IQOYA X/LINK range
IP audio codecs

User manual for:
IQOYA X/LINK-LE, -ST, -DUAL, -AES67

Applies from firmware version 3.12, February 2023

Date	Changes	Document Status
June, 4, 2020	Appendix A,GPIO description: N.C and N.O pins were inverted.	Released
November 2020	New settings for DVB information tables in MPEG-TS streams (Send section).	Released
January 2021	HLS multi bitrate encoding and streaming	Released
March 2021	Added insertion of data associated to an audio program in a TS stream. In contribution, mono is sent to left and right for encoding and decoding; and stereo is mixed down to mono for encoding / decoding.	Released
November 2021	Documentation update for EMC standards and safety rules.	Released
January 2022	Support for Akamai MSL4 CDN (HTTP with and without authentication - Digest SHA-256 & MD5). Icecast/Shoutcast decoding, added support of the permanent redirection error code (301)	Released
July 2022	Possibility to set the WEB Service API protocol to HTTPS (default mode) or HTTP (less security).	Released
December 2022	Configuration of the duplication of the output signal to analog and AES3 outputs Required transcoding channels. This feature is available for all X/LINK flavours with the exception of X/LINK-AES67.	Released
January 2023	Improvement of the management of the front panel vu-metres when importing a configuration with mono channels made from a previous firmware from version 3.05 (audio configuration or full configuration). MPEG-TS streams are now generated by default with the “Maximum_Bitrate_Descriptor” in the PMT, and the “ES_Rate_flag” in the PES.	Released
February 2023	GPIO connectors were inverted in previous manual versions. Fxed	Released

Note regarding the presentation of this document:

IQOYA X/LINK devices feature two modes of use :

- *The 'Program Distribution' mode of use*
- *And the 'Remote Broadcasting' mode of use*

These two modes are described in the WORKING PRINCIPLES chapter.

In this document, the chapters specific to the "Program Distribution" mode of use are presented with a green background and the chapters specific to the "Remote Broadcasting" mode of use are presented with a blue background. The chapters which are relevant for both modes of use are presented on a white background.

TABLE OF CONTENTS

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, this pictogram is used to alert the user to the risk of electric shock.

This pictogram is used to alert the user to important operating or maintenance instructions

This pictogram is used to alert the user that the device has multiple power sources.

WARNING - Risk of electric shock
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!

WARNING - MULTIPLE POWER SOURCES

Power supply
The device is to be connected only to power supplies 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 sockets outlets to which the equipment is connected must be easily accessible.
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.

In Finland and Sweden the equipment is not intended to be connected to a telecom network for the Ethernet ports.

AVERTISSEMENT: risque de choc électrique
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 !

AVERTISSEMENT - Sources d'alimentation multiples

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 !
N'obstruer aucune ouverture 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, les prises de courant sur lesquelles sont branchées l'équipement doivent être aisément accessible
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.

WARNUNG - Stromschlaggefahr
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.

WARNUNG – MEHRERE STROMQUELLEN

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.

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 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.	**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 utiliser 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 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 la reliant à d'autres appareils.	**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 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.

	**Disposal of the device:** European Directive 2012/19 / EU on Waste of Electrical and Electronic Equipment (WEEE), requires that the devices used household goods are not thrown into the normal municipal waste stream. Used devices should be collected separately in order to optimize the rate of recovery and recycling of the materials of which they are made and reduce the impact on human health and the environment.	**Mise au rebut de l’appareil** La directive Européenne 2012/19/UE sur les Déchets des Equipements Electriques et Electroniques (DEEE), exige que les appareils ménagers usagés ne soient pas jetés dans le flux normal des déchets municipaux. Les appareils usagés doivent être collectés séparément afin d’optimiser le taux de récupération et le recyclage des matériaux qui les composent et réduire l’impact sur la santé humaine et l’environnement.

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**
User safety	**EMC**
European Directive *2014/35/UE "Low Voltage Directive* ***Europe****: EN 62368-1 (2014)* ***International****: IEC 62368-1 (2014)*	***European Directive****: EMC 2014/30/UE* ***Radio disturbance*** *:* *International : CISPR32 (2015+AMD1/2019) Class A* *European : EN55032 (2015+A1&A11/2020) Requirements for Information Technology Equipment (ITE)* ***Immunity****: International : CISPR35 (2016)* *IEC 61000-6-2 (2016)* *European : EN55035 (2017+A11/2020) (ITE)* *EN 61000-6-2 (2019)* ***Harmonics****: International : IEC 61000-3-2 (2018+A1:2020)* *European : EN 61000-3-2 (2019)* ***Voltage changes*** *: International : IEC 61000-3-3 (2013+A1:2017+A2:2021)* *European : EN 61000-3-3 (2014+A1/2019)* ***United States****: CFR 47, FCC Part 15, Subpart A (Class A Digital Device) & Industry Canada ICES-003 (Issue 7 – 2020)*

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 equipment's PE and building's PE must be connected (socket outlet with PE required).
- The sockets outlets to which the equipment is connected must be easily accessible
- The maximum altitude to comply with the above standards is 2000m. IQOYA can however operate at higher altitudes.

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.

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

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.

2. KEY HARDWARE FEATURES

2.1. IQOYA X/LINK-LE

- 1U rack
- Two internal redundant power supply units (2x220 VAC, optionally 220VDC / -48VDC)
- 4 Ethernet ports via RJ-45 connectors. 1 x 100 Mbits/s, and 3 x 100/1000 Mbits/s ports
- 2 balanced analog line inputs and outputs
- 1 AES/3 input and output
- Switchable hardware by-pass on inputs to outputs in case of power supply failure
- 1 RS232 port for auxiliary data tunneling
- 8 GPIO's
- External synchronization: AES3 input, PTP clock, Livewire clock
- Status LEDs
- SDHC card reader
- 6.35mm headphones jack with volume knob and codec input/output selection

2.2. IQOYA X/LINK-ST

- 1U rack
- Two internal redundant power supply units (2x220 VAC, optionally 220VDC / -48VDC)
- 4 Ethernet ports via RJ-45 connectors. 1 x 100 Mbits/s, and 3 x 100/1000 Mbits/s ports
- 2 balanced analog line inputs and outputs
- 1 AES/3 input and output
- Switchable hardware by-pass on inputs to outputs in case of power supply failure
- 1 RS232 port for auxiliary data tunneling
- 8 GPIO's
- External synchronization: AES/3 input, PTP clock,
- Front panel LCD display and keypad
- Status LEDs
- SDHC card reader
- 6.35mm headphones jack with volume knob and codec input/output selection

2.3. IQOYA X/LINK-DUAL

- 1U rack
- Two internal redundant power supply units (2x220 VAC, optionally 220VDC / -48VDC)
- 4 Ethernet ports via RJ-45 connectors. 1 x 100 Mbits/s, and 3 x 100/1000 Mbits/s ports
- 4 balanced analog line inputs and outputs
- 2 AES/3 input and output
- Switchable hardware by-pass on the first stereo inputs to outputs in case of power supply failure
- 1 RS232 port for auxiliary data tunneling
- 4 GPIO's if the 10 MHz / 1 PPS external synchro input(s) are used (optional)
- External synchronization: AES/3 input, PTP clock, Livewire clock
- Front panel LCD display and keypad
- Status LEDs
- SDHC card reader

- 6.35mm headphones jack with volume knob and codec input/output selection

2.4. IQOYA X/LINK-AES67

- 1U rack
- Two internal redundant power supply units (2x220 VAC, optionally 220VDC / -48VDC)
- 4 Ethernet ports via RJ-45 connectors. 1 x 100 Mbits/s, and 3 x 100/1000 Mbits/s ports
- 1 RS232 port for auxiliary data tunnelling
- 8 GPIO’s, or 4 GPIO’s if the 10 MHz / 1 PPS external synchro input(s) are used (optional)
- External synchronization: 10 MHz (optional), PTP clock, Livewire clock
- Front panel LCD display and keypad
- Status LEDs
- SDHC card reader
- 6.35mm headphones jack with volume knob and codec input/output selection

3. KEY SOFTWARE FEATURES

3.1. Supported I/O channels

	Number of mono input / output channels of the codec	Type of audio I/Os
X/LINK-ST & X/LINK-LE	2 / 2	Analog, AES/3, AES67, Ravenna, Livewire (standard mode)
X/LINK-DUAL	4 / 4	Analog, AES/3, AES67, Ravenna, Livewire (standard mode)
X/LINK-AES67	From 2 / 2 to 16 / 16	AES67, Ravenna, Livewire (standard mode)

3.2. Standard features

Two modes of use: “Program Distribution” mode and “Remote Broadcasting” mode.

Distribution mode

- Simultaneous encoding, decoding
- Multi-format encoding and multi-protocol streaming of each input.
- Streaming protocols: ARTP, UDP, Icecast/Shoutcast, HLS multi-bitrate
- Support for SIP signalling protocol including SIP presence information
- Support for symmetric RTP
- Contact list management
- Call profile management
- Possibility to place calls choosing the correspondent in an address book and the call profile in a call profile list and to accept or deny incoming calls.

- Support of unicast, multi-unicast, multicast, multi-multicast addressing
- Support of IGMPv3
- MPEG-TS/IP streaming with or without DVB information tables, and insertion of associated program data (serial data and triggers).
- VLAN Tagging + DSCP
- Asymmetric algorithmic encoding/decoding
- 3 decoding priorities per output program, with choice of the audio source on each priority: IP service (RTP, UDP, HTTP), file, playlist and audio input
- Automatic switching to a lower decoding priority in case of upper priority failure
- Possibility to disable/enable any defined priority
- Possibility to stop streaming upon input silence detection with adjustable silence threshold and duration.
- Decoding of a stereo source to a mono output, with possibility to mix left and right channels
- Dual port redundant streaming with optional time diversity up to 3 second
- Selectable FECs for ACIP RTP streams (from +10% to +100% IP bandwidth)
- Pro MPEG Cop#3 FEC for MPEG-TS streams (line, columns)
- Automatic audio format detection on the decoder
- 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
- In-band auxiliary data tunnelling: serial and status (serial via RS232 or UDP, Status via GPIOs or UDP)
- WEB user rights management
- 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 SET, GET, Traps
- WEB Service API (in distribution mode)

Features specific to the contribution mode

- Support of DHCP
- Support for SIP signalling protocol including SIP presence information
- Support for symmetric RTP
- Asymmetric algorithmic encoding/decoding
- Contact list management
- Call profile management
- Possibility to place calls choosing the correspondent in an address book and the call profile in a call profile list and to accept or deny incoming calls.

3.3. Optional software features

- IP streams transcoding. X/LINK-ST, X/LINK-DUAL, X/LINK-AES67
  On IQOYA X/LINK-LE, X/LINK-ST and X/LINK-DUAL, optional transcoding channels (audio buses) can be used to duplicate the decoded audio to both the analog and digital AES3 outputs.
- Multi-protocol streaming: X/LINK-LE
- NTP based audio synchronization

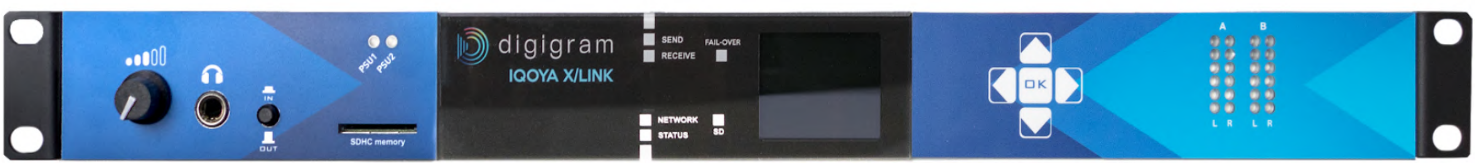
- 1+1 redundancy

2.4 Supported audio algorithms

Included	Otional
• linear 16/20/24-bit PCM • ITU G.711/722 • ISO MPEG-1/2 Layer I, Layer II, Layer III • AAC-LC, HE-AACv1 (LC+SBR), HE-AACv2(LC+SBR+PS), AAC-LD, AAC-ELD • Opus	

3 PHYSICAL INTERFACES

4.1. IQOYA X/LINK-ST, X/LINK-DUAL and X/LINK-AES67 front Panel



4.1.1. LEDs

PSU1	Status of the first internal power supply unit. Blue if PSU is ok. Off for PSU failure.
PSU2	Status of the second internal power supply unit. Blue if PSU is ok. Off for PSU failure.
Network	Green: all the enabled network interfaces are up. Orange: at least one of the enabled network interface is down Red: no network connection on all the network interfaces
Send	Green: “Send” activity is normal. Red: Failure on a sender.
Receive	Green: “Receive” activity is normal Red: at least one active receiver has no audio source
Status	Green flashing if the unit is ok.

Fail-over	Green in case at least one output program has switched to a backup audio source
SD	Flashes when SDHC card is accessed

4.1.2. LCD display and keypad

Next menu or sub-menu

Previous menu or sub-menu

Previous item in the menu, or increase the selected value

Next item in the menu, or decrease the selected value

Validate the selected action

4.1.3. SDHC card reader

Supports SDHC cards, used for:

- saving/loading of the codec configuration
- Storing backup playlists and sound files

4.1.4. Headphones output

Allows audio monitoring.

The push button allows the selection of the audio source to be monitored.

For IQOYA X/LINK-ST and X/LINK-LE: encoder input, decoder output

For X/LINK-DUAL and X/LINK-AES67, it selects the audio that is assigned to vu-meter A or vu-meter B. This assignment is made from the LCD front panel and keypad.

4.1.5. Navigating menus on LCD display

Use the arrow keys to navigate in the menus, and the Ok key to confirm a choice.

System (Home page)	Eth1	Eth2	Eth3	Eth3	Monitoring A(*)	Monitoring B(*)	Status	(**)
Host Name	Enable	Enable	Enable 'yes)	Enable	Ana IN1 L Ana IN1 R	Ana OUT1 L Ana OUT1 R	Clock source: internal, AES IN1, AES IN2, PTP, 10 mHz Valeur (ex: 48000 Hz)	
Device Name	IPv4 @	IPv4 @	IPv4 @	IPv4 @	AES IN1 L AES IN1 R	AES OUT1 L AES OUT1 R	PTP OFF, Sync, Eth, source IP@	
System Time	Speed Mode	Speed Mode	Speed Mode	Speed Mode	Ana IN2 L Ana IN2 R	Ana OUT2 L Ana OUT2 R	SNMP: On / Off	
System Date	Link Status	Status	Status	Status	AES IN2 L AES IN2 R	AES OUT2 L AES OUT2 R	FTP: On/Off	
NTP Server URL1	Mac @	Mac @	Mac @	Mac @	AES67 IN1 L AES67 IN1 R	AES67 OUT1 L AES67 OUT1 R	NTP date and time On/Off	

NTP Server URL2	DHCP	DHCP	DHCP	DHCP	AES67 IN2 L AES67 IN2 R	AES67 OUT2 L AES67 OUT2 R	Audio Synchro on NTP On / Off - Sync / Not sync
Serial Number	Subnet mask	Subnet mask	Subnet mask	Subnet mask	AudioBus1 L AudioBus1 R	AudioBus1 L AudioBus1 R	Audio synchro on PTP On / Off - Sync / Not sync
Firmware version	Gateway	Gateway	Gateway	Gateway	AudioBus2 L AudioBus2 R	AudioBus2 L AudioBus2 R	Clock source: internal, AES IN1, AES IN2, PTP, 10 mHz Valeur (ex: 48000 Hz)
Analog bypass / AES bypass	Primary DNS	Primary DNS	Primary DNS	Primary DNS			
Apply factory settings	Alternate DNS	Alternate DNS	Alternate DNS	Alternate DNS			
Restart							
Halt							
Remount SD card							
Unmount SD card							
Copy config to SD							
Restore config from SD							
Firmware update							
Screen Dimmer							

(*) Note about Monitoring.
The name of the inputs and outputs displayed on the LCD screen are the names configured from the input and output settings WEB pages.

(**) **The following menus are only available in "Remote Broadcasting" mode of use:**

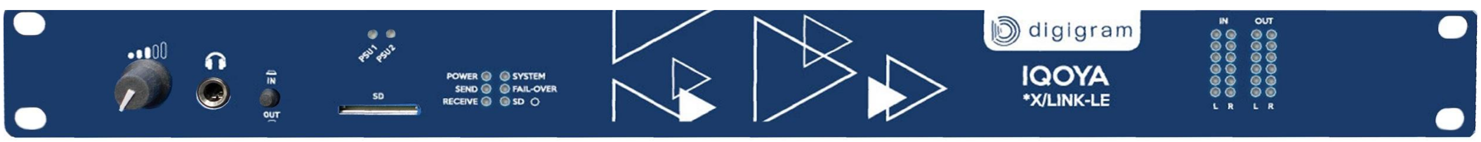
Select codec	(1) **Call C<#N>**	(2) **Contacts C<#N>**	(3) **Recent calls C<#N>**	(4) **Profiles C<#N>**
Codec instance #1: <I/O channels>: <Contact name> <SIP address or RTP listening port> *Ok key leads to submenu (1)*	CALL/ HANGUP <Contact to be called>	*Contact entry #1:* <Contact Name> <SIP address or IP address (****)>	*Recent call #1:* <Name of the remote> <SIP address or IP address (****)>	*Call profile #1:* <Call profile name>
Codec instance #2: <I/O channels>: <Contact name> <SIP address or RTP listening port> *Ok key leads to submenu (1)*	SELECT CONTACT (***) <Selected contact> *Ok key leads to submenu (2)*	*Contact entry #2:* <Contact Name> <SIP address or IP address (****)>	*Recent call #2:* <Name of the remote> <SIP address or IP address (****)>	*Call profile #2:* <Call profile name>
⋮	RECENT CALLS (***) <Selected recent call> *Ok key leads to submenu (3)*	⋮	⋮	⋮
Codec instance #N: <I/O channels>: <Contact name>	SELECT PROFILE (***) <Selected profile>	*Contact entry #N:* <Contact Name>	*Recent call #N:* <Name of the remote>	*Call profile #N:* <Call profile name>

<SIP address or RTP listening port> *Ok key leads to submenu (1)*

Ok key leads to submenu (4)	<SIP address or IP address (****)>	<SIP address or IP address (****)>	
LAST MESSAGE <Message following a call failure>			
CONTACT NAME <Contact name of this codec>			
REGISTRATION NAME <SIP address of this codec>			

(***) Items not available during a communication.
(****) Depending on whether the contact is accessible via SIP or Symmetric RTP

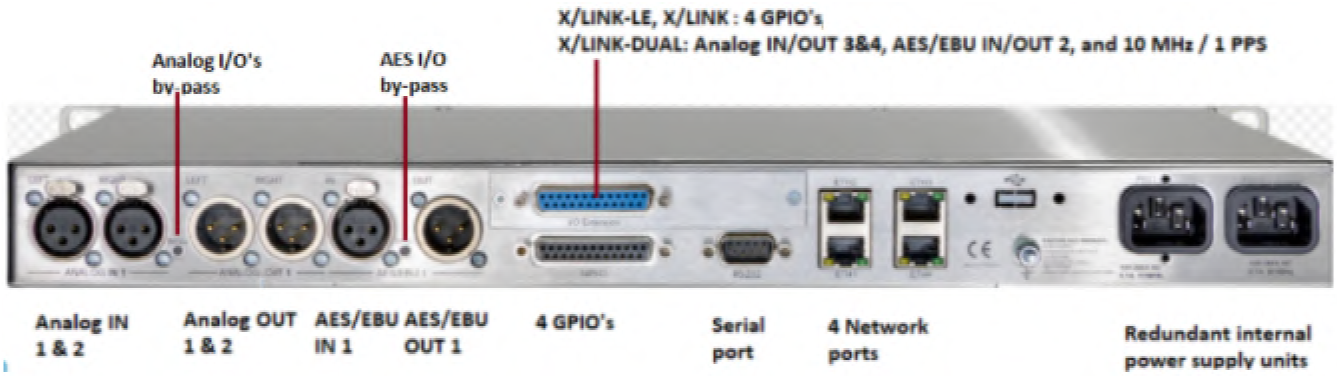
4.2. IQOYA X/LINK/LE front Panel



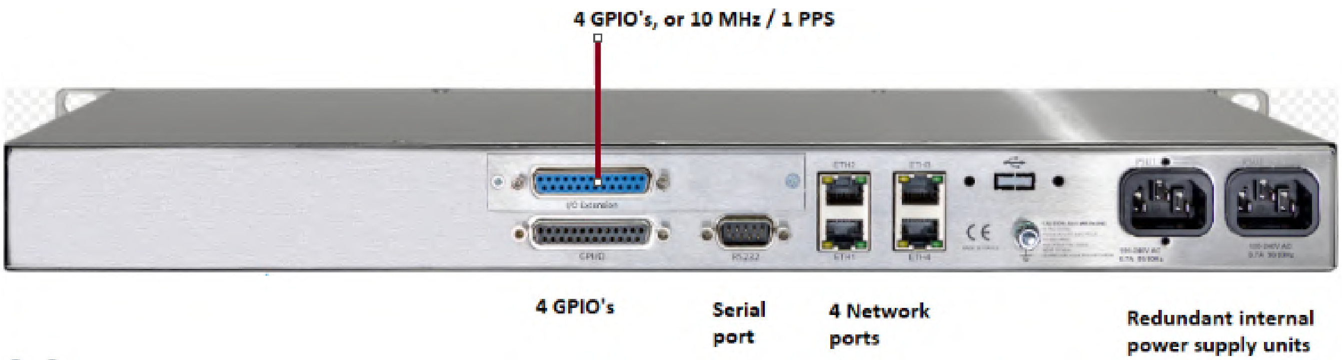
4.2.1. LEDs

PSU1	Status of the first internal power supply unit. Blue if PSU is ok. Off for PSU failure.
PSU2	Status of the second internal power supply unit. Blue if PSU is ok. Off for PSU failure.
Power	Green if internal power is ok
Send	Green: Send activity is normal. Red: at least one active sender has a failure
Receive	Green: Receive activity is normal Red: at least one active receiver has no audio source
System	Green flashing if unit is ok.
Fail-over	Green in case at least one output program has switched to a backup audio source
SD	Flashes when SDHC card is accessed

3.3 IQOYA X/LINK-ST, X/LINK-LE, X/LINK-DUAL back Panel



3.4 IQOYA X/LINK-AES67 back Panel



4 WORKING PRINCIPLES

IQOYA X/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 ...

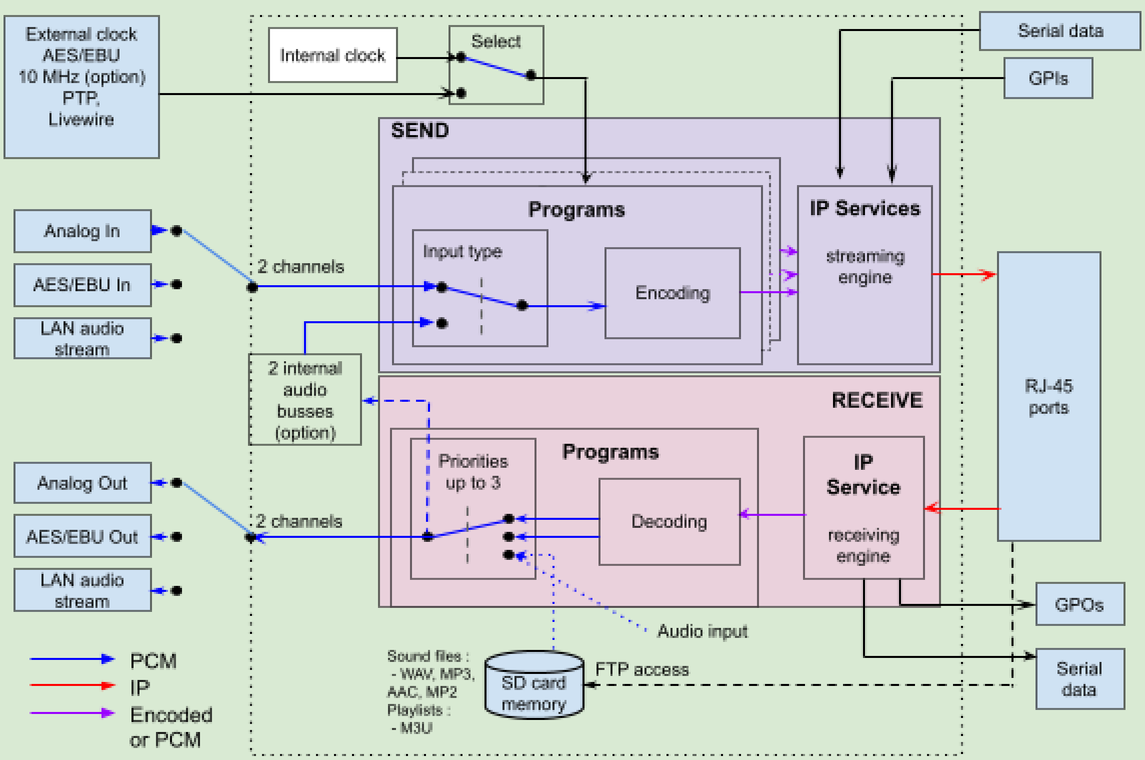
At first power up, the user is prompted to choose the mode of use either from the front panel (except IQOYA X/LINK-LE) or from the configuration web interface. Later it is possible to switch from one mode to another from the configuration web pages.

4.1 Working principles in "Program Distribution" mode of use

4.1.1 IQOYA X/LINK-ST & X/LINK-LE

IQOYA X/LINK allows at the same time:

- Encoding two audio channels in multiple audio formats, and streaming over IP
- Decoding IP audio streams to two output channels
- Transcoding IP audio streams (optional)



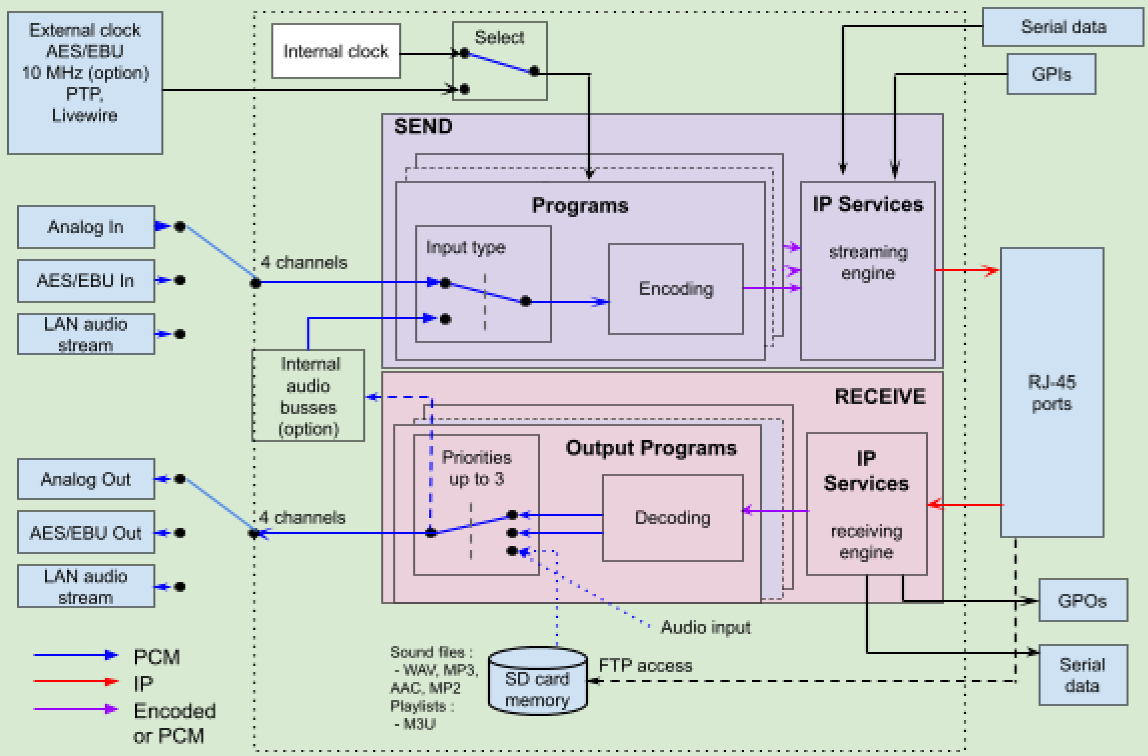
Schematic diagram of IQOYA X/LINK-ST & X/LINK-LE

4.1.2 IQOYA X/LINK-DUAL

IQOYA X/LINK-DUAL allows at the same time:

- Encoding four audio channels in multiple audio formats, and streaming over IP

- Decoding IP audio streams to four output channels
- Transcoding IP audio streams (optional)

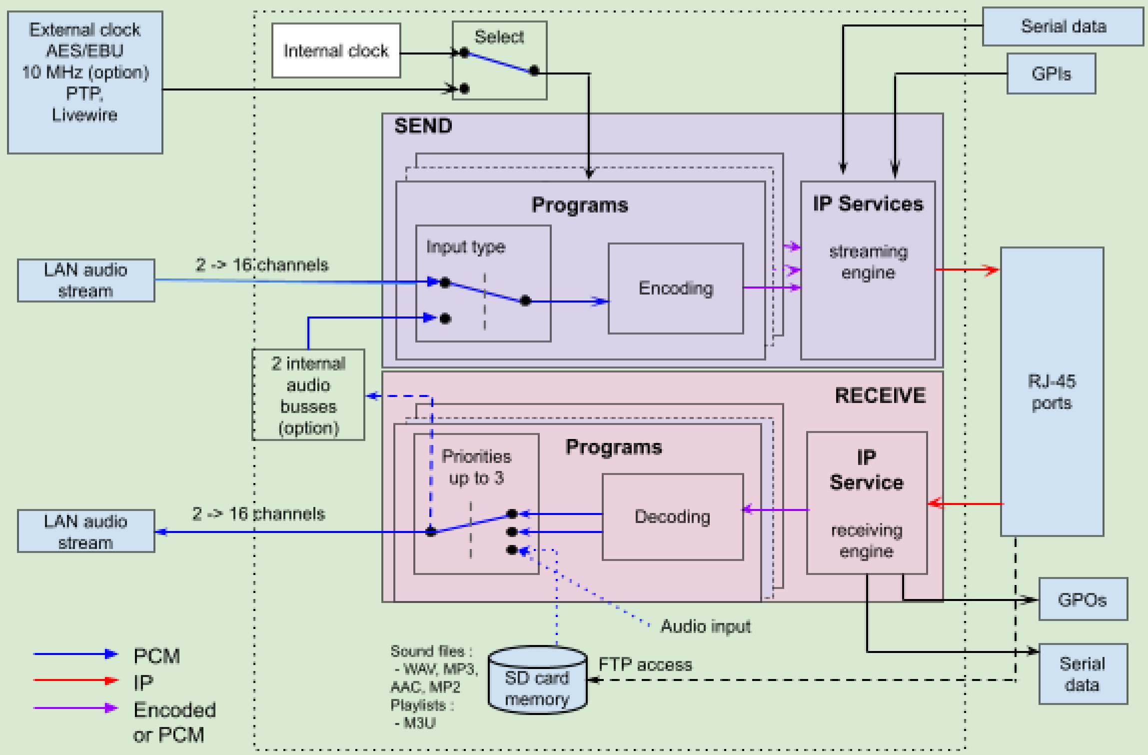


Schematic diagram of IQOYA X/LINK-DUAL

4.1.3 IQOYA X/LINK-AES67

IQOYA X/LINK-AES67 allows at the same time:

- Encoding 16 audio channels in multiple audio formats, and streaming over IP
- Decoding IP audio streams to 16 output channels
- Or transcoding IP audio streams (optional)



Schematic diagram of IQOYA X/LINK-AES67

4.1.4 Audio inputs and outputs

The audio inputs receive the audio signals to be encoded. They can be analog, or AES/3, or LAN audio (RAVENNA or AES67 or Livewire). Each source can be encoded several times at different formats, and streamed to several destinations. Audio samples decoded by X/LINK are played to the selected audio output. An audio output can be analog, or AES/3, or LAN audio (AES67 or RAVENNA or Livewire) .
Note that decoded audio samples can also be sent to internal audio buses, and audio buses can also be sources to be encoded. This optional feature is used for transcoding IP audio streams.

4.1.5 Programs

In the encoding section of IQOYA X/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 inputs that receive the signal to be encoded. The number of inputs is given by the audio mode. Stereo and multi-channel modes refer to consecutive inputs.
- the audio format: algorithm, bitrate, sampling frequency.

On the decoding part of IQOYA X/LINK, a program is the decoding of an audio source to the audio output. 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),
- a HTTP stream (Icecast/Shoutcast),
- audio inputs,
- sound files or playlists stored locally.

4.1.6 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.

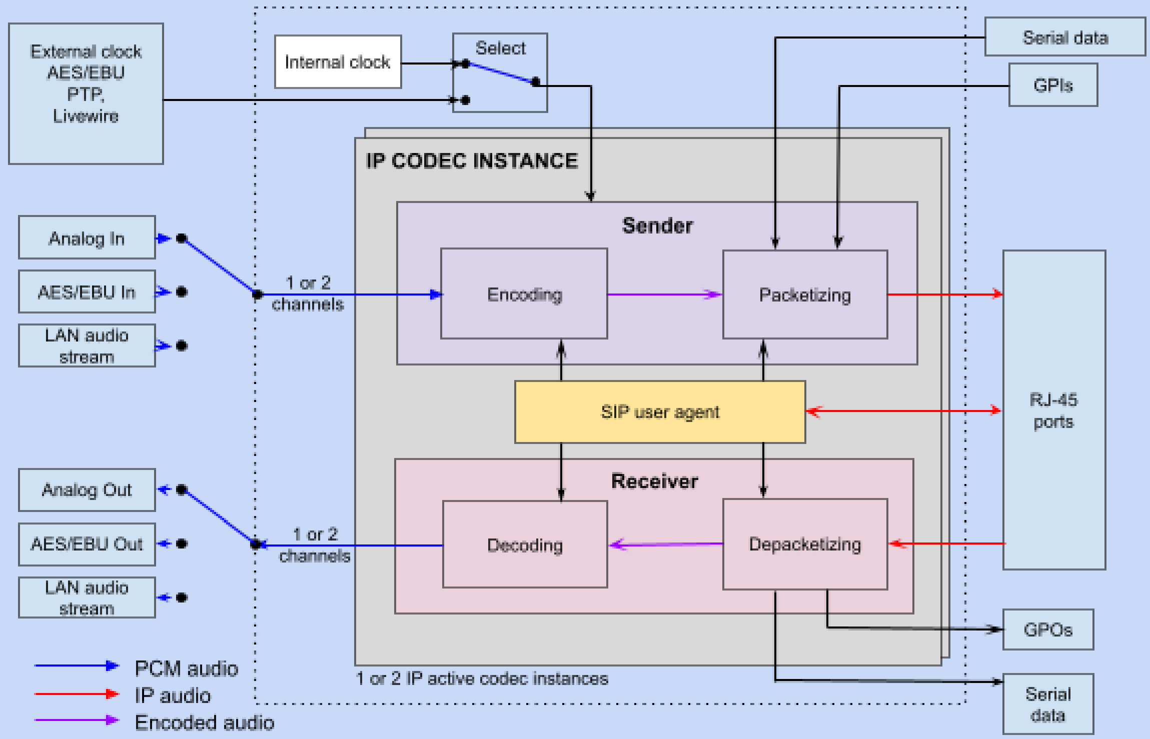
4.1.7 Audio buses

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.

4.2 Working principles in “Remote Broadcasting” mode of use

4.2.1 IQOYA X/LINK-ST & X/LINK-LE

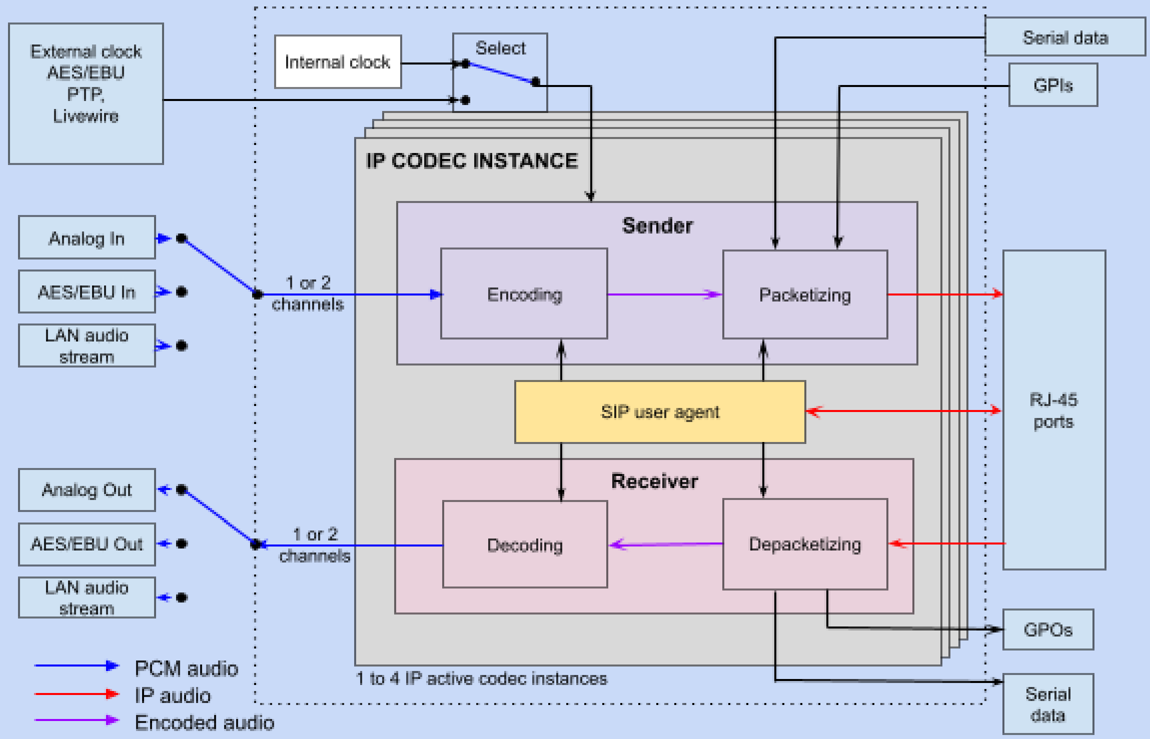
IQOYA X/LINK-ST and IQOYA X/LINK-LE can run one stereo IP codec instance or two mono IP codec instances. Each instance of an IP codec allows to receive, establish or terminate one Symmetric RTP or SIP IP audio connection. Before establishing a connection, the user chooses the recipient in the address book or enters the recipient address manually and chooses the call profile in the call profile list.



Schematic diagram of IQOYA X/LINK-ST & X/LINK-LE

4.2.2 IQOYA X/LINK-DUAL

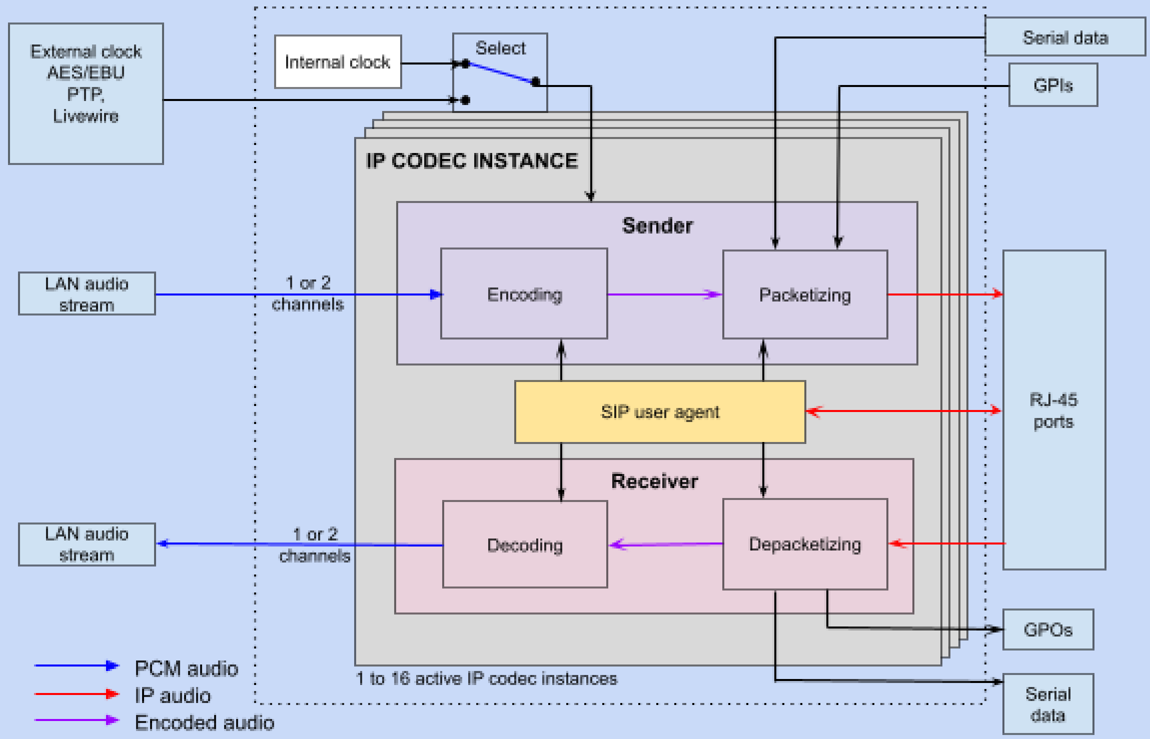
IQOYA X/LINK-DUAL can run two stereo IP codec instances or one stereo and two mono IP codec instances or four mono IP codec instances. Each instance of an IP codec allows to receive, establish or terminate one Symmetric RTP or SIP IP audio connection. Before establishing a connection, the user chooses the recipient in the address book or enters the recipient address manually and chooses the call profile in the call profile list.



Schematic diagram of IQOYA X/LINK-DUAL

4.2.3 IQOYA X/LINK-AES67

IQOYA X/LINK-AES67 can run 8 stereo IP codec instances or 16 mono IP codec instances or any combination of mono and stereo IP codec instances whose total number of audio channels is less than 16. Each instance of an IP codec allows to receive, establish or terminate one Symmetric RTP or SIP IP audio connection. Before establishing a connection, the user chooses the recipient in the address book or enters the recipient address manually and chooses the call profile in the call profile list.



Schematic diagram of IQOYA X/LINK-AES67

4.2.4 Audio inputs and outputs

The audio inputs receive the audio signals to be encoded by the IP codec instance(s). They can be analog, or AES/3, or LAN audio (AES67 or RAVENNA or Livewire).
Audio samples decoded by the IP codec instance(s) are played to the audio outputs. An audio output can be analog, or AES/3, or LAN audio (AES67 or RAVENNA or Livewire).

4.2.5 IP codec instances

A codec instance can establish a connection with a remote IP codec, accept or refuse a connection request from a remote IP codec, or terminate an established connection. The connections can be SIP, direct SIP or symmetrical RTP.
A stereo (resp. mono) codec instance is associated with a stereo (resp. mono) audio input and a stereo (resp. mono) audio output by configuration. Once a connection has been established, the IP codec instance encodes, packetizes and sends over IP to the remote IP codec the audio samples received from the audio input and, at the same time, it depacketizes and decodes the IP audio stream received from the remote IP codec then push the audio samples to the audio output.

4.2.6 Contacts and Address book

A contact is a SIP address (for SIP connections) and/or an IP address (for Symmetrical RTP connections) that has been named. The address book is the list of all the contacts defined on the equipment. Usually the address book of the studio codecs are populated with the addresses of the field codecs and vice versa.

