routing.
NOTE
Ctrl+click on the cell at the intersection of the devices to configure the entire routing.

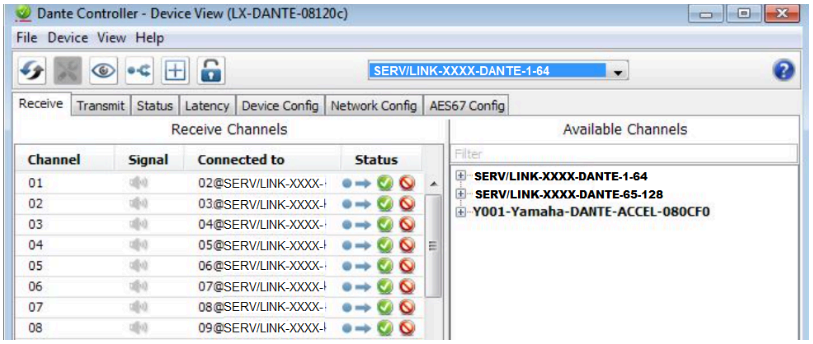
Configuring SERV/LINK Dante interface

SERV/LINK DANTE features one or two Dante modules, each module supporting 64 channels.
To configure the Dante parameters of your SERV/LINK-DANTE, including its name, sample rate, and latency:
1. Start the Dante Controller.
2. Select the device. Either double-click on the device name from the Dante Controller Network View (either in the transmitter row, or the Receiver column), or from the Network View menu, select Device –> Device View (or press Ctrl+D). This opens a Device View window for the selected device.

Device View Window

The Device View has five tabs: Receive, Transmit, Status, Device Config and Network Config. The Receive tab for a configured

SERV/LINK DANTE interface is shown below.



Changing the Device Name

It is possible to rename the SERV/LINK-DANTE interface. To do this open Device View in the Dante Controller, select the interface you wish to modify in the Device View, and change the Device Name in the Device Config tab. If you rename a device, you must re-establish any existing audio routing to and from the re-named device using its new name. Please refer to the Dante Controller User Guide for more information on renaming devices.

Changing Channel Labels

To change channel labels in Dante Controller:

- Open the Device View for the relevant device.
- Click the Receive or Transmit tab (depending on which channel labels you want to edit).
- Double-click the channel label.
- Enter a new value. Tx (transmit) labels must be specific to that device.

Changing the Sample Rate

Audio routing can only be set up between devices that are operating at the same sample rate. It is possible to set the sample rate to any one of the following values: 44.1, 48, 88.2, 96, 176.4, and 192 kHz. However, note that SERV/LINK supports 44.1 and 48 kHz. To do this, open Device View in the Dante Controller, select the SERV/LINK DANTE interface you wish to modify in the Device View, and change the sample rate in the Device Config tab. Pull-up/pull-down (+4.1667, +0.1, - 0.1, and -4.0 %) is also supported. Please refer to the Dante Controller User Guide for more information on changing sample rates.

Setting the Latency

To adjust the latency setting, open a Device View for the selected device in Dante Controller, and select the Device Config tab. This allows several device settings to be viewed and modified. It shows the current receive latency setting and allows the user to change the operating receive latency for the selected device.

The allowed values are:

- 0.15ms (150 microseconds) - a suitable setting for a network containing a single network switch
- 0.25ms - a suitable setting for a network containing 3 network switches
- 0.5ms - a suitable setting for a network containing where the signal path may encompass up to five network switches
- 1.0ms - a suitable setting for a network containing where the signal path may encompass up to ten network switches
- 5.0ms – a safe value for a network of almost any conceivable size

NOTE

Even if you set values suited for your network, various factors may cause some noises. If a value other than the current setting is selected a message will be displayed warning the user that the effect of changing the latency is that any existing audio routing to the device will be temporarily suspended, resulting in some loss of audio data. If you wish to make the change, select “Yes”; otherwise select “No”.

11.3 AES67 compatibility mode

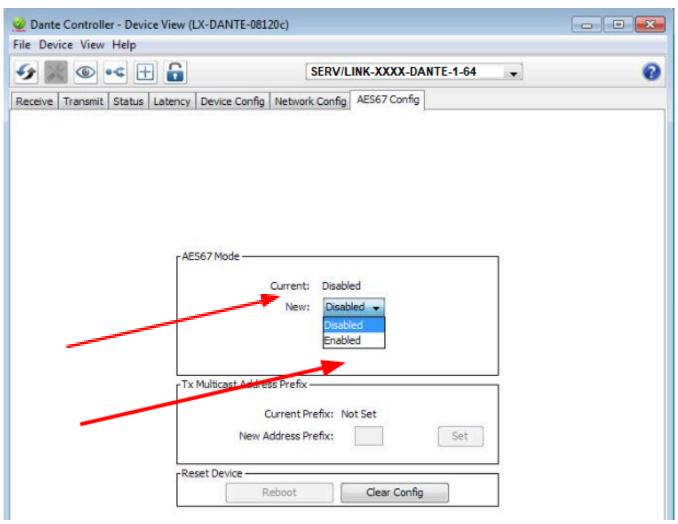
AES67 streams can be exchanged between the SERV/LINK-DANTE and AES67 devices.

The parameters of the AES67 compatible mode are:

- SAP (Session Announcement Protocol) must be supported.
  This is one of four device discovery methods referenced in the AES67 standard. Check if the transmitting non-Dante AES67 device supports SAP. Otherwise Dante Controller cannot discover the audio flows coming from the device. Dante devices support SAP if their AES67 mode is enabled. Even if the non-Dante AES67 device does not directly support SAP, the vendor may have a software conversion tool available, providing wider compatibility.
- Multicast flow only (unicast is not supported yet).
- Up to 8 channels per flow.
- Multicast IP addresses must be in the range of 239.69.0.0 - 239.69.255.255/16.
  In addition, the same network address range must be specified for other AES67 devices transmitting to Dante devices.
- 48 kHz sampling rate only.
  The Dante audio flows from/to the device will also be limited to 48 kHz if the device is engaged in AES67 mode.
- 24 bit (L24) only for transmission and reception.
- 2 milliseconds fixed latency for reception.
- Primary network only .
- 100 Mbps or slower connections are not supported.

It is necessary to enable the AES67 mode of the AES67 interface, through the Dante controller.

Select the SERV/LINK-DANTE interface, and go to the “AES67 Config” tab.



Apply “Clear Config” and reboot the Dante interface when enabling or disabling AES67 mode.

Common sense precautions should also avoid audio mishandling:

- If you modify the device labels, reboot the Dante interface after label modifications to update the AES67 signal name announcement
- Do not change the bit depth (default 24 bit) from Dante Controller or the console after enabling the AES67 mode.
- AES67 is Multicast only : create AES67 multicast transmission streams in Device View as usual, but tick “AES67 flow” first.
- In the Routing tab on Dante Controller, the non-Dante AES67 devices will appear in blue as the transmitters.

Clocking considerations

- AES67 mode on a Dante device enables both IEEE 1588 Precision Time Protocol (PTP) v1 and v2. A single clock domain must be created across both PTP v1 and v2 devices:
- Standard Dante devices support PTP v1 only
- AES67-enabled Dante devices support PTP v1 and PTP v2
- AES67 devices support PTP v2 only
- PTP v1 and v2 are not inter-compatible. One AES67-enabled Dante device will act as the boundary clock between PTP v1 and v2, bridging the two clock domains

Enabling the Dante interface as Master for both Dante and AES67.

- Enable the Dante interface “Preferred Master” status
- Disable “Preferred Master” for all Dante devices that have AES67 disabled.
- Disable “Sync to External” for all devices.
- Assign a PTPv2 priority level of between 128 and 255 for all non-Dante devices

If another AES67 device (Grandmaster Clock, or another AES67 device) is the Master ,

- Make sure the PTP v2 Master has a priority of between 1 and 100, and is using the “Media Profile” clock settings.
- Disable “Preferred Master” and “Sync To External” for all Dante devices.
- One AES67-enabled Dante device will automatically be selected as the Boundary clock, becoming the Dante Master.
- Make sure the Master Clock device is set to use the “Media profile” (not the “Default profile” because Dante devices do not support the Default profile.

12 High Availability (1+1 redundancy)

IQOYA SERV/LINK supports 1+1 redundancy, meaning that two similar units can run together with one being a “hot” backup of the other.

This redundancy is available through a software option named “High Availability” which has to be installed on the two units working in redundant mode.

12.1 What is High Availability (HA)?

High Availability is a network configuration ensuring that planned and unplanned outages do not interrupt significantly the IP audio services. In a HA configuration, the IQOYA SERV/LINKs are deployed in pairs called HA pairs. The devices of a HA pair share a floating IP address (virtual IP address = VIP). Thanks to this VIP, the HA pair is seen as a single device. The VIP is the IP address to be used to reach the HA pair.

Within the HA pair, one IQOYA SERV/LINK operates in the **active** mode and the other IQOYA SERV/LINK operates in the **standby** mode:

- **Active**: The active member of the HA pair is the system actively processing the audio signals. The active member continuously monitors itself for internal processes health. If the active member detects a condition that can interrupt the IP audio services, it hands over its role as the active member of the HA pair to the standby member.
- **Standby**: The standby member of the HA pair is the backup system. The standby member audio configuration is fully synchronized with the active member configuration, but it does not actively process the audio signals. Synchronization of the configuration is based on **the FTP protocol, which must be activated on both units**.
  The standby member monitors the status of the active member and it can assume the active role without the active system having to instruct it to do so. When the standby system assumes the active role, it can notify a network management system through an SNMP trap.

The communication between the active member and standby member (life line, transfer of the configuration via FTP) goes through an IP connection, as explained here-below.
In the event of an outage of the active member, the IP audio services are interrupted for less than 5 seconds.

12.2 Installation of a HA pair of IQOYA SERV/LINK

⚠ **The two IQOYA SERV/LINKs must be identical : same model, same type and same number of I/Os.**

One network interface of each IQOYA SERV/LINK unit must be dedicated to the HA service (devices advertise their current state and health to one another - life line). The network interface can be a dedicated Eth interface, or a logical interface belonging to a VLAN. We however recommend using a direct network link to connect these interfaces, in order to avoid going through switch(s) which could be points of failure.
This network interface is called the HA network interface.
Another network interface must be dedicated to all the other IP services and in particular the IP audio services. It is called the audio network interface. The floating virtual IP address will be shared by these audio network interfaces for receiving streams.

To install a HA pair of IQOYA SERV/LINK, follow these steps:

1. Choose and set the IP configuration of the HA network interface on both devices. The two HA interfaces must be in the same subnetwork. We recommend a direct link between them, and to set them at 100 Mbits/s.
2. Choose and set the IP configuration of the audio network interface on both devices. The two audio interfaces must be in the same subnetwork.

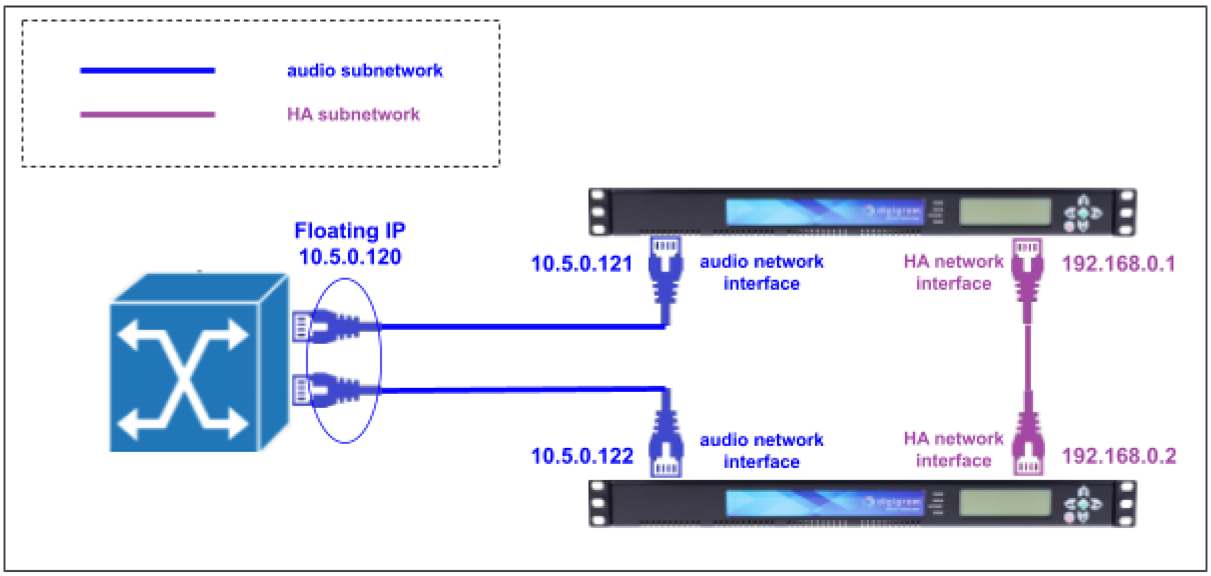
⚠ **The subnetwork of the HA interfaces and the subnetwork of the audio interfaces must be separated subnetworks.**

3. Choose a free IP address in the same subnetwork as the audio network interfaces. This IP address will be the floating IP Address (VIP) shared by the two audio network interfaces.

4. Establish the physical network connections.

   ⚠ **The connection between the HA network interfaces must be very reliable. For better reliability, it is strongly recommended to establish the physical connection between the HA network interfaces using only a network cable (as in the example below). However the physical connection between the HA network interfaces can be established via network switches provided that these switches are themselves part of a network HA system (like STP for instance).**
   It is also strongly recommended to force the HA network link at 100M Full Duplex.

Here is an example of an installation:



12.2.1 In what kind of application the HA service is to be used?

HA service is to be used anytime a 1+1 redundancy is required for encoding, for decoding, and for transcoding.
The floating IP address (VIP) is useful for decoding and transcoding applications, as it allows the two units to receive unicast streams. It is also useful for accessing the WEB pages of the active unit.

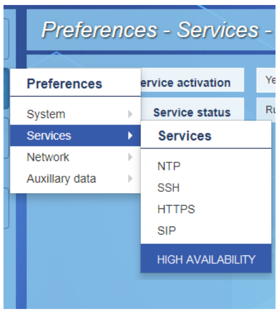
12.2.2 Configuration of the HA service

The configuration of the HA service is done from one and only one of the two IQOYA SERV/LINK to be paired. The device from which you configure the HA service will get the active role.

⚠ **The device on which you configure the HA service will push its audio configuration to the paired device, overwriting the current audio configuration of the paired device. This uses the FTP protocol, which must be enabled and configured on both units.**

To configure the HA service follow these steps:

- Connect to one of the IQOYA SERV/LINK. The IQOYA SERV/LINK you will connect to will get the active role once the configuration is complete.
- Go to the HA service configuration page. The configuration page of the HA service is accessible from the menu: Preferences→Services→High Availability



The configuration page of the High Availability service looks like this:

Set "*Service activation*" to "*Yes*" and enter the other parameters. The parameters to be filled in are described in the table below:

Parameter	**Description**	**Possible values**	**Default value**
Service activation	Activate/deactivate the HA service	Yes/No	No
IP address of the paired device	This is the IP address of the HA network interface of the device to pair with (standby unit).	Any valid unicast IP address	∅

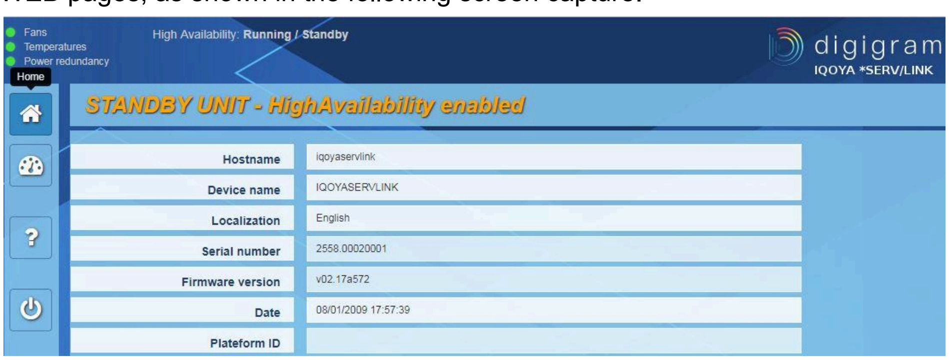
Floating IP address	This is the virtual IP address thanks to which the pair of devices is seen from the audio network perspective as a single device.	Any valid unicast IP address which is not used by another device on the network.	∅
ID	Value which identifies a HA service used by a pair of devices.. In case several pairs of devices run with the HA service, each HA service must be configured with a unique ID.	From 1 to 256	No default value. Value to be entered.
Election of the active device	If equal to “Preferably active”, the device from which you configure the High Availability service will always try to get the active role. This means that if it fails, the other device becomes Active, and when it is connected again it becomes Active and the other device goes back to Standby mode. If equal to “No device has priority”, the role of the devices will change only in case of an outage of the active device.	No device has priority/Preferably active	No device has priority
SNMP	If set to Yes, all the above variables can be read from an SNMP GET command, and trap is enabled. Make sure the SNMP service is correctly configured.	Yes - No	No

The following parameter is a status

Service status	Gives the current status of the HA service (read only parameter).	Running/Stopped	N/A

⚠ **Once the configuration is complete and the HA service is active, it is recommended to use the floating IP address to connect to the configuration Web GUI of the HA pair. If you connect to the configuration Web GUI of the standby device (using the IP address of its audio network interface) you will get the message “This is the standby device. Please connect to the active device to change the configuration”.**

When a device is in standby mode, its configuration cannot be changed, and some menus are no longer accessible from its WEB pages, as shown in the following screen capture.



When a change of configuration is applied to the Active unit, this configuration is uploaded onto the standby unit (through FTP). The standby unit loads this configuration when it has to become active.

When the HA service is running, the IP addresses of the involved SERV/LINK cannot be changed from the front panel display.

What happens if the HA service is stopped?
The Virtual IP address is no longer reserved.
Each SERV/LINK unit becomes autonomous.

Does enabling the HA service stop the audio activity?
Audio activity is not stopped in the "preferred active" mode.
Audio activity is stopped and restarted in the "No device has priority" mode (in fact, both units are shortly set to the standby status for the election of the active device).

What happens if the network link used by the HA service fails?
The Active unit remains active.
The unit that was in standby mode, loads the configuration of the active unit, and becomes active.
As a result, the two units are active.
This is why it is recommended to dedicate physical Eth ports to the High Availability lifeline.