4.2.6 Call profiles and Call profile list

A call profile is a named set of audio and network parameters used to define the characteristics of a connection and applied at connection establishment. The call profile list is the list of all the call profiles defined on the equipment. The parameters of a call profile are:

- The audio encoding format of the sent stream
- The payload type of the outgoing audio stream
- The packet size of the outgoing audio stream
- The FEC (Forward Error Correction) scheme or dual streaming scheme of the outgoing audio stream
- The outgoing stream QoS (Quality of Service)
- The size of the jitter buffer recommended by the caller to the callee
- The jitter buffer size of the caller
- The audio encoding format expected for the stream sent by the remote
- The payload type expected for the stream sent by the remote
- The FEC (Forward Error Correction) scheme or dual streaming scheme expected for the stream sent by the remote

5. INSTALLATION

5.1. Grounding the IQOYA X/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.
The equipment’s PE (protective earth) and building’s PE must be connected (use of a socket outlet with PE is required).
The sockets outlets to which the equipment is connected must be easily accessible.
Once IQOYA is installed and properly grounded, you can connect the Eth ports and audio I/Os as required for your installation.

5.2. Connecting IQOYA X/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 X/LINK Eth ports are:

- Eth1: 192.168.0.100
- Eth2: 192.168.1.100
- Eth3: 192.168.2.100
- Eth4: 192.168.3.100

In case you do not know the IP addresses of the IQOYA X/LINK unit you want to connect to, you can read its IP addresses from the front panel (see paragraph “LCD display and keypad”), except for IQOYA X/LINK-LE where the IP addresses are written on the inserted SDHC card at startup (the SD card is not delivered by Digigram).

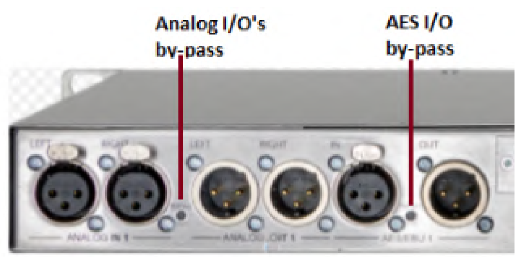
Make sure all other devices connected to this LAN are in the same subnet and have different IP addresses (this includes the PC from which you will connect to the IQOYA codec to configure).

WARNING:

- Eth1, Eth2, Eth3 and Eth4 must belong to different subnetworks.
- Eth1, Eth3 and Eth4 are Gbits interfaces.
- Eth2 is a 100 Mbits/s interface. **It is recommended to use one of the other interfaces for LAN audio connectivity (AES67, RAVENNA, Livewire).**

5.3. Enabling / disabling the hardware bypass function

IQOYA X/LINK, X/LINK-LE and X/LINK-DUAL allow for the hardware bypass of audio inputs to audio outputs in case of power supply failure. This concerns analog inputs & outputs 1&2, and AES/3 input & output 1.

<table><tr><td>
</td><td>Enabling / disabling the hardware pass</td></tr></table>

5.4. Powering up IQOYA X/LINK

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

5.5 Steps to follow to configure IQOYA X/LINK in “Program Distribution” mode of use

Set the global parameters of your IQOYA X/LINK
If IQOYA is used for encoding:

- Adjust the parameters of the audio inputs: type (analog, AES3, AES67, RAVENNA, Livewire), and gain.
- Declare the programs (encodings)
- Declare the IP services to be streamed over IP (IP audio streams)

If IQOYA is used for decoding:

- Adjust the parameters of the audio outputs: type (analog, AES3, AES67, RAVENNA, Livewire), and gain.
- Declare the IP services to be received from the network (IP audio stream)
- Declare the output program(s)
- Check the status and metrics on the output programs.

If IQOYA is used for transcoding:

- Declare the IP services to be received from the network
- 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
- Check the status and metrics on the output programs.

5.6 Steps to follow to configure IQOYA X/LINK in “Remote Broadcasting” mode of use

- Set the network parameters of your IQOYA X/LINK
- Adjust the parameters of the audio inputs: name, type (analog, AES3, AES67, RAVENNA, Livewire), and gain.
- Adjust the parameters of the audio outputs: name, type (analog, AES3, AES67, RAVENNA, Livewire), and gain.
- Declare the SIP accounts
- Declare the IP codec instances

6. Accessing IQOYA X/LINK WEB pages

From a WEB browser, connect to the IQOYA X/LINK WEB pages:

- for a network connection through Eth1 port, enter https://192.168.0.100 (this is the default IP address of Eth1).
- for a network connection through Eth2 port, enter https://192.168.1.100 (this is the default IP address of Eth2).
- for a network connection through Eth3 port, enter https://192.168.2.100 (this is the default IP address of Eth3).
- for a network connection through Eth4 port, enter https://192.168.3.100 (this is the default IP address of Eth4).

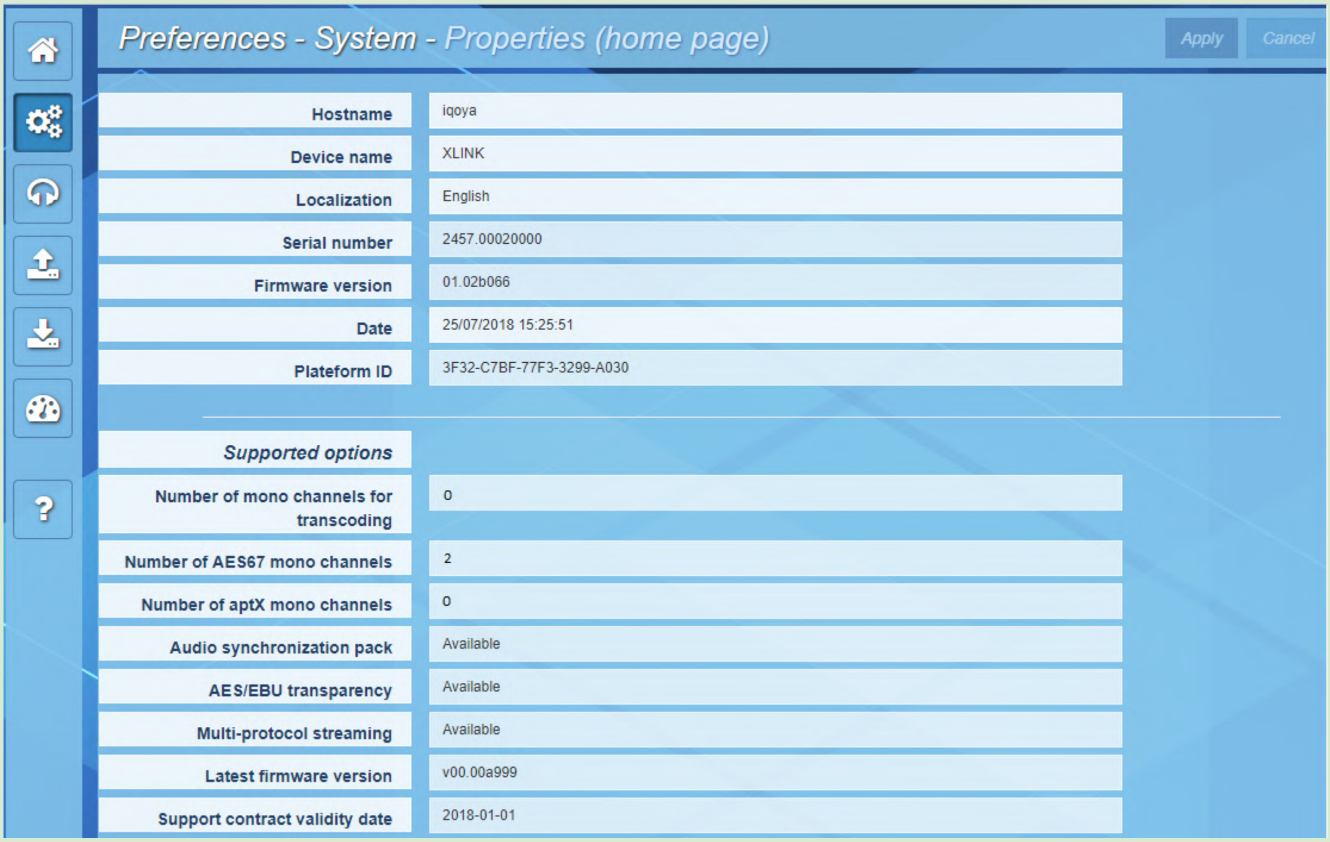
The WEB browser displays a message about security certificate. Select the option that allows to continue with this WEB server.
Enter the requested username and password. The default administrator login is:
username = iqoya
password = iqoya

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

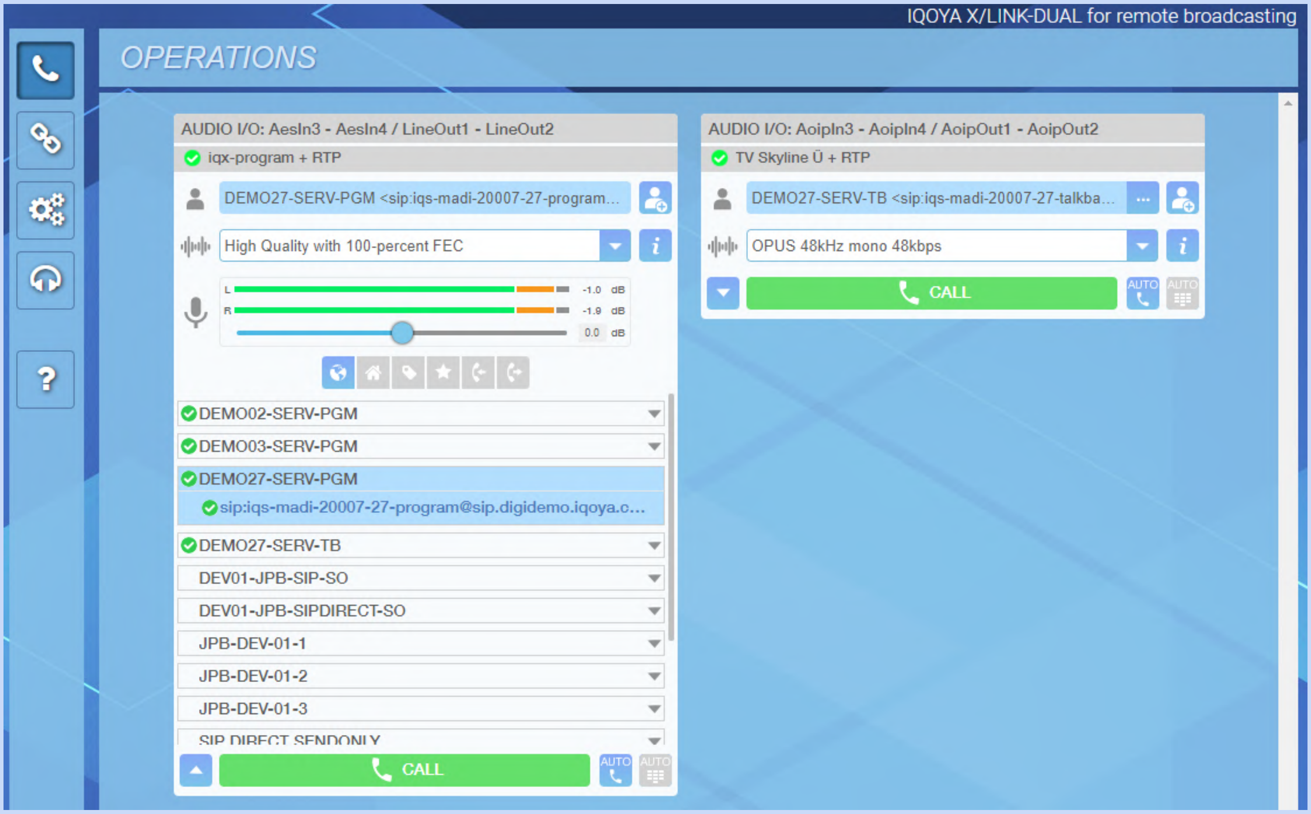
- **“Administrator” category**
  A user from the “Administrator” category has all the access rights on the WEB pages.
  The login to the embedded WEB server as “Administrator” is:
  - username: iqoya
  - default password: iqoya
  Username and password can only be modified when logged as Administrator. 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.

Once the login has passed:

- In “Program Distribution” mode of use the “Properties” WEB page is displayed. This is the home page.



- In “Remote Broadcasting” mode of use the “Operations” WEB page is displayed. This is the home page.



7. Configuration from the WEB pages

Click on the “value” field of a 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” on the top right of the page to confirm the settings, or “Cancel to ignore the changes.

8. WEB pages organization

8.1 WEB pages organization in “Program Distribution” mode of use

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

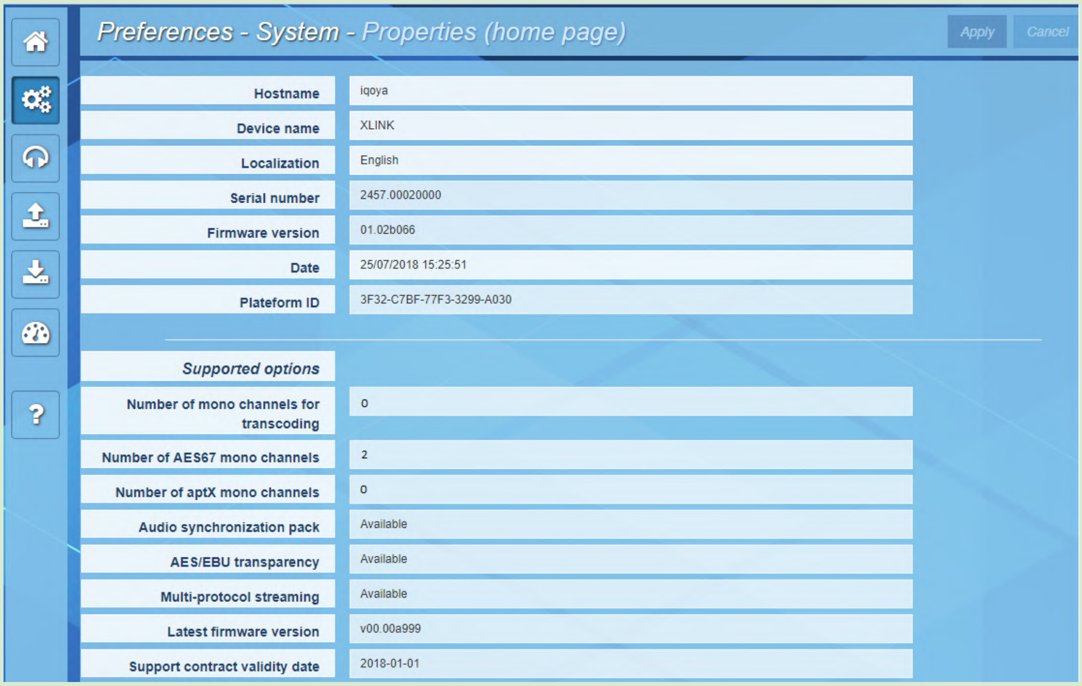
Icon	Category	Description
	Home page	Displays the properties of the unit as well as its software options
	Preferences	Global parameters of the unit.
	Audio I/Os	Audio inputs and outputs parameter settings: name, type selection, audio level adjustment, vu-meters
	Encoders	Settings of programs (encodings 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	About IQOYA X/LINK and this user manual.

8.1.1 “Preferences” category of menus

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

8.1.1.1 Preferences –> System

8.1.1.1.1 Preferences –> System –> Properties

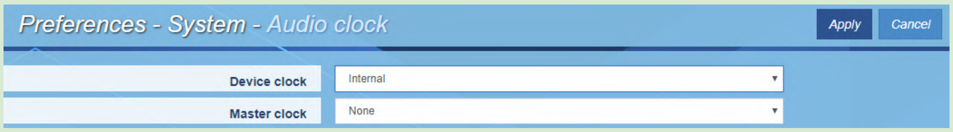


Parameter	Read/Write	Meaning
Hostname	R / W	Logical name given to the device on the network.
Device Name	R / W	Name given to the equipment
Localization	R / W	Language
Serial number	R	Serial number of the unit. This number is set in factory and cannot be changed.
Firmware version	R	Version of the firmware 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.
Supported Options		
Number of mono channels for transcoding	R	Number of mono channels supported for transcoding through internal buses.
Number of AES67 mono channels	R	Number of mono input and output channels on AES67 or Ravenna, or Livewire
Number of aptX mono channels	R	Number of mono channels to be processed in aptX
Audio synchronous pack	R	Value 1: the codec features the audio synchronization via NTP Value 0 : the option is not installed.
AES/3 transparency	R	Value 1: the codec allows for AES transparency transport. Value 0; the option is not installed.

Multiprotocol streaming	R	Value 1: the codec features the multiprotocol streaming. Value 0: the option is not installed
Latest firmware version	R	Maximum firmware version number authorized by the ongoing support contract.
Support contract validity date	R	Defines the date until when the firmware can be updated/upgraded according to the purchased support contract.

8.1.1.1.2 Preferences –> System –> Audio Clock

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



Device clock

The clock source can be:

- Internal: on-board clock
- Extracted from an AES/3 input (not available on X/LINK-AES67)
- A PTP clock (AES67, RAVENNA)
- A Livewire clock

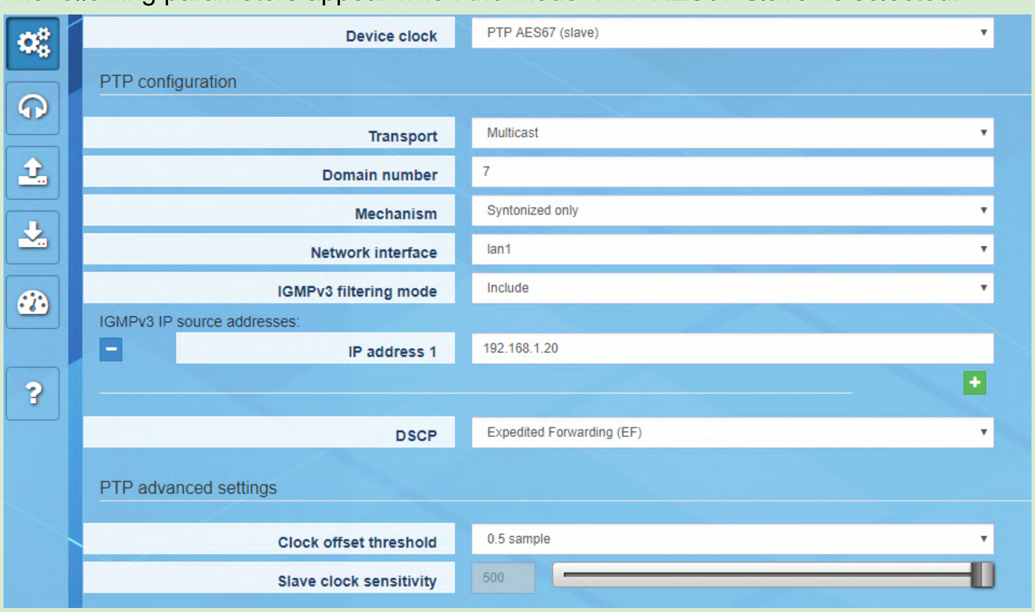
The clock sampling frequency value is set from Preferences–>Audio setup.

Master clock

Allows defining if the codec generates a PTP clock.

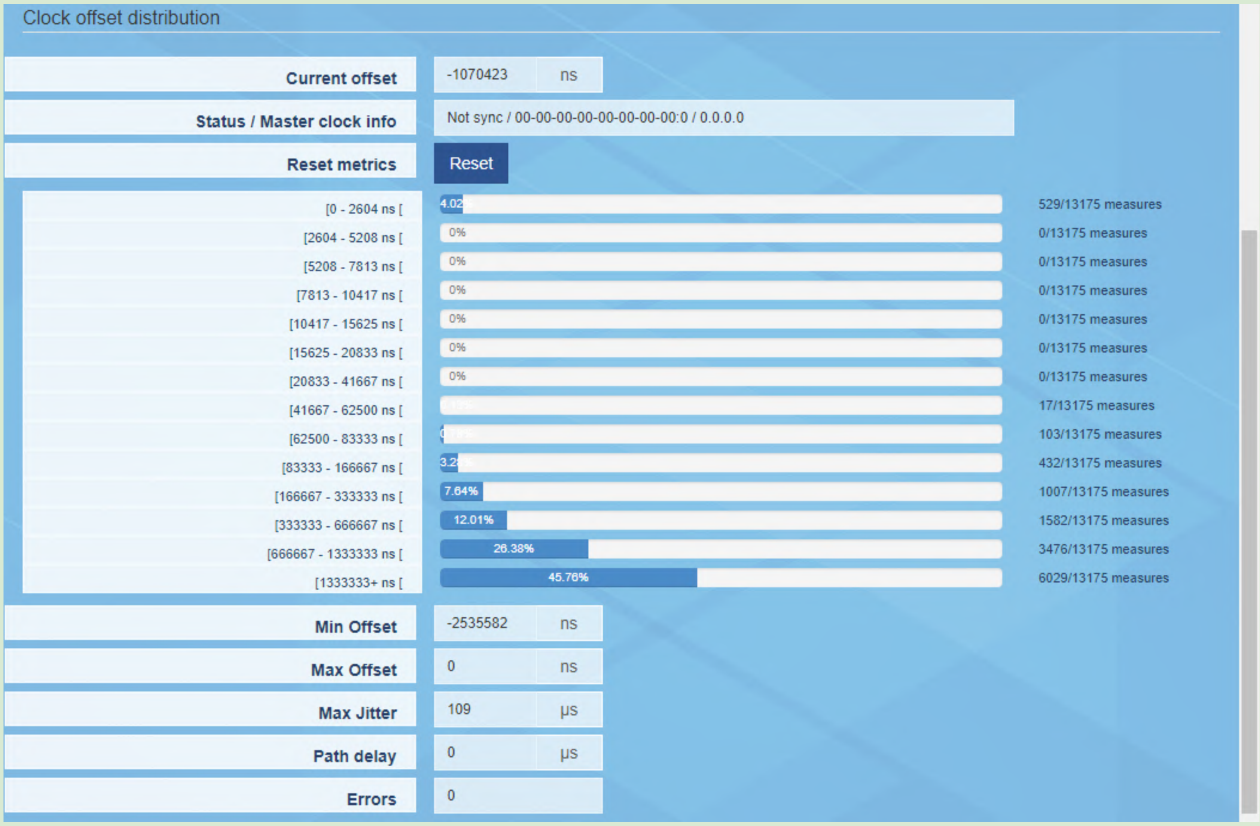
8.1.1.1.2.1 PTP clock source

The following parameters appear when the mode “PTP AES67 Slave” is selected:



Transport	R/W	Allows specifying if the PTP clock is unicast or multicast.
Domain number	R/W	PTP clock domain number (from 0 to 128)
Mechanism	R/W	**Syntonized**: means that IQOYA's clock is the same as the Grandmaster PTP, but they are not synchronous (delay between the two clocks). Synchronous clock is obtained thanks to E2E or P2P modes, which serve to compensate the delay between Grandmaster PTP clock and IQOYA. **E2E** is a more universal setting (it consists of requests and answers between the node (IQOYA) and the Grandmaster PTP clock unit). **P2P** provides higher clock sync precision but requires full PTP support from all participating switches (between IQOYA and related clock master.) In case the PTP clock is generated by an IQOYA, the PTP mechanism must be the same as in the IQOYA master: syntonized.
Network interface	R/W	Select the network interface that receives the PTP
IGMPv3 filtering mode	R/W	Off: X/LINK subscribes to the multicast PTP clock which can be generated by any source IP address. Include: X/LINK subscribes to the multicast PTP clock which is generated only by the listed source IP addresses. Exclude: X/LINK subscribes to the multicast PTP clock which is generated by any source IP address, with exception of the listed IP addresses..
IGMPv3 IP source addresses		
IP address x	R/W	Allows declaring the source IP addresses to be included or excluded. Click on to add an IP@ to the list.
DSCP	¨R/W	QoS assigned to the PTP frames. Select the value from the drop down list. For optimal QoS on PTP, "Expedited forwarding (EF)" value is recommended.
PTP advanced settings		
Clock offset threshold	R/W	This parameter defines the condition for being synchronized to the PTP clock. The lower the value, the better the phase with the PTP clock. Lower values require a deterministic network. For networks that introduce an erratic jitter to the PTP frames, the value must be increased. Default value is 0.5 sample. It can be increased up to 64 samples.
Slave clock sensitivity	R/W	It defines the sensibility of the slave clock to the PTP packet jitter. Enter a value between 500 (for a high sensitivity) and 100 (for a low sensitivity). Default value is 500

The *clock offset distribution* section displays information about the received PTP clock.

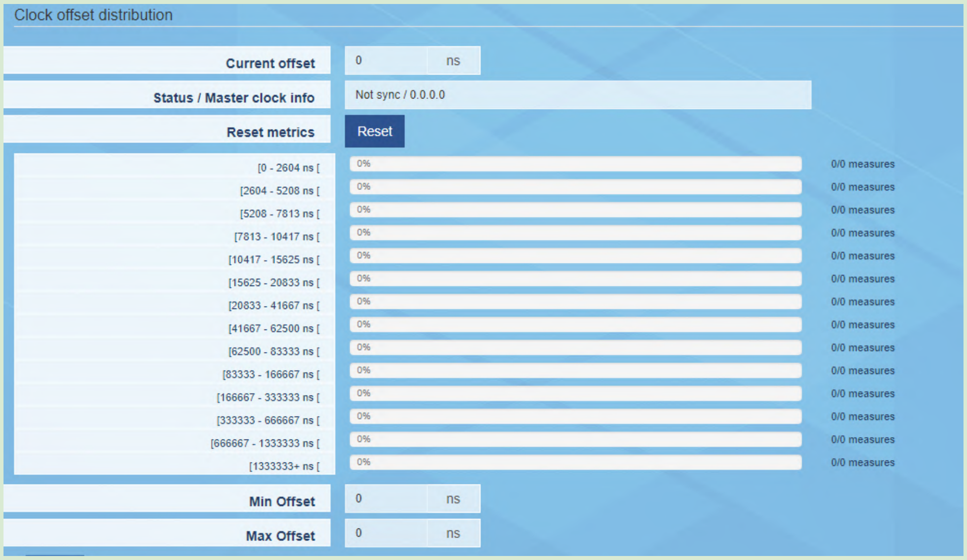


8.1.1.1.2.2 Livewire (Slave)

The following parameters appear when the mode “Livewire Slave” is selected:

Network interface	R/W	Select the network interface that receives the livewire clock.
IGMPv3 filtering mode	R/W	Off: X/LINK subscribes to the Livewire clock which can be generated by any source IP address. Include: X/LINK subscribes to the Livewire clock which is generated only by the listed source IP addresses. Exclude: X/LINK subscribes to the Livewire clock which is generated by any source IP address, with exception of the listed IP addresses..
IGMPv3 IP source addresses		
IP address x	R/W	Displayed if IGMPv3 filtering mode is set to “Exclude” or “Include”. Allows declaring the source IP addresses to be included or excluded. Click on to add an IP@ to the list.

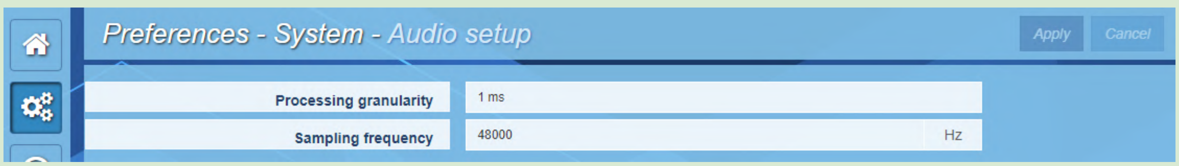
The *clock offset distribution* section displays information about the received Livewire clock.



Click on "Apply" to confirm your choice.

8.1.1.1.3 Preferences –> System –> Audio setup

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



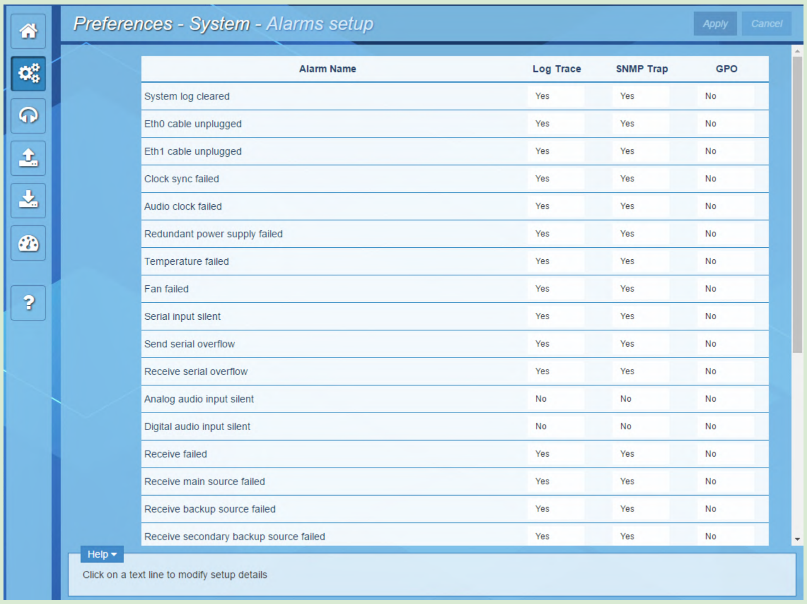
Click on a parameter field to be able to change the values.

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. Note: 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 frequency (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.

Click on "Apply" to confirm your changes.

8.1.1.1.4 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



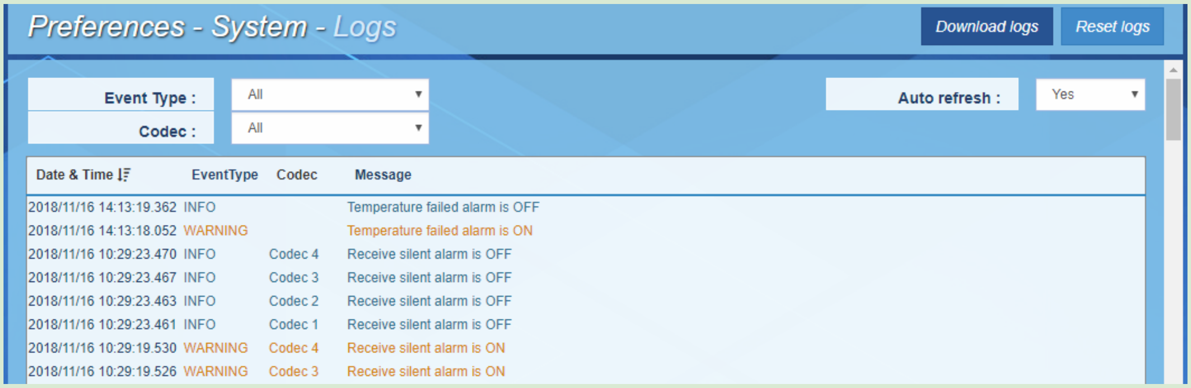
Click on a parameter field to be able to change the values.
Click on “Apply” to confirm your changes.

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	No data received on serial port
Send serial overflow	Serial data can't be all sent
Receive serial overflow	Received serial data can't be all extracted and output
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 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 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.
Receive silent	Audio in the IP stream is silent according to the silence criteria.

8.1.1.1.5 Preferences –> System –> Logs

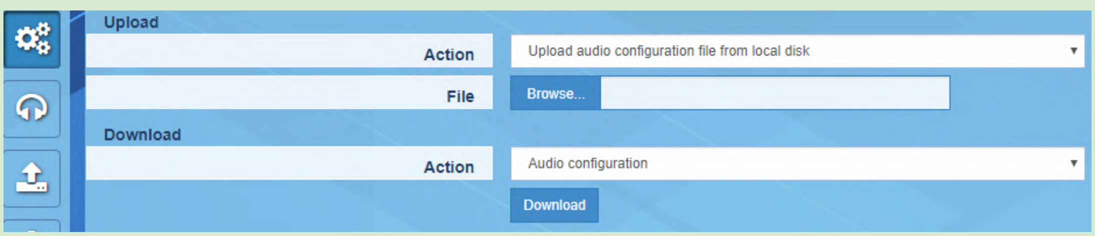


This page allows viewing and downloading the log file of IQOYA X/LINK. This log file gives information about the internal behaviour of IQOYA, and is useful for advanced diagnostics. Traces of enabled alarms are written into this log file (alarm ON, alarm OFF). This log file is stored internally and is 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: The page content is refreshed automatically if this parameter is set to “Yes”.
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 remotely downloading the log traces.

8.1.1.1.6 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:

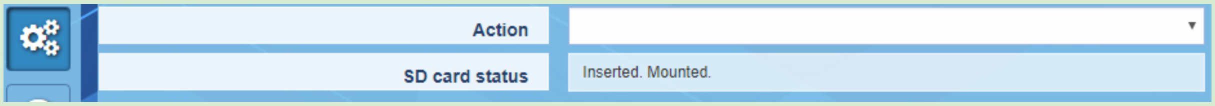
- The audio configuration only (includes the programs and IP services)
- The full codec configuration

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.

8.1.1.1.7 Preferences –> System –> SD card

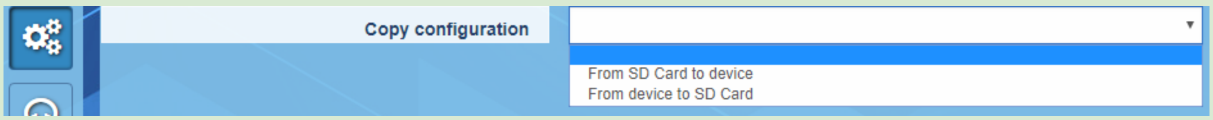
This page allows:

- mounting an SDHC card if it is inserted while the unit is running,
- unmounting it before removing it from the front panel.
- Viewing the SDHC card status: mounted/unmounted



8.1.1.1.8 Preferences –> System –> SD card backup

The codec configuration can be saved to SDHC card or loaded from it.



- From the "Copy configuration" field, select whether the configuration has to be copied from the SDHC card to IQOYA's internal memory or from the internal memory to the SDHC card.

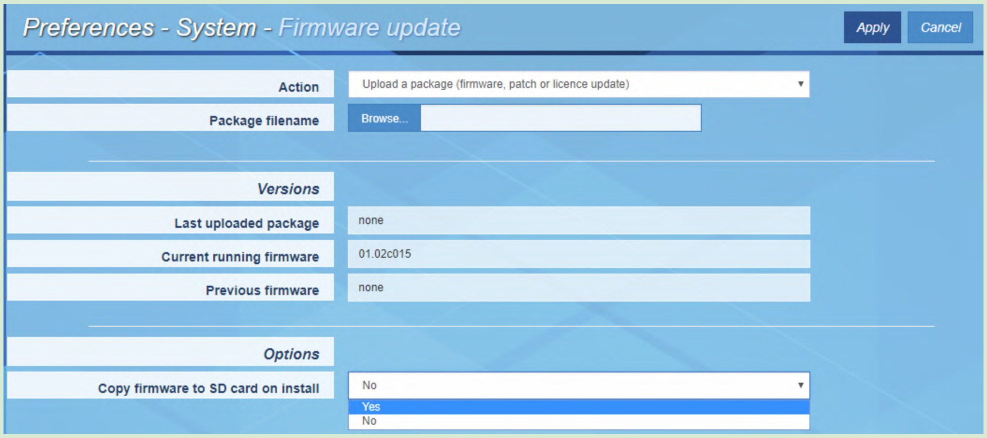
Notes:

- Audio activity is stopped when the configuration is loaded from the SDHC card.
- The unit is restarted to apply the new configuration.
- On the SDHC card, the configuration file "IQOYA_Configuration_save.tar" is stored in folder \IQOYA_LINK\Config.
- The current configuration of the IQOYA codec can also be displayed from a WEB browser by selecting the file \IQOYA_LINK\ Config.html, accessible via FTP.
- The configuration saved on the SDHC card can be loaded from the IQOYA X/LINK front panel LCD display and keyboard (menu System)
- This configuration on SDHC card can also be loaded when starting IQOYA with the SD card inserted. The file "/SDCARD/iqoya_link/run_once/ boot_commands.txt" must contain the following line:
  `RESTORE_FULLCONFIG_FROMSD=Yes`

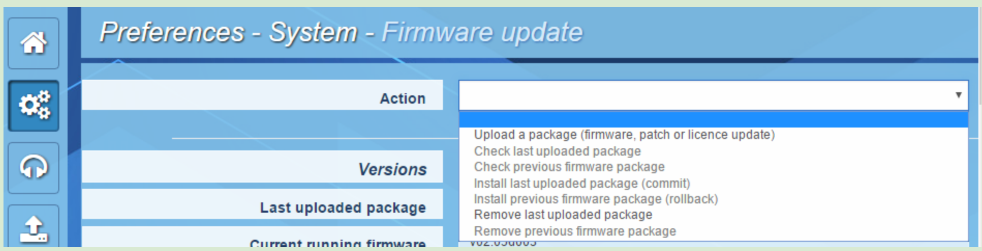
8.1.1.1.9 Preferences –> System –> Firmware update

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 users 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 upload 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.

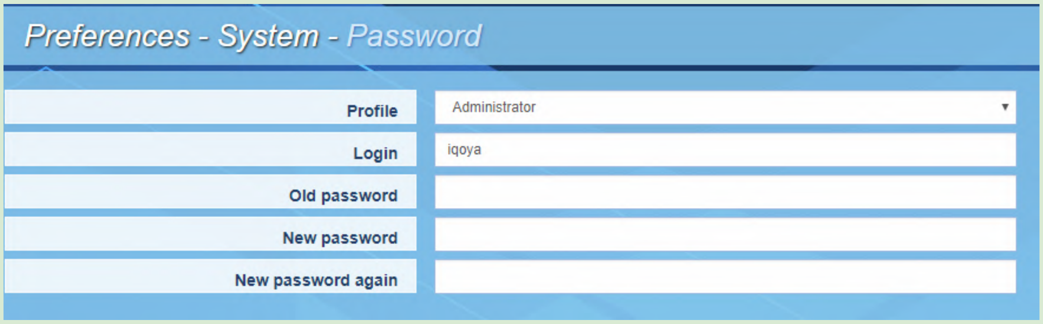
Copy firmware to SD card on install
Set to Yes, this parameter allows copying to the SD card the firmware to be installed to facilitate a future possible firmware rollback. Exemple:

1) Firmware to upload and apply = version A
   Copy to SD card set to Yes
2) Firmware to upload and apply=: version B
   Copy to SD card set to Yes

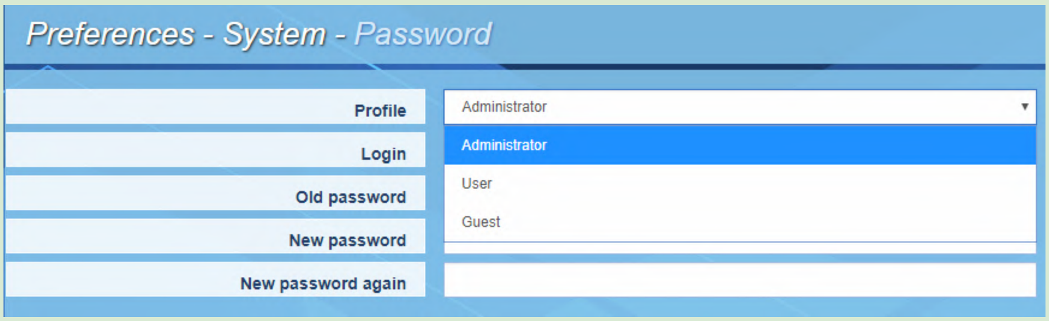
=> Current firmware = version B / Previous firmware = version A
At this point version A can be re-installed without the upload phase.

8.1.1.1.10 Preferences -> System -> Password

This page allows changing the username and 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.

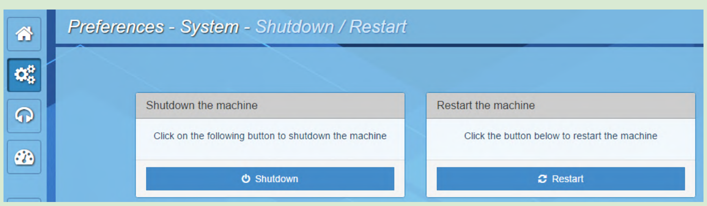


Login: allows configuring the username to be used in order to log to the WEB GUI with the selected profile.

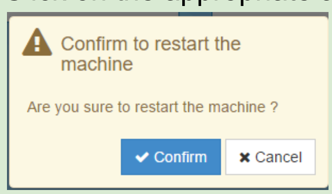
Old password: Type the current password
New password: Type the new password
New password again: confirm the new password
Click on “Apply” to confirm the changes.

8.1.1.1.11 Preferences -> System -> Shutdown / Restart

This page allows you to restart or shutdown IQOYA.



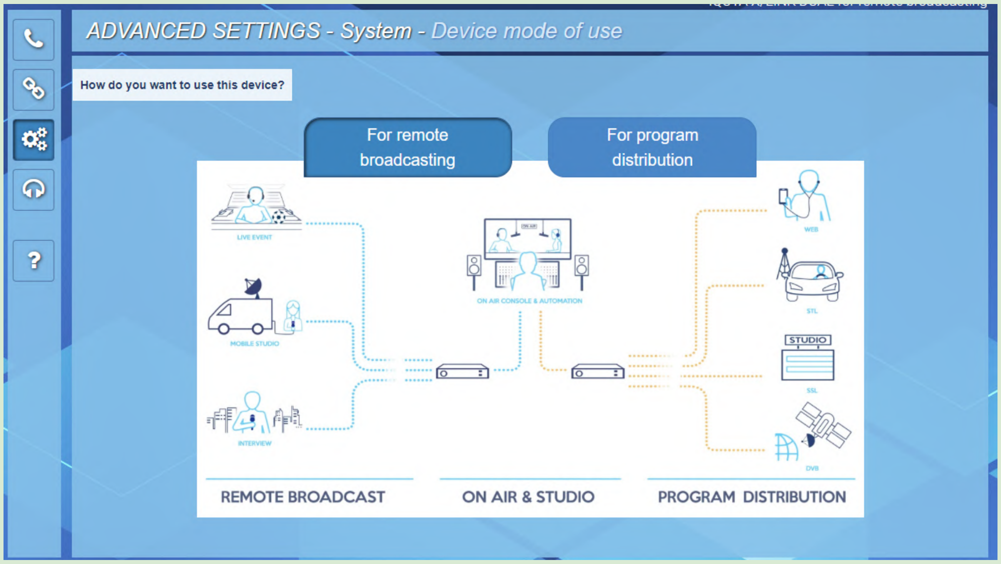
Click on the appropriate action.

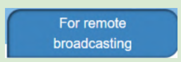


Confirm or cancel your choice through the displayed confirmation window.

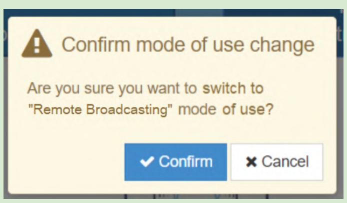
8.2.3.1.11 Preferences –> System –> Switch mode of use

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



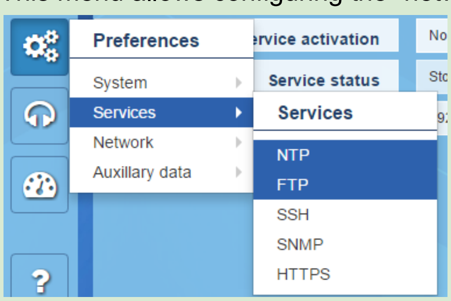
To switch to “Remote Broadcasting” mode of use, click 

button then confirm your choice through the displayed confirmation window:



8.1.1.2 Preferences –> Services

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



8.1.1.2.1 Preferences –> Services –> NTP

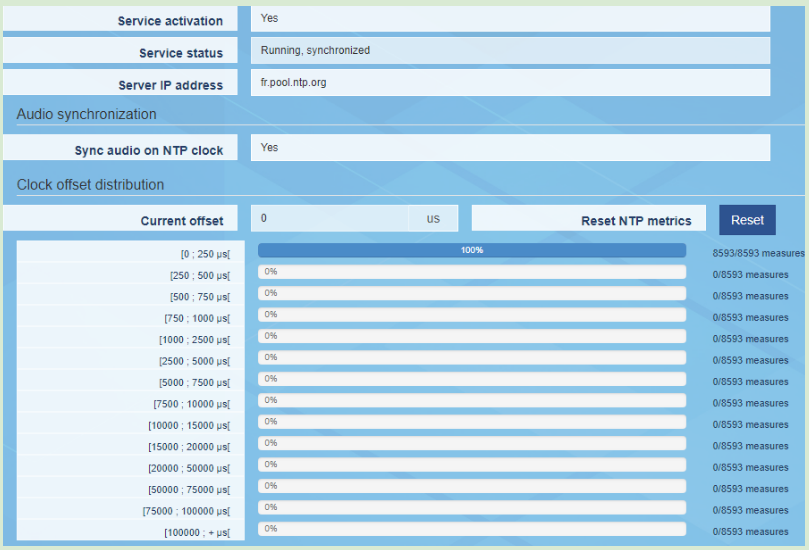
This page allows:

- 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 then 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".

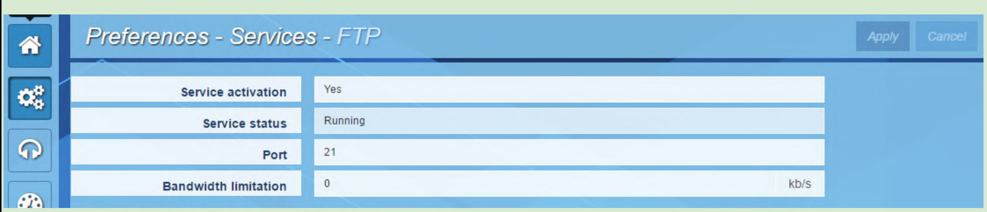
For activation of the NTP based audio synchronization, select "Yes" for parameter "**Sync audio on NTP clock**".



Once IQOYA is synchronized on the NTP server, the field "Service status" displays "Running, synchronized".
This requires that the software option is installed on the IQOYA X/LINK, as well as on the associated IQOYA decoders.

8.1.1.2.2 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.



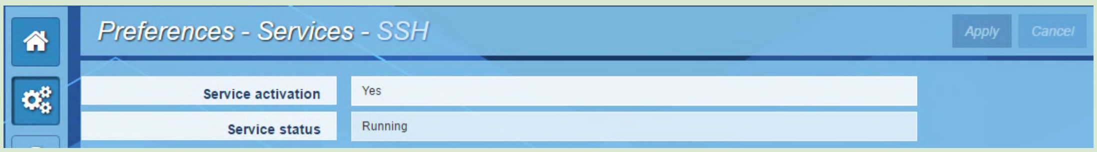
Click on the "Service activation" field. Select "Yes" to enable the FTP service, "No" to disable it.
If necessary, you may change the port used for FTP (default value is 21).
Parameter "Bandwidth limitation" allows limiting the network bandwidth of the FTP traffic.
Click on "Apply" to confirm the changes.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

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

Note that backup playlists and sound files have to be stored in the folder DEVICE_STORAGE.

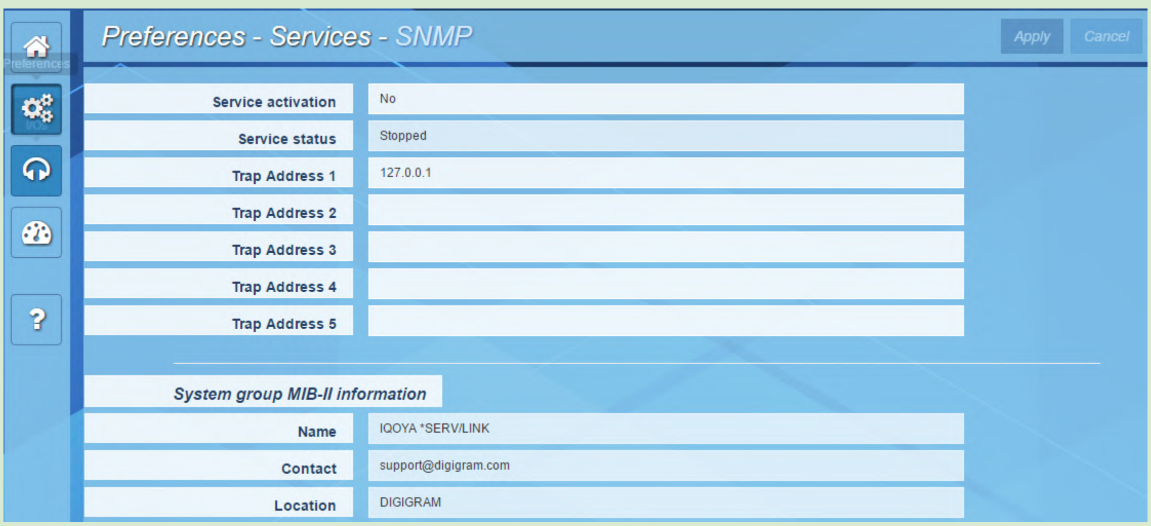
8.1.1.2.3 Preferences –> Services –> SSH

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



8.1.1.2.4 Preferences –> Services –> SNMP

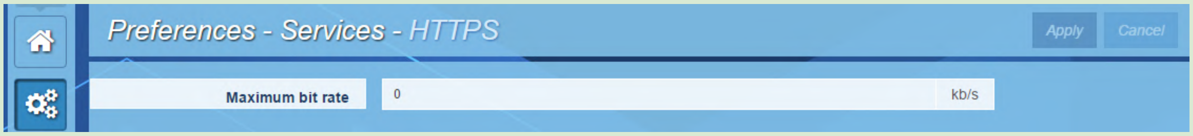
This page allows setting the SNMP parameters.
It also displays the System group MIB-II information.



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

8.1.1.2.5 Preferences –> Services –> HTTPS

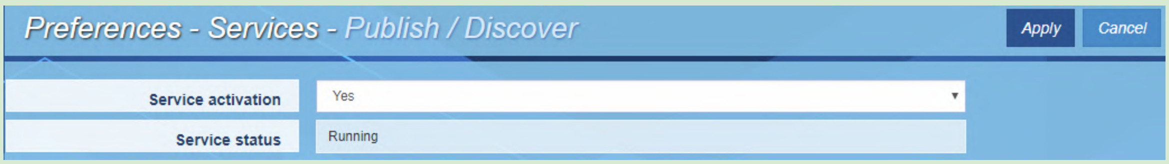
This page allows setting a bandwidth limitation to the HTTP traffic.
In case the IP audio stream takes almost all the available network bandwidth, the HTTP traffic generated when accessing the WEB pages may disturb the IP audio frames transmission, because the total bandwidth necessary for the IP audio 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.
The smaller the value, the longer it takes to load the WEB page!
Click on “Apply” to confirm the settings.

8.1.1.2.6 Preferences –> Services –> Publish / Discover

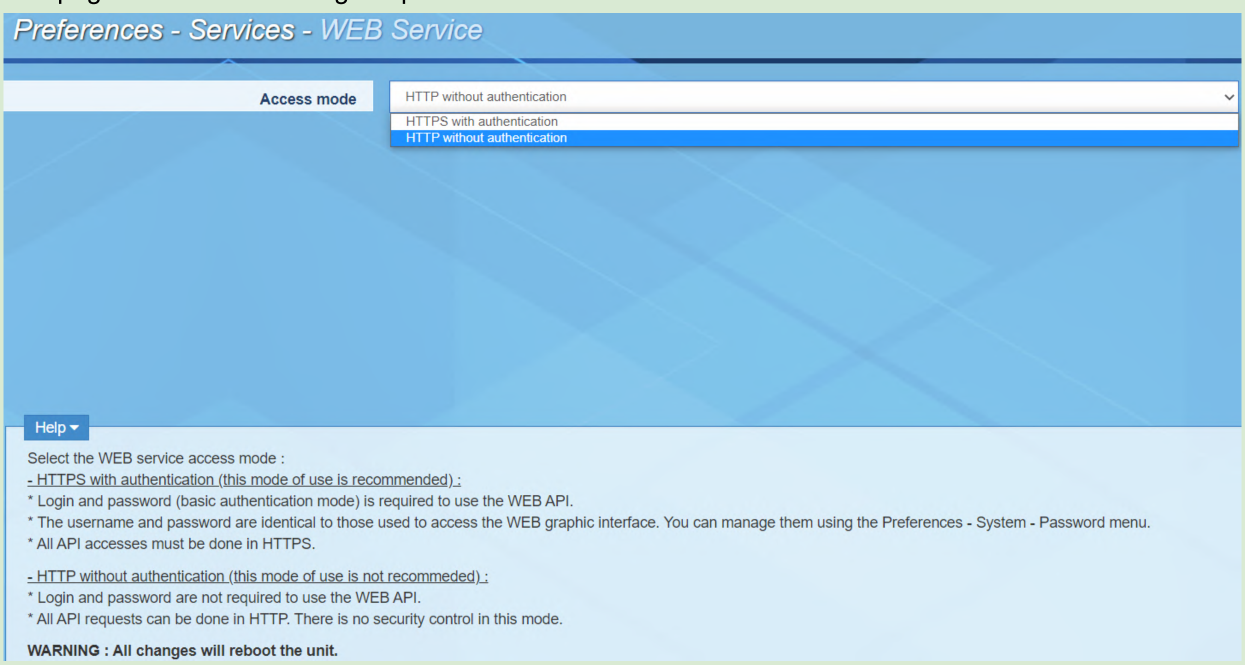
This page allows for enabling the automatic discovery and publishing of AES67 or RAVENNA streams.



In case you do not use AES67 or RAVENNA audio I/Os, there is no need to activate this service.

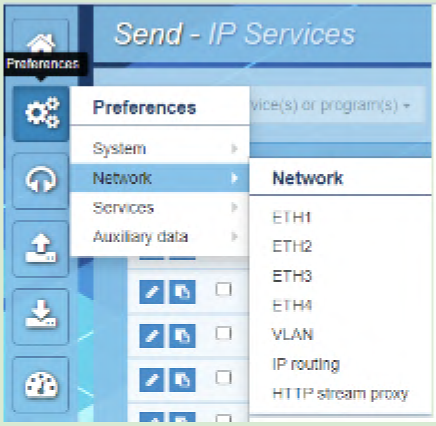
8.1.1.2.6 Preferences –> Services –> WEB Service

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



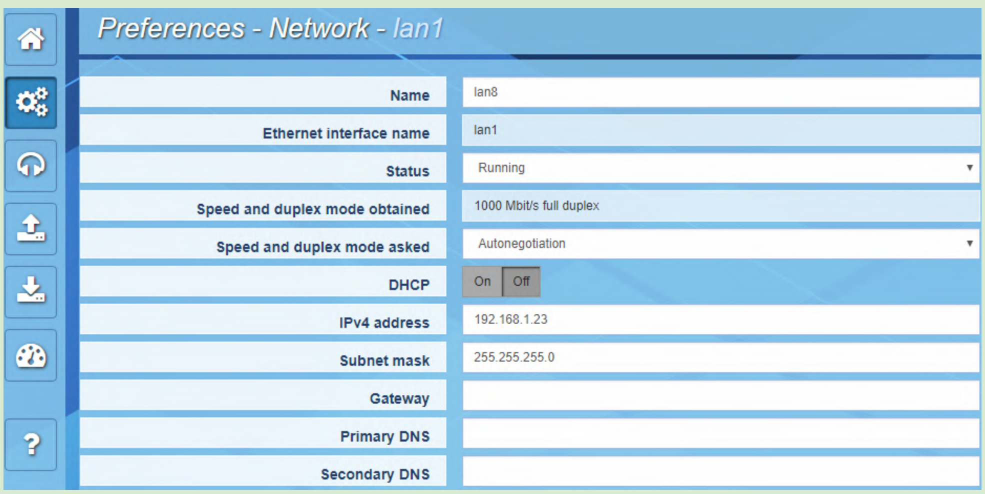
Default protocol is HTTPS (authentication through the username and password used to access the WEB pages).
HTTP can be selected, in which case no authentication is required; this last mode is less secure than the HTTPS mode, and is not recommended.
Changing the WEB Service API access protocol restarts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

8.1.1.3 Preferences –> Network

	This menu allows accessing the network configuration of IQOYA.

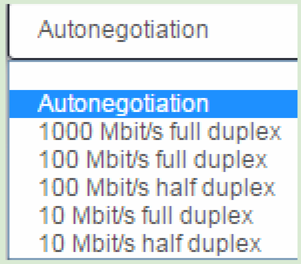
8.1.1.3.1 Preferences –> Network –> Eth1 (-> Eth4)

These pages allow configuring the four network ports of IQOYA X/LINK.



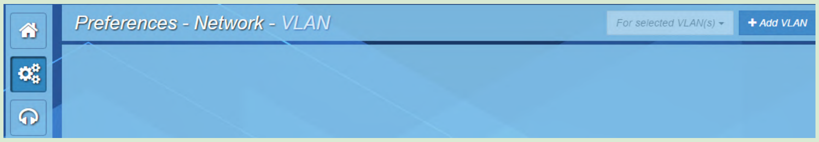
Click on a parameter field (“Status” for instance) to enter the editing mode.

Parameter	**Type**	**Description**
Name	R/W	Allows giving a name to the interface. This is the name displayed in the WEB pages typically for selecting the ethernet interface.
Ethernet interface name	Read	Displays the “real low level” name of the ethernet ports, as they can be read from th IQOYA back panel. This parameter can’t be changed.
Status	Read/Write	This parameter allows enabling/disabling the interface Default value=Running Possible values: Running: ethernet port is enabled. Stopped: ethernet port is disabled
Speed and duplex mode obtained	Read	Displays the current speed and mode of the ethernet interface.
Speed and duplex mode asked	Read/Write	Allows selecting the working mode of the ethernet interface. Possible values are as

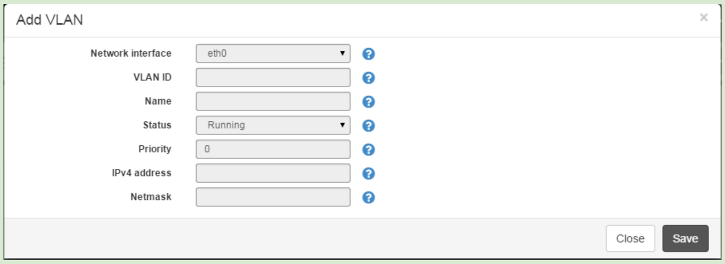
		follows:  We recommended to avoid the “Auto-negotiation” mode. Select the mode supporte by the network node connected the IQOYA.
DHCP	Read/Write	Allows enabling/disabling DHCP (Dynamic Host Configuration Protocol). Default value is OFF (disabled). Click on “On” to enable DHCP. This mode disables the following parameters.
IPv4 address	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Default value is:192.168.0.100 for Eth1, 192.168.1.100 for Eth2, 192.168.2.100 for Eth3, 192.168.3.100 for Eth4 Enter the IP address of this ethernet interface. **DHCP On** Displays the IP address automatically set by DHCP.
Subnet mask	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Enter the mask of the subneworkt this ethernet port belongs to. **DHCP On** Displays the subnetwork mask automatically set by DHCP.
Default gateway	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Enter the default gateway IP address. Streams sent beyond the subnets configured on LAN1 to 4 will pass through this gateway except if specific routing rules has bee defined in the IP routing page. ⚠ Only one default gateway must be configured for all the ethernet interfaces. I several gateways has to be used, one can be set as default gateway, the others mu be the subject of routing rules in the IP routing page. **DHCP On** Displays the default gateway IP address automatically set by DHCP.
Primary DNS	Read if DHCP is On Write if DHCP is Off	**DHCP Off** Enter the IP address of the primary DNS (if any). **DHCP On** Displays the IP address of the DNS automatically set by 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 automatically set by DHCP (may be empty).

8.1.1.3.2 Preferences –> Network –> VLAN

This page allows declaring VLANs on the ethernet interfaces. No VLAN is declared by default.
Multiple VLANs can be declared for each ethernet interface.



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



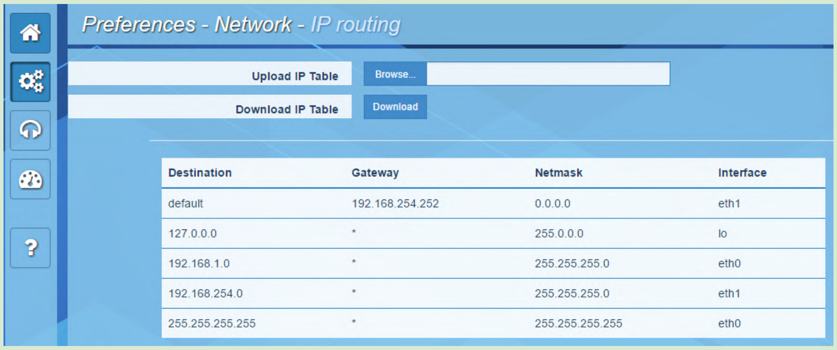
Parameter	Type	Description
Network interface	Read/Write	Select the network interface that will support the VLAN (ETH1 to ETH4)
VLAN ID	Read/Write	Enter the VLAN ID in the range 1--4094. Avoid ids 1002 to 1005 which are reserved.
Name	Read/Write	Enter a logical name for this VLAN
Status	Read/Write	Allows 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 ethernet port within this VLAN. If no value is entered, the IP address is the IP address of the selected ethernet port.
Netmask	Read/Write	Enter the netmask for this VLAN interface. If no value is entered, the netmask is the same as the selected ethernet 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.

8.1.1.3.3 Preferences –> Network –> IP routing

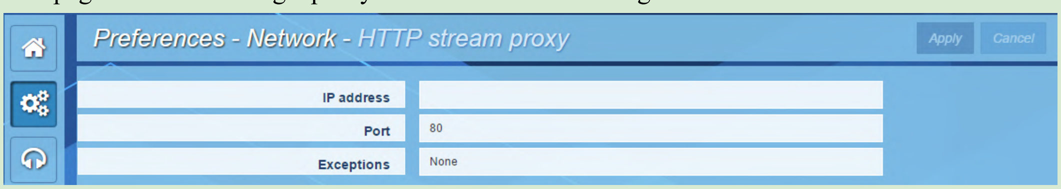
This page allows viewing the current IP routing table, downloading it, and uploading a modified IP routing table.



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. **Only the additional routes must appear in this file. Routes to directly accessible subnets are not present in this file and need not be added to this file.**
Note: In case you use more than one ethernet interface, do not declare several gateways. Declare instead one default gateway, for instance on Eth0, and declare routes on other ethernet interfaces through this routing table.

8.1.1.3.4 Preferences –> Network –> HTTP stream proxy

This page allows declaring a proxy used for HTTP streaming.



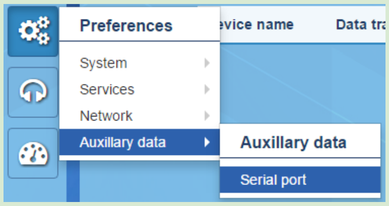
Parameter	Type	Description
IP address	Read/Write	IP address (or domain name) of the HTTP proxy.
Port	Read/Write	TCP Port for the HTTP proxy (80 by default)
Exceptions	Read/Write	Default is None. Select “Locals” to bypass the HTTP stream proxy for local IP addresses.

8.1.1.4 Preferences –> Auxiliary data

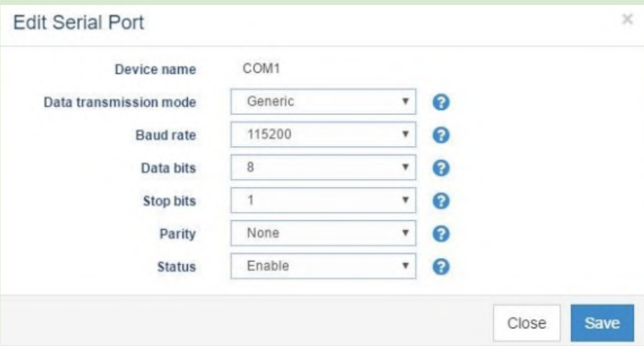
This section allows configuring the tunnelling of serial data and status data.

8.1.1.4.1 Preferences –> Auxiliary data –> Serial port

This page allows enabling/disabling the RS232 port, and setting its 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.

8.1.1.4.2 Preferences –> Auxiliary data –> GPIO

X/LINK offers the possibility to use physical GPIOs, or virtual GPIOs through UDP ports. 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 GPO’s.

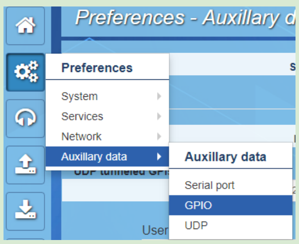
Virtual GPIO allow third party applications to send/receive status information via IP to/from IQOYA.32 virtual GPI status can be tunneled.

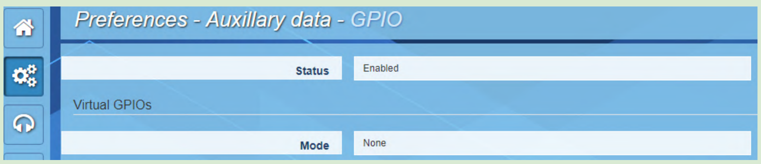
Structure of a virtual status information frame over UDP

	0 1 2 3 4 5 6 7 8 9 10 11 12 13 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 31	
32-bit word 1:	Version number (4 bits) = 0000	User ID (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 validation mask validates the GPI statuses to be taken into account.

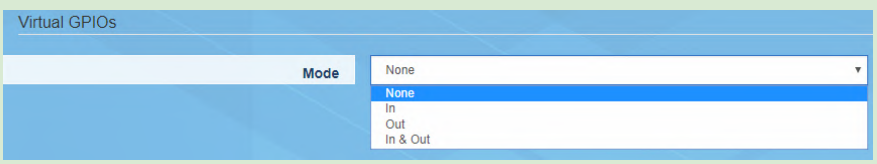
The page Preferences –> Auxiliary data –> GPIO allows enabling/disabling the in-band tunneling of GPI status information to GPO.





Status: Select enable to activate the status tunneling.

To declare virtual GPI's to be tunneled, select "In" from 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".

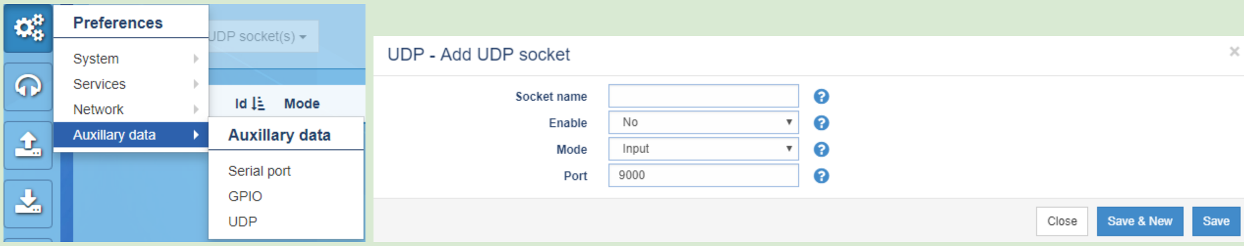


The following screen capture corresponds to the mode "In & Out".

Parameter	Type	Description
UDP Tunneled GPIs		
User ID	Read/Write	Allows defining a group of Virtual GPIs (among 32 possible tunneled GPIs) sent by an application. The 32 virtual GPIs can be shared between several applications. The User ID identifies one given application.
UDP GPI1	Read/Write	Click on [+] to declare an additional input status. 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 so that the decoder does not miss a status change.
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.

8.1.1.4.3 Preferences –> Auxiliary data –> UDP

This page allows defining the UDP ports used for receiving and /or sending serial data over IP.

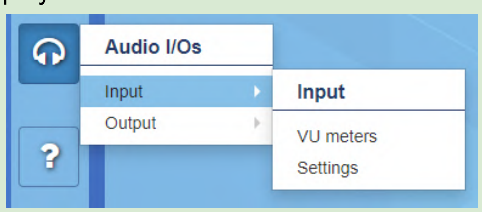


Parameter	Type	Description
Socket name	Read/Write	Name given to the UDP socket. This name allows selecting the socket for tunneling data, in the Send−>IPService and Receive−>Program pages.
Enable	Read/Write	Yes: socket is enabled. No, socket is disabled.
Mode	Read/Write	Input: IQOYA reads the data to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

Serial data received via a UDP port are 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, serial data has to conform to the standard ICY-metadata syntax.

8.1.2 Audio I/Os category of menus

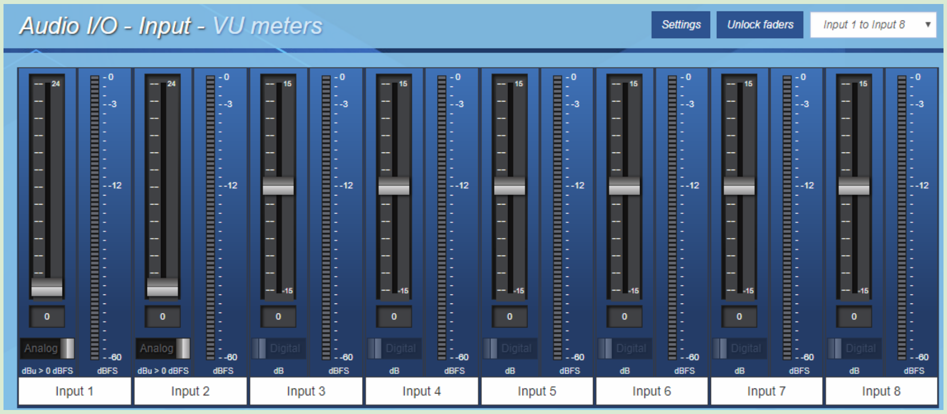
This category gathers all the menus allowing for the configuration of the inputs that can be encoded, and the outputs that play decoded audio.



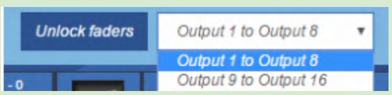
8.1.2.1 Audio I/O −> Input

8.1.2.1.1 Audio I/O −> Input −> VU meters

This page displays the level of the signals incoming on the inputs (Line analog, AES/3, or MADI depending on the X/LINK configuration).

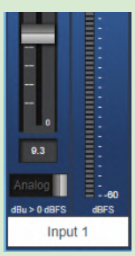


Displayed VU-meters unit is dBfs.
For a X/LINK with more than 8 mono channels (X/LINK-AES67 with additional optional I/O channels), the group of channels to be displayed is selectable from the top right menu.

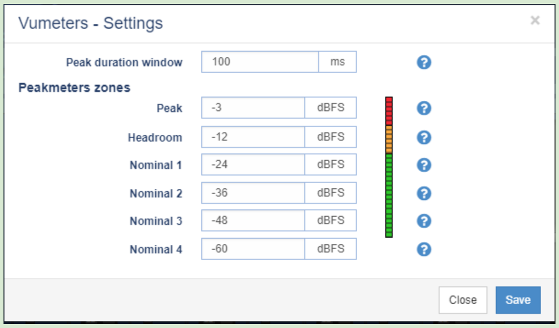


Select “Unlock faders” to change the input gains.
If the X/LINK features analog inputs, it is possible to adjust both the analog input gain and the digital input gain.

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 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

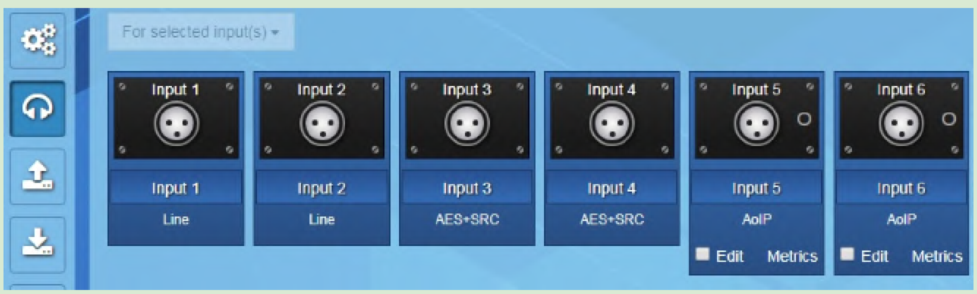
Vu-meters settings
Click on the “Settings” button to adjust the bargraph display and the front panel LED vu-meters display (red zone, orange zone, and green zones).

	**Peak duration window**: duration of the display of the peak levels (from 20ms to 10000ms) **Peak**: Level value in dBfs above which the vu-meter is red **Headroom**: Level value in dBfs above which the vu-meter is orange. **Nominal 1**: Level value in dBfs above which the LED right below the headroom LED is highlighted in green. **Nominal 2**: Level value in dBfs above which the 3rd LED from the bottom is highlighted in green. **Nominal 3**: Level value in dBfs above which the 2nd LED from the bottom is highlighted in green. **Nominal 4** :Level value in dBfs above which the 1rst LED from the bottom is highlighted in green.

8.1.2.1.2 Audio I/O –> Input –> settings

This page allows the following:

- Selection of the input signals to be allocated to the encoder inputs
- naming of the encoder inputs
- Configuration of the input AES67, or RAVENNA, or Livewire AoIP streams



This page displays all the inputs proposed by your IQOYA.
The audio sources to be encoded (input Programs) are selected among these inputs.

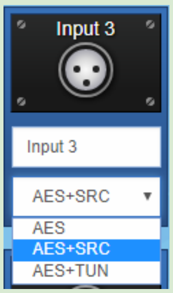
	Displayed mono inputs	Number of mono inputs that can be selected for encoding
X/LINK-ST & X/LINK-LE	2 analog, 2 on AES/3, 2 AoIP(*)	2
X/LINK-DUAL	4 analog, 4 on AES/3, 4 AoIP(*)	4
X/LINK-AES67	AoIP(*)	2 (basic version Up to 16 depending on the installed software option.

(*) AES67, RAVENNA, Livewire

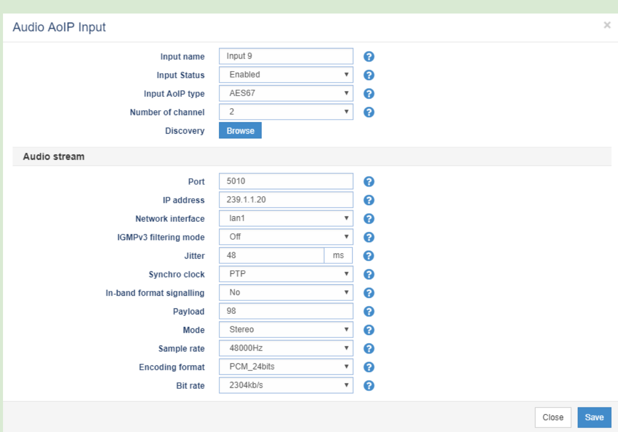
Analog line input settings

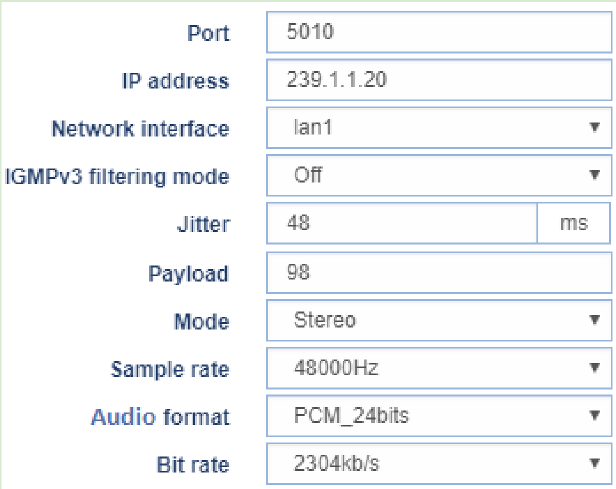
	Click on the “Input” field to rename the input. The new name will appear in other WEB pages (Input Program). Audio levels are adjustable from the VU-Meters page.

AES/3 input settings

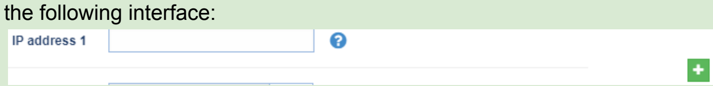
	Click on the “Input” field to rename the input. The new name will appear in other WEB pages (Input Program). Audio levels are adjustable from the VU-Meters page. The AES/3 input features a hardware sample rate converter, which is useful when the AES/3 input is not synchronous of the selettect sampling clock source. To enable the hardware SRC, select AES+SRC. To disable the hardware SRC, select AES. For AES transparent transport, select AES+TUN

AoIP input settings

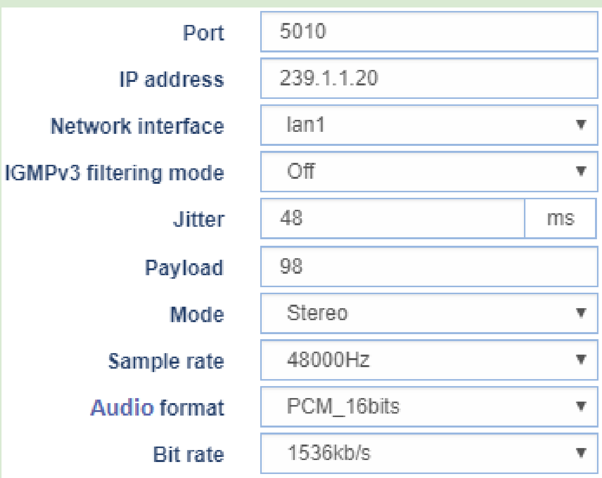
	Click on the “Input” field to rename the input. The new name will appear in other WEB pages (Input Program). Audio levels are adjustable from the VU-Meters page. Click on”***Edit***” to be able to configure the input AoIP stream, as described below. **Click on “Metrics”** to display the metrics on the configured AoIP stream. This is useful to get the minimum jitter value to be entered in the parameters. LED: if an AoIP stream is configured and it is well received the LED is green; The LED is red if the stream is not received, and grey if the stream reception is disabled.
	**Input Name**: the same as described above. **Input Status**: Enable/disable. **Input AoIP type**: AES67, RAVENNA, or Livewire **Number of channels**: defines the number of audio channels to be extracted from this AoIP stream When the AES67 or RAVENNA type is selected, the ***Browse*** button allows discovering the available AES67 or RAVENNA streams on all the networks. The list of parameters below is then filled in according to the selected AoIP stream.



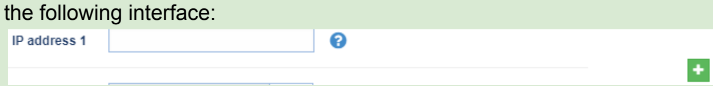
Settings for an AES67 stream
Most of the parameters below are automatically filled in after having browsed and selected an AES67 stream.
Port: UDP port number for receiving the stream
IP address: multicast or unicast IP@
Network interface: network interface (LAN or VLAN) used for receiving the stream.
IGMPv3 filtering: Allows including or excluding source IP addresses of the multicast stream.
If *Include* or *Exclude* value is selected, the list of IP addresses can be entered via the following interface:



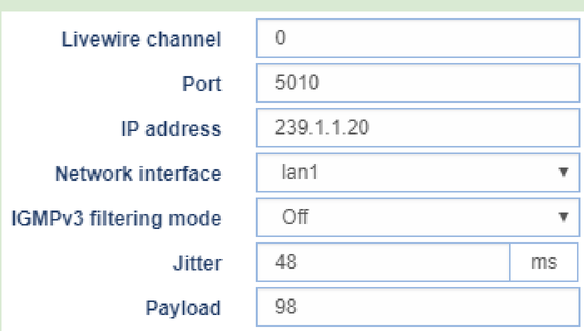
Jitter: enter the jitter value. This value must be at least equal to the jitter value reported from the Metrics on the stream.
Payload: Set to 98.
Mode: defines the number of channels to be considered from the stream (Mono or stereo).
Sample rate: 32 kHz, 44.1 kHz, or 48 kHz
Audio Format: PCM 12, 16, 20 or 24 bits



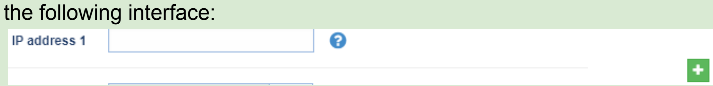
Settings for a RAVENNA stream
Most of the parameters below are automatically filled in after having browsed and selected a RAVENNA stream.
Port: UDP port number for receiving the stream
IP address: multicast or unicast IP@
Network interface: network interface (LAN or VLAN) used for receiving the stream.
IGMPv3 filtering: Allows including or excluding source IP addresses of the multicast stream.
If *Include* or *Exclude* value is selected, the list of IP addresses can be entered via the following interface:



Jitter: enter the jitter value. This value must be at least equal to the jitter value reported from the Metrics on the stream.
Payload: Set to 98.
Mode: defines the number of channels to be considered from the stream (Mono or stereo).
Sample rate: 32 kHz, 44.1 kHz, or 48 kHz
Audio Format: PCM 12, 16, 20 or 24 bits



Settings for a Livewire stream
Livewire channel:
Port: UDP port number for receiving the stream
IP address: multicast or unicast IP@
Network interface: network interface (LAN or VLAN) used for receiving the stream.
IGMPv3 filtering: Allows including or excluding source IP addresses of the multicast stream.
If *Include* or *Exclude* value is selected, the list of IP addresses can be entered via the following interface:

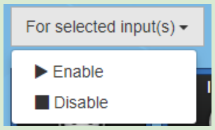


Jitter: enter the jitter value. This value must be at least equal to the jitter value reported from the Metrics on the stream.
Payload: Set to 98.

It is possible to enable or disable the reception of one or several declared AoIP input streams.

- Select the declared input streams though the check box on the left of “Edit”

- Select Enable or Disable from the “For selected input(s)” list box on the top left.



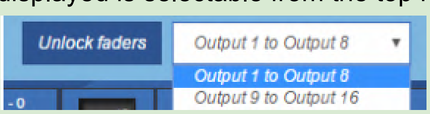
Click on “Apply” to confirm the changes.

8.1.2.2 Audio I/O –> Output

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

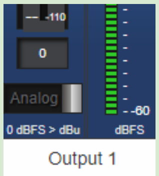
This page displays the level of the output signals.

Displayed VU-meters unit is dBfs.
For a X/LINK with more than 8 mono channels (X/LINK-AES67 with additional optional I/Os), the group of channels to be displayed is selectable from the top right menu.



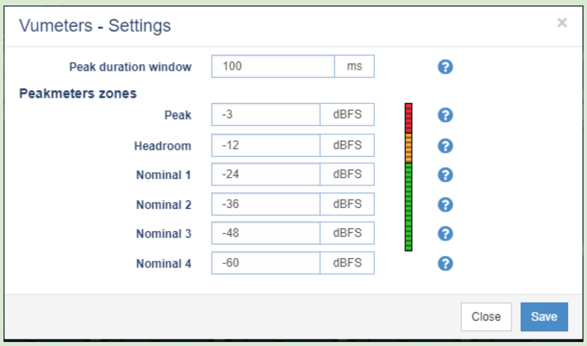
Select “Unlock faders” to change the output gains.
If the X/LINK features analog outputs, it is possible to adjust both the analog output gain and the digital output gain.
Selection of analog or digital gain is done thanks to the selector below the fader.

	When Digital is selected, a digital gain/attenuation is applied to the output signal.

	When Analog is selected, 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.

Vu-meters settings

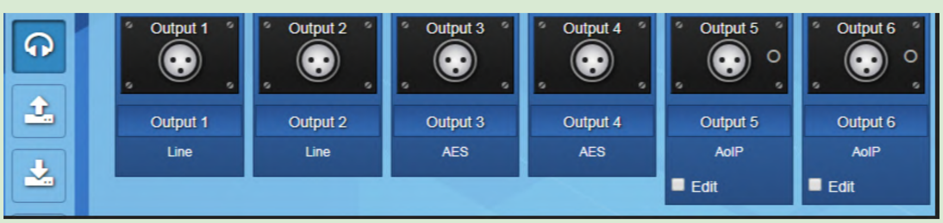
Click on the “Settings” button to adjust the bargraph display and the front panel LED vu-meters display (red zone, orange zone, and green zones).

	**Peak duration window**: duration of the display of the peak levels (from 20ms to 10000ms) **Peak**: Level value in dBfs above which the vu-meter is red **Headroom**: Level value in dBfs above which the vu-meter is orange. **Nominal 1**: Level value in dBfs above which the LED right below the headroom LED is highlighted in green. **Nominal 2**: Level value in dBfs above which the 3rd LED from the bottom is highlighted in green. **Nominal 3**: Level value in dBfs above which the 2nd LED from the bottom is highlighted in green. **Nominal 4** :Level value in dBfs above which the 1rst LED from the bottom is highlighted in green.

8.1.2.2.2 Audio I/O –> Output –> settings

This page allows the following:

- assign a physical output or AoIP output to a decoder output signal
- naming of the encoder outputs
- Configure the AoIP output(s)



This page displays all the inputs proposed by your IQOYA.
The audio sources to be encoded (input Programs) are selected among these inputs.

	Displayed mono outputs	Number of mono outputs that can be selected for output programs destinations
X/LINK-ST & X/LINK-LE	2 analog, 2 on AES/3, 2 AoIP(*)	2

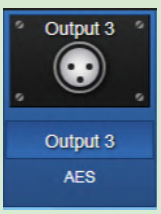
X/LINK-DUAL	4 analog, 4 on AES/3, 4 AoIP(*)	4
X/LINK-AES67	AoIP(*)	From 2 to 16 depending on the software option installed.

(*) AES67, RAVENNA, Livewire

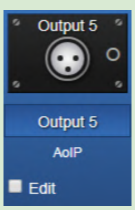
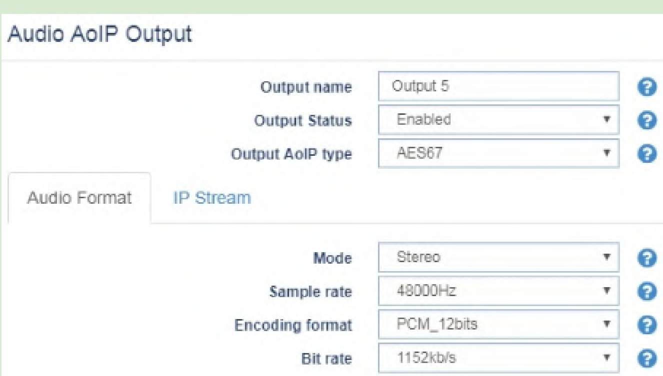
Analog line output settings

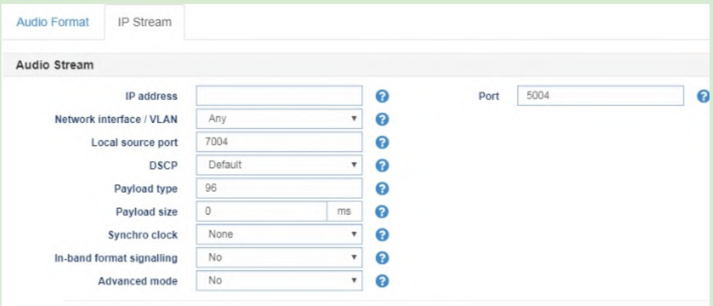
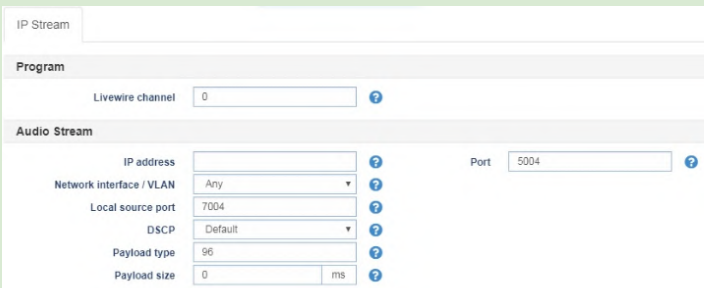
	Click on the “output” field to rename the output. The new name will appear in other WEB pages (output Program). Audio levels are adjustable from the VU-Meters page.