What to do when a unit is in fault?
The faulty unit has to be isolated from the network to diagnose the cause of the problem.
Once this is done, it is necessary to reboot the SERV/LINK, and then reconnect it to the network.

13 Specifications

CONFIGURATION

Dimensions	19", 1RU, 43.9 cm depth
Weight	7.9 kg
Power supply	2 hot swappable redundant PSU 90-264VAC Optionally, 2 hot swappable redundant PSU -48VDC
Temperature	Operating 0°C 50°C, Storage -20°C TO 70°C
Humidity	Humidity 85% non-condensing
Power consumption	Max 120W

CONNECTIVITY

WAN Ethernet ports	2 x 10/100/1000 Mbps RJ-45
AES67 Ethernet ports(*)	2 x Gbps RJ-45 (64-channel version) / 4 x Gbps RJ-45 (128-channel version)
Dante Ethernet ports(**)	2 x Gbps RJ-45 (Primary and secondary)
Analog and AES/EBU audio inputs(***)	Female XLR on breakout cables
Analog and AES/EBU audio outputs (***)	Male XLR on breakout cables
MADI IN/OUT(****)	Multimode optical fibre (to be used with 50/125-µm or 62.5/125-µm fibres).
AES/EBU Sync	Female XLR
Word Clock Sync	BNC 75 Ohms
Optional RS232 ports (optional)	8 x SubD-D 9 per breakout cable
GPIO (Optional)	16 GPIs TTL compatible on terminal block 16 relay GPOs on terminal block

(*) Only for AES67 and AES67-MADI versions
(**) Only for the Dante version
(***) Analog only for "Analog & AES/EBU" versions
(***) only for MADI and for AES67-MADI versions

ANALOG INPUTS

Only for "Analog & AES/EBU" versions

Type	Balanced
A/D converter resolution	24 bits

Maximum input level/ impedance	+24 dBu/ >10 kΩ
Adjustable input gain	from –94.5dB à +24 dB

ANALOG OUTPUTS

Only for “Analog & AES/EBU” versions

Type	Balanced
D/A converter resolution	24 bits
Maximum input level/ impedance	+24 dBu/ <100 Ω
Adjustable output gain	from –94.5dB à +24 dB

AES/EBU INPUTS

Only for “Analog & AES/EBU” versions and for “AES/EBU” versions

Hardware sample rate converters	Sample rate conversion = 7.5:1 to 1:8, up to 192 kHz
Programmable input gain	from –15 dB to +15 dB

AES/EBU OUTPUTS

Only for “Analog & AES/EBU” versions and for “AES/EBU” versions

Sample rate	32 kHz, 44.1 kHz, or 48 kHz

SYNCHRONIZATION INPUTS

AES11(*)	Only on analog and AES/EBU I/O versions, and AES/EBU I/O versions
Word Clock	32, 44.1 kHz 48 kHz

(*) Only for “Analog & AES/EBU” versions and for “AES/EBU” versions

MADI INPUTS/OUTPUTS

Sample Rate	32, 44.1 kHz 48 kHz
MADI OUT channel mode (*)	64/64 channels mode

(*)Only for “MADI” versions and for “AES67-MADI” versions

ANALOG AUDIO PERFORMANCES

Only for “Analog & AES/EBU” versions

Dynamic range (A-weighted)	Analog In: >104 dB / Analog Out: >106 dB
THD + noise 1 kHz at –1 dBfs	Analog In: <–97 dB / Analog Out: <–96 dB
Channel phase difference: 20/20kHz	0.2° / 2°

Crosstalk (Analog in or out) 1 kHz at 22 dBu	<–100 dB - 15 kHz at 22 dBu: <–85 dB

AES67 / RAVENNA / MADI parameters

RAVENNA / AES67	Up to 128/128 I/O (Mono) channels at 44.1 kHz or 48 kHz (64/64 I/O on each Gigabit Ethernet interface)
MADI	64/64 I/O (Mono) at 48 kHz 32/32 I/O (Mono) at 96 kHz
Word Clock	One BNC used either as Word Clock input or Word Clock output. Configured through WEB pages. Input : TTL , impedance selectable by jumper (75 Ohms / HighZ). Output : Max 5 Vpp, 75 Ohms output impedance
Synchronization	Word clock, 44.1 or 48kHz, with selectable input impedance (High Z / 75 ohms) (can be used if SERV/LINK delivers the master clock to the AES67/Ravenna network).
Sampling frequencies	Internal clock: 44.1 kHz, 48 kHz and 96 kHz (MADI) From PTP clock or Word Clock: 44.1 kHz, 48 kHz and 96 kHz (MADI) From MADI: 44.1 kHz, 48 kHz, 96 kHz
Clock sources	PTPv2 (IEEE1588-2008) from network, or internal clock or Word Clock or MADI input Local clock eligible as GrandMaster PTP Local clock precision : better than 10 ppm

RAVENNA / AES67

Supported audio payload formats	PCM16 / PCM24 / PCM32 / AM824 (PCM24+AES3 channel status)
Packet size	From 128 down to 1 (ultra-low latency profile) audio samples per RAVENNA packet
Clock source	PTPv2 (IEEE1588-2008) from network or internal clock or Word Clock or MADI input Local clock eligible as GrandMaster PTP Local clock precision : better than 10 ppm
AES67 compliance	Full compliance in all respects with AES67 Support of SMPTE 2022-7 seamless protection switching
Control	HTTP (web pages from embedded server) EMBER+
Routing	Zero latency on-board routing matrix between RAVENNA, PC Rec/Play and optional MADI channels

Dante: IP audio

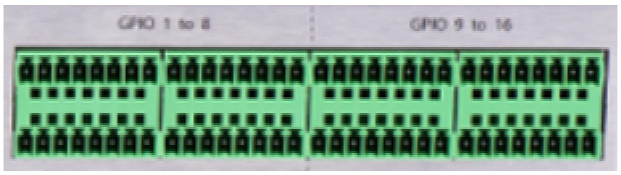
IP audio transport	Dante Audio over IP, AES67
Redundancy	Glitch-free Dante audio redundancy using Primary and secondary Ethernet networks
Clock synchronization	Master or slave

Dante: Audio

Audio Channels	8 / 8 to 128 / 128 I/O channels
Supported Sample Rates	44.1, 48 kHz
Sample bit-depth	24 bit PCM Audio

14 Appendix A: GPIOs description

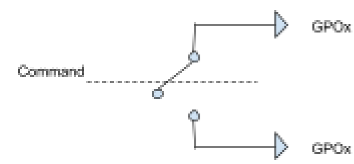
Physical GPIO's are a hardware option. When this option is installed, GPIO's are located on the rear panel, and accessible via Terminal Block connectors.



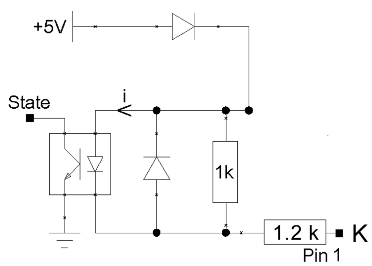
GPO's are as follows

Each GPO consists of a siThe GPI's are compatible TTL 5 V.

They do not require any external power.

GPI status is "open" (1) when pin K is not connected to the ground for at least 20ms. GPI status is "closed" (0) when pin K is connected to the ground for at least 20ms.	

GPI's are as follows



Connector GPIO 1 to 8

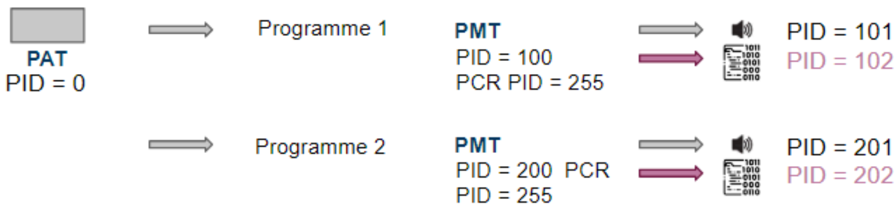
GND	GPI1	GND	GPI2	GND	GPI3	GND	GPI4	GND	GPI5	GND	GPI6	GND	GPI7	GND	GPI8
GP01	GP01	GP02	GP02	GP03	GP03	GP04	GP04	GP05	GP05	GP06	GP06	GP07	GP07	GP08	GP08

Connector GPIO 9 to 16

GND	GPI9	GND	GPI10	GND	GPI11	GND	GPI12	GND	GPI13	GND	GPI14	GND	GPI15	GND	GPI16
GP09	GP09	GP010	GP010	GP011	GP011	GP012	GP012	GP013	GP013	GP014	GP014	GP015	GP015	GP016	GP016

15 Appendix B: MPEG-TS auxiliary data

Structure of an MPEG-TS MPTS stream that includes 2 audio programs, each program having its auxiliary data.