AES/3 output settings

	Click on the “output” field to rename the output. The new name will appear in other WEB pages (Output Program). Audio levels are adjustable from the VU-Meters page. For AES transparent transport, select AES+TUN by clicking on the *AES* field

AoIP output settings

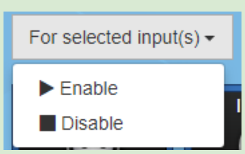
	Click on the “output” field to rename the output. The new name will appear in other WEB pages (Output Program). Audio levels are adjustable from the VU-Meters page. Click on”***Edit***” to be able to configure the output AoIP stream, as described below.
	**Output Name**: the same as described above. **Output Status**: Enable/disable. **Output AoIP type**: AES67, RAVENNA, or Livewire Audio Format Tab ***Sample rate***: 32 kHz, 44.1 kHz, or 48 kHz ***Audio Format***: PCM 12, 16, 20 or 24 bits

	**Settings for an AES67/RAVENNA output stream** ***IP address***: multicast or unicast destination IP@ ***Port***: destination UDP port number Local Source Port ***Network interface***: network interface (LAN or VLAN) used for sending the stream. ***DSCP:*** Value for the QoS of the stream. ***Payload type***: Set to 98 for PCM. ***Payload size***: When set to 0, the payload size is equal to the processing granularity (Preferences/Audio setup). Set 1 ms for 48 samples at 48 kHz (interoperable AES67 profile).
	**Settings for a Livewire output stream** **Livewire channel:** number of the Livewire channel ***IP address***: multicast or unicast IP@ ***Port***: UDP port number for receiving the stream ***Network interface***: network interface (LAN or VLAN) used for receiving the stream. ***Local source port***: Local UDP port used to send the stream ***DSCP:*** Value for the QoS of the stream. ***Payload type***: Set to 98 for PCM. ***Payload size***: It is recommended to set this value to 5, which corresponds to the Livewire "standard" mode (240 samples per packet).

It is possible to enable or disable the sending of one or several declared AoIP output streams.

- Select the declared output streams though the check box on the left of "Edit"

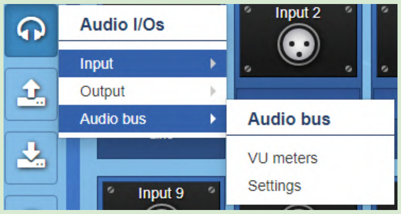
- Select Enable or Disable from the "For selected input(s)" list box on the top left.



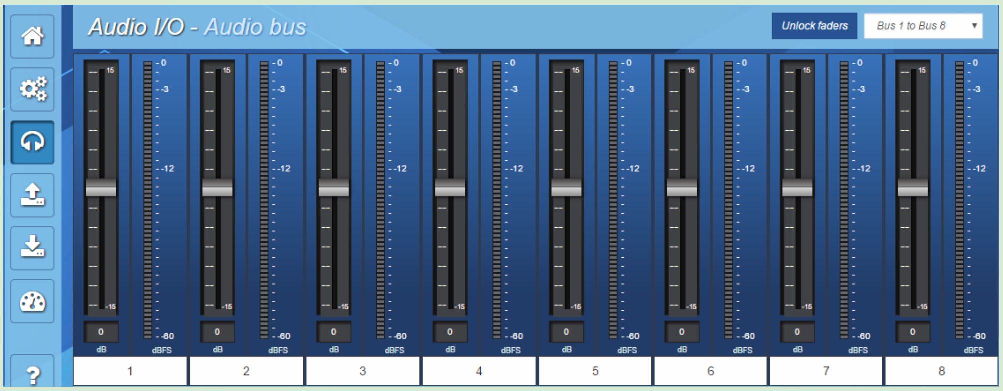
Click on "Apply" to confirm the changes.

8.1.2.3 Audio I/O –> Audio Bus

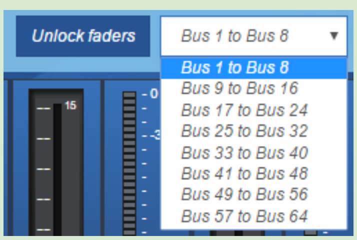
Audio buses are useful for the transcoding of IP streams, or for mixing several decoded IP streams. An audio bus can be the destination of one or several output programs, and the source of input programs. The number of available audio buses is defined by the license. Audio buses are optional.



8.1.2.3.1 Audio I/O -> Audio Bus -> Vu meters



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



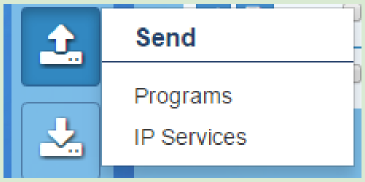
8.1.2.3.2 Audio I/O -> Audio Bus -> Settings

Click on the “name” field to rename the audio bus. The new name will appear in other WEB pages.
Right below the name of the bus, its type can be selected:

- AudioBus + Tun: select this option in case AES transparency is needed (transport of samples and user bits)
- AudioBus: default value. Only audio samples are transported.

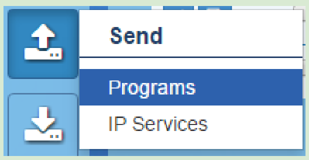
8.1.3 “Send” category of menus

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



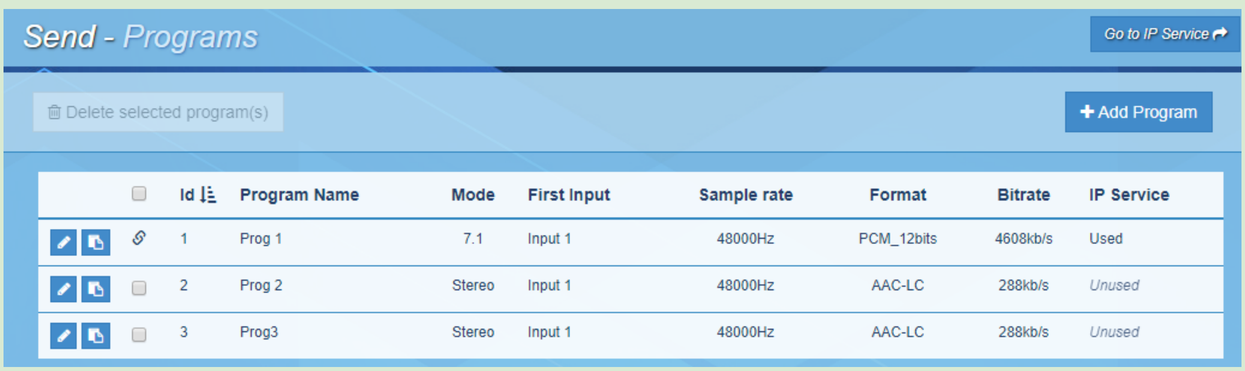
8.1.3.1 Send –> Programs

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

.



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.

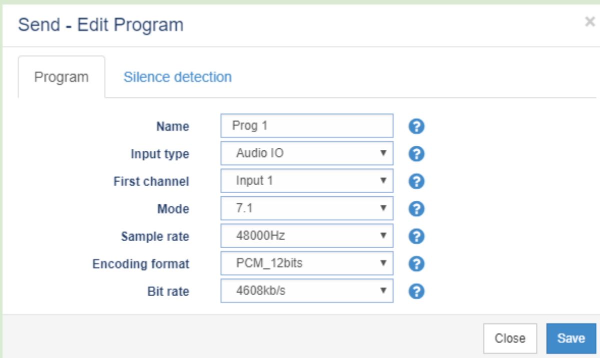
If a Program is used in at least one IP Service, the icon  is displayed on the left of its name, and “Used” appears in the column “IP Service”. The IP services that use this program are listed when moving the mouse above “Used”.

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 

.



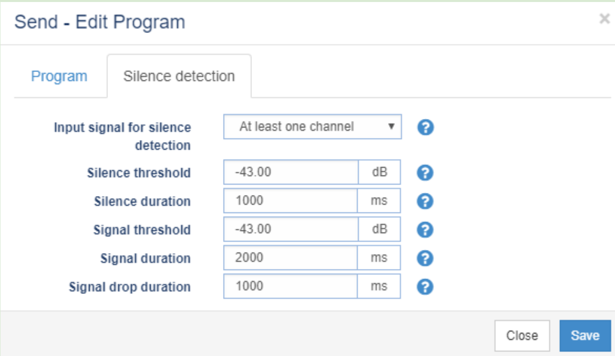
Give a unique name to the program.
Click on “Save” to confirm the parameters.
Click on “Close” to discard the changes.
Click on “Save & New” to confirm the settings,and duplicate them so that to create a new program with similar settings, except the name.

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, or an AoIP input
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.

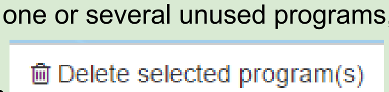
Silence detection parameters
Click on the “Silence detection” tab to set the criteria for silence detection on this program. An alarm is signalled when silence is detected, and it is reset when signal is detected again.
It is also possible to automatically stop/start the streaming upon silence/signal 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 set 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, silence condition is reached.
Silence threshold & Silence duration	Silent audio is defined through these two parameters, expressed in dBfs. When 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 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 the Save button to confirm the new Program.

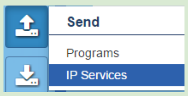
To delete one or several unused programs, select them by clicking on the icon on the left of their names, and click on the button



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.

8.1.3.2 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 

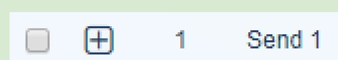
, or directly from the icon “Go to IP Service” on the top right of the Programs page Go to IP Service.

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.

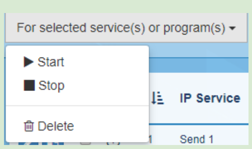
The program(s) carried by an IP service can be displayed by clicking on the icon  right on the left of the IP service name (an IP service can contain several programs in case of MPEG-TS MPTS encapsulation).

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

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.

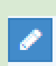
To start, stop, or delete an IP service, check the box on the left of its name 

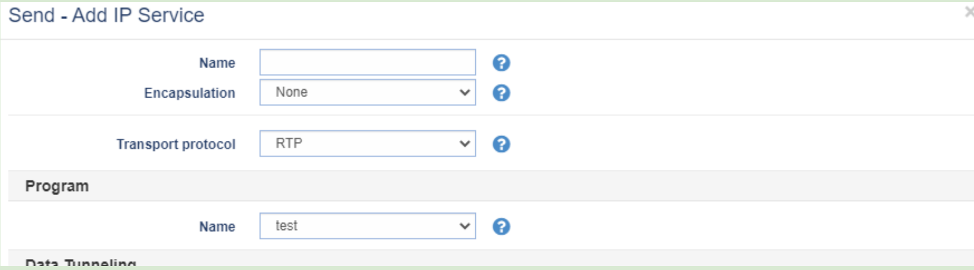
, and select the appropriate action:

For selected service(s) or program(s) ▾ 

.

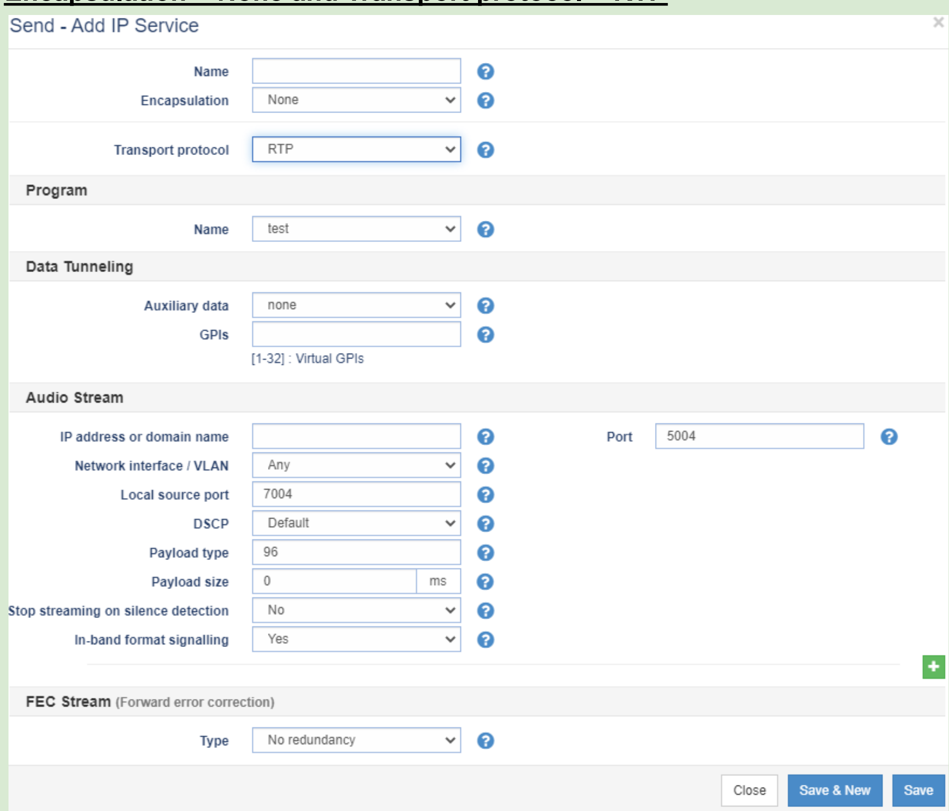
Note that a list of consecutive service can be selected by clicking on the first service check box, and shift clicking on the check box of the last service of the list.
Non consecutive services can be selected by CTRL clicking on their check boxes.

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 are 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 to select 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 tunnelled in-band.
GPIs	Read/Write	If there is a GPIO hardware option installed, enter a list of GPI numbers whose status is to be tunnelled in-band. Numbers start from 1 and must be separated by commas
Audio stream		
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 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 VLAN. In case the target IP address is multicast, select the Eth interface or the 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, or NTP. NTP can be selected if the option “NTP based audio synchro” is installed.
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

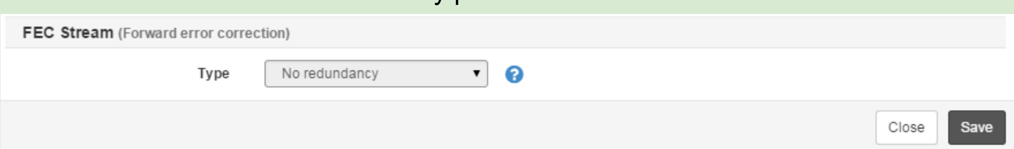
		(PCM, apt-X, G7xx), and 256 frames for framed audio formats (MPEG, AAC). In MPEG Layer 2 48 kHz, this corresponds to 6 seconds (6 000 000 microseconds).
In-band format signalling	Yes/No	**Yes**: the description of the audio format is inserted in the IP audio stream so that the decoder can automatically 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.

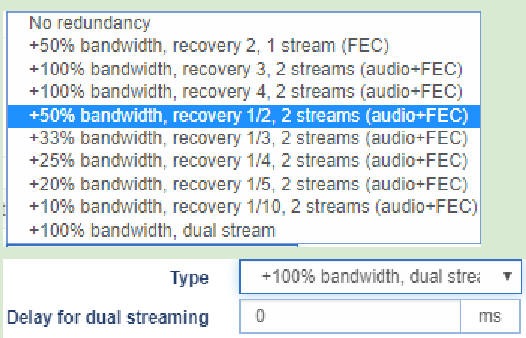
Click on the icon on the bottom right of the page to add an IP destination.

Enter the new target IP address, UDP port, and the network interface through which the stream is sent.

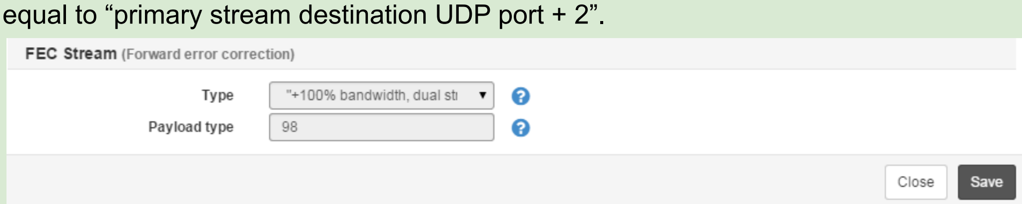
Click on the icon to remove a destination.

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.

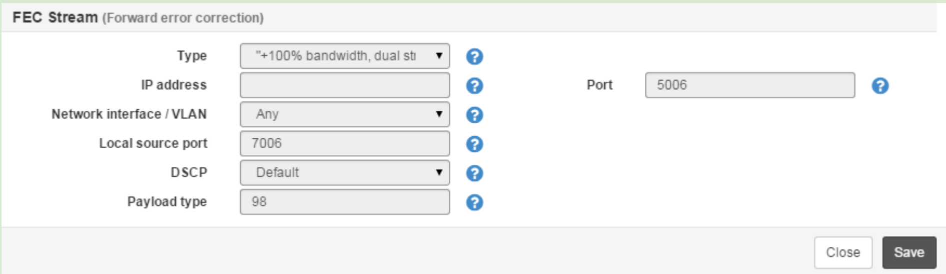


	FEC on 1 stream means that additional data is sent in the IP audio stream (in-band). FEC on 2 streams means additional data are sent as a second IP stream. Dual stream FEC means that the IP steam is duplicated. When no delay is selected, primary stream and redundant stream are sent at the same time.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



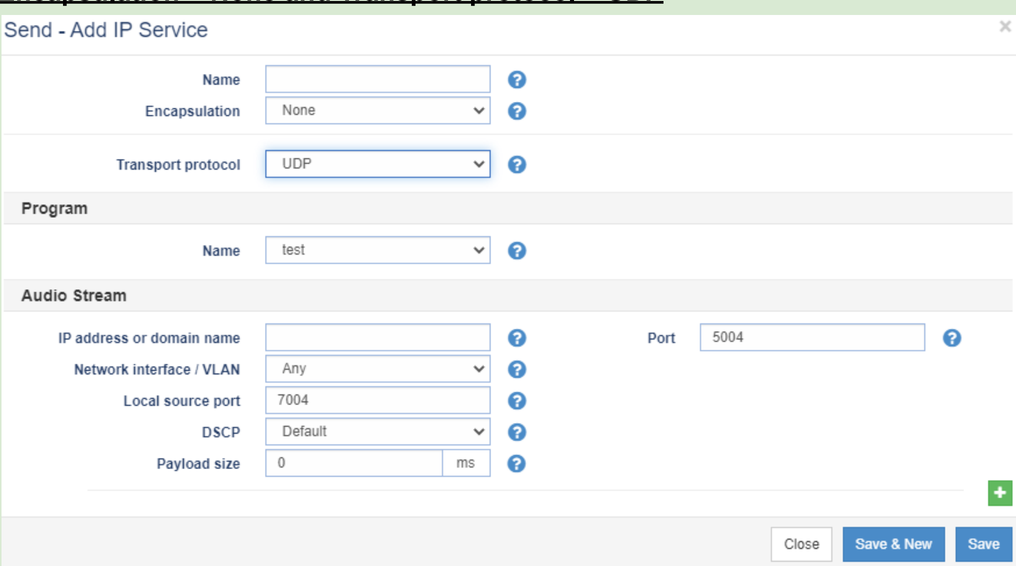
In case parameter “In-band format signalling” is set to “No”, the destination IP address and UDP port of the FEC stream can be configured.



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 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. 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 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.

Encapsulation = None and Transport protocol = UDP

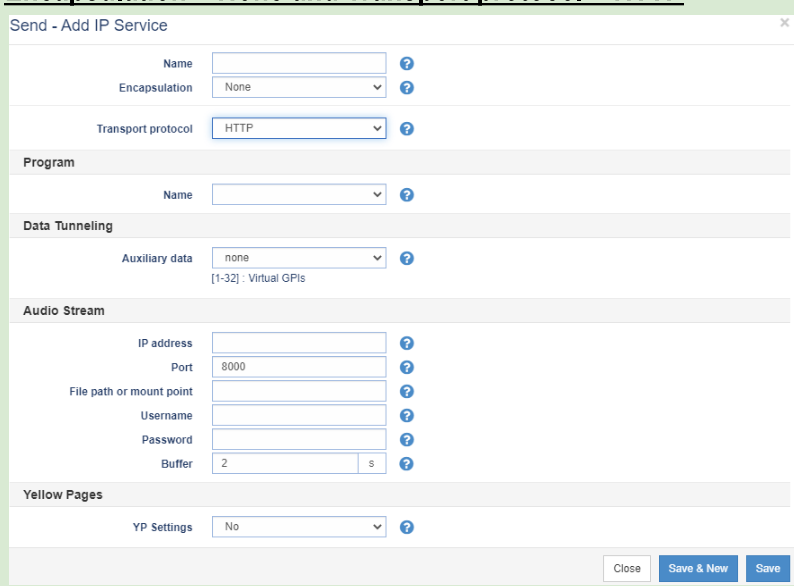


Parameter	Type	Description
Program		
Name	Read/Write	Select the Program to be streamed from the list of Programs. A Program can be used by several IP services.
Audio stream		
IP address or Domain name	Read/Write	Destination IP address (unicast or multicast) or domain name.
Network interface / VLAN	Read/Write	Displayed if a multicast destination IP address has been entered. Select the network interface or VLAN that is to be used for streaming.
Port	Read/Write	Enter the destination UDP port.
Local source port	Read/Write	Local UDP port number of IQOYA SERV/LINK. In case a lot of streams are generated, it is recommended that they are not all streamed through the same local port.
DSCP	Read/Write	Select the quality of service (QoS) class of the 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.

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

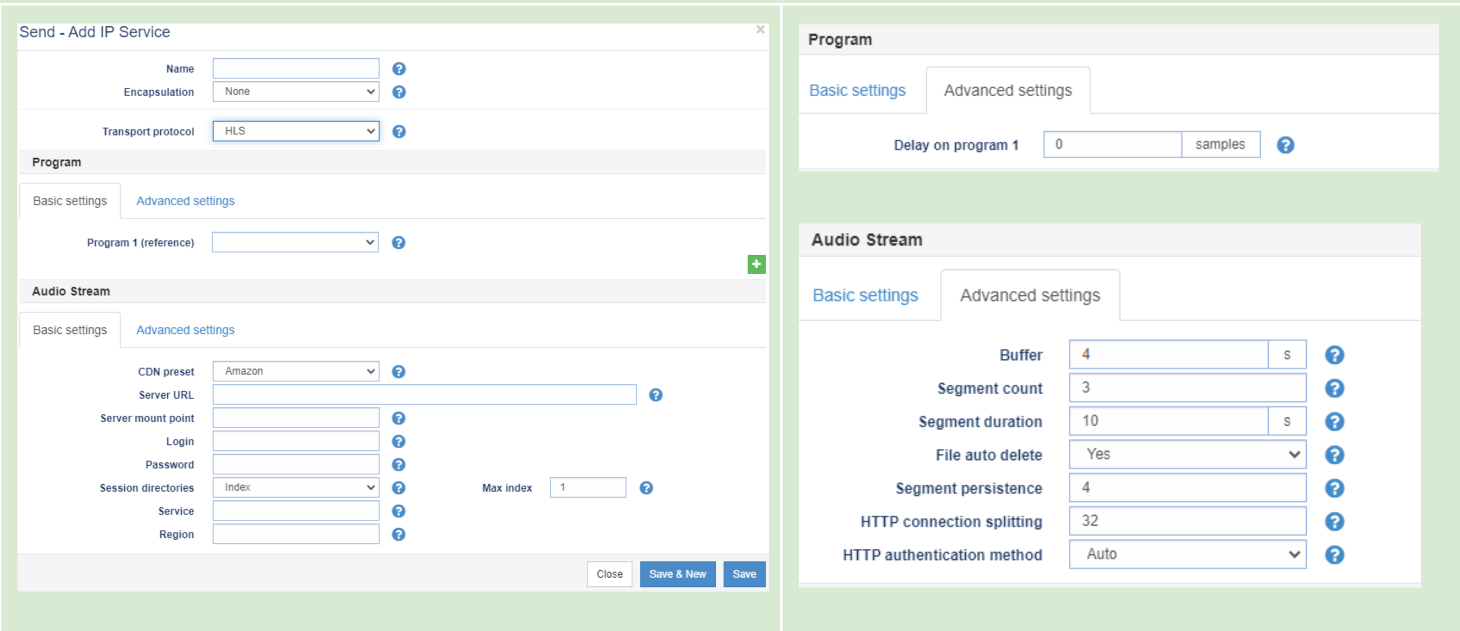


Parameter	Type	Description
Program		
Name	Read/Write	Select the Program to be streamed from the list of Programs. A Program can be used by several IP services.
Data tunnelling (This section is displayed only if tunnelling has been activated from Preferences->Auxiliary data->GPIO)		
Auxiliary data	Read/Write	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 3.12, it is possible to enable the descriptor “Maximum_Bitrate_Descriptor” in the PMT, and the flag “ES_Rate_flag" in the PES. See appendix H.



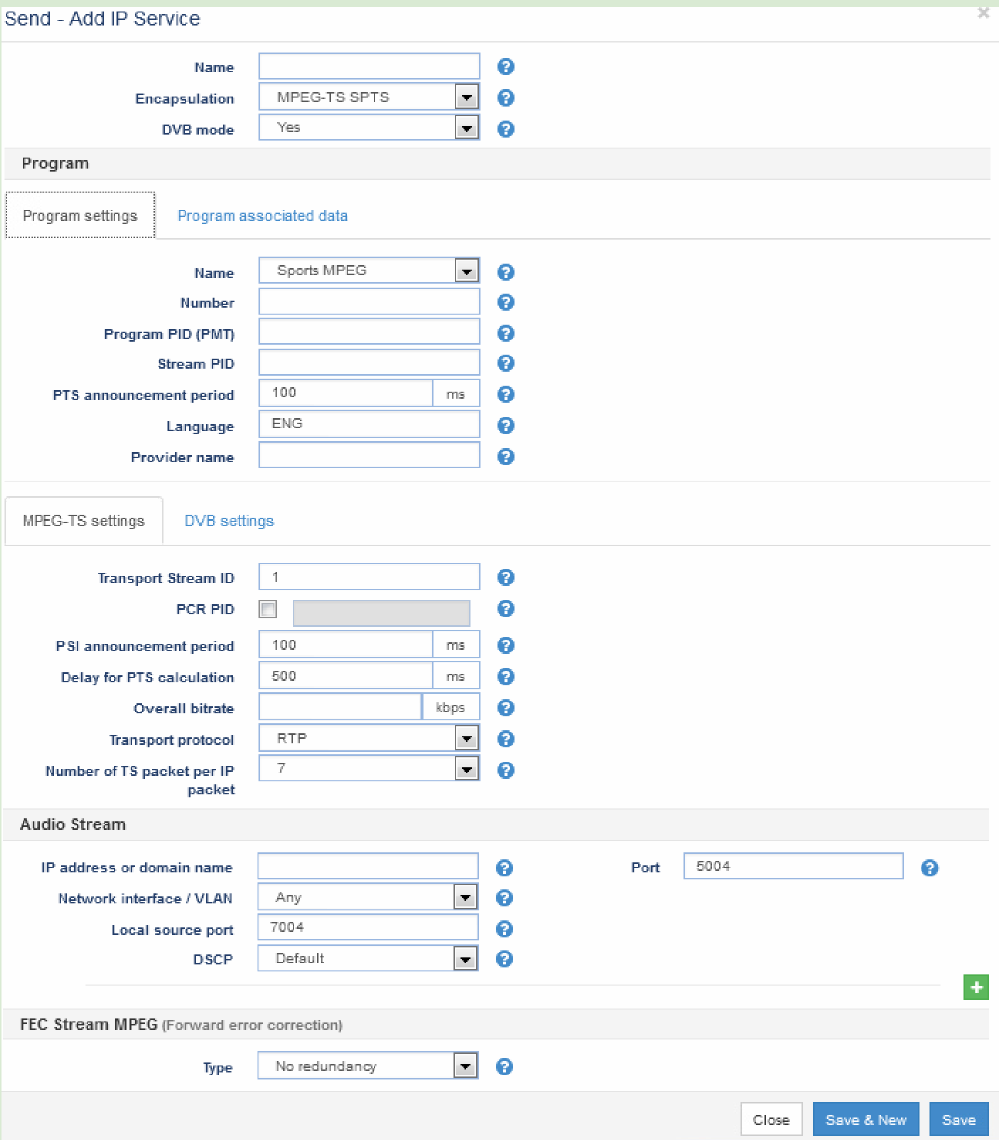
Parameter	Type	Description
Program: Basic settings		
Program 1(reference)	Read/Write	Select the Program to be streamed from the list of Programs. Program can be in MPEG Layer 2, MPEG Layer 3, and AAC-ADTS formats. For multi bitrate HLS, up to 5 additional audio Programs can be selected. They all must 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) - Akamai (MSL4): for Akamai CDN (with digest authentication) 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	Read/Write	Transport protocol used to push HLS data to the origin server. Select HTTP or HTTPS.
Network port	Read/Write	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. Only 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 - Akamai edge grid v1 - Digest

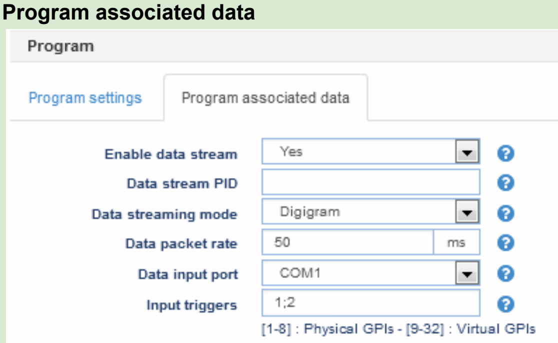
Click on "Save" to confirm the settings. Click on "Save & New" to confirm the settings and create a new IP service with the same parameters.
Click on "Close" to discard the settings

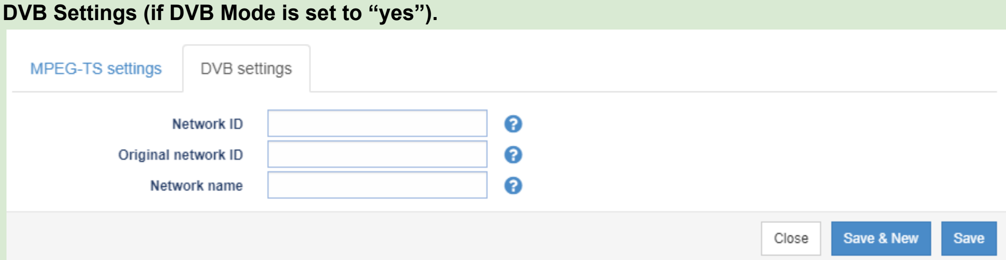
Encapsulation = MPEG-TS SPTS / MPEG-TS MPTS

Note: Since version 3.12, 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 H.



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. The following formats are supported in the MPEG-TS stream: MPEG Layer2, MPEG Layer 3, AAC. The name of the selected program is the name on 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 100 to 700 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 G for the structure of the data packet
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 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.

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

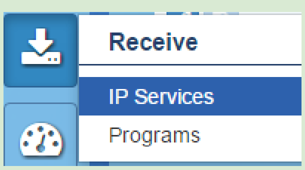
Click on the icon  on the bottom right of the page to add IP destinations.

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.

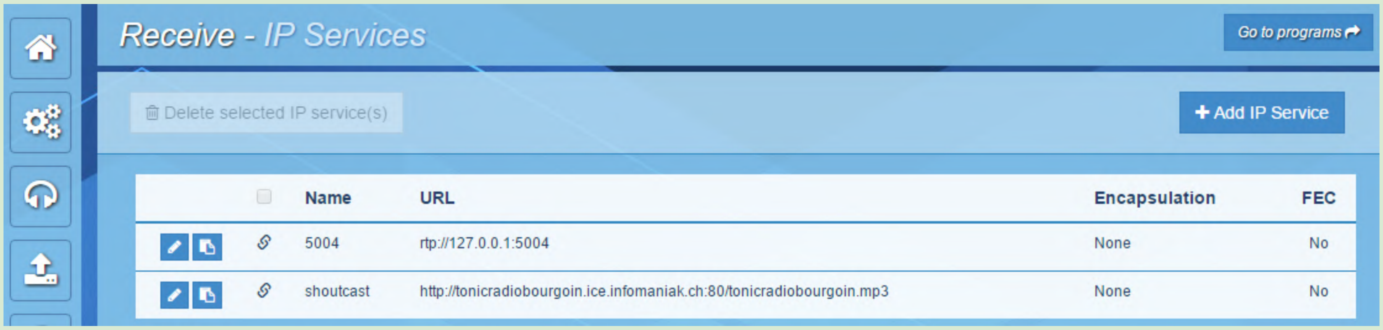
8.1.4 “Receive” category of menus

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.

8.1.4.1 Receive –> IP services



This page allows declaring and viewing the IP services to be received 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.

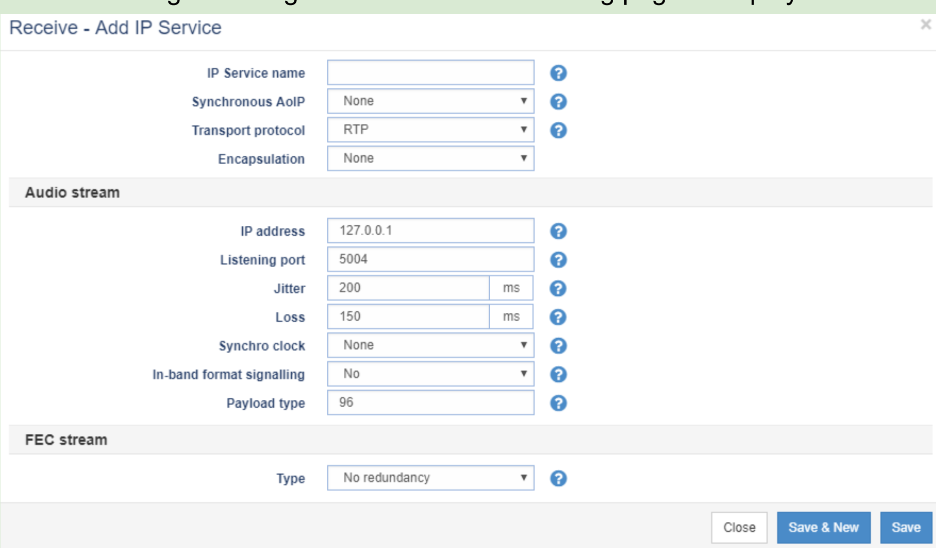
To remove an IP service, click on the icon on the left end of its line, and select Delete selected IP service(s) .

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

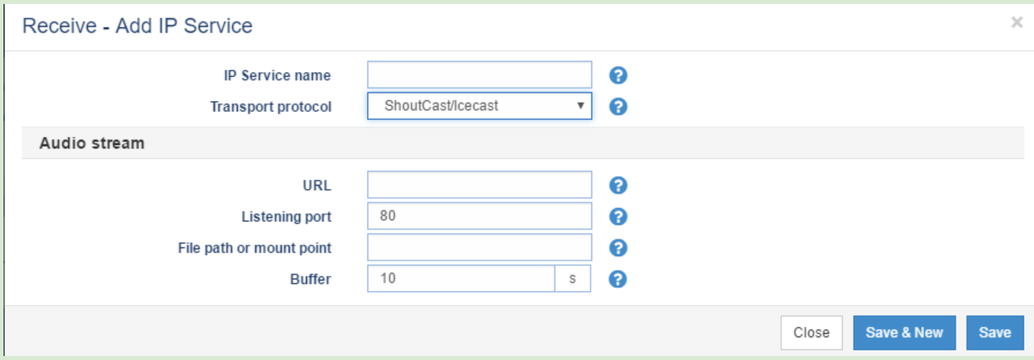
To declare a new IP service, click on the button + Add IP Service .

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.



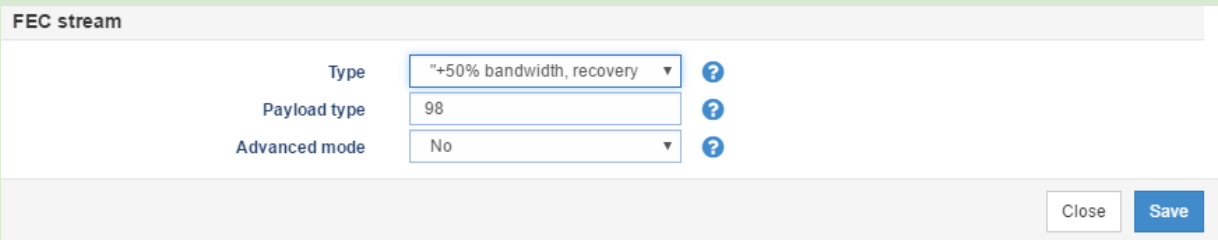
In case Icecast/Shoutcast is selected for the transport protocol (for WEB radio), the following parameters are displayed.



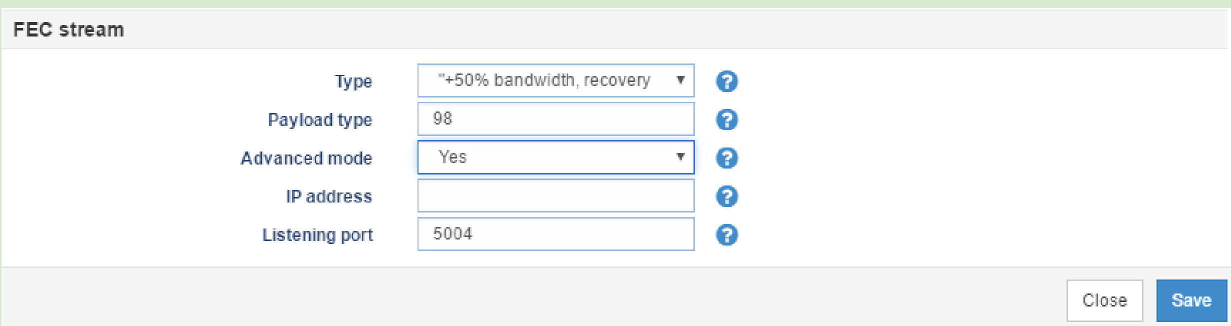
Parameter	Type	Description
IP service name	Read/Write	Name given to this IP service. This is the name that can be selected in the source of a decoding priority of an output program.
Transport protocol	Read/Write	Values: RTP, UDP, Icecast/Shoutcast
Encapsulation	Read/Write	Only displayed if the selected transport protocol is different from Icecast/Shoutcast. Values are: None, MPEG-TS SPTS or MPEG-TS MPTS.
Audio stream		
IP address	Read/Write	For RTP and UDP protocols only. In unicast, set this parameter to 127.0.0.1, otherwise enter the multicast IP address to listen to.
listening port	Read/Write	For RTP and UDP protocols, value of the UDP port to listen to. For Icecast/Shoutcast, value of the TCP port to listen to.
Jitter	Read/Write	For RTP and UDP protocols only. 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	Read/Write	For the RTP protocol only. 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 buffer of jitter is then flushed, and filled again with received packets; this allows resynchronization on the incoming IP audio stream, but this generates a silence longer than the consecutive packet lossed. 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	Read/Write	For the RTP protocol only. Select NTP in case the audio synchronization on NTP is used for decoding this stream (optional feature).
In-band format signalling	Read/Write	For the RTP protocol only. 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	Read/Write	For the RTP protocol only. 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).
URL	Read/Write	Only displayed when Transport Protocol is set to Icecast/Shoutcast. URL of the Icecast/Shoutcast server. Example: streamer.mysite.com.

File path or mount point	Read/Write	Only displayed when Transport Protocol is set to Icecast/Shoutcast File path of the source of a Shoutcast server. File path of the source or mount point of an Icecast server.
Buffer	Read/Write	Only displayed when Transport Protocol is set to Icecast/Shoutcast 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.

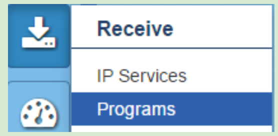


FEC stream parameters	Type	Description
Type	Read/Write	Select the FEC that is configured on the encoder of the received IP stream.
Payload type	Read/Write	Enter the same FEC payload type that is configured on the encoder of the received IP stream.
Advanced mode	Read/Write	Select “Yes” if the FEC destination 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.

8.1.4.2 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. The switching between priorities depends an criteria named “Backup switching criteria”.

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 

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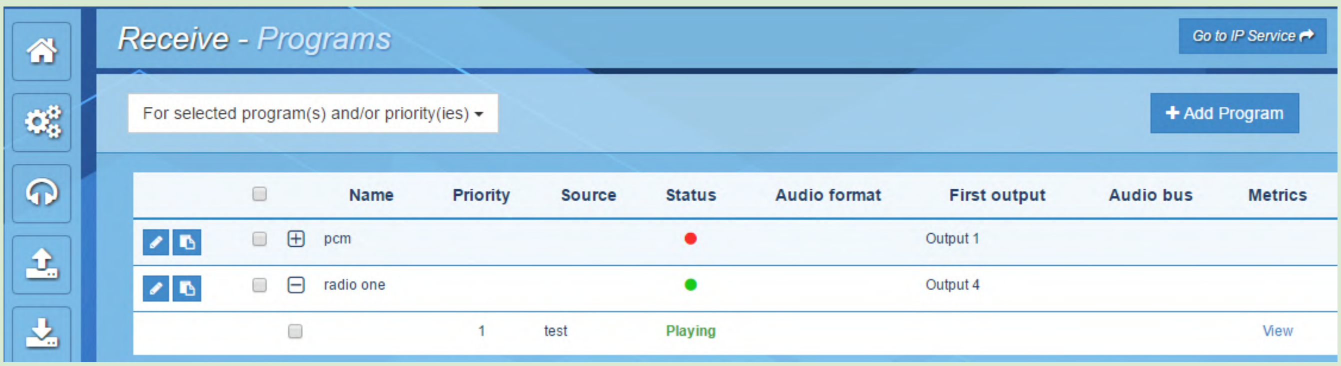
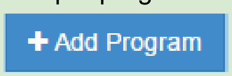
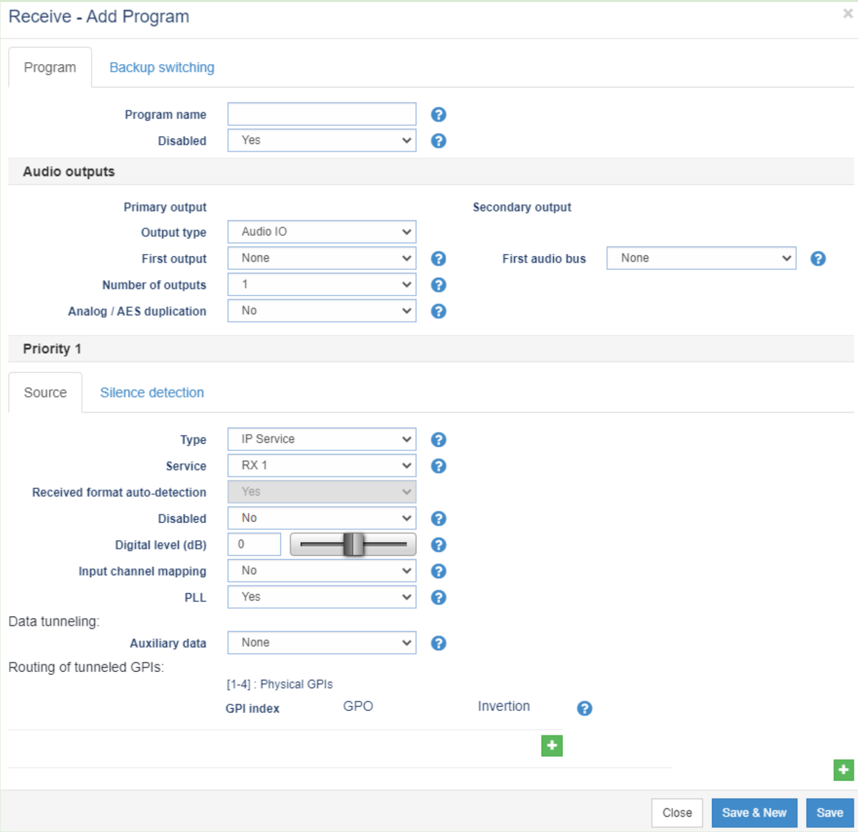


The “Programs” page displays the declared output programs.

To declare a new output program, click on 

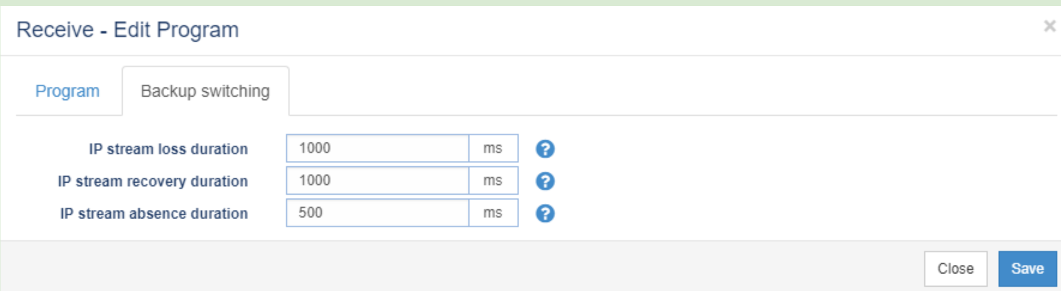
, or create it from an existing one by selecting the icon  on the left of the program.



Program Parameter	Type	Description
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 section		
Primary output: output type	Read/Write	Audio I/O: to select a physical output (not available on X/LINK-AES67) AoIP: to select a declared output stream (AES67/RAVENNA/Livewire) Audio Bus: to select an audio bus, typically used for transcoding (optional)
Primary output: first output	Read/Write	Select the first audio output associated with this program. Note that it is necessary to select an audio output that is not assigned to another active program (otherwise an error is displayed). In case two programs are assigned to the same output, only one can be started.
Primary output: Number of outputs	Read/Write	Set this parameter to 1 for mono, 2 for stereo, 6 for 5.1, 8 for 7.1 (these last two cases are possible on X/LINK-AES67 with multiple channels).
Primary output: Analog / AES duplication	Read/Write	No: the program is decoded to the selected audio outputs. Yes: the program is decoded to both the analog and the AES3 outputs. This feature requires that an audio bus is selected from the parameter “Secondary output -> First audio bus”.

Secondary output: First audio bus	Read/Write	This parameters allows selecting a secondary output to duplicate the decoded audio. When the primary output is an AoIP output (like AES67/Ravenna), the secondary output can be an analog or AES3 output. If a software option for transcoding channels has been installed, it is possible to select an internal audio bus as a secondary output, when the primary output is set to an analog or AES3 output. This serves in the following cases: • The audio that is decoded to an audio output can at the same time be routed to an audio bus, in order to be re-encoded and streamed. • The audio decoded to the analog or AES3 outputs, is to be duplicated to both the analog and AES3 outputs (the parameter Analog / AES duplication must be set to Yes).

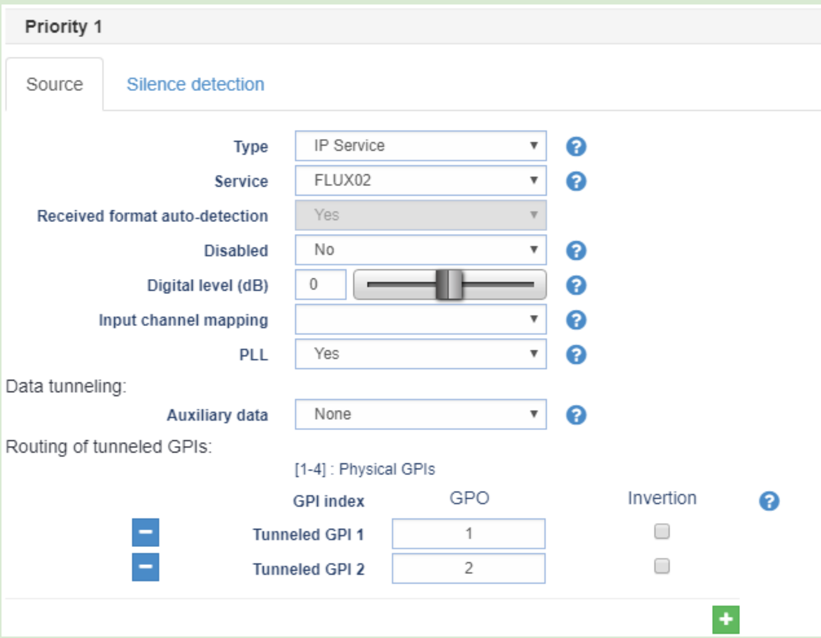
Backup switching criteria



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.

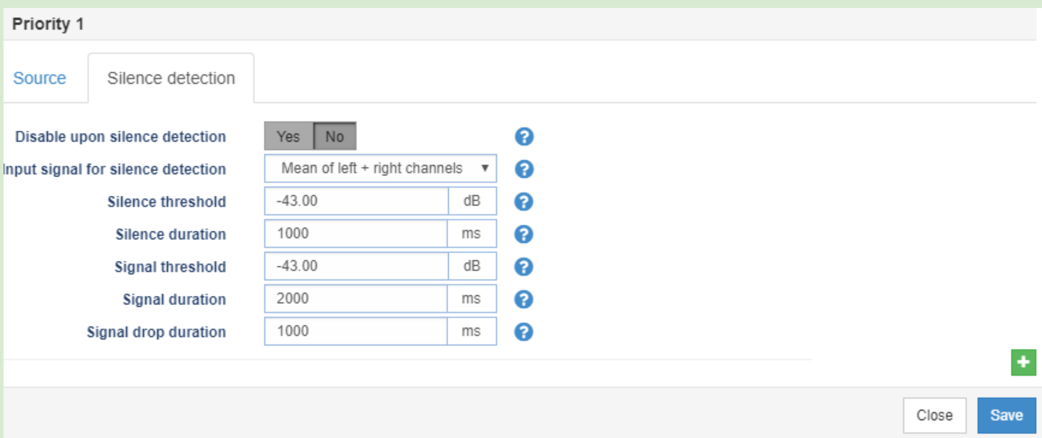
Source Type = IP service, and IP service is RTP



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).
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	Displayed when the audio source includes more audio channels than the output. Select how the channels of the selected source are to be 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	To be set to “Yes” in most of the use cases. It allows synchronisation of the incoming stream to the internal clock, thus guaranteeing a constant latency with the encoder.

		Has to be set to “No” typically when AES transparency is required between the encoder and the decoder (this requires also that the encoder and the decoder use clocks that have the same frequency).
Data tunneling: Auxiliary data	Read/Write	In case serial data are tunneled in-band, select the output port. It can be the RS232 COM port, or a UDP port if it has been declared from the menu “Preferences/Auxiliary data/UDP”.
Routing of tunneled status data: Status data indexes	Read/Write	 Enter the GPO number (starting from 1) that will reflect the tunneled GPI status. Click on “Add tunneled GPI” to route another tunneled status.
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.

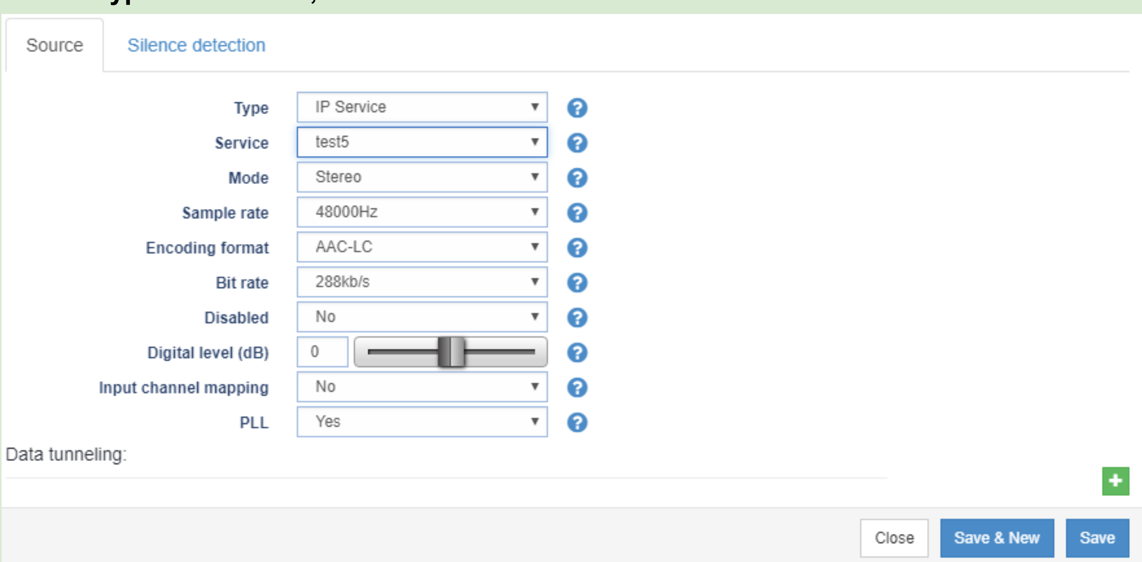
Silence detection parameters for the decoding priority



Priority Source Parameter		**Type**
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.

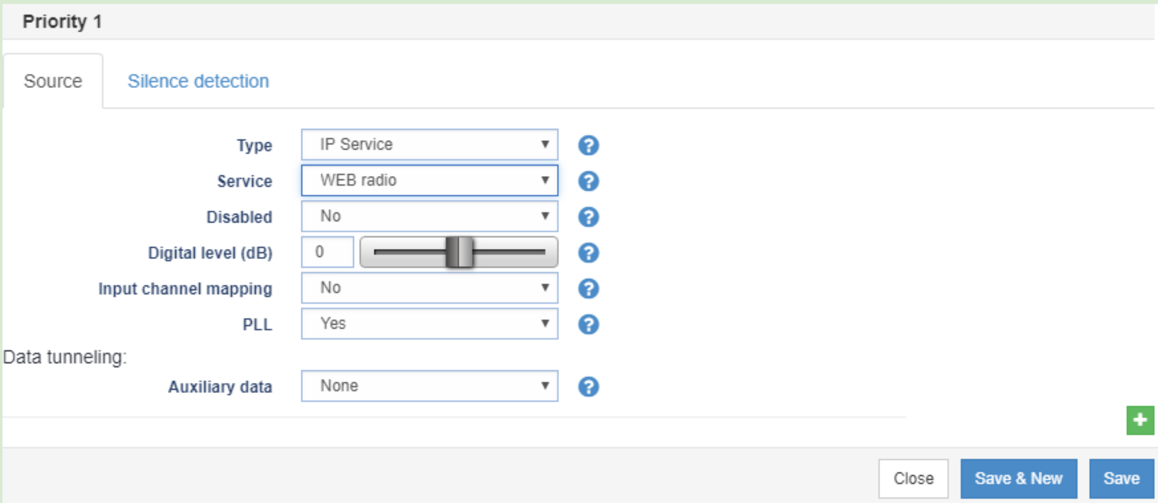
		- Right channel only: compares the right channel samples to the silence threshold. In case the sample values are always lower to the silence threshold during the defined silence duration, silence condition is reached. **- Left and right channels**: compares both the left and right channel samples to 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 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 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



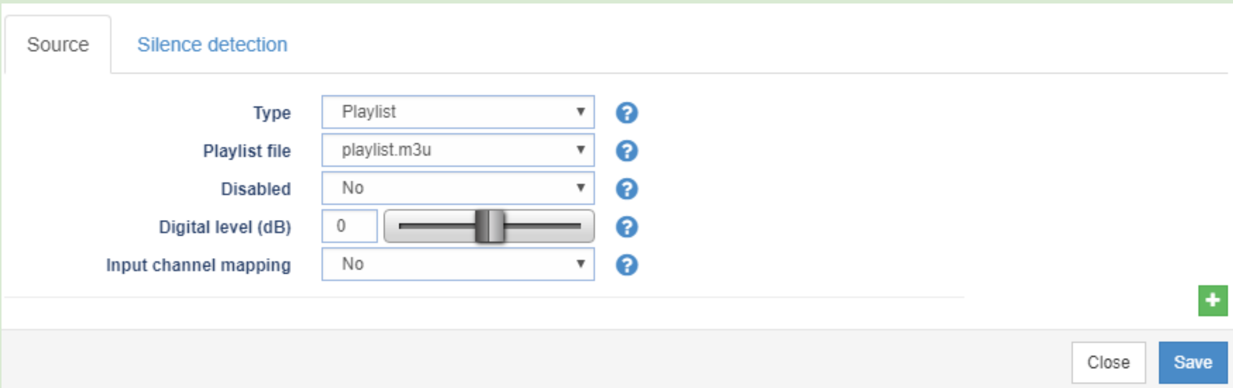
In UDP mode, the audio format has to be declared.

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



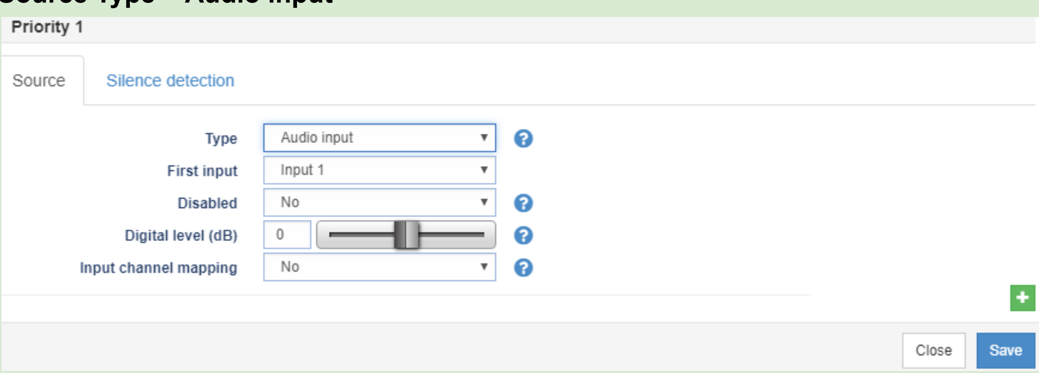
Parameter	Type	Description
Type	Read/Write	IP Service
Service	Read/Write	Select the IP service from the list of declared IP services. (IP services must have been declared first from the IP Services page).
Disabled	Read/Write	Set this parameter to “Yes” to disable this decoding priority.
Digital Level (dB)	Read/Write	Digital gain applied to the audio samples on this priority.
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 required that the encoder and the decoder have clock sources having the exact same sampling frequency)
Data tunneling: Auxiliary data	Read/Write	In case serial data are tunneled in-band, select the output port. It can be the RS232 COM port, or a UDP port if it has been declared from the menu “Preferences/Auxiliary data/UDP”.

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.

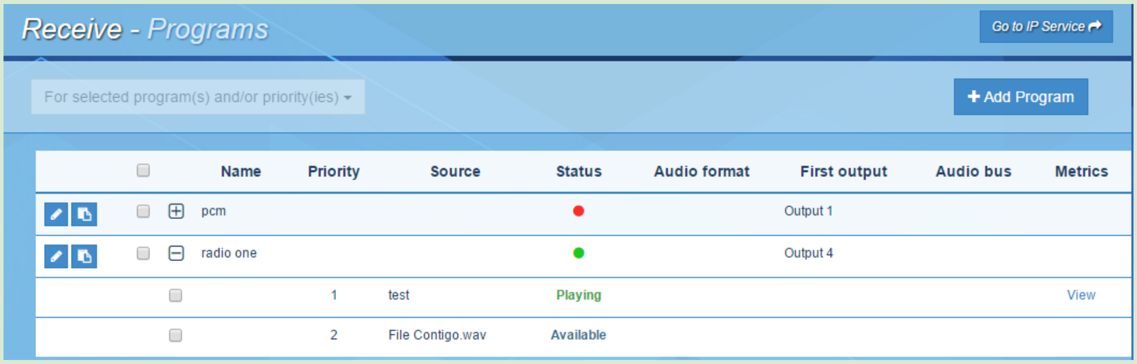
Source Type = Audio input



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.

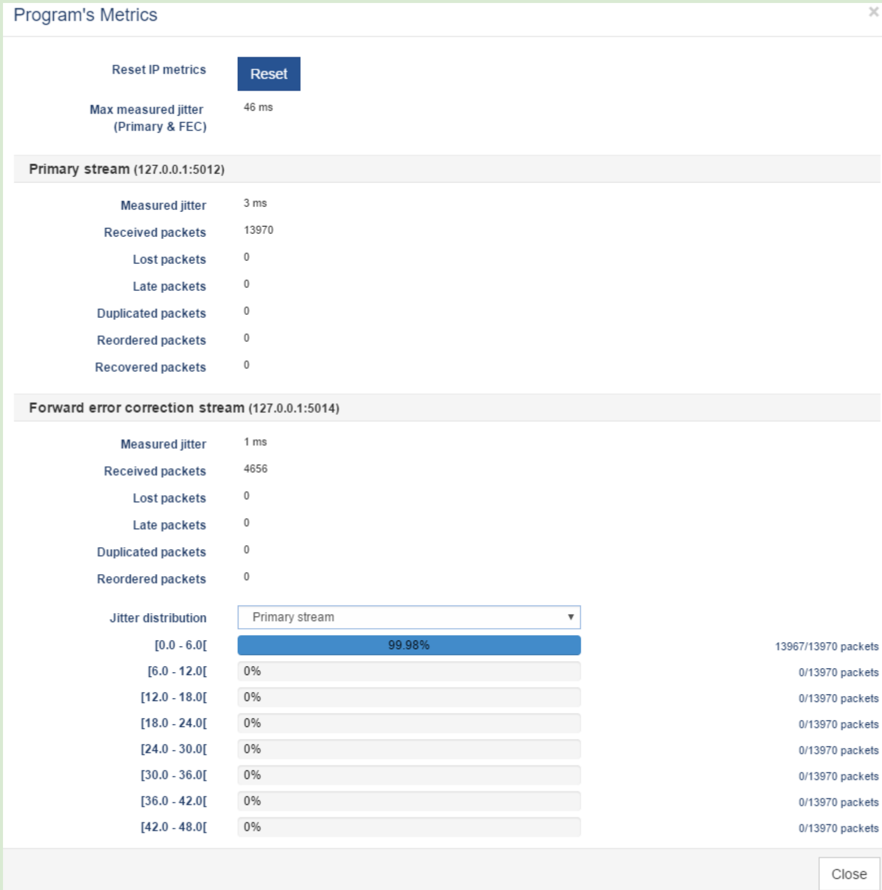
To view the content of a program, click on ⊞ on the left of its name.



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)

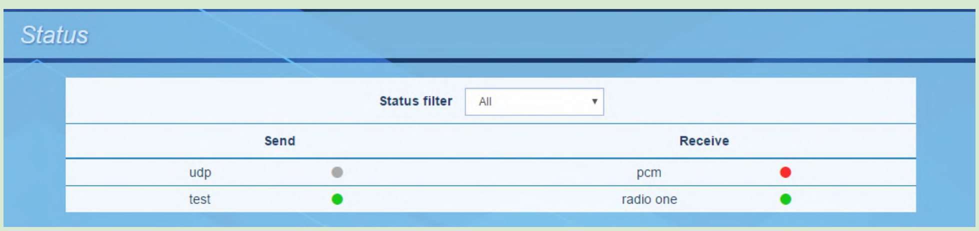
Variable	**Meaning**
Max Measured jitter	Displayed only if an FEC stream is received. Defines the minimum jitter to be configured in Receive−>IP Services (it includes the primary stream and the FEC stream). On unmanaged networks, we recommend configuring a higher value as the jitter may evolve and reach 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 also sure that the jitter value set in Receive−> IP services is higher than the max measured jitter.

8.1.5 Status

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).

8.2 WEB pages organization in “Remote Broadcasting” mode of use

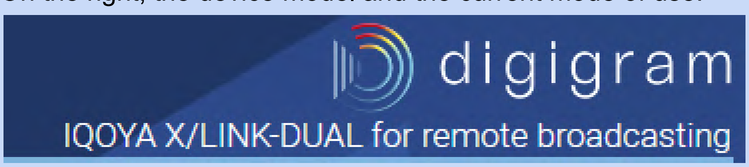
The WEB pages are organized in categories which are always accessible from the left side of the WEB pages.

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 X/LINK and this user manual.

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

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

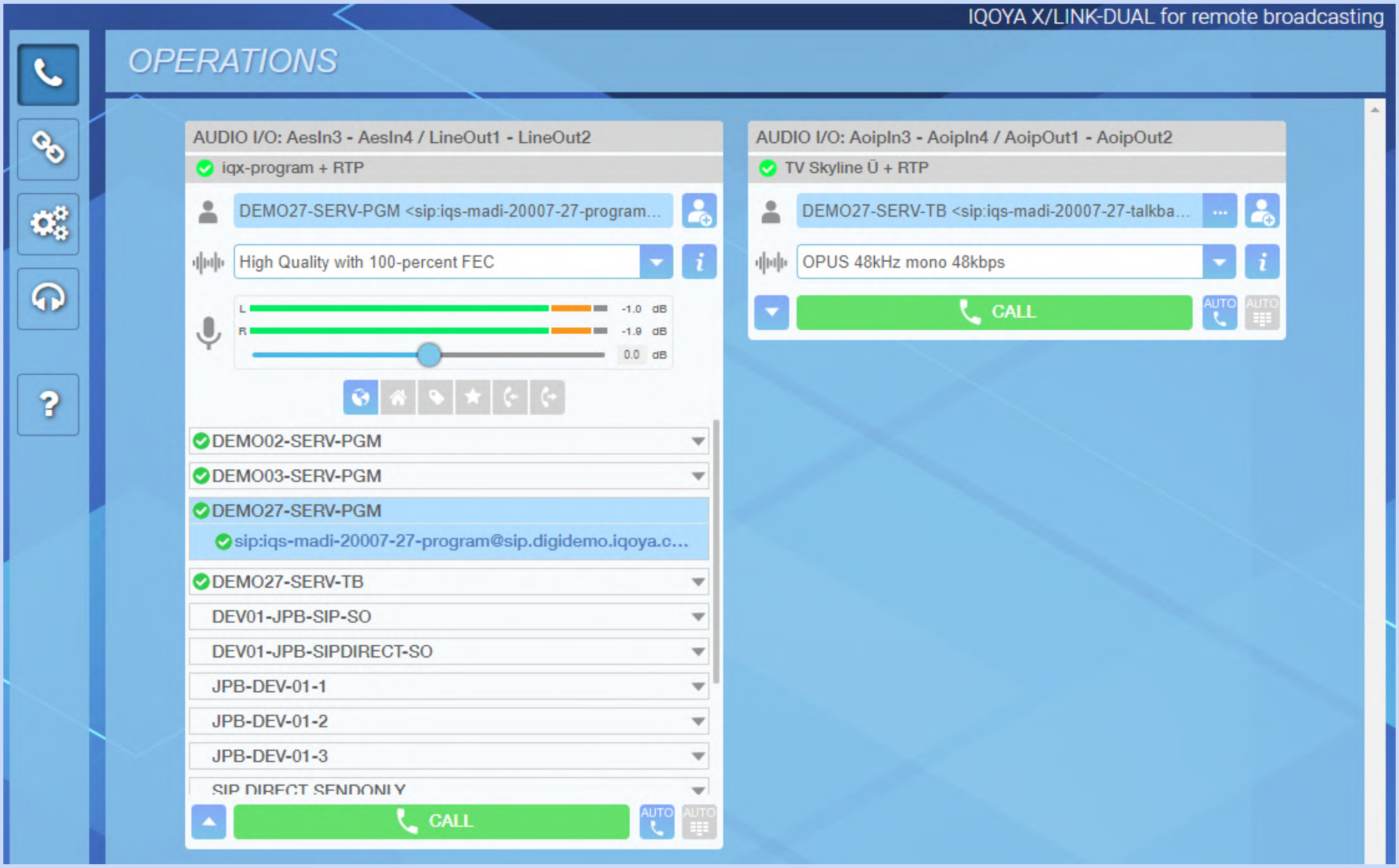
  The led is green when the two redundant power supplies work correctly,
  The led is red when one of the two redundant power supplies is out of order.
  The redundant power supplies are hot swappable.
- On the right, the device model and the current mode of use:



8.2.1 “Operations” page

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

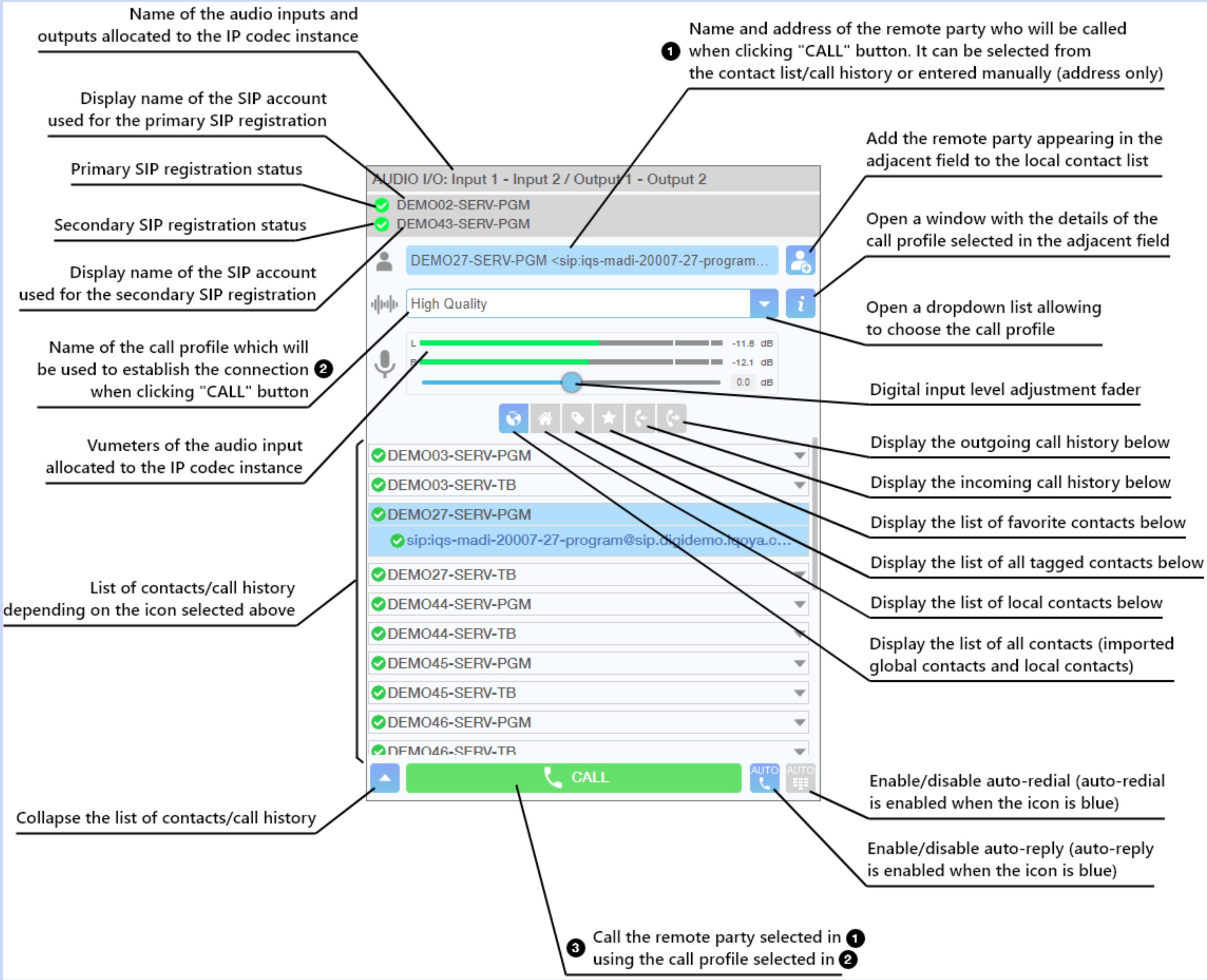
In the example below, the call window of the first IP codec instance is expanded while the call window of the second codec instance is collapsed:



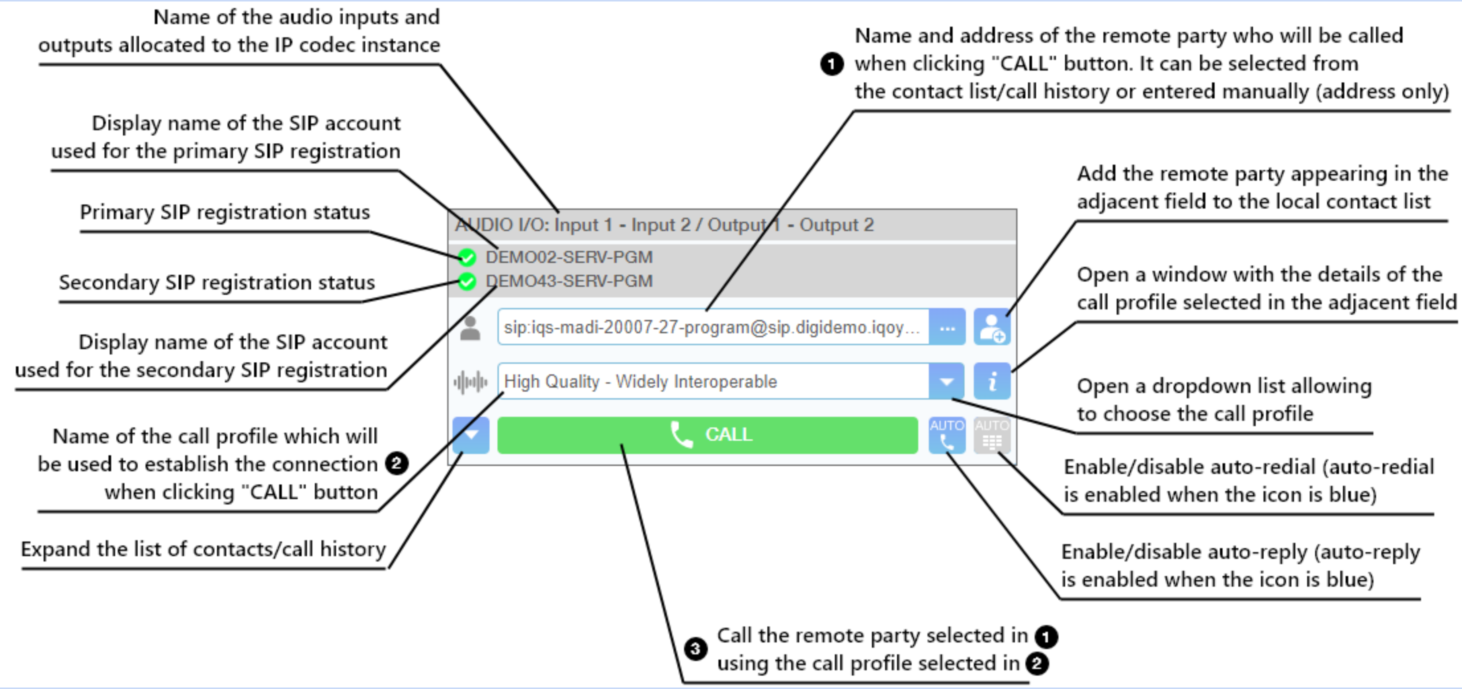
Each call window can be reopened in an independent window by double clicking on its title bar.

8.2.1.1 Call window when no communication is in progress

Call window (expanded version):



Call window (collapsed version):



8.2.1.2 Place a call

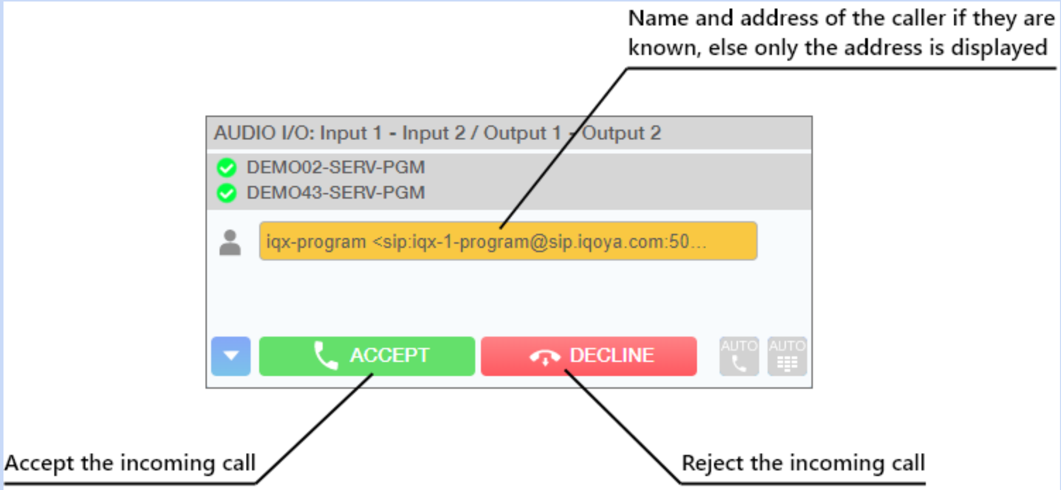
Please refer to the image of the previous paragraph for references to the graphical interface.
To place a call the user (1) select a remote party in the contact list/call history or enter the remote party address, (2) select a call profile and (3) press the CALL button. The CALL button is greyed out 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

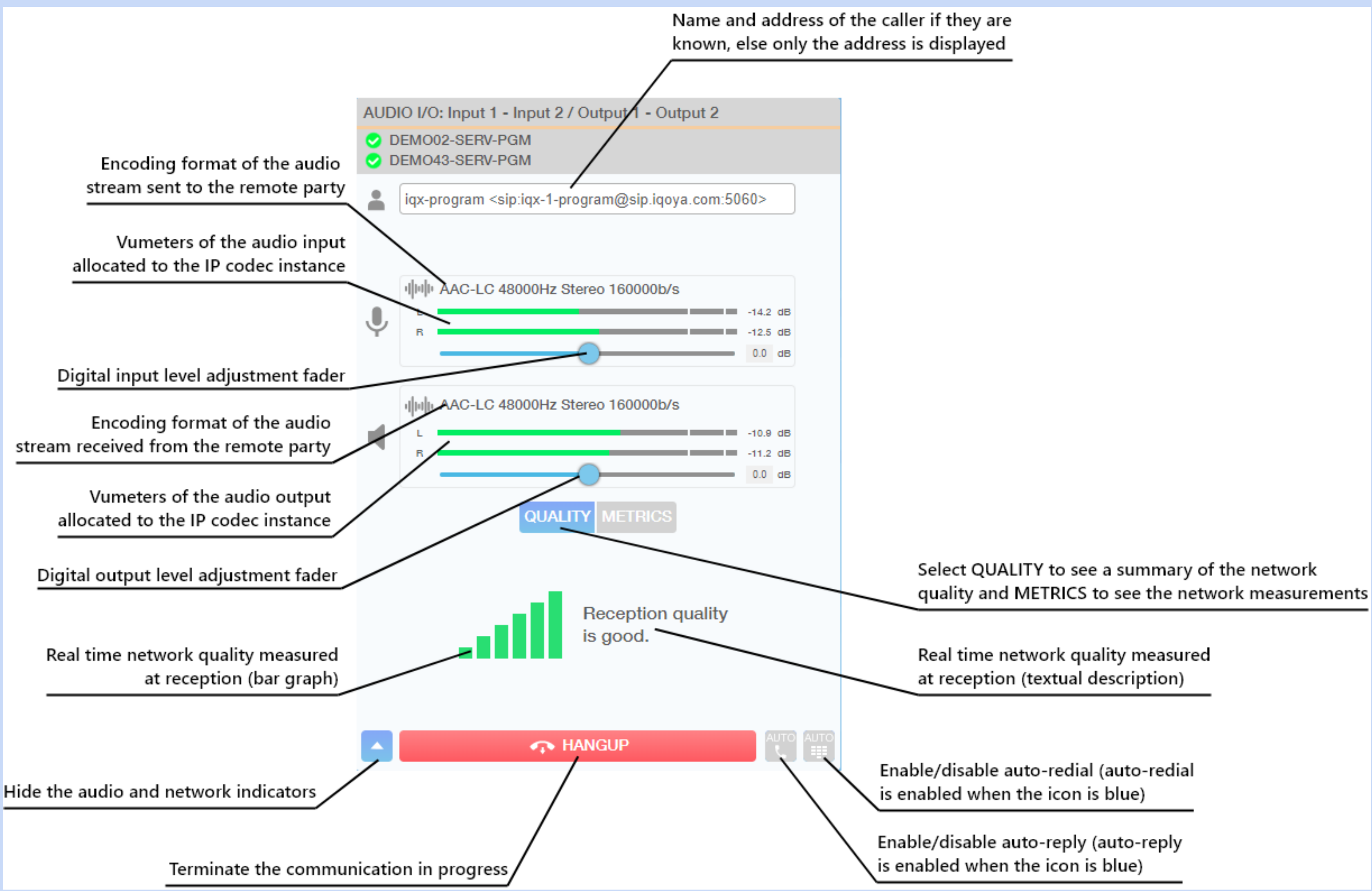
8.2.1.3 Accept or reject a call

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

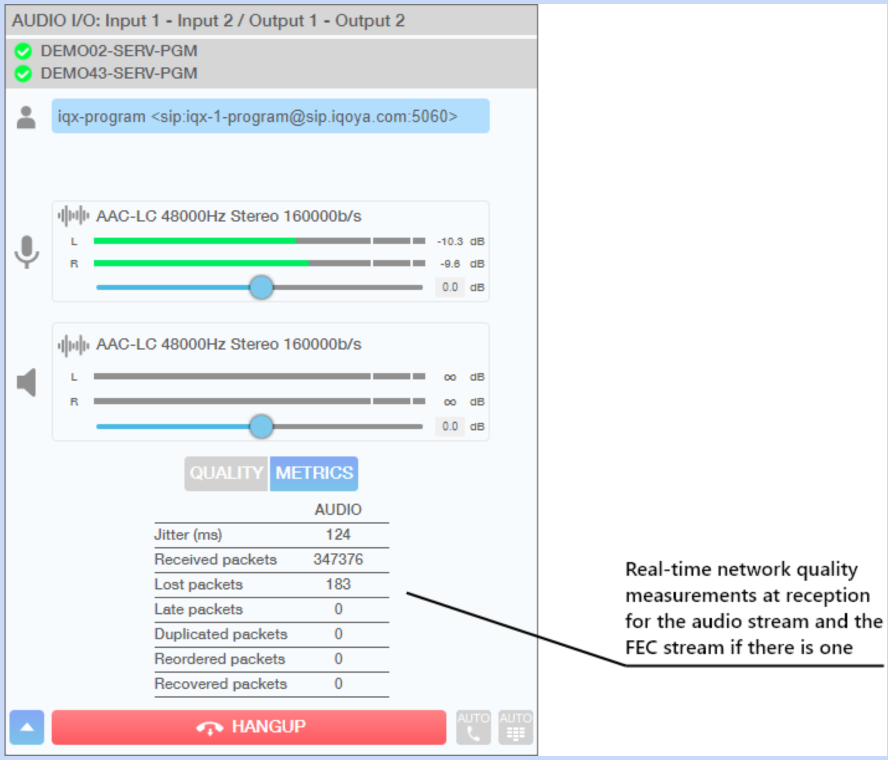


8.2.1.5 Call window when a communication is in progress

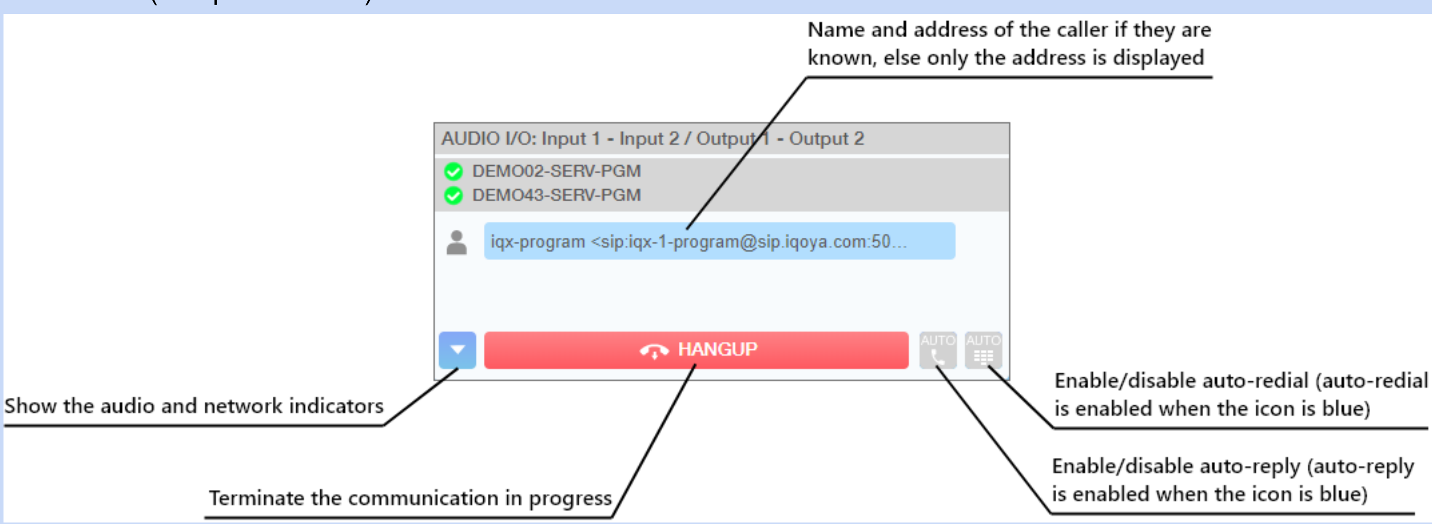
Call window (expanded version with network quality selector in QUALITY position):



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



Call window (collapsed version):

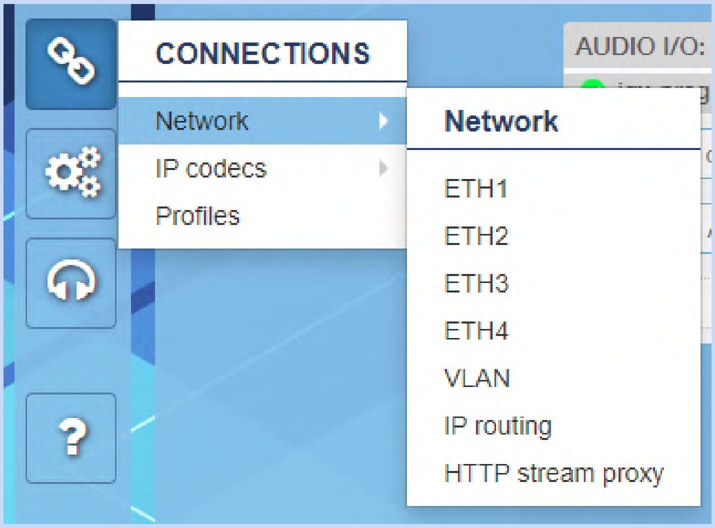


8.2.1.6 Hang up a call

Click the HANGUP button to terminate the communication.

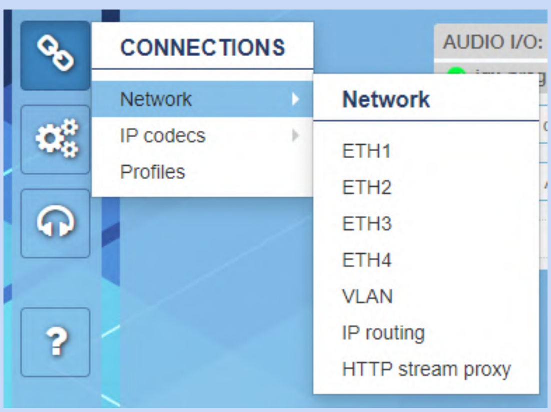
When auto-redial is activated on the caller's side, only the caller can terminate the communication. If the callee hangs up, the communication is automatically re-established by the caller device.