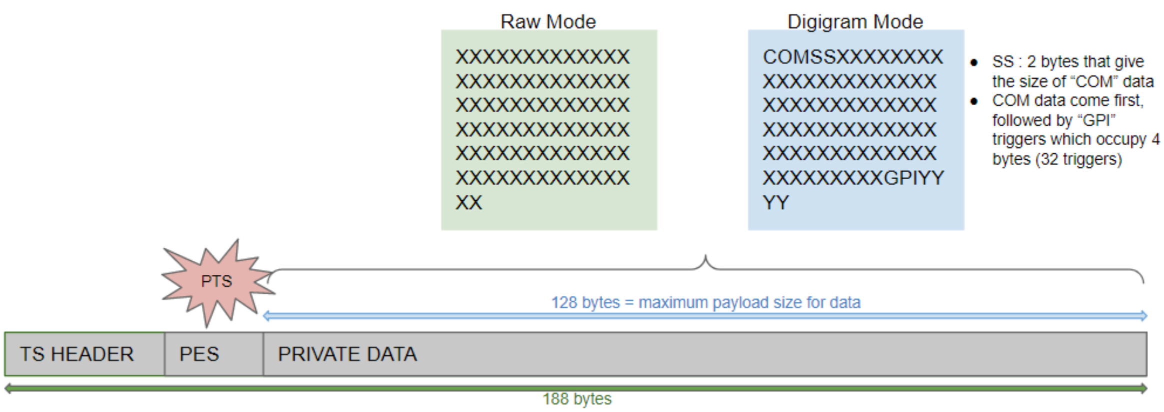


PAT: Program Association Table.
Gives the list of programs

PMT: Program Map Table.
One PMT per audio program
It lists all the elementary streams associated to a program.

Elementary streams:
Audio
Associated data

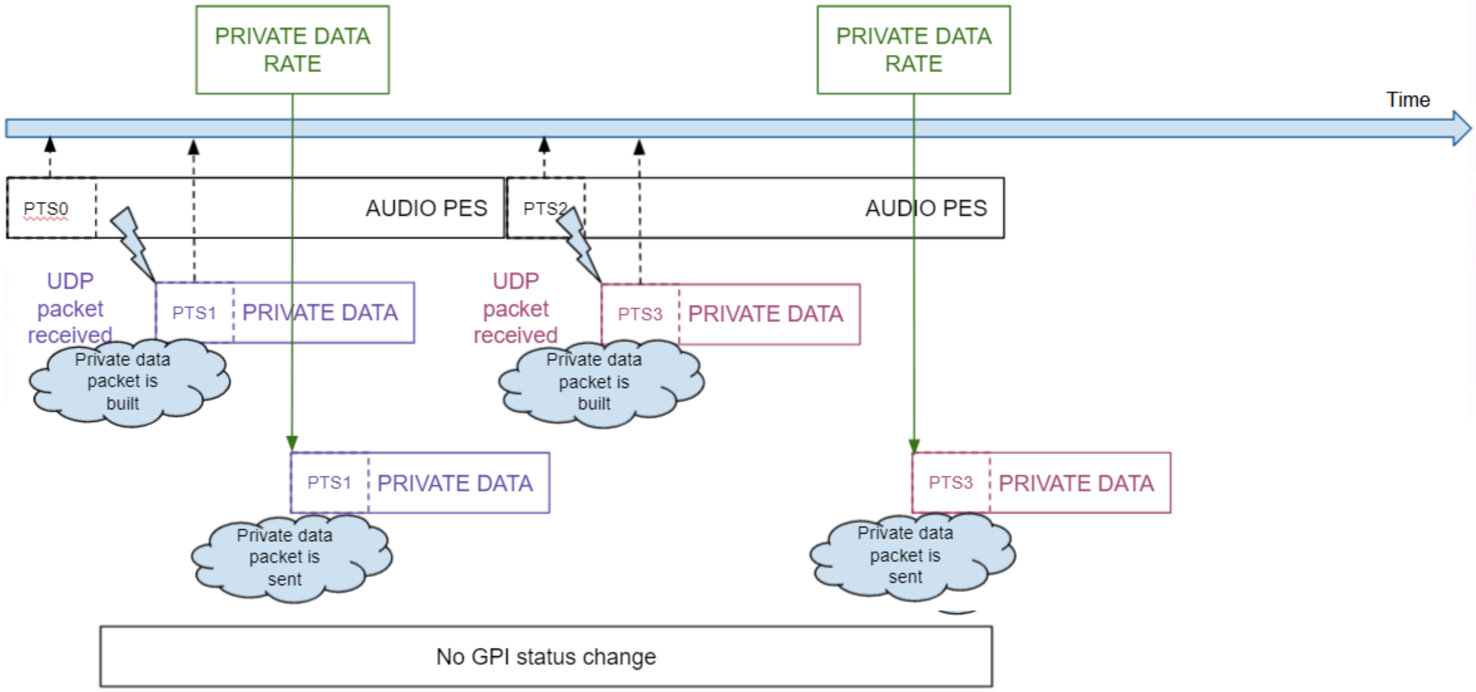
Auxiliary data packet structure



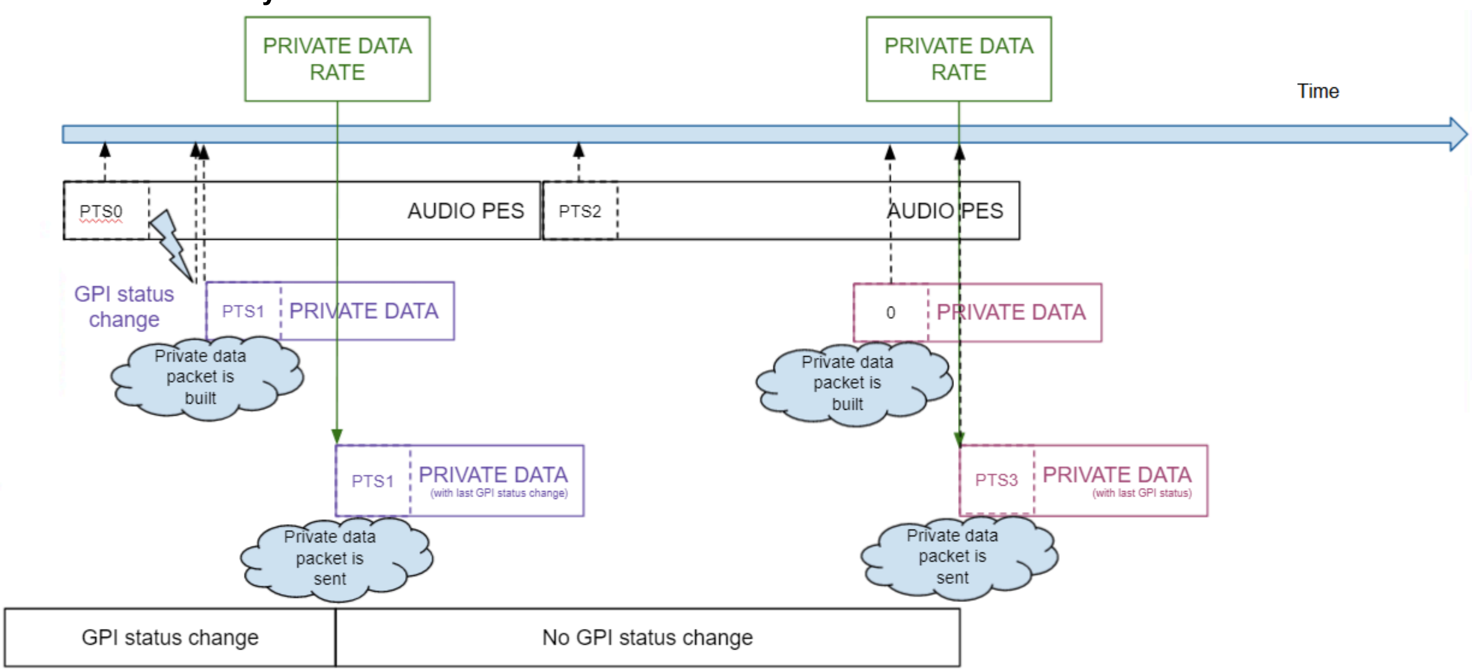
In raw mode, the packet only includes data coming from a serial port or from UDP (text, characters, commands).
In Digigram mode, the packet contains data coming from a serial port or from UDP (text, characters, commans), as well as triggers coming from physical or virtual GPI's.
128 bytes are allocated for data. Data packets are always 188 bytes long, but data can be less than 128 bytes.