8.2.2 “Connections” category of menus



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

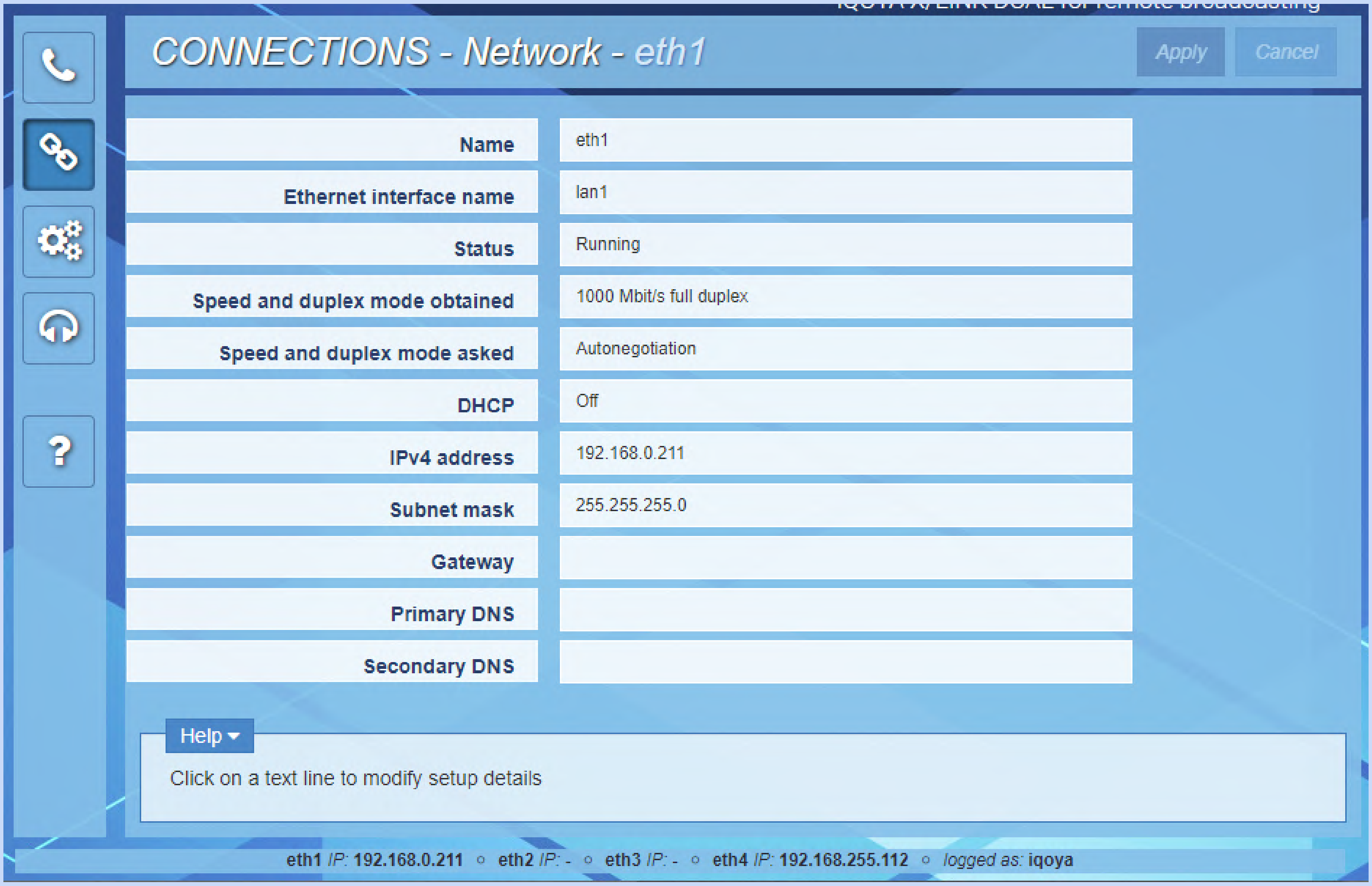
8.2.2.1 Connections –> Network



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

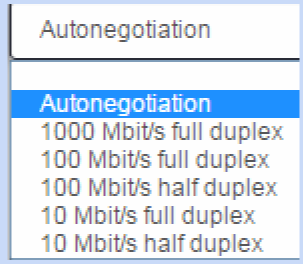
8.2.2.1.1 Connections –> Network –> ETHx

These pages allow configuring the four network ports of IQOYA X/LINK.



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

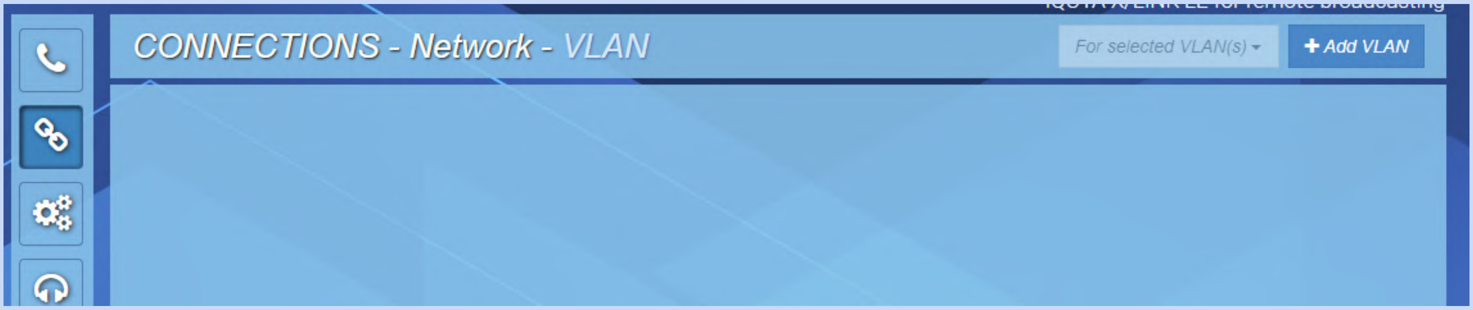
Parameter	Type	Description
Name	R/W	This is the logical name of the ethernet interface which will be used in all the graphic user interfaces and in particular in the web pages. The names in the factory configuration are ETH1 to ETH4.
Ethernet interface name	Read	Displays the physical name of the ethernet ports. This parameter can’t be changed.
Status	Read/Write	This parameter allows enabling/disabling the interface Default value=Running Possible values: Running: ethernet port is enabled. Stopped: ethernet port is disabled
Speed and duplex mode obtained	Read	Displays the current speed and mode of the ethernet interface.
Speed and duplex mode asked	Read/Write	Allows selecting the working mode of the ethernet interface.

		Possible values are as follows:  We recommend avoiding the “Auto-negotiation” mode. Select the mode supported by the network node connected to the IQOYA X/LINK.
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 5 following parameters.
IPv4 address	Read only if DHCP is On Read/Write if DHCP is Off	**DHCP Off** Default value is: 192.168.0.100 for Eth1, 192.168.1.100 for Eth2, 192.168.2.100 for Eth3, 192.168.3.100 for Eth4 Enter the IP address of this ethernet interface. **DHCP On** Displays the IP address automatically set by DHCP.
Subnet mask	Read only if DHCP is On Read/Write if DHCP is Off	**DHCP Off** Enter the mask of the subnetwork this ethernet port belongs to. **DHCP On** Displays the subnetwork mask automatically set by DHCP.
Default Gateway	Read only if DHCP is On Read/Write if DHCP is Off	**DHCP Off** Enter the default gateway IP address. Streams sent beyond the subnets configured on LAN1 to 4 will pass through this gateway except if specific routing rules have been defined in the IP routing page. Only one default gateway must be configured for all the ethernet interfaces. If several gateways have to be used, one can be set as default gateway, the others must be the subject of routin rules in the IP routing page. **DHCP On** Displays the gateway IP address automatically set by DHCP.
Primary DNS	Read only if DHCP is On Read/Write if DHCP is Off	**DHCP Off** Enter the IP address of the primary DNS (if any). **DHCP On** Displays the IP address of the DNS automatically set by DHCP.
Secondary DNS	Read only if DHCP	**DHCP Off**

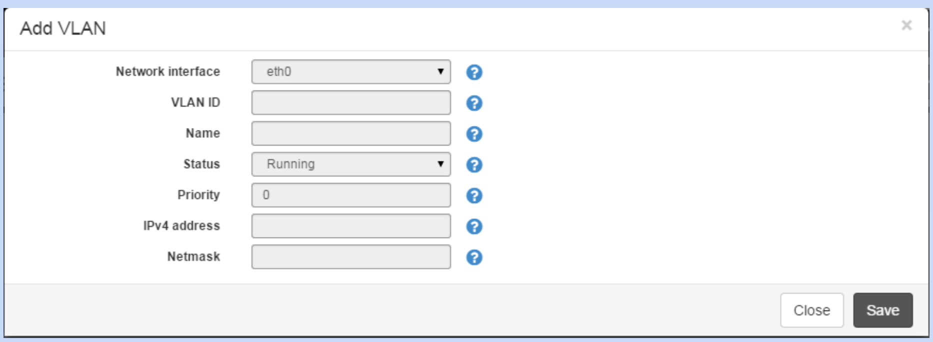
	is On Read/Write if DHCP is Off	Enter the IP address of the secondary DNS (if any). **DHCP On** Displays the IP address of the secondary DNS automatically set b DHCP (may be empty).

8.2.2.1.2 Connections –> Network –> VLAN

This page allows declaring VLANs on the ethernet interfaces. No VLAN is declared by default.
Multiple VLANs can be declared for each ethernet interface.



Click on the "+Add VLAN" button to declare a new VLAN.

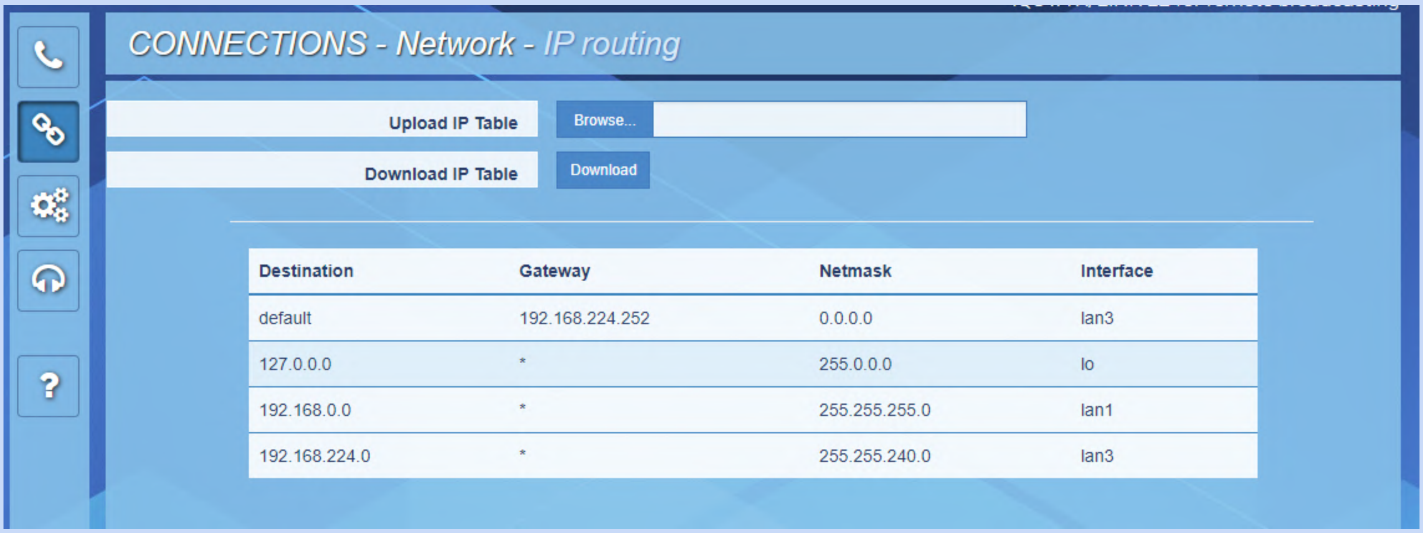


Parameter	Type	Description
Network interface	Read/Write	Select the network interface that will support the VLAN (ETH1 to ETH4)
VLAN ID	Read/Write	Enter the VLAN ID in the ranges 1-4094. Avoid using ids 1002 to 1005 which are reserved.
Name	Read/Write	Enter a logical name for this VLAN
Status	Read/Write	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 ethernet port in this VLAN. If no value is entered, the IP address is the IP address of the selected ethernet port.
Netmask	Read/Write	Enter the netmask for this VLAN interface. If no value is entered, the netmask is the same as the selected ethernet port netmask.

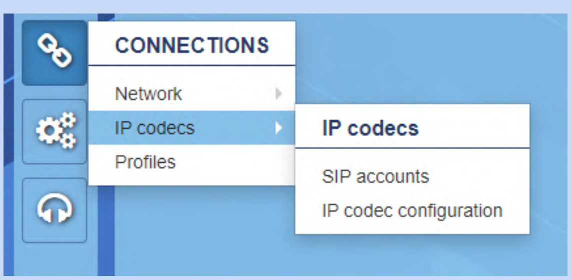
8.2.2.1.3 Connections –> Network –> IP routing

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. **Only the additional routes must appear in this file. Routes to directly accessible subnets are not present in this file and need not be added to this file.**
Note: In case you use more than one ethernet interface, do not declare several gateways. Declare instead one default gateway, for instance on Eth0, and declare routes on other ethernet interfaces through this routing table.

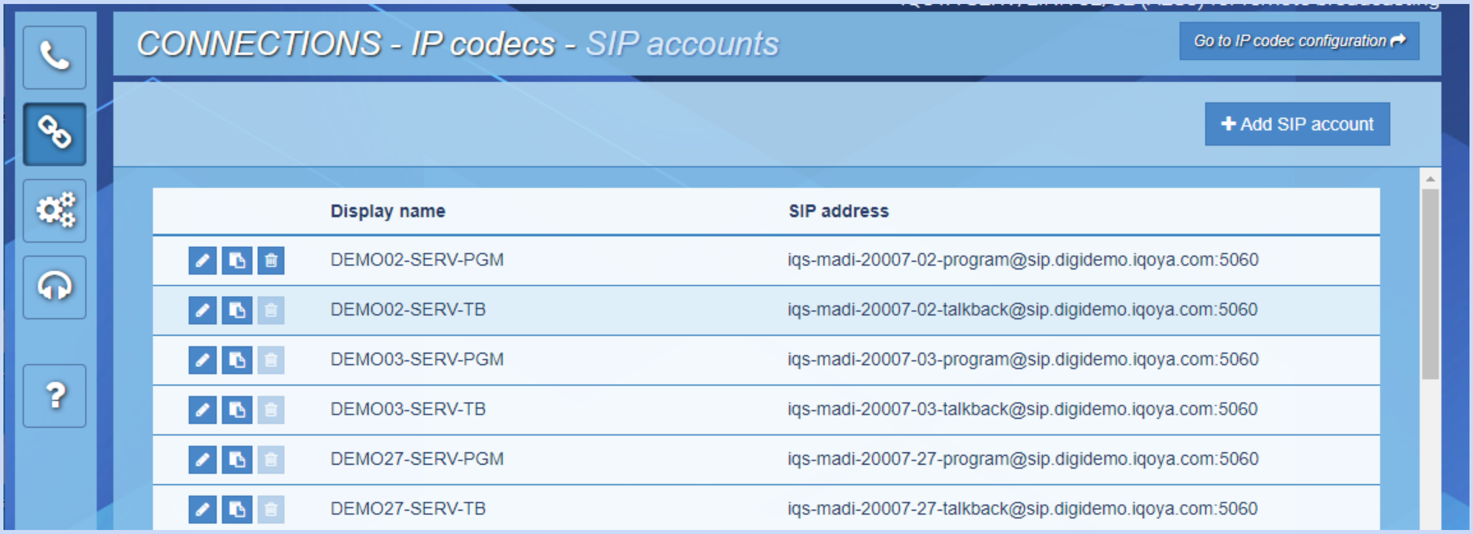
8.2.2.2 Connections –> IP codecs



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

8.2.2.2.1 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.



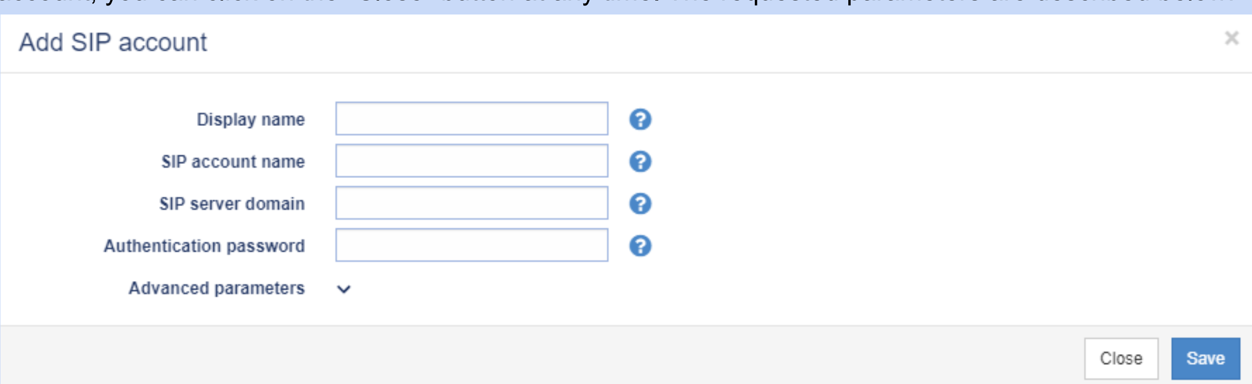
The shortcut 

allows you to quickly jump to the IP codec configuration page described below.

8.2.2.2.1.1 Declare a new SIP account

To declare a new SIP account, click on 

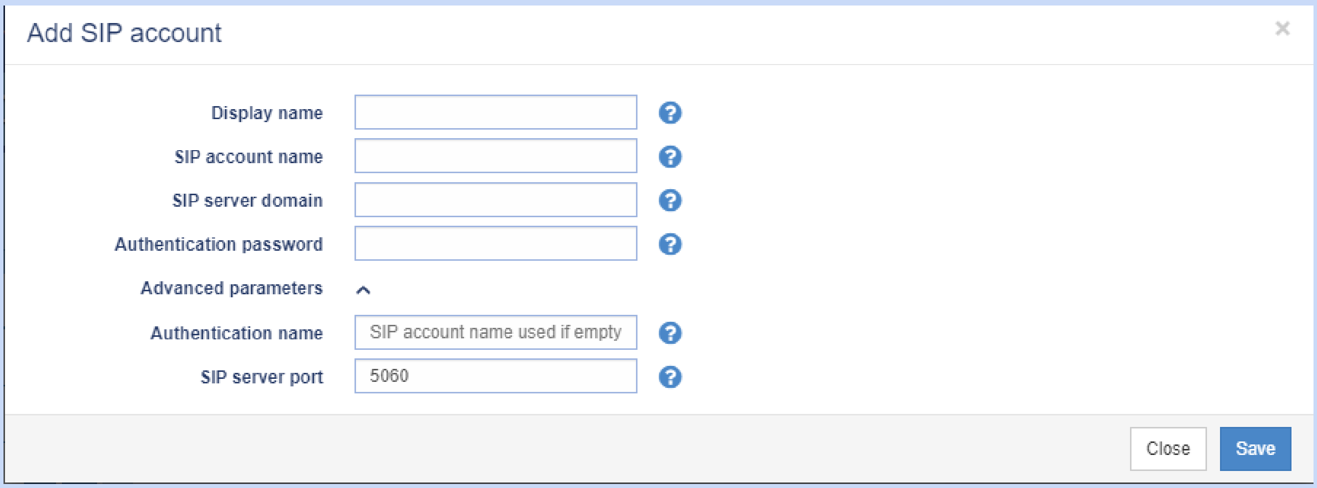
, or create it from an existing one by clicking the icon  on the its left. Then provide the requested parameters and click on the "Save" button. To cancel the declaration of a new SIP account, you can click on the "Close" button at any time. The requested parameters are described below:



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.

8.2.2.2.1.2 Edit a SIP account

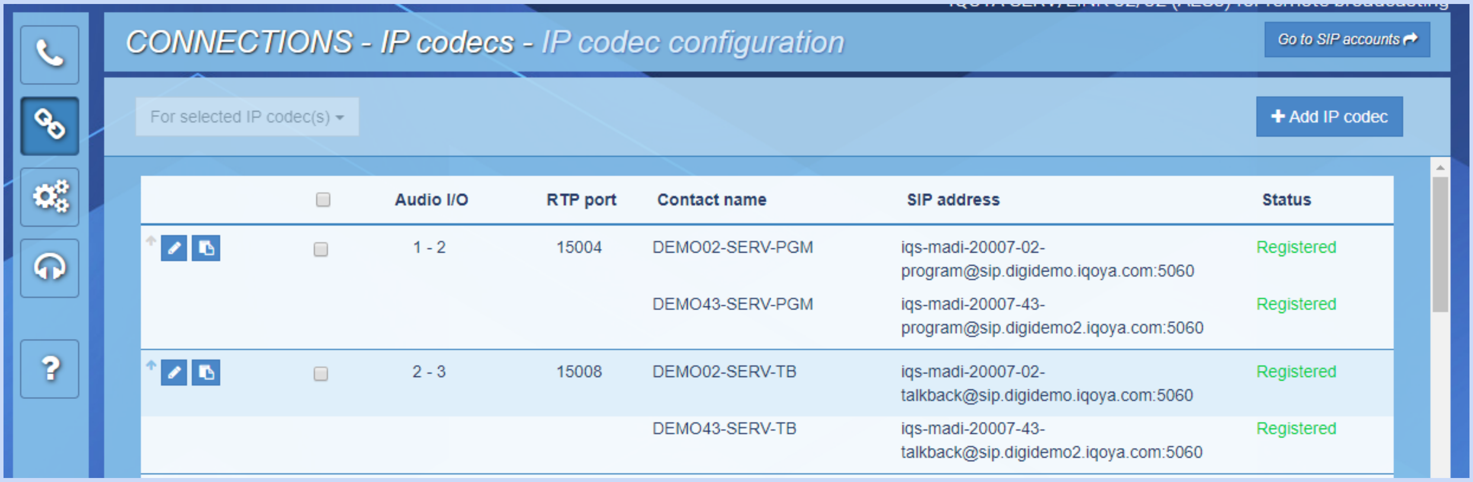
To edit an existing SIP account, click the icon on its left.
The edit page is identical to the add page described in the previous paragraph.

8.2.2.2.1.3 Delete a SIP account

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

8.2.2.2.2 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:

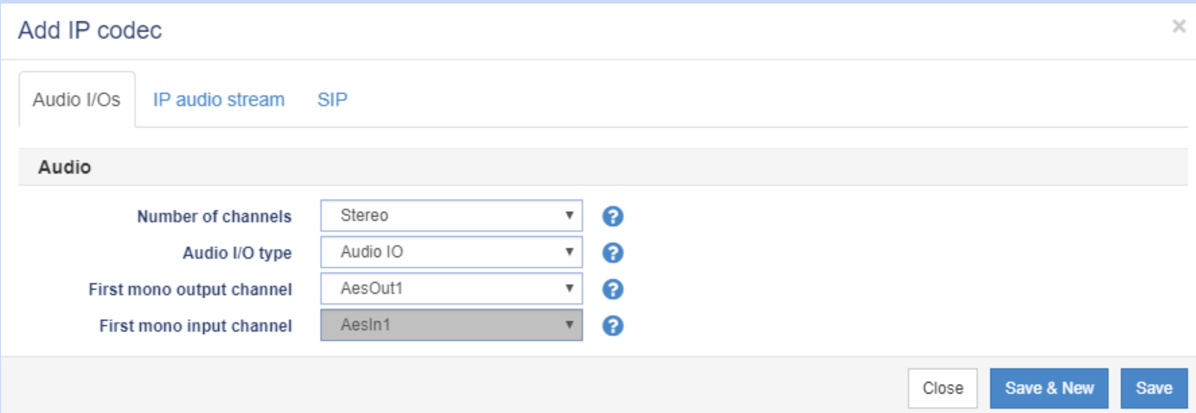
- **Audio I/O**: The audio I/Os associated with the IP codec instance.
- **RTP port**: The port used by the the IP codec instance to listen to the IP audio stream coming from the remote party.
- **Contact name**: The display names of the SIP accounts used to register the IP codec instance with SIP servers (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.

8.2.2.2.2.1 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 its left. 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.

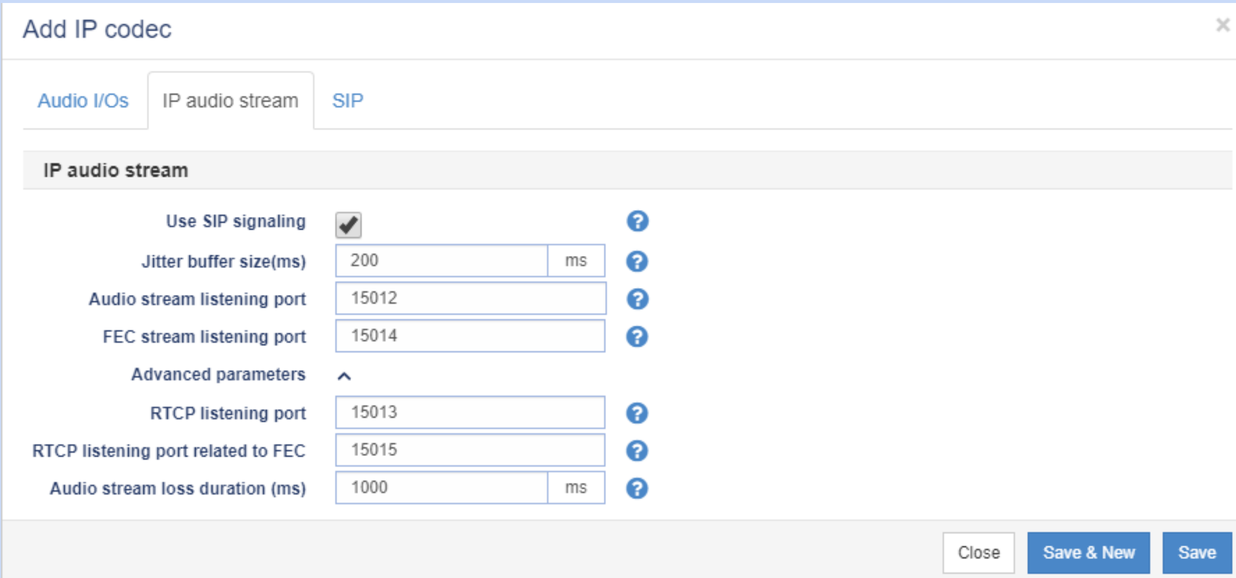
The requested parameters are described below:

- Parameters related to the audio I/Os



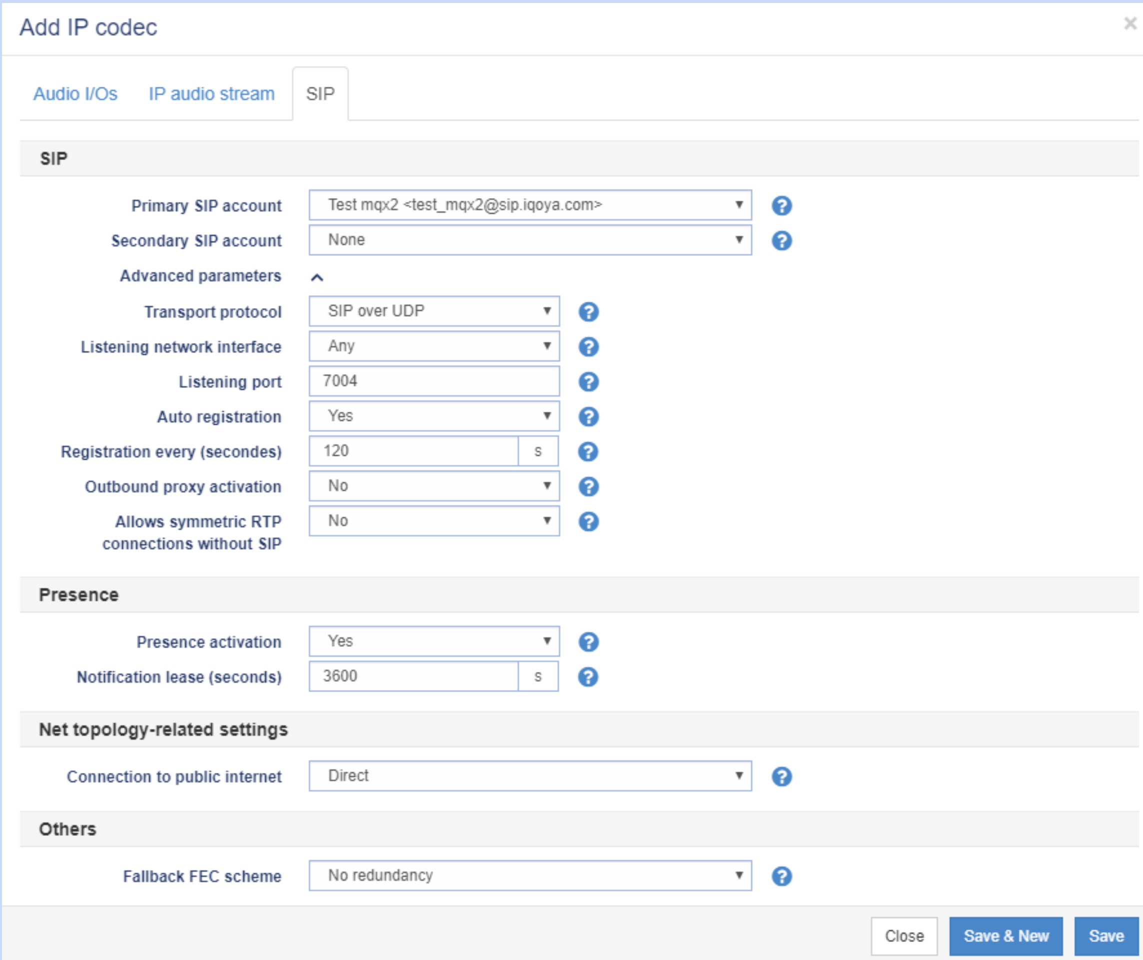
IP codec parameter	**Type**	**Description**
Number of channels	Read/Write	Number of audio channels managed by the IP codec instance. It can be Mono or Stereo.
Audio I/O type	Read/Write	Type of the audio I/Os allocated to the IP codec instance. It can be “Audio IO” for Analog or AES/3 I/Os or “AoIP” for 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



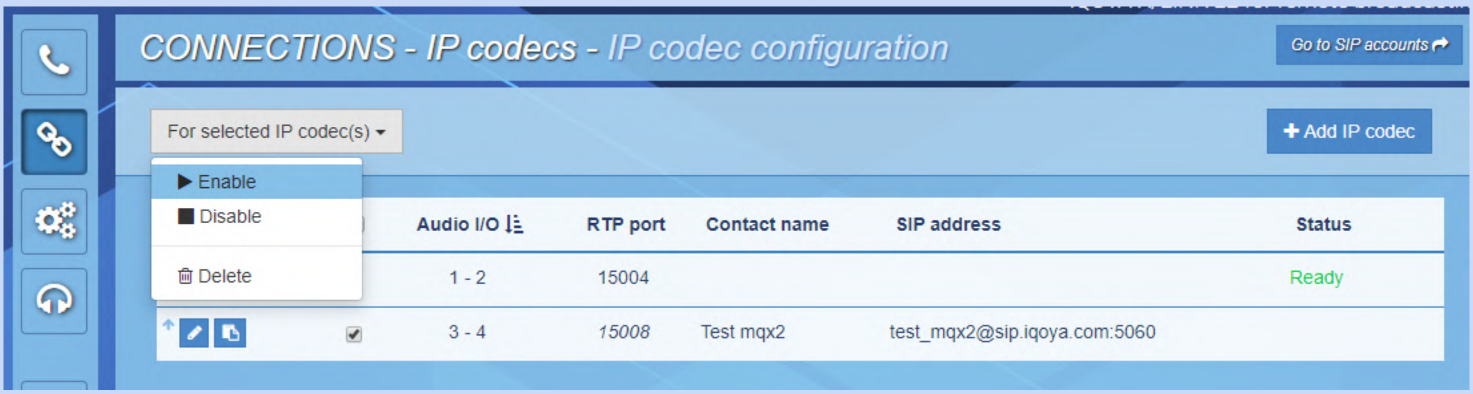
IP codec parameter	Type	Description
SIP section		
Primary SIP account	Read/Write	Primary SIP account to be used by the codec instance to register with a SIP server
Secondary SIP account	Read/Write	The codec instance can register on 2 SIP servers at the same time. So if one SIP infrastructure breaks down, 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 settings		
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.

8.2.2.2.2.2 Enable IP codec instances

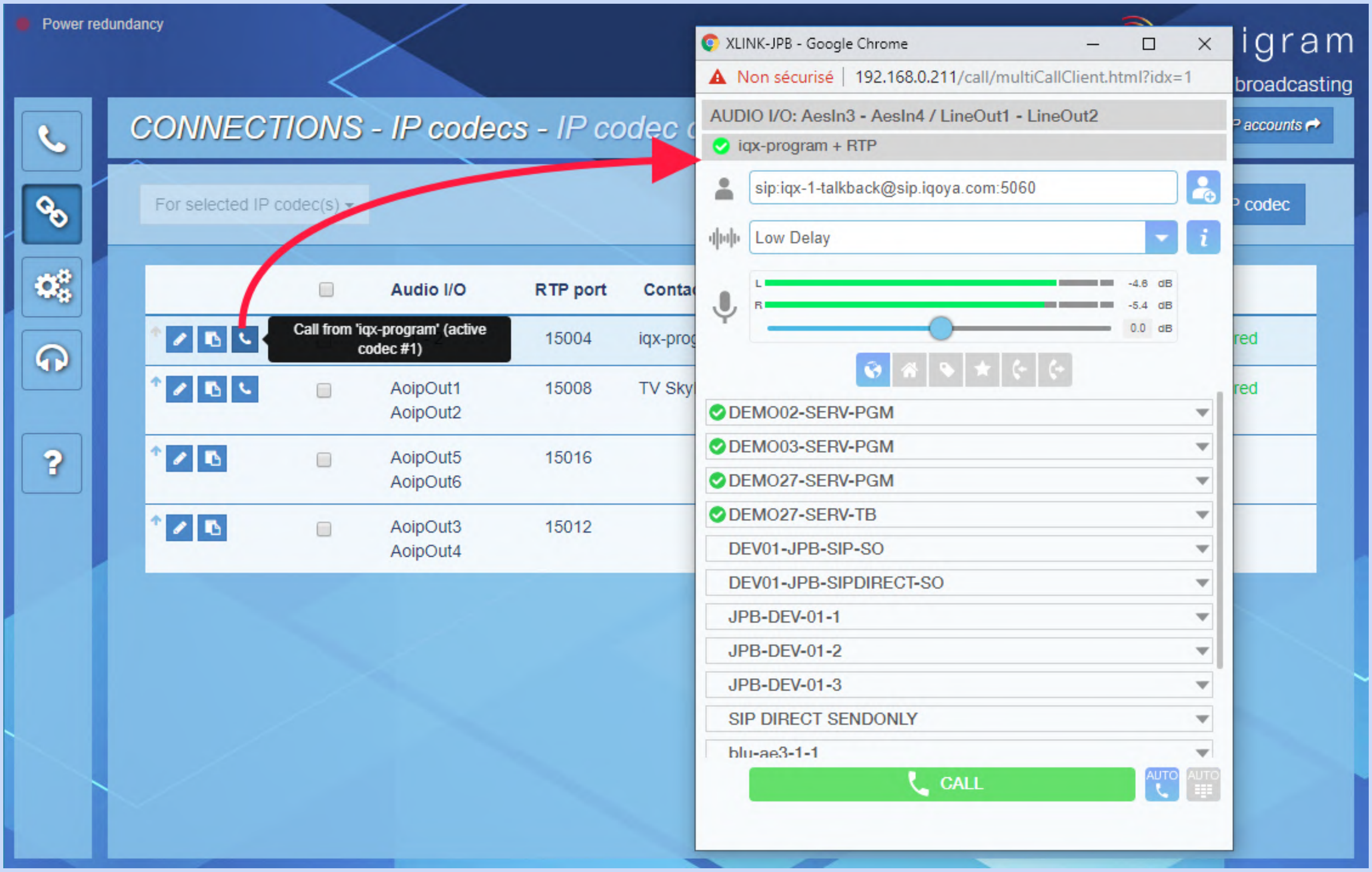
After creation, the IP codec instances must be enabled to appear in the “Operations” page.



To enable IP codec instances, select them by 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.

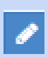
The enabled IP codec instances appear on the “Operations” page in the same order as they appear in the list. It is possible to reorder the list clicking the “up” icon present at the beginning of each line:

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:

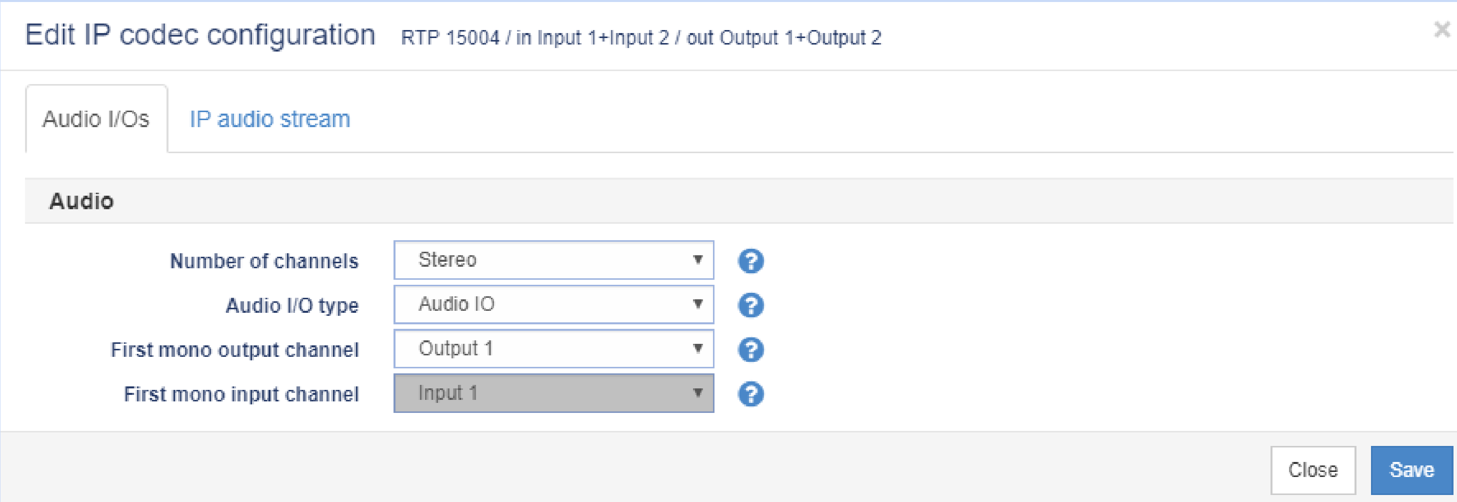


8.2.2.2.2.3 Edit an IP codec instance

It is possible to edit an existing IP codec instance.

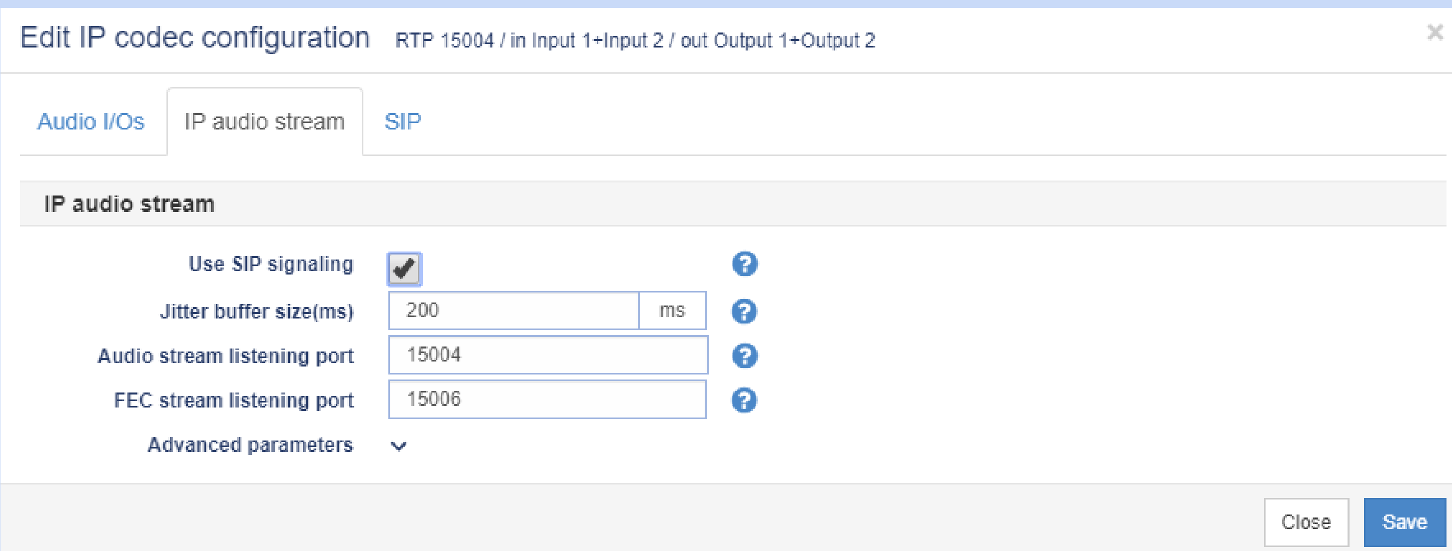
Click the pencil icon  on the line of the IP codec instance you want to modify to start editing. The edition gives access to the same settings pages as the creation:

- Parameters related to the audio I/Os



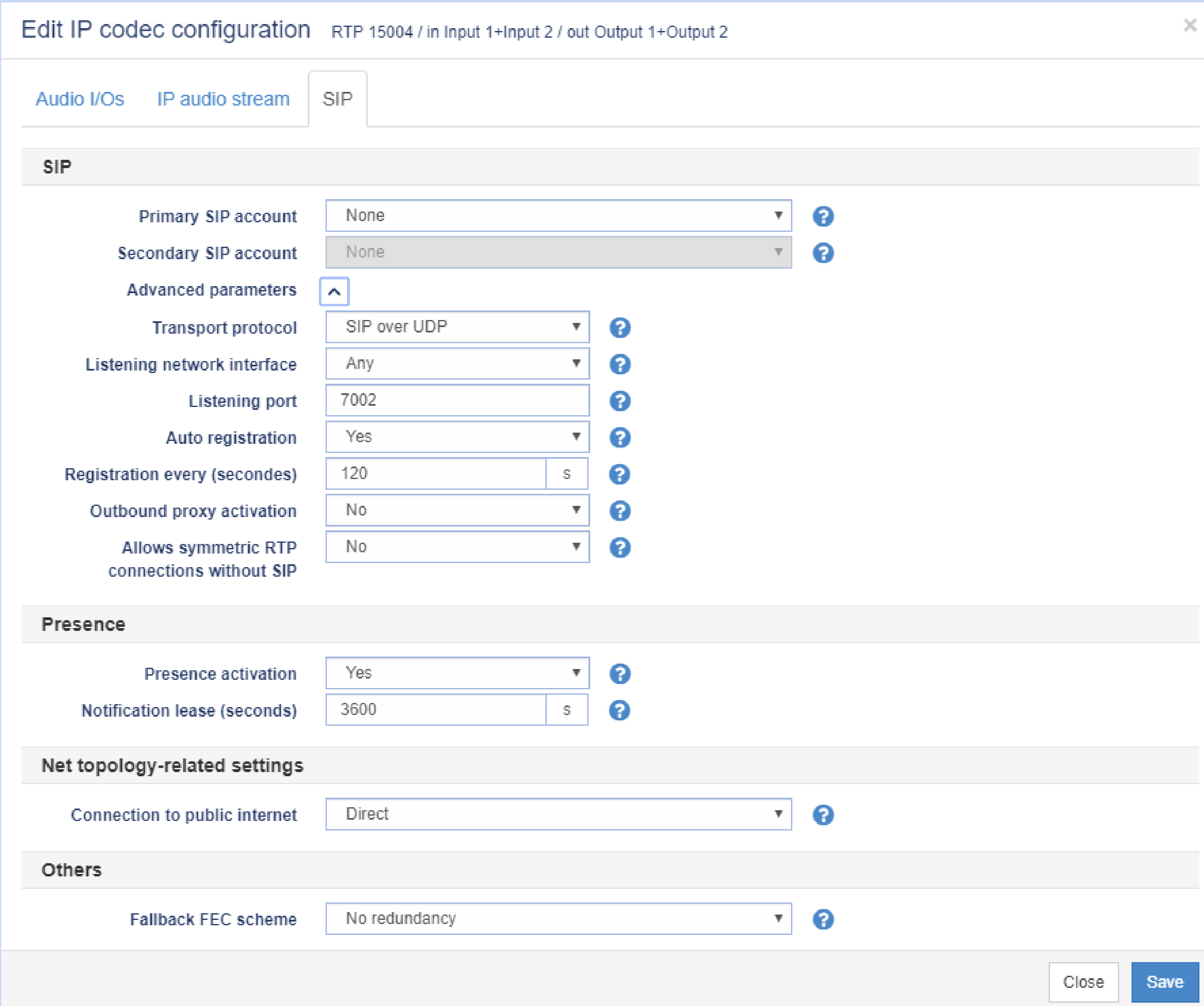
Refer to the paragraph "Create a new IP codec instance" to know the meaning of each parameter.