16 Appendix C: Time-stamping of private data in a TS stream

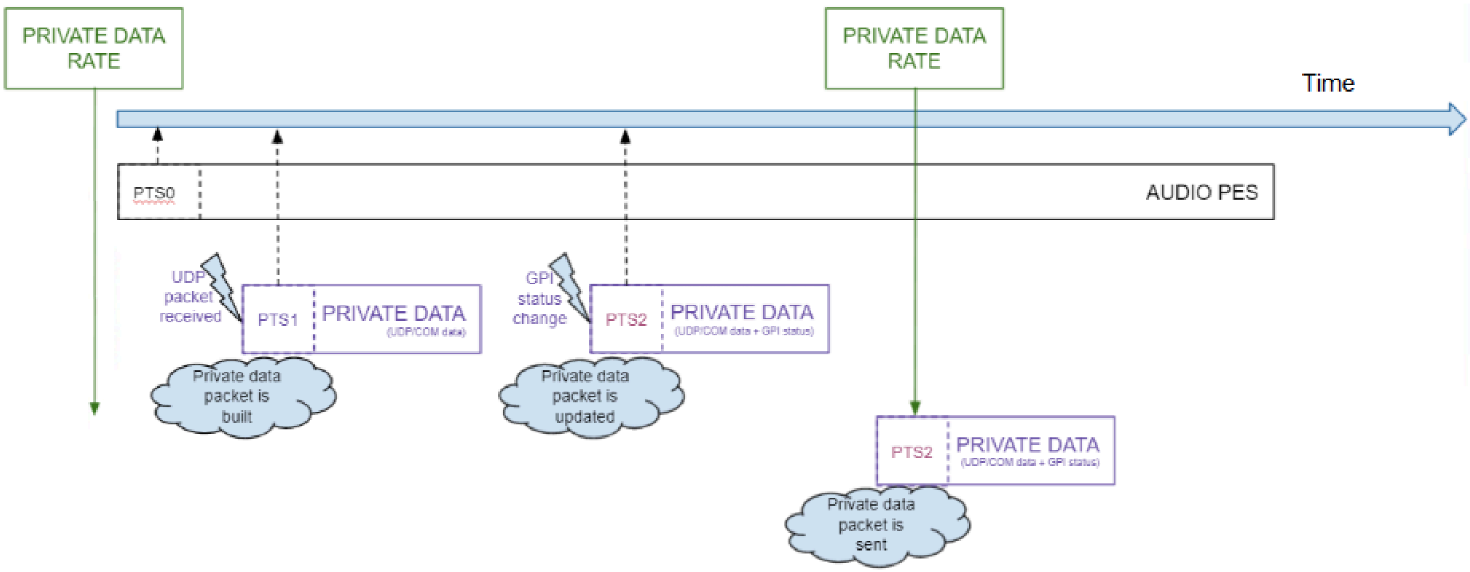
Insertion of serial data only



Insertion of GPI's only

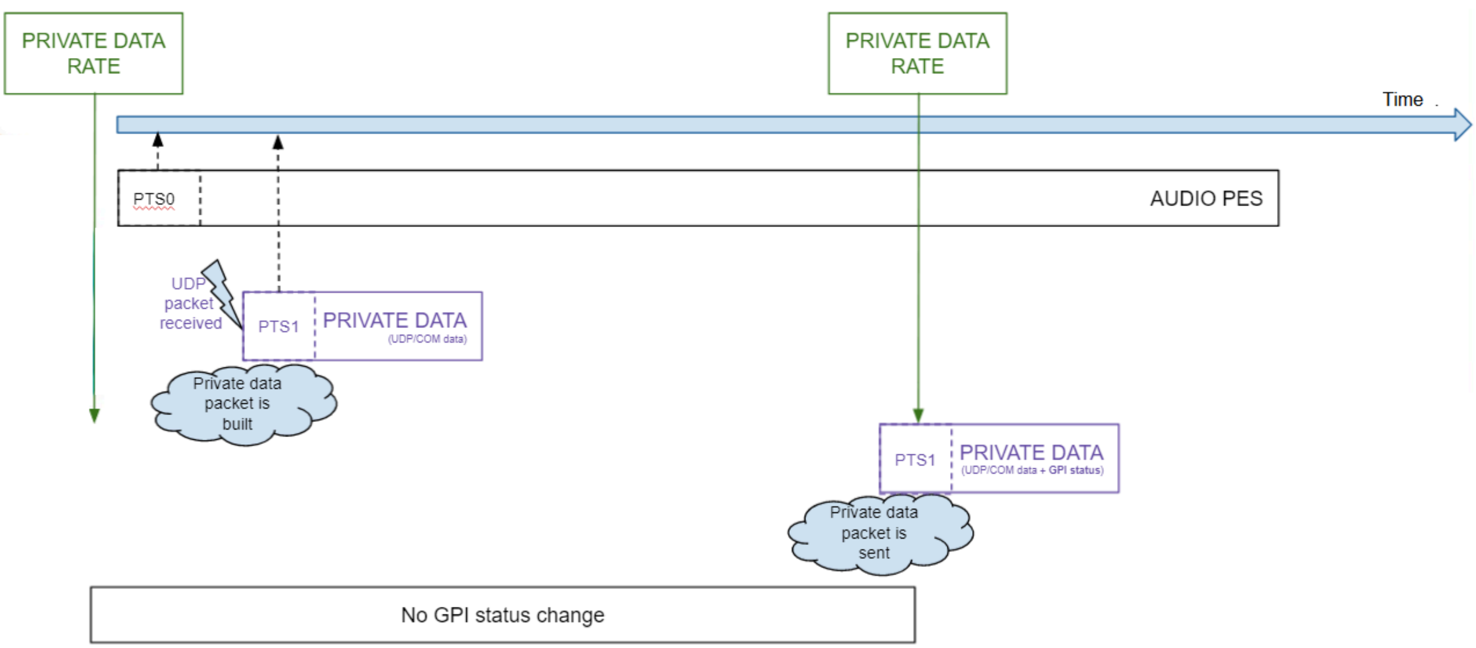


Insertion of serial data and GPI's: case 1



We see here that the time-stamp of the private data packet corresponds to the time of a GPI status change.

Serial data and GPI's: case 2



We see here that if there is no GPI status change between two private data packets, and if serial data are received during this interval of time, the time-stamp of the private data packet corresponds to the time serial data are received.

17 Appendix D: Modifying Maximum_Bitrate_Descriptor and ES_Rate_flag via the configuration file

Since version 3.11, the MPEG-TS streams generated by IQOYA by default contain the descriptor "Maximum_Bitrate_Descriptor" in the PMT, and the flag "ES_Rate_flag" in the PES.
These fields are by default not present in the generated HLS streams.

These settings can be modified via the IQOYA configuration file. Here is the procedure.

- Download the configuration file after having configured the MPEG-TS and/o HLS streams.
  Go to Preferences-> System-> Download/Upload -> Download / Audio configuration.
- Edit the downloaded configuration file: IqoyaServLink_DISTRIB.cfg

- **For MPEG-TS**, these parameters can be disabled from each [TRANSMITTER] section.
  Check if entries MTSProgramDescriptorFlag and MTSStreamPesFlag exist. If they are not present (by default) and you want to disable the "Maximum_Bitrate_Descriptor" and the flag "ES_Rate_flag", create them as follows:
  MTSProgramDescriptorFlag=0
  MTSStreamPesFlag=0

  If MTSProgramDescriptorFlag is removed or if it is set to 1, this enables the Maximum_Bitrate_Descriptor in the generated MPEG TS.
  If MTSStreamPesFlag is removed or if it is set to 4, this enables the ES_Rate_flag in the generated MPEG TS.

- **For HLS**, these parameters can be enabled from each [TRANSMITTER] section.
  Find the entries MTSProgramDescriptorFlag and MTSStreamPesFlag in each [TRANSMITTER] section. If they are not present (by default) and you want to enable the "Maximum_Bitrate_Descriptor" and the flag "ES_Rate_flag", create them as follows:
  MTSProgramDescriptorFlag=1
  MTSStreamPesFlag=4

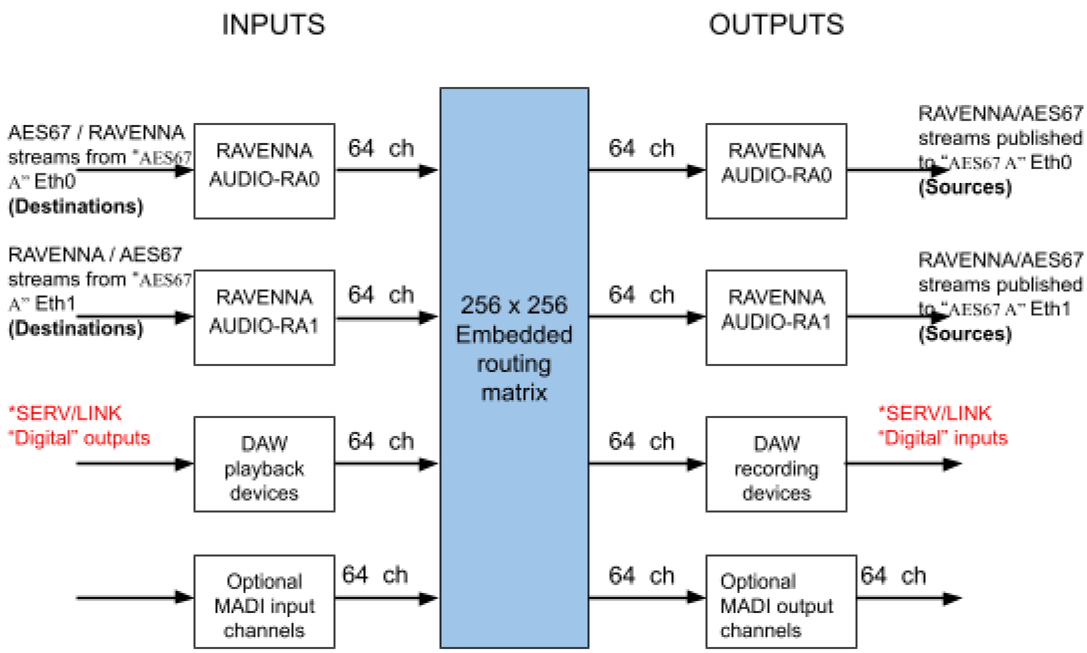
  If MTSProgramDescriptorFlag is set to 1, this enables the Maximum_Bitrate_Descriptor in the generated MPEG TS. Reset it to 0 or remove it to disable it.
  If MTSStreamPesFlag is set to 4, this enables the ES_Rate_flag in the generated MPEG TS. Reset it to 0 or remove it to disable it.