- 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 each parameter.

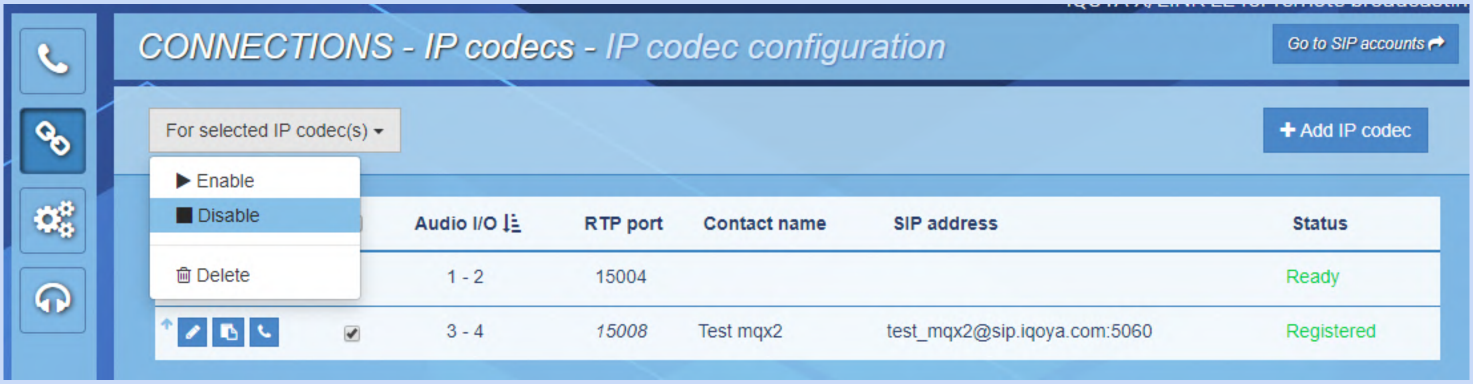
- Parameters related to SIP



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

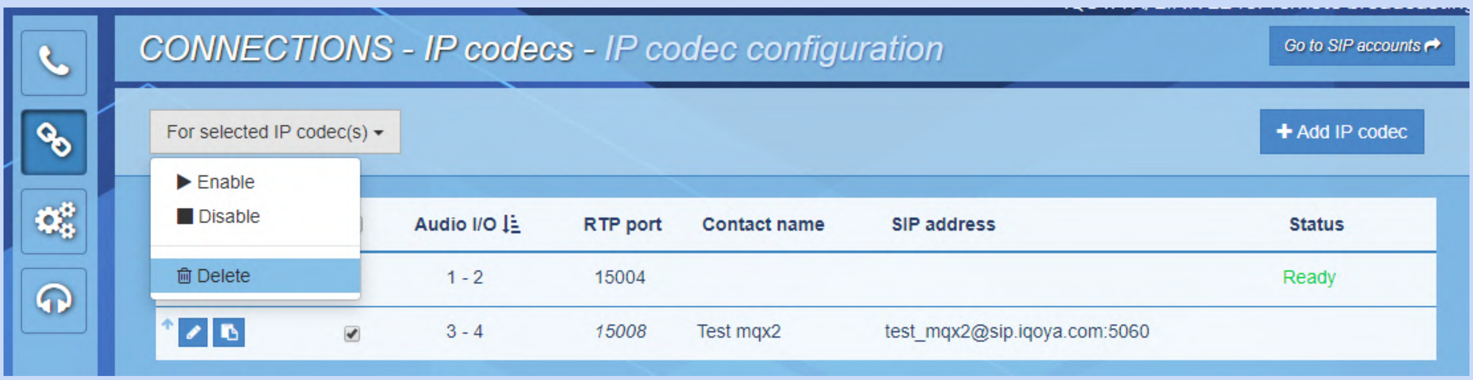
8.2.2.2.2.4 Disable IP codec instances

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



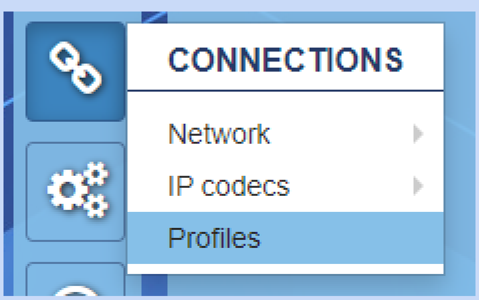
8.2.2.2.2.5 Delete IP codec instances

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



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

8.2.2.3 Connections –> Profiles



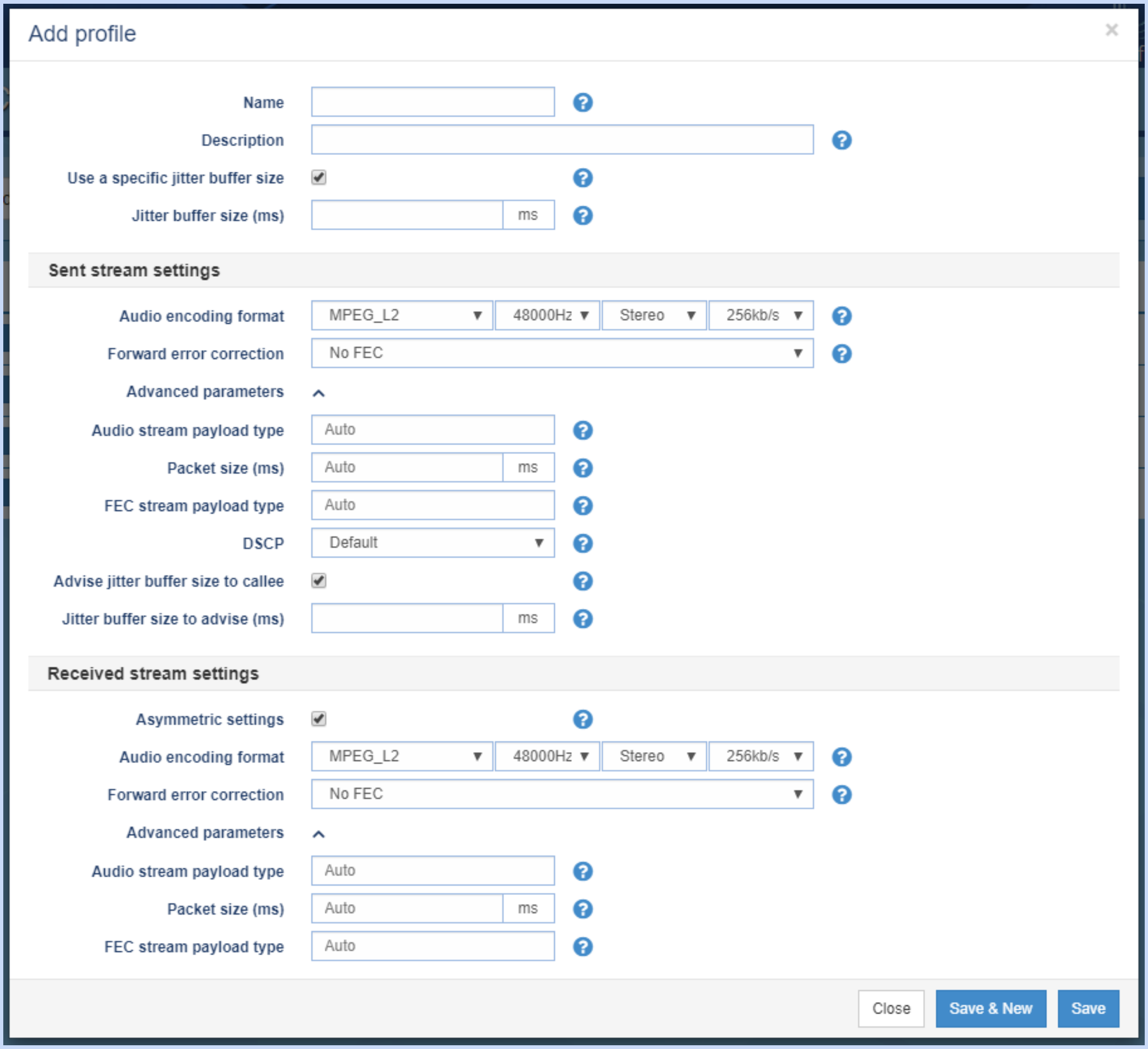
This menu gives access to the call profile management page. This page allows you to add, modify or delete call profiles stored only locally on the device.

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

8.2.2.3.1 Add a call profile

To create a new call profile, click on 

, or create it from an existing one by clicking the icon  on its left. 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 allowing to correct them. This parameter allows 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)

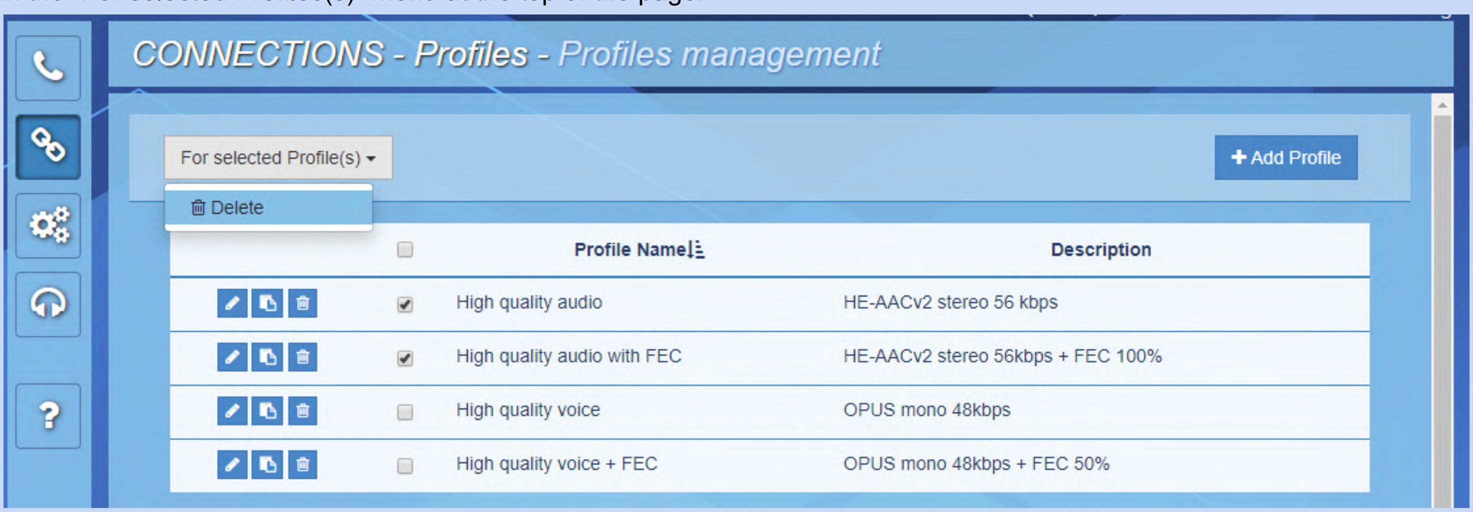
		• +25% bandwidth, recovery 1/4, 2 streams (audio + FEC) • +20% bandwidth, recovery 1/5, 2 streams (audio + FEC) • +10% bandwidth, recovery 1/10, 2 streams (audio + FEC) 'recovery N' means that up to N consecutive lost IP packets can be reconstructed thanks to the FEC scheme, 'recovery 1/N' means one 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.

8.2.2.3.2 Edit a call profile

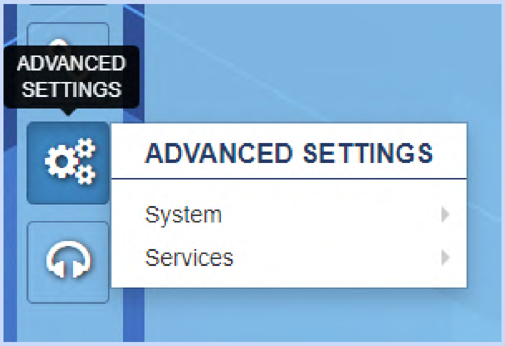
To edit an existing call profile, click the icon on its left.
The edit page is identical to the add page described in the previous paragraph.

8.2.2.3.2 Delete call profiles

To delete a call profile, click the icon on its left.
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:



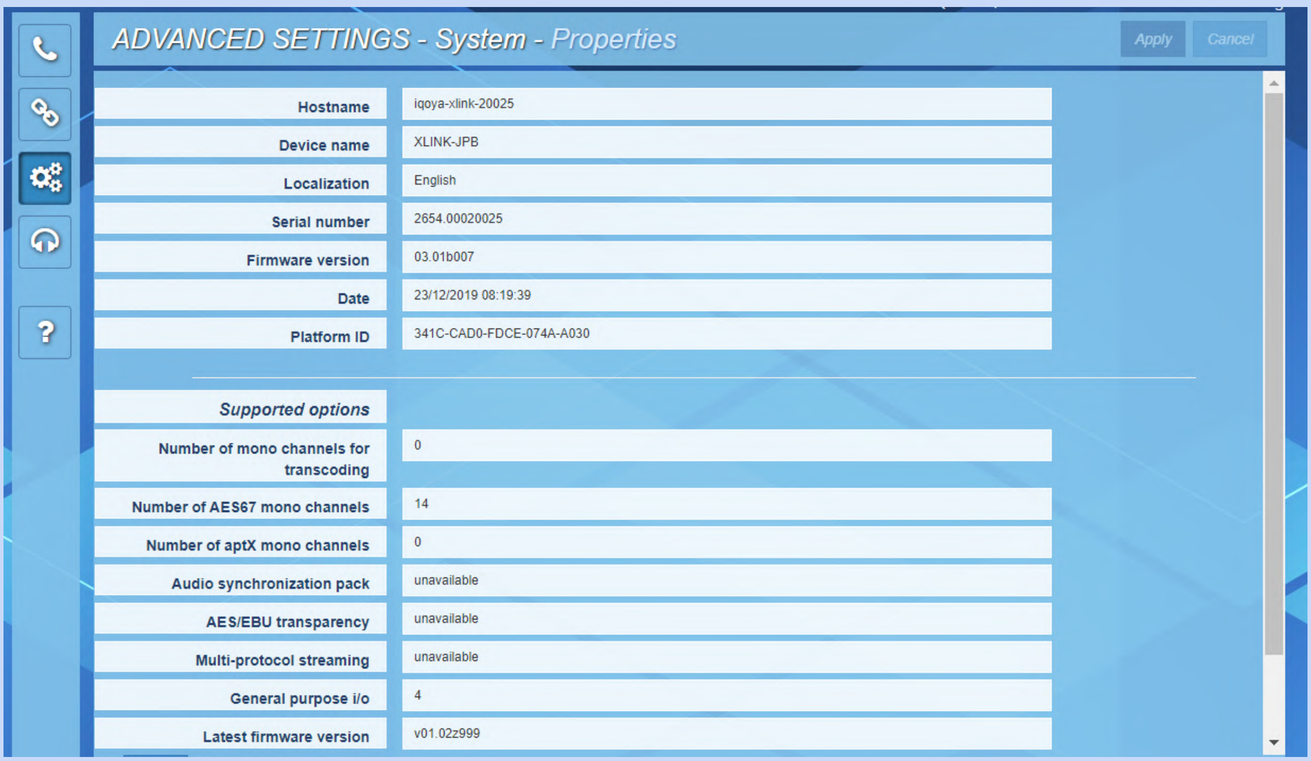
8.2.3 “Advanced settings” category of menus



8.2.3.1 Advanced settings –> System

8.2.3.1.1 Advanced settings –> System –> Properties

This page displays the system properties:



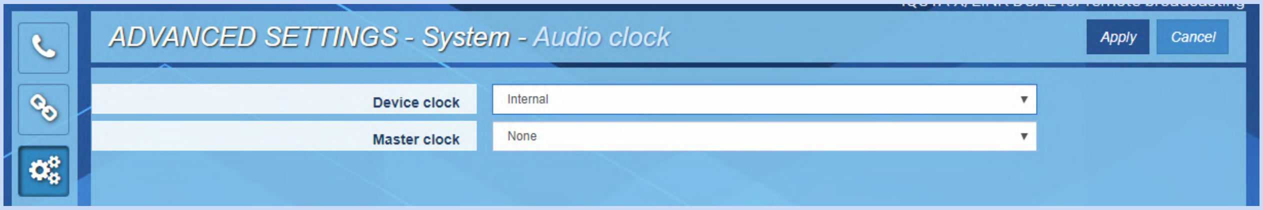
Description of the parameters:

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 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.
Supported Options		
Number of mono channels for transcoding	R	Number of mono channels supported for transcoding through internal buses.
Number of AES67 mono channels	R	Number of mono input and output channels on AES67 or Ravenna, or Livewire
Number of aptX mono channels	R	Number of mono channels to be processed in aptX
Audio synchronous pack	R	Value 1: the codec features the audio synchronization via NTP, Value 0 : the option is not installed.
AES/3 transparency	R	Value 1: the codec allows for AES transparency transport. Value 0; the option is not installed.
Multiprotocol streaming	R	Value 1: the codec features the multiprotocol streaming. Value 0: the option is not installed
Latest firmware version	R	Maximum firmware version number authorized by the ongoing support contract.
Support contract validity date	R	Defines the date until when the firmware can be updated/upgraded according to the purchased support contract.

8.2.3.1.2 Advanced settings –> System –> Audio Clock

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



Device clock

The clock source can be:

- Internal: on-board clock
- Extracted from an AES/3 input (not available on X/LINK-AES67)
- A PTP clock (AES67, RAVENNA)
- A Livewire clock

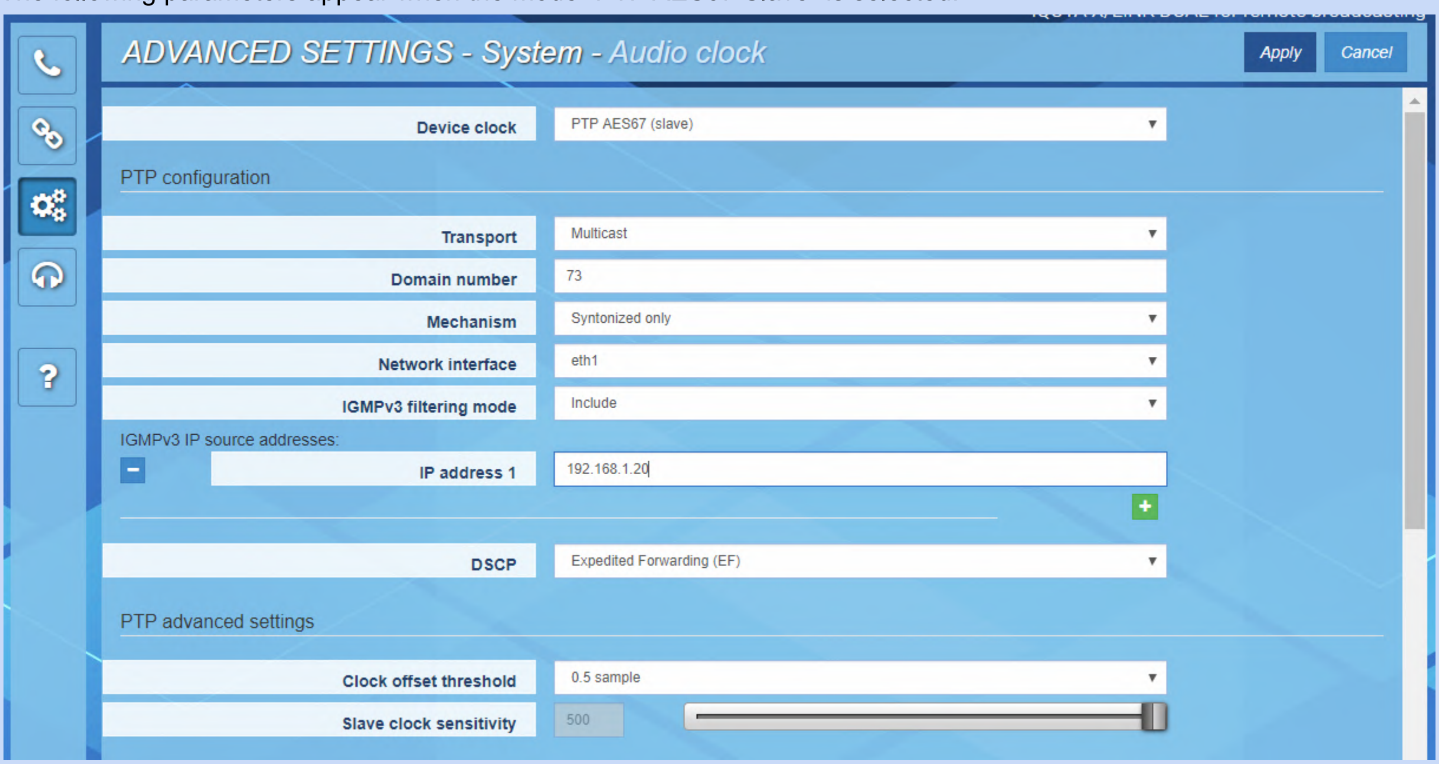
The clock sampling frequency value is set from Preferences–>Audio setup.

Master clock

Allows defining if the codec generates a PTP clock.

8.2.3.1.2.1 PTP clock source

The following parameters appear when the mode “PTP AES67 Slave” is selected:

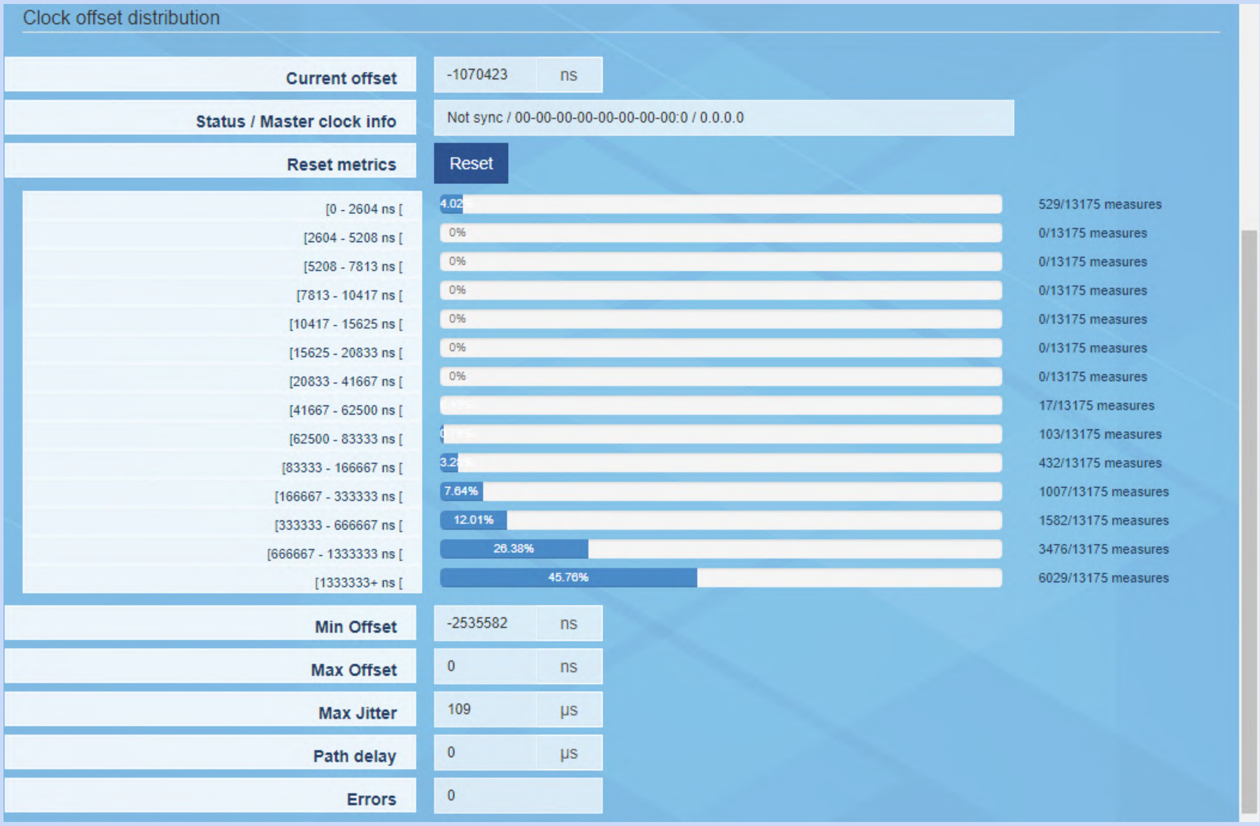


Description of the parameters:

Parameter	**Read/ Write**	**Meaning**
Transport	R/W	Allows specifying if the PTP clock is unicast or multicast.
Domain number	R/W	PTP clock domain number (from 0 to 128)
Mechanism	R/W	**Syntonized**: means that IQOYA's clock is the same as the Grandmaster PTP, but they are not synchronous (delay between the two clocks). Synchronous clock is obtained thanks to E2E or P2P modes, which serve to compensate for the delay between Grandmaster PTP clock and IQOYA. **E2E** is a more universal setting (it consists of requests and answers between the node (IQOYA) and the Grandmaster PTP clock unit). **P2P** provides higher clock sync precision but requires full PTP support from all participating switches (between IQOYA and related clock master.) In case the PTP clock is generated by an IQOYA, the PTP mechanism must be the same as in the IQOYA master: syntonized.
Network interface	R/W	Select the network interface that receives the PTP
IGMPv3 filtering mode	R/W	Off: X/LINK subscribes to the multicast PTP clock which can be generated by any source IP address. Include: X/LINK subscribes to the multicast PTP clock which is generated only by the listed source IP addresses. Exclude: X/LINK subscribes to the multicast PTP clock which is generated by any source IP address, with exception of the listed IP addresses..
IGMPv3 IP source addresses		
IP address x	R/W	Allows declaring the source IP addresses to be included or excluded. Click on + to add an IP@ to the list.
DSCP	¨R/W	QoS assigned to the PTP frames. Select the value from the drop down list. For optimal QoS on PTP, "Expedited forwarding (EF)" value is recommended.
PTP advanced settings		
Clock offset threshold	R/W	This parameter defines the condition for being synchronized to the PTP clock. The lower the value, the better the phase with the PTP clock. Lower values require a deterministic network. For networks that introduce an erratic jitter to the PTP frames, the value must be increased. Default value is 0.5 sample. It can be increased (up to 64 samples).

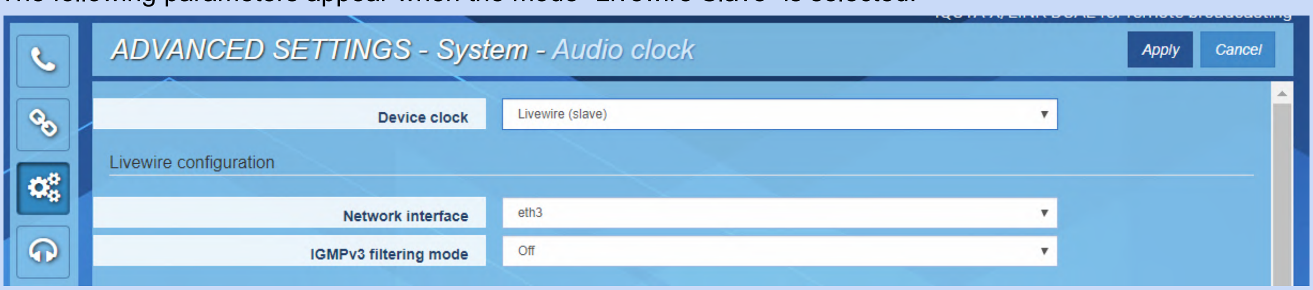
Slave clock sensitivity	R/W	It defines the sensibility of the slave clock to the PTP packet jitter. Enter a value between 500 (for a high sensitivity) and 100 (for a low sensitivity). Default value is 500

The *clock offset distribution* section displays information about the received PTP clock.



8.2.3.1.2.2 Livewire (Slave)

The following parameters appear when the mode “Livewire Slave” is selected:

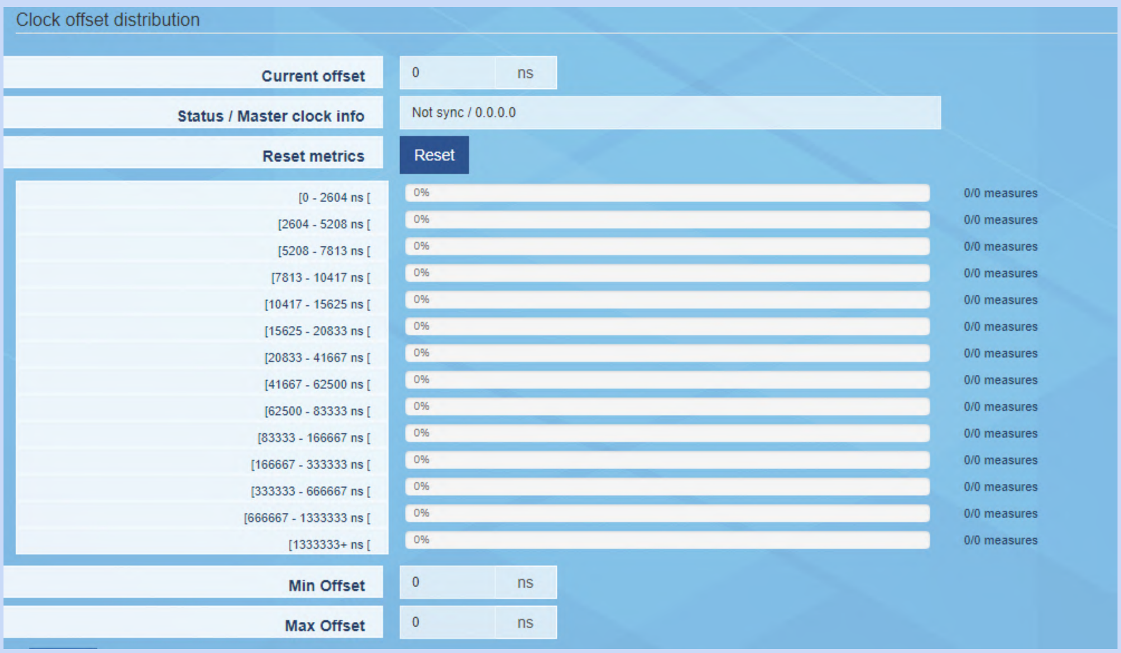


Description of the parameters:

Parameter	Read/ Write	Meaning

Network interface	R/W	Select the network interface that receives the livewire clock.
IGMPv3 filtering mode	R/W	Off: X/LINK subscribes to the Livewire clock which can be generated by any source IP address. Include: X/LINK subscribes to the Livewire clock which is generated only by the listed source IP addresses. Exclude: X/LINK subscribes to the Livewire clock which is generated by any source IP address, with exception of the listed IP addresses..
IGMPv3 IP source addresses		
IP address x	R/W	Displayed if IGMPv3 filtering mode is set to “Exclude” or “Include”. Allows declaring the source IP addresses to be included or excluded. Click on to add an IP@ to the list.

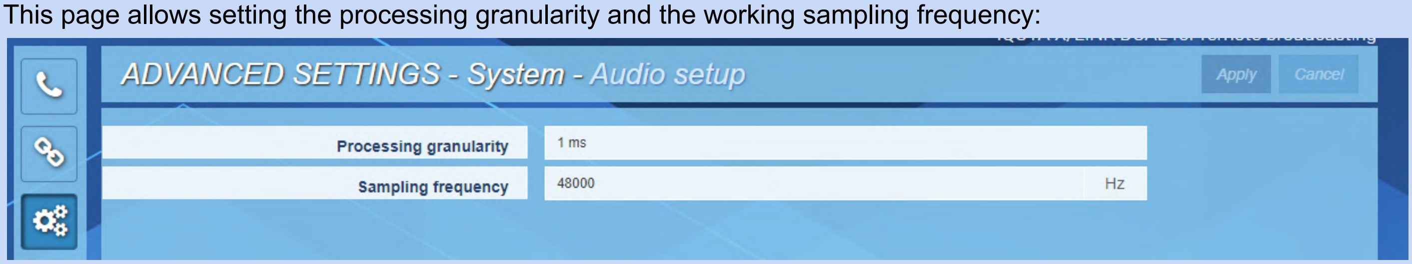
The *clock offset distribution* section displays information about the received Livewire clock.



Click on “Apply” to confirm your choice.

8.2.3.1.3 Advanced settings -> System -> Audio setup

This page allows setting the processing granularity and the working sampling frequency:

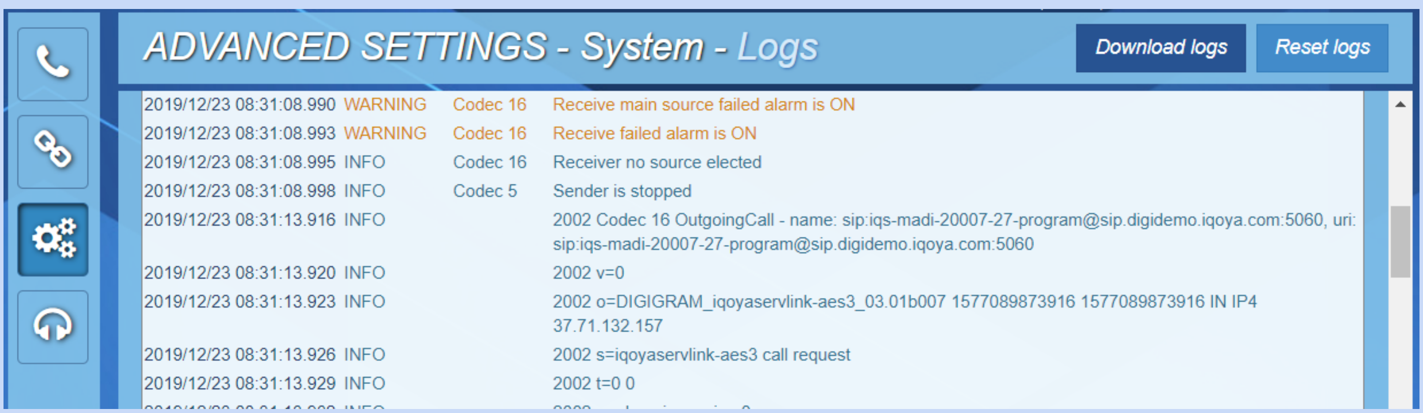


Click on a parameter field to be able to change the values.

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. Note: 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.

.
Click on “Apply” to confirm your changes.

8.2.3.1.4 Advanced settings –> System –> Logs

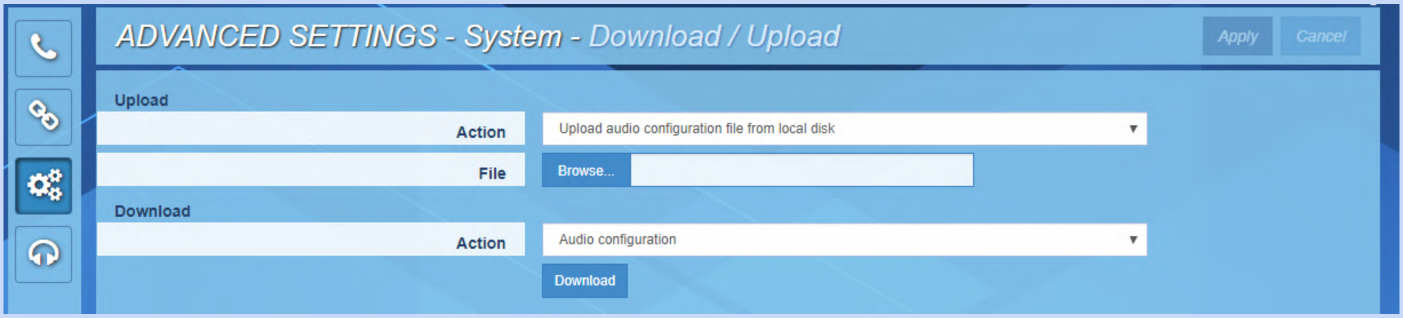


This page allows viewing and downloading the log file of IQOYA X/LINK. This log file gives information about the internal behaviour of IQOYA, and is useful for advanced diagnostics. Traces of enabled alarms are written into this log file (alarm ON, alarm OFF). This log file is stored internally and is 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: The page content is refreshed automatically if this parameter is set to “Yes”.
Date & Time: clicking on this icon allows sorting 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 remotely downloading the log traces.

8.2.3.1.5 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:

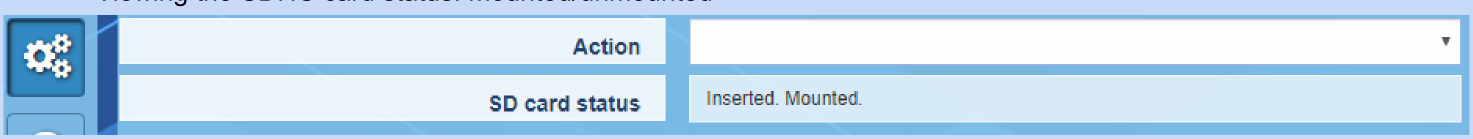
- The audio configuration only (includes the programs and IP services)
- The full codec configuration
- The connection book: The connection book is the concatenation of the contact list and the call profile list.

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

8.2.3.1.6 Advanced settings –> System –> SD card

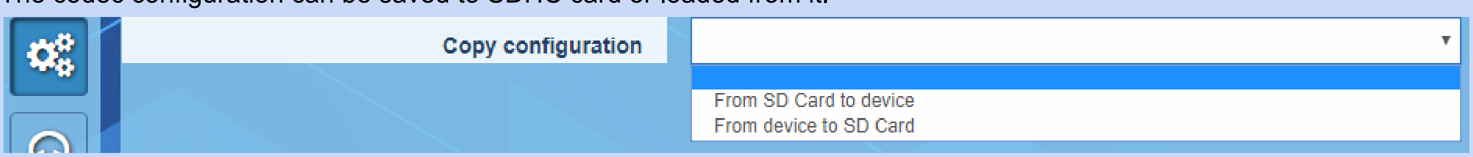
This page allows:

- mounting an SDHC card if it is inserted while the unit is running,
- unmounting it before removing it from the front panel.
- Viewing the SDHC card status: mounted/unmounted



8.2.3.1.7 Advanced settings –> System –> SD card backup

The codec configuration can be saved to SDHC card or loaded from it.



- From the “Copy configuration” field, select whether the configuration has to be copied from the SDHC card to IQOYA’s internal memory or from the internal memory to the SDHC card.

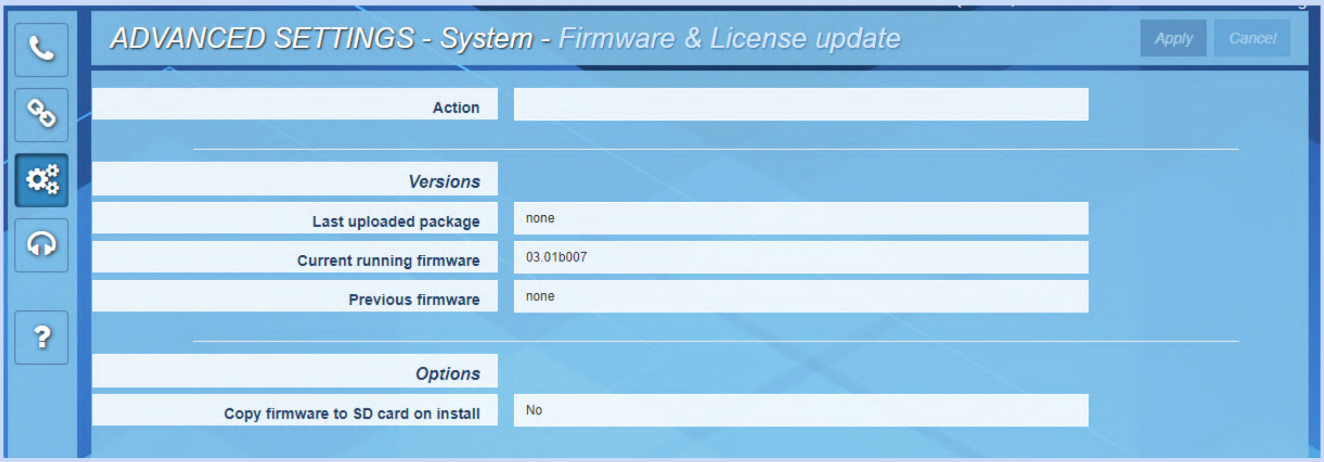
Notes:

- Audio activity is stopped when the configuration is loaded from the SDHC card.
- The unit is restarted to apply the new configuration.