- Save your modifications
- Upload the new configuration file.
  Go to Preferences-> System-> Download/Upload -> Upload / Upload audio configuration file from local disk.
  Select the file. The new settings are applied right after the cfg file is uploaded.

18 Appendix E: AES67 configuration for SERV/LINK AES67 units sold before 2025.

18.1 Principles

18.1.1 Embedded routing matrix



The inputs of the on-board routing matrix are:

- 64 channels coming from the input block "RAVENNA AUDIO-RA0". This block extracts the audio channels from the RAVENNA / AES67 streams that are declared to be received through the network interface AES67 A Eth0. These streams to be received are named "Destinations".
- 64 channels coming from the block "RAVENNA AUDIO-RA1". This block extracts the audio channels from the RAVENNA / AES67 streams that are declared to be received through the network interface AES67 A Eth1. These streams to be received are named "Destinations".
- 64 channels coming from the block "DAW". This block receives the audio channels coming from the output programs of IQOYA SERV/LINK.
- 64 channels coming from the MADI input for SERV/LINK AES67/MADI.

The outputs of the routing matrix are:

- 64 channels assigned to the output block "RAVENNA AUDIO-RA0". These channels are the audio sources of the AoIP streams published to the network through AES67 A Eth0. these streams are named "Sources".
- 64 channels assigned to the output block "RAVENNA AUDIO-RA1". These channels are the audio sources of the AoIP streams published to the network through AES67 A Eth1. these streams are named "Sources".
- 64 channels assigned to the output block "DAW". These channels correspond the audio inputs of IQOYA SERV/LINK.
- 64 channels assigned to the MADI output for SERV/LINK AES67/MADI.

18.2 Sources

Sources is the name given to the audio streams published to the network by AES67 / RAVENNA capable devices. For IQOYA SERV/LINK, sources are published to the network through the network interfaces Eth0 and Eth1.
Streams which audio channels come from the output block “RAVENNA AUDIO-RA0” are sent through AES67 A Eth0.
Streams which audio channels come from the output block “RAVENNA AUDIO-RA1” are sent through AES67 A Eth1.

18.3 Destinations

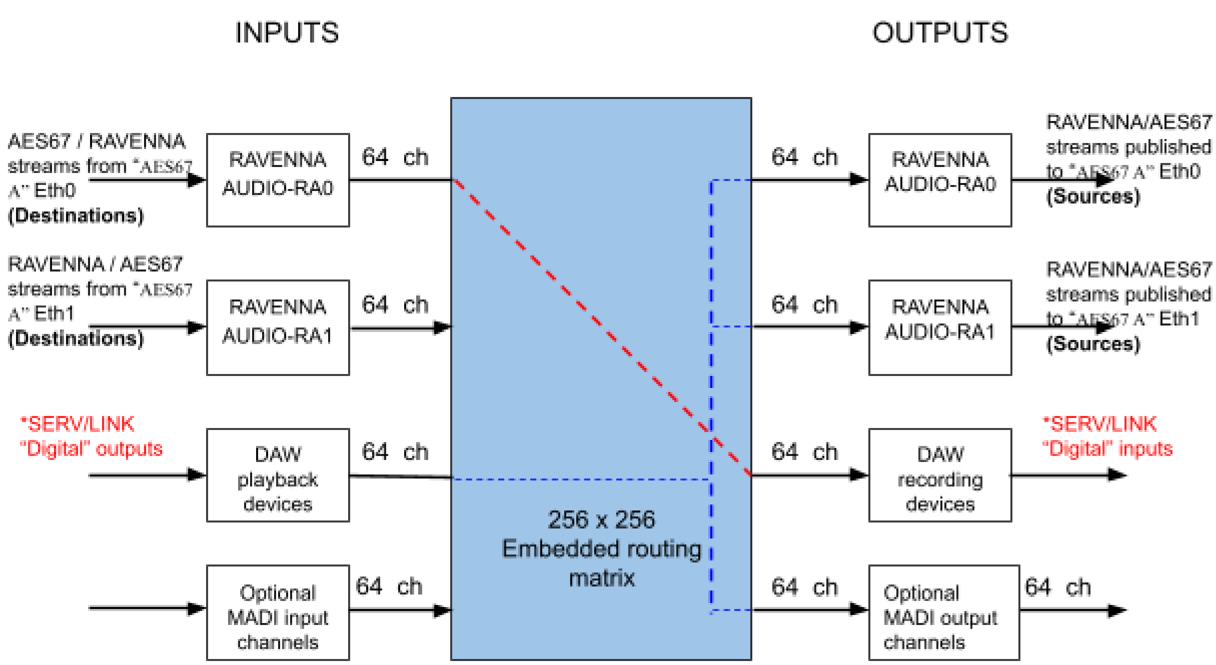
Destinations are the AoIP streams received from the network.
Destinations can be received on Eth0 and Eth1 ports.
The audio channels extracted from the Destinations received through AES67 A Eth0 are available from the input block “RAVENNA AUDIO-RA0”.
The audio channels extracted from the Destinations received through AES67 A Eth1 are available from the input block “RAVENNA AUDIO-RA1”.

18.4 Clock

IQOYA SERV/LINK can be slaved to the PTP clock, or master PTP clock on the network.
Slave to PTP clock: the source clock is the PTP clock.
Master PTP Clock: the clock source has to be internal, WordClock, or MADI, and the card is elected as Master PTP on the network if its PTP clock has a higher priority than other potential Master PTP clock units on the network.

Default routing matrix

The matrix default routing is as follows:



18.5 Configuration of the AoIP parameters

The AES67 network parameters of SERV/LINK can be configured through its embedded WEB server, which is accessible from any of the two AoIP AES67 ports.

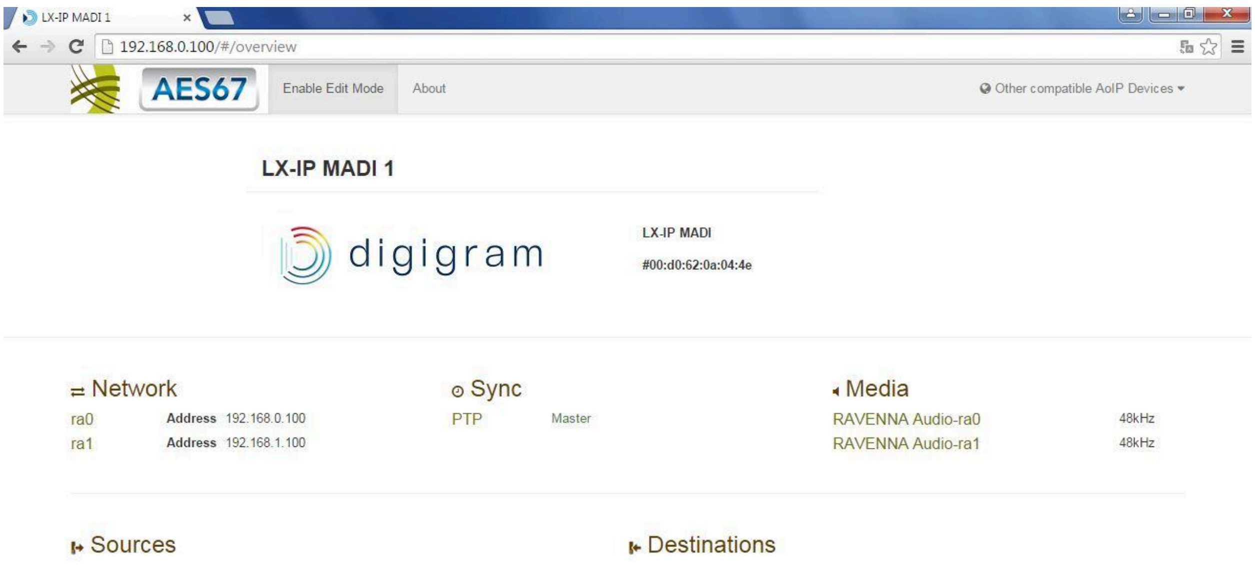
In case of an IQOYA SERV/LINK 7272-AES67 platform, It is necessary to configure the AES67 parameters of the first group of 64 channels from one of the two “Eth A” ports, and to configure the AES67 parameters of the second group of 64 channels from one of the two “Eth B” ports

18.5.1 Default IP addresses of AES67 Eth ports

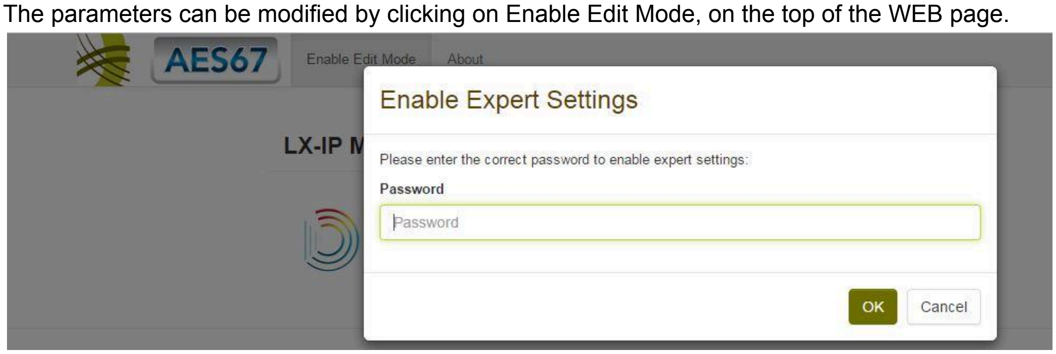
Bank A: Eth0	Bank A: Eth1	Bank B: Eth0	Bank B: Eth1
192.168.2.100	192.168.3.100	192.168.4.100	192.168.5.100

From a PC connected to the AoIP network, and set to an IP address that belongs to the same subnetwork as an AES67 interface of SERV/LINK, enter in your WEB browser the IP address of the AES67 interface the PC has access to.