- On the SDHC card, the configuration file “IQOYA_Configuration_save.tar” is stored in folder \IQOYA_LINK\Config.
- The current configuration of the IQOYA codec can also be displayed from a WEB browser by selecting the file \IQOYA_LINK\ Config.html, accessible via FTP.
- The configuration saved on the SDHC card can be loaded from the IQOYA X/LINK front panel LCD display and keyboard (menu System)
- This configuration on SDHC card can also be loaded when starting IQOYA with the SD card inserted. The file “/SDCARD/iqoya_link/run_once/ boot_commands.txt” must contain the following line:
  `RESTORE_FULLCONFIG_FROMSD=Yes`

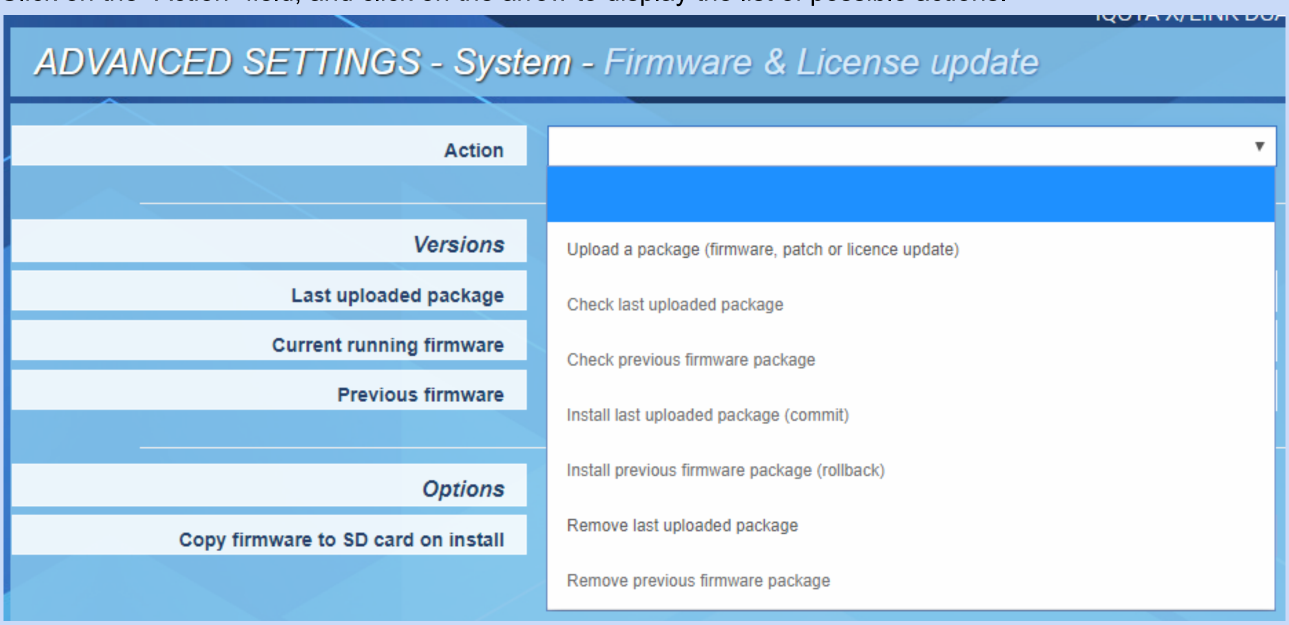
8.2.3.1.8 Advanced settings -> System -> Firmware & License update

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:

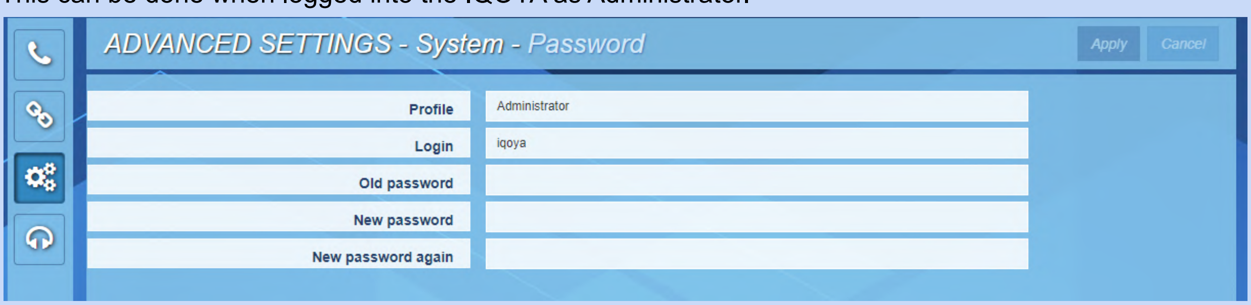
- **Check previous firmware package**: this allows checking that the previous firmware version that is stored locally is correct.
- **Check last uploaded package**: this allows checking that the last uploaded firmware version is correct. This operation is done automatically during the upload phase.
- **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.

Copy firmware to SD card on install
Set to Yes, this parameter allows copying to the SD card the firmware to be installed to facilitate a future possible firmware rollback. Exemple:

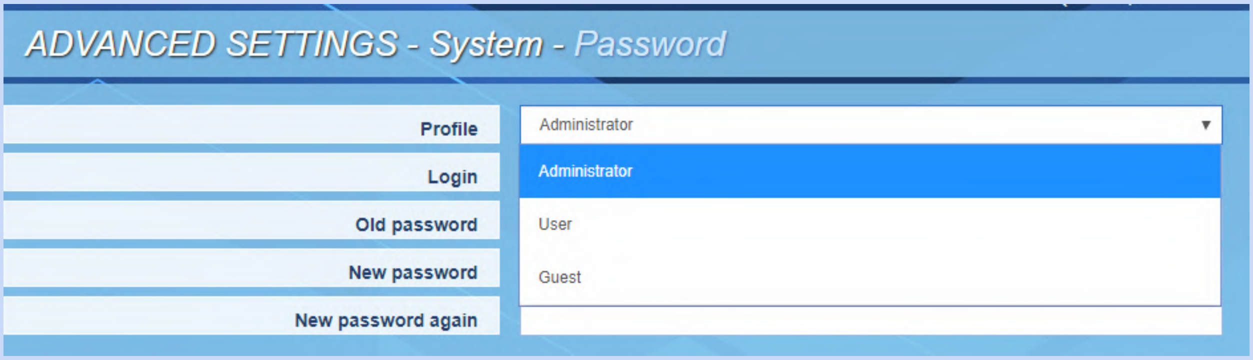
- Firmware to be upload and applied: version A
- Copy to SD card set to Yes
- Firmware to upload and applied: version B
- Copy to SD card set to Yes
  => Current firmware = version B / Previous firmware = version A
  At this point version A can be re-installed without the upload phase.

8.2.3.1.9 Advanced settings –> System –> Password

This page allows changing the username and 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.

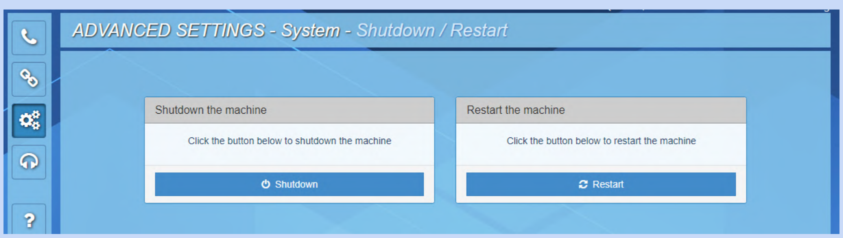


Login: allows configuring the username to be used in order to log to the WEB GUI with the selected profile.

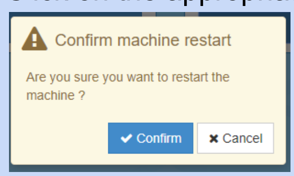
Old password: Type the current password
New password: Type the new password
New password again: confirm the new password
Click on "Apply" to confirm the changes.

8.2.3.1.10 Advanced settings -> System -> Shutdown / Restart

This page allow to restart or shutdown IQOYA.



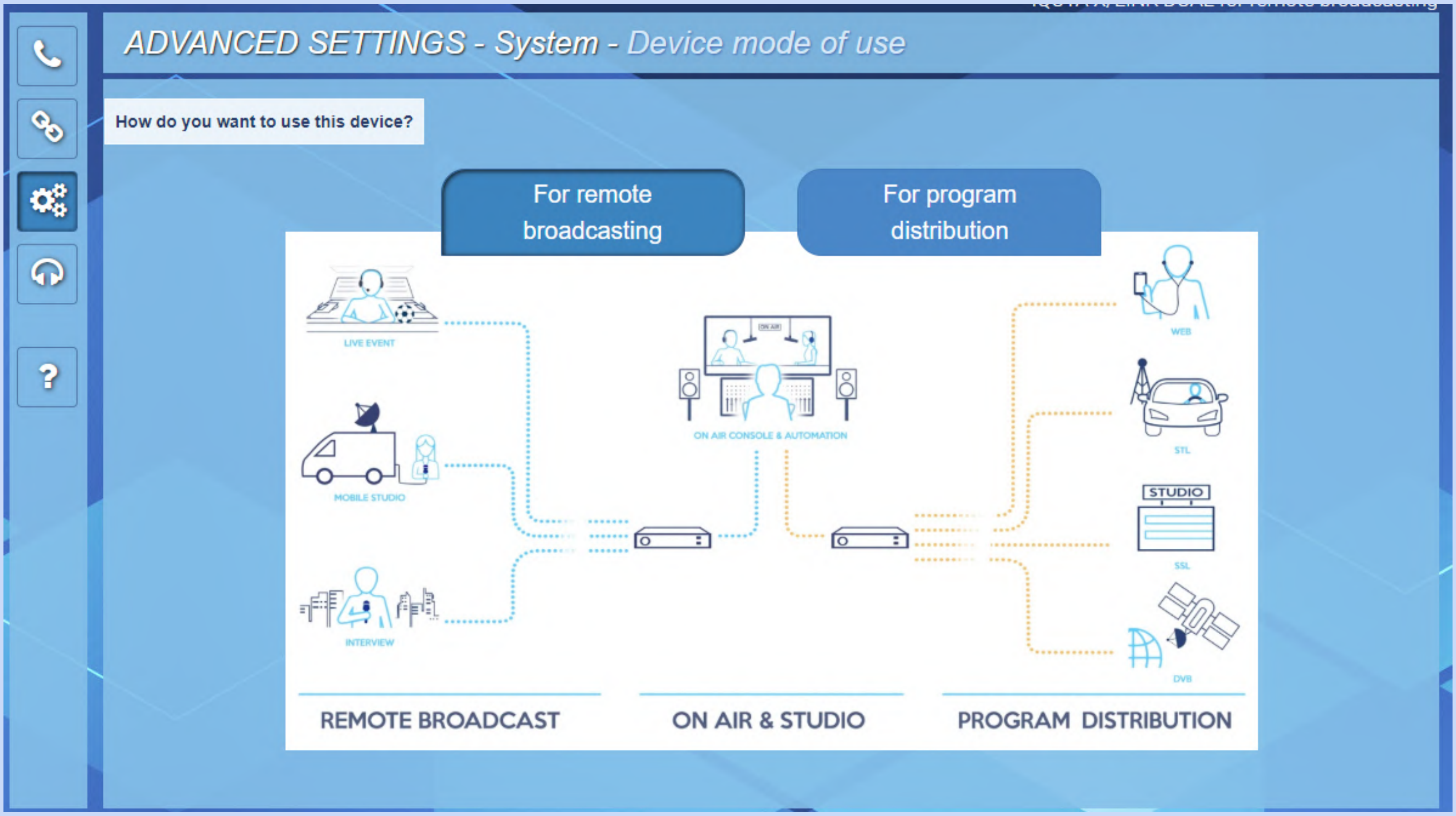
Click on the appropriate action.

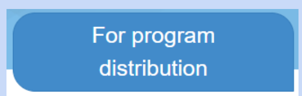


Confirm or cancel your choice through the displayed confirmation window.

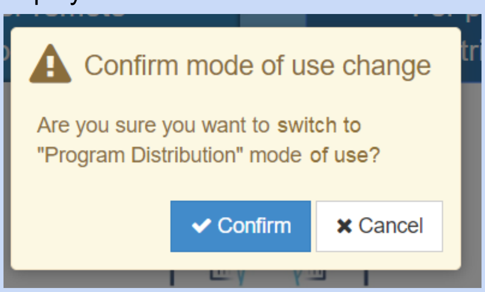
8.2.3.1.11 Advanced settings -> System -> Switch mode of use

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:



8.2.3.2 Advanced settings –> services

8.2.3.2.1 Advanced settings –> services–> NTP

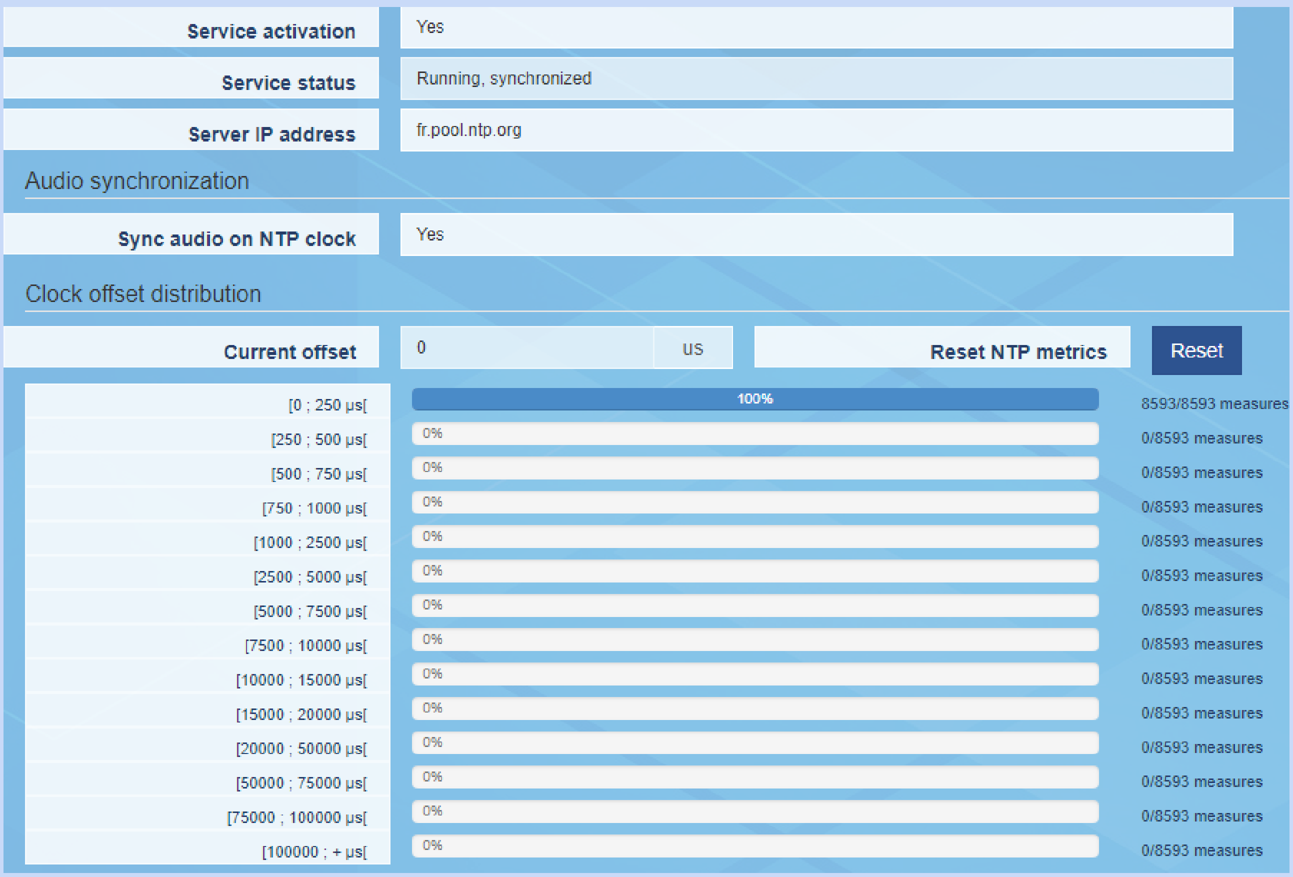
This page allows:

- 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 then 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”.

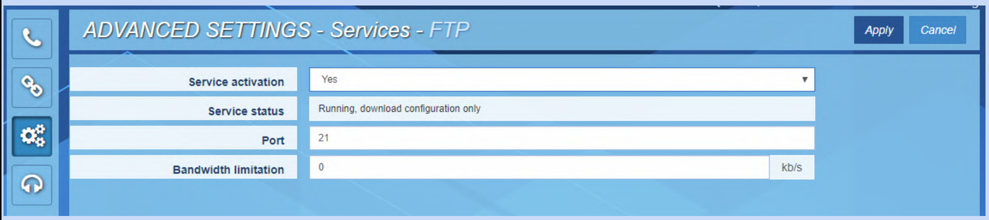
For activation of the NTP based audio synchronization, select “Yes” for parameter “**Sync audio on NTP clock**”.



Once IQOYA is synchronized on the NTP server, the field “Service status” displays “Running, synchronized”.
This requires that the software option is installed on the IQOYA X/LINK, as well as on the associated IQOYA decoders.

8.2.3.2.2 Advanced settings –> 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.

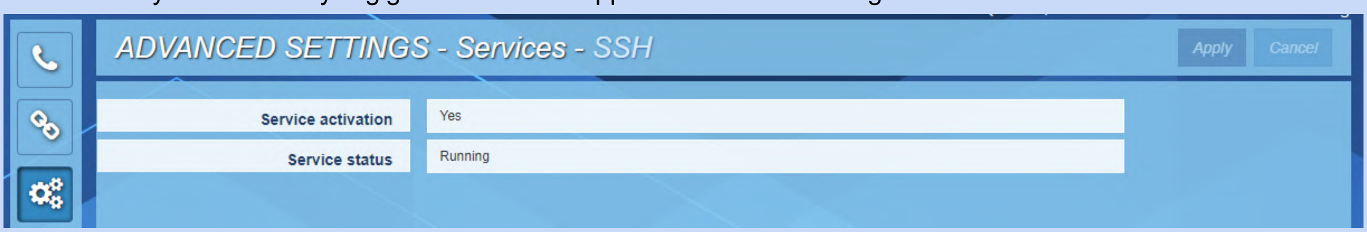


Click on the “Service activation” field. Select “Yes” to enable the FTP service, “No” to disable it.
If necessary, you may change the port used for FTP (default value is 21).
Parameter “Bandwidth limitation” allows limiting the network bandwidth of the FTP traffic.
Click on “Apply” to confirm the changes.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

Note that backup playlists and sound files have to be stored in folder DEVICE_STORAGE.

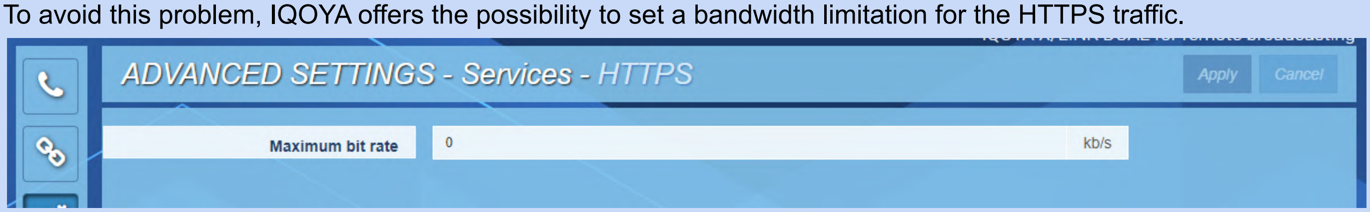
8.2.3.2.3 Advanced settings –> services–> SSH

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



8.2.3.2.4 Advanced settings –> 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.

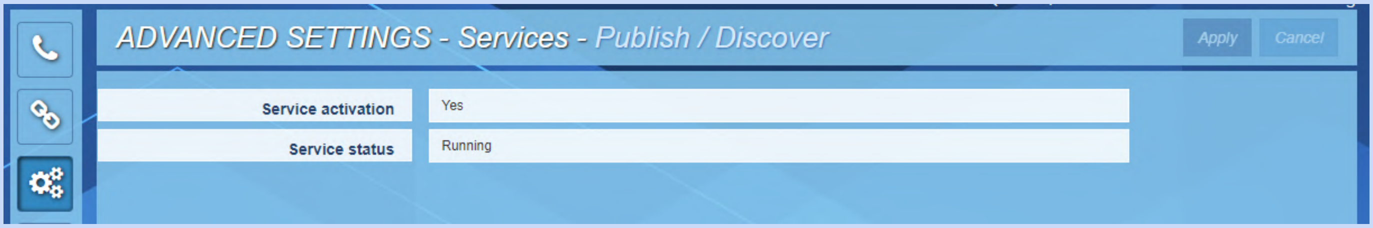


Click on the “Maximum bit rate” field, and enter the maximum bit rate allowed for HTTPS traffic.
Default value is 0, which means no limitation on HTTPS traffic.
The smaller the value, the longer it takes to load the WEB page!

Click on “Apply” to confirm the settings.

8.2.3.2.5 Advanced settings –> services–> Publish / Discover

This page allows enabling the automatic discovery and publishing of AES67 or RAVENNA streams.



In case you do not use AES67 or RAVENNA audio I/Os, there is no need to activate this service.

8.2.4 Audio I/Os category of menus

This category of menus and the pages they allow to reach are identical in "Remote Broadcasting" mode of use and in "Program Distribution" mode of use. Please refer to their descriptions in the “Program Distribution” section of this manual, paragraph 8.1.2 Audio I/Os category of menus.

9. Managing sound files and playlists via FTP

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

- username: ftp
- password: iqoya

  These are default username and passwords. Note that the username and password may be changed.

Playlists (.m3u) and sound files must be stored in folder “SDCARD”.

10. Specifications

10.1. IQOYA X/LINK-LE and X/LINK-ST

10.1.1. CONFIGURATION

Dimensions	19”, 1RU
Weight	~ 3.1 kg (~6.85 lbs)
Power supply	2 internal redundant PSU 100-250VAC, Optional: 100-240VAC / -48VDC
Temperature / Humidity non-condensing	Operating: 0°C – 50°C / 0% – 95% Storage: -5°C – 70°C / 0% – 95%
Power consumption	Max 21W

10.1.2. CONNECTIVITY

<table>
<tr><th></th><th>X/LINK-ST
X/LINK-LE</th><th>X/LINK-DUAL</th><th>X/LINK-AES67</th></tr>
<tr><td>WAN / LAN Ethernet ports</td><td colspan="3">1 x 100 Mbps (Eth2) + 3 x 10/100/1000 Mbps RJ-45 (Eth1, 3,4)</td></tr>
<tr><td>Analog and AES/3 audio inputs</td><td colspan="2">Female XLR on breakout cables</td><td></td></tr>
<tr><td>Analog and AES/3 audio outputs</td><td colspan="2">Male XLR on breakout cables</td><td></td></tr>
<tr><td>Serial data</td><td colspan="3">1 x RS232 port SubD-D 9</td></tr>
<tr><td>GPIO’s</td><td>8 Opto-Isolated GPIs (4 with factory option “Sync option for X/LINK”)
8 relay GPOs (4 with factory option “Sync option for X/LINK”):
: - 3 SPDT outputs: common, norm. open, .norm. closed
- max 220 VDC/250 VAC,
- max 60 W, 62.5 VA
- max. continuous/switching current: 2 A/3 A</td><td>4 Opto-Isolated GPIs
4 relay GPOs:
: - 3 SPDT outputs: common, norm. open, .norm. closed
- max 220 VDC/250 VAC,
- max 60 W, 62.5 VA
- max. continuous/switching current: 2 A/3 A</td><td>8 Opto-Isolated GPIs (4 with factory option “Sync option for X/LINK”)
8 relay GPOs (4 with factory option “Sync option for X/LINK”):
: - 3 SPDT outputs: common, norm. open, .norm. closed
- max 220 VDC/250 VAC,
- max 60 W, 62.5 VA
- max. continuous/switching current: 2 A/3 A</td></tr>
</table>

10.1.3. ANALOG INPUTS

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67
Type	2 balanced	4 balanced	-
A/D converter resolution	24 bits		-
Maximum level/ impedance	+24 dBu/ >10 kΩ		-
Adjustable gain	From –94.5dB to +24 dB; 0.5 dB steps Maximum sensitivity: 0 dBU input signal –> 0 dBfs		-
Adjustable digital gain	From –15 dB to +15 dB; 0.1 dB steps		-

10.1.4. ANALOG LINE OUTPUTS

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67
Type	2 Line balanced	4 Line balanced	-
D/A converter resolution	24 bits		-
Maximum input level/ impedance	+24 dBu/ <100 Ω		-
Adjustable analog gain	From –94.5dB to +24 dB; 0.5 dB steps		-
Adjustable digital gain	From –15 dB to +15 dB; 0.1 dB steps		-

10.1.5. AES3 INPUTS

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67
Type	1 balanced. Zin = 110 Ohms	2 balanced. Zin = 110 Ohm	-
Hardware sample rate converters	Sample rate conversion = 7.5:1 to 1:8, up to 192 kHz		-
Adjustable digital gain	from –15 dB to +15 dB		-

10.1.6. AES/3 OUTPUTS

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67
Type	1 balanced. Zout = 110 Ohms	2 balanced. Zout = 110 Ohms	-
Sample rate	32 kHz, 44.1 kHz, or 48 kHz		-

10.1.7. AES67/RAVENNA

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67
Inputs / outputs	2 mono channels (1 stereo)	4 mono channels (2 stereo)	2 mono to 16 mono (1 stereo to 8 stereo)
Sample rate	44.1 kHz, or 48 kHz -		
PTP slave	Yes		
PTP Master	Yes		
Clock source	PTPv2 (IEEE1588-2008) from network or internal clock or Word Clock or local clock eligible as GrandMaster PTP		
Samples per packet	48 / 192		
Audio payload formats	PCM16 / PCM24 / PCM32 / AM824 (PCM24+AES3 channel status)		

10.1.8. Livewire

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67
Inputs / outputs	2 mono channels (1 stereo)	4 mono channels (2 stereo)	2 to 16 mono channels (1 to 8 stereo)
Sample rate	48 kHz		
Mode	Standard (240 samples)	Standard (240 samples)	Standard (240 samples)

10.1.9. HEADPHONES OUTPUT

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67

Type	1 balanced (6.35mm jack)
Power	max 2x50 mW / 2x32 ohms load

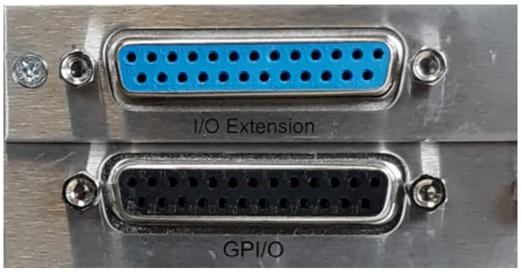
10.1.10. OTIONAL SYNCHRONIZATION INPUTS

	X/LINK X/LINK-LE	X/LINK-DUAL	X/LINK-AES67
10 MHz / 1 PPS	BNC connectors on breakout cable		

10.1.11. ANALOG AUDIO PERFORMANCES

Frequency response	20 Hz-20 kHz +/- 0,1 dB at 48 kHz
Signal to Noise	>108 dBA
Dynamic range (A-weighted)	Analog In: >104 dB / Analog Out: >106 dB
THD + noise 20-20kHz at –1 dBfs	<–90 dB
Channel phase difference: 20/20kHz	0.1° / 0.27°
Crosstalk (Analog in or out) 1 kHz at 22 dBu	1 kHz: < -120 dB 10 kHz: <-110 dB 20 kHz: <-107 dB
Internal clock precision	Better than 10 PPM

11. APPENDIX A: GPIO's CONNECTORS



IQOYA X/LINK codecs provide four of eight GPIs and GPOs on two female Sub-D 25 connectors.
The lower connector named "GPIO" provides 4 GPIO's.

The upper connector named "I/O Extension" provides 4 additional GPIO (except on X/LINK-DUAL), when it is not used for other types of I/Os

GPIO pinout

Lower Sub-D 25

Pin	13	12	11	10	9	8	7	6	5	4	3	2	1
GPIO	-	-	-	GPO_4	GPI_4	GPO_3	GPO_3	-	GPO_2	GPI_2	GPO_1	GPO_1	-
Label	unused	GND	GND	N.C.	K	N.O.	COM	unused	N.C.	K	N.O.	COM	unused

Pin	25	24	23	22	21	20	19	18	17	16	15	14
GPIO	-	-	GPO_4	GPO_4	-	GPO_3	GPI_3	GPO_2	GPO_2	-	GPO_1	GPI_1
Label	unused	GND	N.O.	COM	unused	N.C.	K	N.O.	COM	unused	N.C.	K

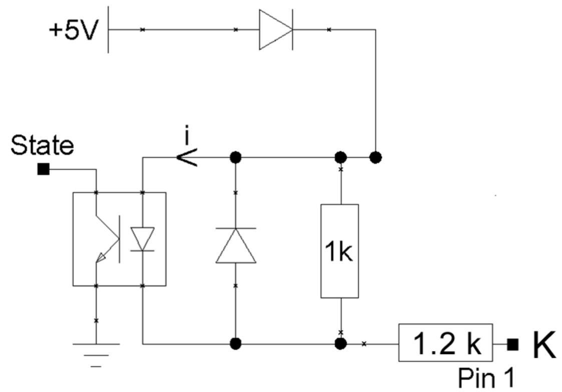
Upper Sub-D 25

Pin	13	12	11	10	9	8	7	6	5	4	3	2	1
GPIO	-	-	-	GPO_8	GPI_8	GPO_7	GPO_7	-	GPO_6	GPI_6	GPO_5	GPO_5	-
Label	unused	GND	GND	N.C.	K	N.O.	COM	unused	N.C.	K	N.O.	COM	unused

Pin	25	24	23	22	21	20	19	18	17	16	15	14
GPIO	-	-	GPO_8	GPO_8	-	GPO_7	GPI_7	GPO_6	GPO_6	-	GPO_5	GPI_5
Label	unused	GND	N.O.	COM	unused	N.C.	K	N.O.	COM	unused	N.C.	K

GND: connected to ground
N.C.: contact normally closed
N.O.: contact normally open
COM: common contact
unused: not used, DO NOT CONNECT!
K: optocoupler cathode

General Purpose Inputs (GPIs)



The IQOYA X/LINK 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. (ground is on pins 11-12-24 on each connector).

GPI optocoupler specifications

Minimum current imin to switch GPI	1 mA
Maximum current i_{max} supported	60 mA
Maximum voltage V_K supported	11 V_{DC}

General Purpose Outputs (GPOs)

The IQOYA X/LINK GPO's are opto-isolated SPDT type relays (Single Pole, Double Throw).

Each GPO features 3 pins:

- COM : Common
- N.C. : normally closed
- N.O. : normally open

According to the status applied to the GPO, pin N.C. is connected to pin COM, or pin N.O. is connected to pin COM.

GPIO tunneling in direct mode (status not inverted)

When GPI tunneling is enabled, an open GPI (pin K not connected to the ground) is reflected on the distant GPO by pin N.O connected to pin COM.

GPIO tunneling in inverted mode (status inverted)

An "open" GPI (pin K not connected to the ground) is reflected on the distant GPO by pin N.O connected to pin COM. Pin N.C is left unconnected.

A "closed" GPI (pin K connected to the ground) is reflected on the distant GPO by pin N.C. connected to pin COM. Pin N.O is left unconnected.

Alarms notification

Alarms can also be notified on GPOs. See chapter “Alarms management”.

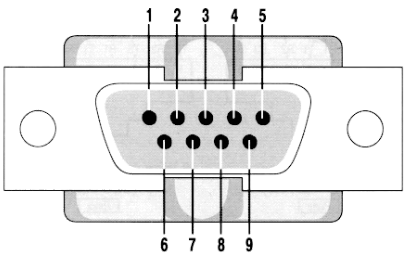
GPO relay specifications

Maximum power switching capability	60 W/62.5 VA
Maximum switching current	5 A_{DC}
Maximum carrying current	2 A_{DC}
Maximum switching voltage*	220 V_{DC} /0.24A-60 W 250 V_{AC} /0.25-62.5 V_{AC} 125 V_{AC} /0.5A-62.5 V_{AC} 30 V_{DC} /2 A-60 W
Typical life expectancy (switching max power)	10^6 operations

****Note: The maximum voltage makes it possible to control devices (up to 60 W. max) directly through the power outlet.***

12. APPENDIX B: SERIAL PORT (RS232 ON DB9)

Pin	Description
1	Not connected
2	RxD (received data)
3	TxD (transmitted data)
4	Not connected
5	Signal ground
6	Not connected
7	RTS (request to send)
8	CTS (clear to send)
9	Not connected



IQOYA X/LINK codecs provide an RS232 serial port on a male DB-9 connector on the back panel. Use this port to connect any compatible device. For pinout allocation details, please refer to the figure and table above.

The port may be used for tunneling serial data between encoder and decoder (RDS data, commands). Set-up is done through a web browser .

13. APPENDIX C: TYPICAL LATENCY VALUES

The back to back latency between two IQOYA X/LINK devices depends on the selected audio format, the network quality, and the enabled functionalities (backup, half/full duplex, FEC).

See the table underneath for maximum latency values in half-duplex, using neither failover configurations nor FEC, with the jitter buffer size set to 0, and with an optimized network.

Audio Type	Audio format	Latency (processing granularity set to 1ms)	Description
PCM	24 bit	9ms	
MPEG Layer II	256 kbps	90ms	
MPEG Layer III	128 kbps	152ms	
AAC-4 LC	256 kbps	105ms	
AAC-4 LC+SBR	96 kbps	210ms	aka HEv1
AAC-4 LC+SBR+PS	56 kbps	251ms	aka HEv2
AAC-4 LD	160 kbps	51ms	
AAC-4 ELD	160 kbps	45ms	
Opus	256 kbps	73ms	

All measurements taken on stereo samples at 48 kHz

Impact of the processing granularity

Add about 4 ms to the latency each time the processing granularity is increased of 1ms.

Impact of the network on latency

Latency highly depends on the quality of the network. Network jitter and packets loss typically have a direct impact on latency.

- Network jitter compensation is achieved by buffering audio data on the decoder. A good quality network generally offers a low jitter, then requiring low buffering on the decoder, which means a low increase of latency. But a network with a high jitter requires increasing the decoder buffering accordingly, leading to a significant increase of latency.
- In case of packets loss on the network, it is necessary to enable an FEC, which allows recovering lost packets thanks to redundant frames. FEC increases the latency.

Impact of features on the latency

The amount of features used in IQOYA directly impacts the latency. For a given audio format, the lowest latency is obtained in half duplex mode, with no backup defined and no FEC. As soon as one of these features is used, the latency increases a bit.

APPENDIX D: AAC SETTINGS FOR STEREO SAMPLES

AAC type	Sampling frequency (Hz)	Audio bit rate (bit/s)	IP stream bit rate (bit/s)
AAC-LC	16000	32000 – 39999	8250+ Audio bit rate
AAC-LC	22050	32000 – 39999	11369+ Audio bit rate
AAC-LC	24000	32000 – 39999	12375+ Audio bit rate
AAC-LC	32000	40000 – 320000	16500+ Audio bit rate
AAC-LC	44100	40000 – 320000	22739+ Audio bit rate
AAC-LC	48000	40000 – 320000	24750+ Audio bit rate

HE-AACv1 (SBR)	16000		
HE-AACv1 (SBR)	22050		
HE-AACv1 (SBR)	24000		
HE-AACv1 (SBR)	32000	24000 – 96000	8250+ Audio bit rate
HE-AACv1 (SBR)	44100	24000 – 96000	11369+ Audio bit rate
HE-AACv1 (SBR)	48000	24000 – 96000	12375+ Audio bit rate

HE-AACv2 (SBR+PS)	16000		
HE-AACv2 (SBR+PS)	22050		
HE-AACv2 (SBR+PS)	24000		
HE-AACv2 (SBR+PS)	32000	14000 – 56000	8250+ Audio bit rate
HE-AACv2 (SBR+PS)	44100	18000 – 56000	11369+ Audio bit rate
HE-AACv2 (SBR+PS)	48000	18000 – 56000	12375+ Audio bit rate

AAC type	Sampling frequency (Hz)	Audio bit rate (bit/s)	IP stream bit rate (bit/s)
AAC-LD	16000		
AAC-LD	22050		
AAC-LD	24000	80000 – 111999	24750 + Audio bit rate
AAC-LD	32000	112000 – 320000	33000 + Audio bit rate
AAC-LD	44100		45478 + Audio bit rate
AAC-LD	48000		49500 + Audio bit rate

AAC-ELD	16000		
AAC-ELD	22050		
AAC-ELD	24000	64000 – 97999	24750 + Audio bit rate
AAC-ELD	32000	64000 – 135999	33000 + Audio bit rate
AAC-ELD	44100	76000 – 256000	45478 + Audio bit rate
AAC-ELD	48000	98000 – 256000	49500 + Audio bit rate

AAC-ELD + SBR	16000		
AAC-ELD + SBR	22050		
AAC-ELD + SBR	24000		
AAC-ELD + SBR	32000		
AAC-ELD + SBR	44100	48000 – 96000	45478 + Audio bit rate
AAC-ELD + SBR	48000	48000 – 96000	49500 + Audio bit rate

14. APPENDIX E: AVAILABLE FEC's FOR RAW RTP STREAMING (NO TS ENCAPSULATION)

FEC (Forward Error Correction) is a mechanism which consists in sending redundant information (redundant frames) to the decoder so that it can compensate packet transmission errors on unreliable networks.

An FEC can be selected when defining the parameters of the stream to be generated (Send page) and/or to be received (Receive page).

FEC requiring no additional stream

Redundant frames are sent in the same stream as the IP audio stream.

The FEC to be selected is "**+50% bandwidth, recovery 2, 1 stream (audio)**".

Its characteristics are: +50% bandwidth, additional delay of 2 frames, recovers 1 lost packet at 100%, recovers 2 consecutive lost packets at 75%.

FEC requiring an additional stream

Standard FECs

Redundant frames are sent as a second stream of data. The used UDP port is: port of the IP audio stream + 2.

Selectable FECs are:

- **+100% bandwidth, recovery 3, 2 streams (audio + FEC)**
  +100% bandwidth, additional delay of 1 frame, recovers 2 consecutive lost packet at 100%, recovers 3 consecutive lost packet at 75%
- **+100% bandwidth, recovery 4, 2 streams (audio + FEC)**
  +100% bandwidth, additional delay of 3 frames, recovers 3 lost packet at 100%, recovers 4 consecutive lost packet at 80%
- **+50% bandwidth, recovery 1/2, 2 streams (audio + FEC)**
  +50% bandwidth, additional delay of 1 frame, recovers 1 lost packet over 2 consecutive packets.
- **+33% bandwidth, recovery 1/3, 2 streams (audio + FEC)**
  +33% bandwidth, additional delay of 2 frames, recovers 1 lost packet over 3 consecutive packets.
- **+25% bandwidth, recovery 1/4, 2 streams (audio + FEC)**
  +25% bandwidth, additional delay of 3 frames, recovers 1 lost packet over 4 consecutive packets.
- **+20% bandwidth, recovery 1/5, 2 streams (audio + FEC)**
  +20% bandwidth, additional delay of 4 frames, recovers 1 lost packet over 5 consecutive packets.
- **+10% bandwidth, recovery 1/10, 2 streams (audio + FEC) –** From firmware 2.31
  +10% bandwidth, additional delay of 9 frames, recovers 1 lost packet over 10 consecutive packets.

Redundant dual streaming

Redundant dual streaming is activated by selecting an appropriate "Dual stream" FEC. A dual stream FEC consists in considering the redundant stream as an FEC.
In addition, the duplicated stream can be delayed to offer time diversity, thus avoiding that a network disturbance affects the same frames on the primary stream and on the FEC stream. Selectable delay is from 0 to 3000 ms, by steps of 100 ms.

Notes:

- When in-band audio format signaling is enabled, FEC stream is sent to the same IP address as the primary stream, and on UDP port + 2.

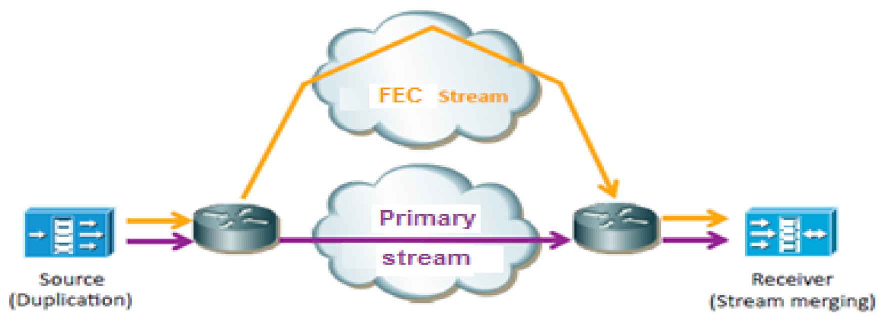
- When in-band audio format signaling is disabled, it is possible to define the destination address and port of the FEC stream.

15. APPENDIX F: REDUNDANT DUAL STREAMING

Spatial diversity

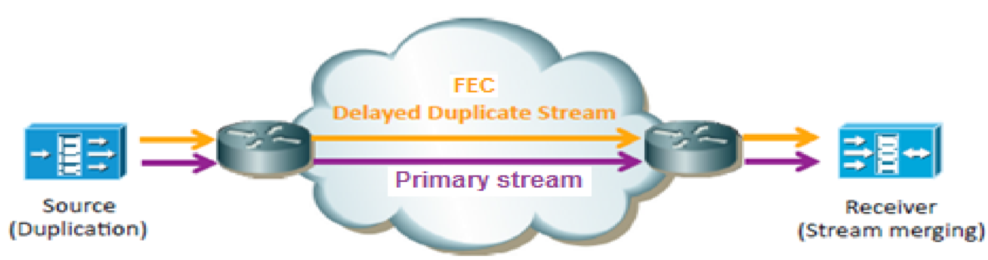
IQOYA X/LINK can be configured to send the same AoIP stream on two distinct networks, typically through Eth0 and Eth1 interfaces. On the decoding side, IQOYA automatically synchronizes both received streams. Using separate network paths ensures that potential network failures are statistically uncorrelated, enabling the reconstruction of a unique unperturbed stream.

Terminology used for the two redundant streams is: primary stream, and FEC “dual” stream for the duplicate stream.



Time diversity

IQOYA doesn’t only propose passive duplication as on most codecs. It also allows delaying the duplicate stream compared to the primary stream. Although the primary stream and the FEC stream are configured to use different networks, it is quite common that some network components are common to both networks (last mile router for instance). The selected delay avoids that temporary failures occurring on common network components impact both a primary frame and its duplicate frame.



Multicast and unicast can be used for redundant dual streaming, and different UDP ports can also be used for the primary stream and the FEC stream.

A typical redundant dual streaming configuration is as follows:

- Enter the destination IP address and UDP port of the primary stream. The IP address can be the public IP address of the Eth interface of the IQOYA that decodes the stream, or a multicast address. Select the IP interface used to send the stream in case of multicast.
- Select a “Dual stream” FEC, with or without time delay. Enter if necessary the destination IP address and UDP port of the FEC stream. The IP address can be the public IP address of another Eth interface of the IQOYA that decodes the stream, or a multicast address. Select the IP interface used to send the FEC stream.

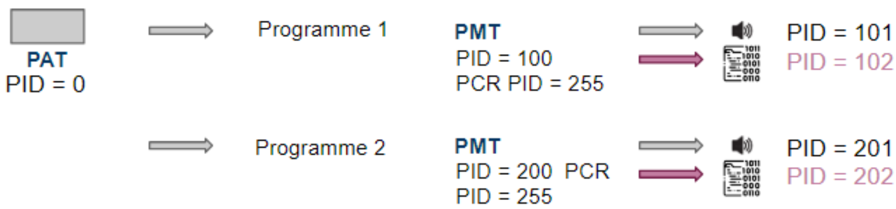
Notes:

- When in-band audio format signaling is enabled, FEC stream is sent to the same IP address as the primary stream, and on UDP port + 2.
- When in-band audio format signaling is disabled, it is possible to define the destination address and port of the FEC stream.

16. APPENDIX G: MPEG-TS AUXILIARY DATA

Structure of an MPEG-TS MPTS stream

including 2 audio programs, each program having its auxiliary data.

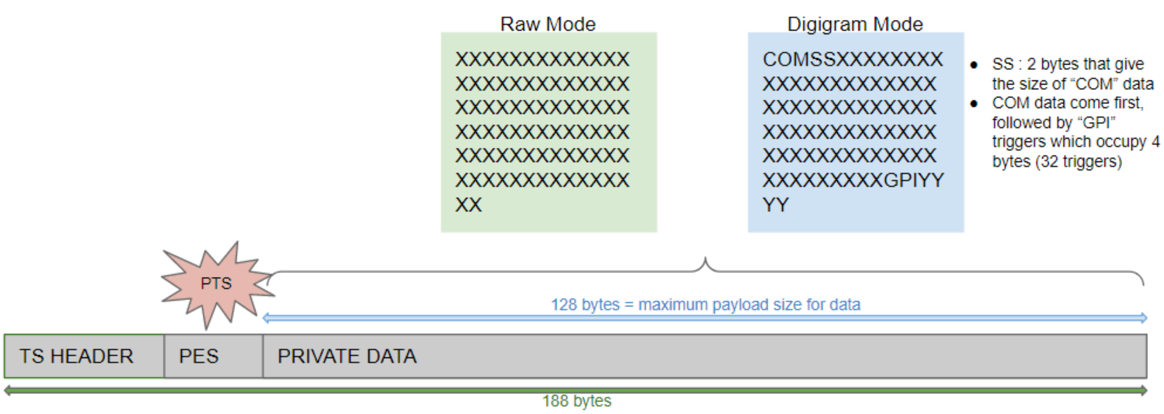


PAT: Program Association Table.
Gives the list of programs

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, commans).
In Digigram mode, the packet can contain 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 H: Modifying Maximum_Bitrate_Descriptor and ES_Rate_flag via the configuration file

Since version 3.12, 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.