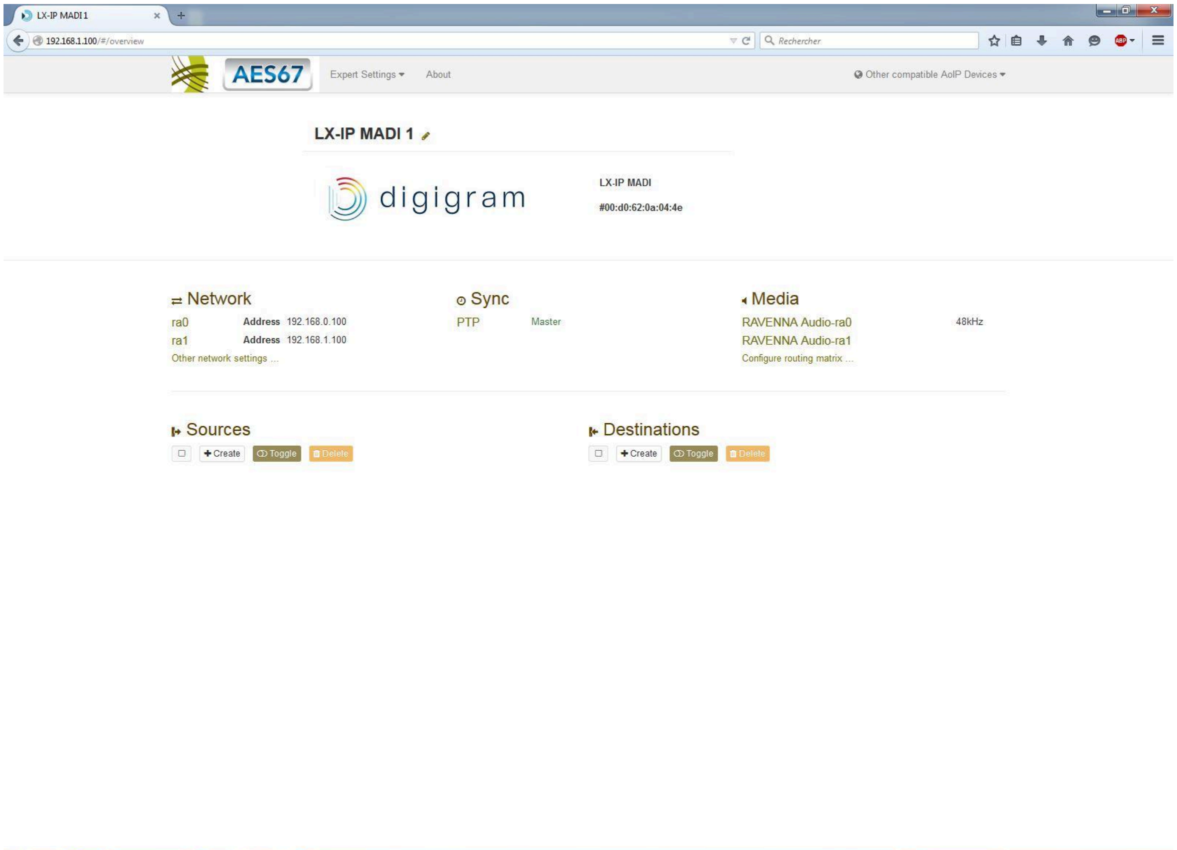
The following page is displayed.



The parameters can be modified by clicking on Enable Edit Mode, on the top of the WEB page.



The requested password is: ravenna.
The following page is displayed;



18.5.2 Network settings

Click on “ra0” to configure the network parameters of network port Eth0.

	**Configuration**: Static or DHCP If configuration is set to static, the following two parameters must be set. **Address**: IP address of the Eth0 interface. **Network mask**: mask of the network the AES67A Eth0 interface belongs to.

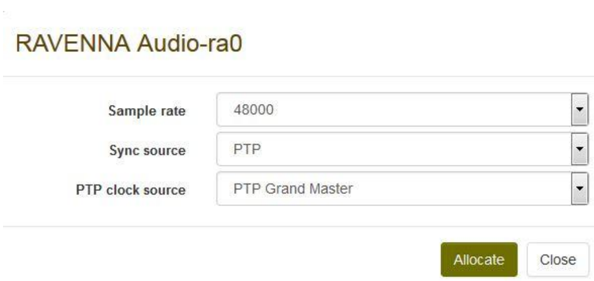
Similarly, click on “ra1” to configure the network parameters of network port AES67A Eth1.

Click on “other network settings” to configure the Hostname, and the gateway.

Hostname: This is the tname of the RAVENNA node, used to identify the device at a network level. This name must be unique on the network,; no spaces are allowed, no special characters.
Gateway: specify the network gateway IP address if there is one.
Default TTL: Time To Live value. It defines the number of routers the generated frames can cross..

18.6 Global audio and clock settings

The sampling frequency, the network packet size as well as the clock source can be configured by clicking on “RAVENNA Audio-ra0”, from the “Media” section.
The sampling frequency and the network packet size can be configured by clicking on “RAVENNA Audio-ra1”.

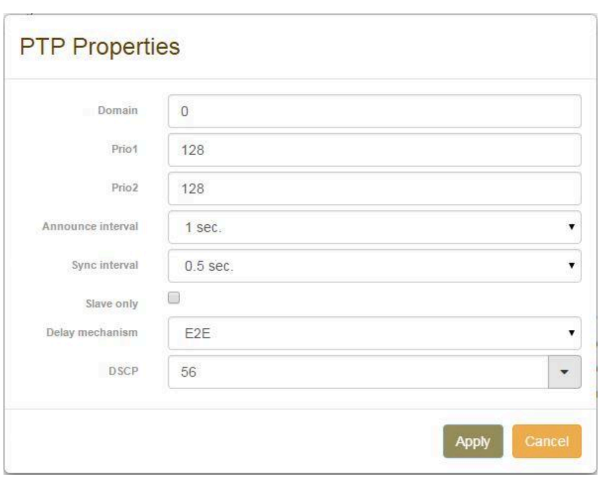


Audio block size: this defines the number of samples per channel per network packet; the smaller the
frame size, the lower the latency but the more susceptible to drop-outs. Values are from 1 to 128 samples.
Sample rate: this is the audio sample rate: 48000 Hz or 44100 Hz.
Sync Source: this defines the clock source of SERV/LINK. The clock source can be:

- PTP: SERV/LINK can be PTP Master or slaved to PTP
- WordClock: received on the WordClock input of the card
- MADI: the card automatically synchronizes on the detected MADI signal.

PTP Clock source: This defines the source of the PTP clock of the card.
See “Use cases for clock configuration” below for setting the clock parameters.

In case the clock source is configured to PTP , it is necessary to set the PTP clock parameters, by clicking on “PTP”, from the “Sync” section



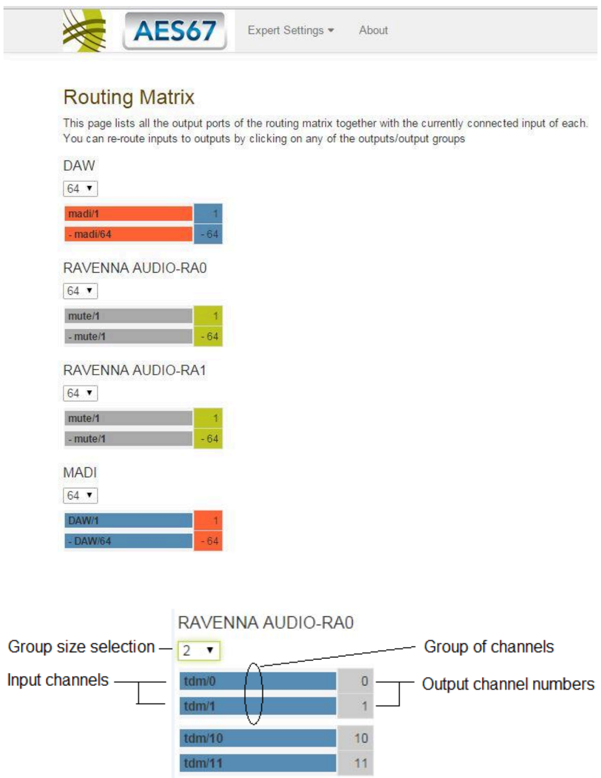
Slave only: when this option is checked, SERV/LINK is forced to run in PTP slave mode.
In case this option is not check, the SERV/LINK card can be elected as Master PTP depending on its priority. A device is elected as master PTP if the priority of its PTP clock is the highest.
Domain: Time domain for PTP. This MUST be set to match the domain number of the related PTP Grandmaster.
Prio1: Internal PTP setting. This parameter is used to control the priority of Grandmaster selection
Prio2: Sames as Prio1
Announce Interval: in seconds (1, 2, 4, 8 or 16). In Slave mode, this MUST be set to match the Announce Interval of the related PTP master clock. In Master mode, this determines the desired Announce Interval.
Sync interval: in seconds (0.5, 1 or 2). In Slave mode, this MUST be set to match the Sync Interval of the related PTP master clock. In Master mode, this determines the desired Sync Interval.
Delay request mechanism: End to End (E2E) or Peer to Peer (P2P). This MUST be set to match the related PTP master clock. While E2E is a more universal setting, P2P provides higher clock sync precision but requires full
support from all participating switches (between the node and related clock master.)

Use cases for clock configuration

SERV/LINK use cases	Sync source	PTP clock source	PTP properties	Destinations properties
Slave to the PTP clock from the network. This PTP clock is delivered by another device acting as Grand Master	PTP	PTP Grand Master	Slave only checked	
Master PTP on the network, with synchro source from the WordClock input	PTP	WordClock IN	“Slave only” unchecked	
Master PTP on the network, with synchro source from internal clock of SERV/LINK	PTP	Internal	“Slave only” unchecked	
Master PTP on the network, with synchro source from MADI	PTP	MADI	“Slave only” unchecked	
Slave to MADI	MADI	NA	NA	Syntonized checked
Slave to WordClock	WordClock	NA	NA	Syntonized checked
Slave to the internal clock	Internal	NA	NA	Syntonized checked

18.7 Routing matrix

The embedded matrix can be configured by clicking on "Configure routing matrix".



This view displays the three available output ports of the matrix, and the audio inputs that are allocated to each port.
Each output port is 64 channels.

DAW : This section allows for the selection of the audio inputs that will be published to the Digital Audio Workstation software devices (DirectSound, ASIO, Alsa).
RAVENNA AUDIO-RA0: This section allows for the selection of the audio inputs that will be published to the network as IP audio streams through Eth0.
RAVENNA AUDIO-RA1: This section allows for the selection of the audio inputs that will be published to the network as IP audio streams through Eth1.
MADI (optional): This section allows for the selection of the audio inputs that will be published to the MADI output.

The allocated inputs can be configured channel per channel (select 1 in the drop down menu), per stereo channels, (select 2 in the drop down menu), per group of 8 consecutive channels (select 8 in the drop down menu), or per group of 64 consecutive channels (select 8 in the drop down menu).

The input channels are then listed by groups (1, 2, 8, or 64), with the number of the associated output channels in front of the input channel name.

This is an example of input channels allocated per groups of 8 channels to the the DAW output port.

The selected inputs are routed the the PC devices a software application can record from.

To configure the group of audio inputs to be allocated to a group on the output port, click on the group of channels.

The inputs can then be selected on the right part of the window. Inputs can be selected from the four input ports:

DAW: This section allows for the selection of the audio inputs that come from the playback devices.
RAVENNA AUDIO-RA0: This section allows for the selection of the audio inputs that come from AoIP streams on Eth0.
RAVENNA AUDIO-RA1: This section allows for the selection of the audio inputs that come from AoIP streams on Eth1.
MADI (optional): This section allows for the selection of the audio inputs that come from the MADI input.

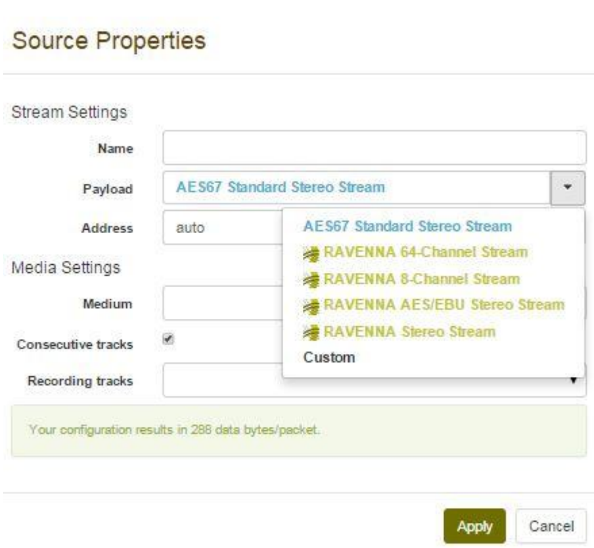
Note that for a given group, all the inputs come from the same input port.

Mute: Allows muting all the channels of the group.

Click on  on the top of the wEB page to display the main WEB page.

18.8 Creation of IP streams published to the network

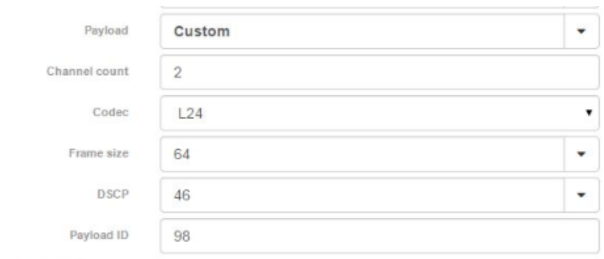
To declare an RAVENNA/AES67 stream to be published to the network, click on “Create source” from the “Sources” section.



Stream settings
Name: Enter a name to identify the stream to be generated on the network (e.g. From My Automation). A default name is automatically entered, and is taken from the RAVENNA node Name (in the headline).
Payload: This is an internal RTP value which informs subscribers about the nature of the content. Possible values are:

- AES67 Standard Stereo Stream: AES67stereo stream
- RAVENNA 64-Channel stream: RAVENNA stream with 64 channels.
- RAVENNA 8-Channel stream: RAVENNA stream with 8 channels.
- RAVENNA AES/EBU Stereo Stream: AM824 format = 24-bit audio + 8-bit meta data as used with AES/EBU
- RAVENNA Stereo Stream: 24-bit samples
- Custom: allows defining an other format configuration (payload, number of channels, frame size, DSCP, and Payload ID)
  Note for DSCP: Select a QoS (Quality of Service) value from the drop-down menu - EF
  (46), AF41 (34), AF31 (26) or Standard (0); the default value is EF (46). This should match the priority settings used in your network for preferred real-time media packet forwarding.

Address: auto. This automatically assigns a multicast IP address to the generated stream.

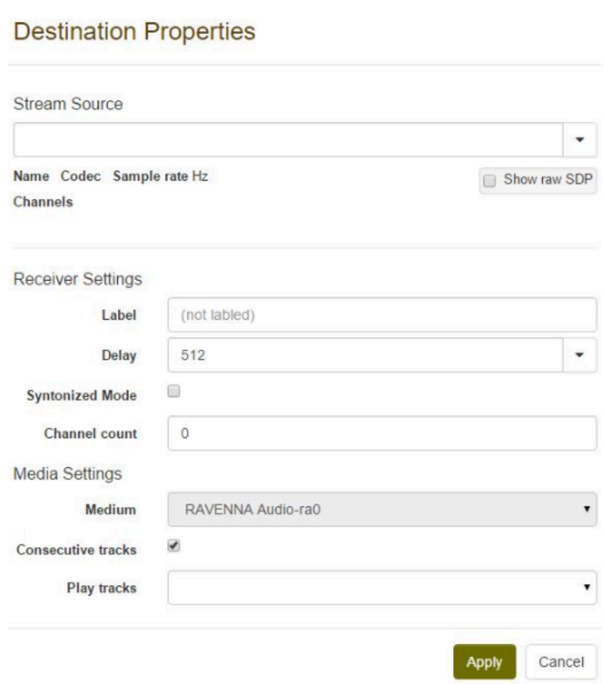


Media Settings
Medium: Select here the input port of the matrix from which the audio channel(s) are to be selected.
Consecutive tracks: If checked, channels of the stereo or multichannel stream are taken consecutively from the input port group that is selected. Otherwise, channels can be selected one by one from **recording tracks** fields.

Recording tracks: allows selecting the channels available from the selected input port of the matrix.

Receiving IP streams from the network

To declare an RAVENNA/AES67 stream to be received from the network, click on "Create destination" from the "Destinations" section.



Sream Source: This drop down menu displays the list of available IP streams published on the RAVENNA/AES67 network. Select a stream from this list.
Show raw SDP: Allows displaying the details of the stream description.

Receive settings
Label: displays the label of the selected stream.
Delay: Enter the amount of delay to be applied, in samples, before samples are played out (forwarded to
the internal audio interface). The delay is referenced to the sampling time at the sender. Thus, it needs to be large enough to cover all possible influences, such as the packet assembly delay at the sender (frame size), transport delay, maximum packet jitter and packet
disassembly delay at the receiver.
TIP: Set the delay to be larger than the frame size specified by the sender. As a general rule, the
delay value should be 2 x sender's frame size. So, if the sender's frame size = 128, set the delay =
256. If you experience drop-outs, increase the delay time.
Syntonized mode: This mode is to be checked in case the **Sync source** is not PTP.

Channel count: The channel count determines the number of channels to be routed from the selected stream to the internal audio interface. If 0 is entered, all available channels are used upon subscription.

Media Settings
Medium: Select the output group of the routing matrix
Consecutive tracks: If checked, channels of the stream are played out on consecutive channels of the selected group. Otherwise, output channels can be selected one by one from **Play tracks** fields.
Play tracks: selection of the tracks of the output group.

Click on **Apply** to confirm the setting changes.