Bridge-IT II and Bridge-IT XTRA II Codec User Manual

Software Version: 3.10.xx
Manual Version: v1_2_20250305
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Table of Contents

1 How to Use the Documentation

Overview of this User Manual

Use this manual to learn how to:

- Connect the codec to an IP network and configure peer-to-peer, multicast or multi-unicast connections.
- Configure the codec over a LAN or USB cable.
- Adjust audio and other settings within the codec.
- Configure automatic SDHC card backup.

Please read Getting Connected Quickly for an overview of how to configure the codec using 'programs' to store connection settings.

Manual Conventions

Warnings: Instructions that, if ignored, could result in death or serious personal injury caused by incorrect operation of the equipment. These must be observed for safe operation.

Cautions: Instructions warning against potential hazards, or to detail practices that must be observed for safe operation and to prevent damage to equipment or personnel.

Important Note: Information you should know to connect and operate your codec successfully.

Typographic Conventions

- Codec software elements are in Arial bold, e.g. **Contacts**
- Codec hardware elements are in bold Capitals, e.g. **KEYPAD**

Help Button

Press the Information button on the codec when navigating codec menus to display a help dialog on the **OLED SCREEN** suggesting actions or providing information for a menu item.

2 Warnings and Safety Information

THUNDERSTORM AND LIGHTNING WARNING:
DO NOT USE Tieline codecs during thunderstorms and lightning. You may suffer an injury using a Tieline codec, or any device connected to a LAN connection during a thunderstorm. This can lead to personal injury and in extreme cases may be fatal. Protective devices can be fitted to lines, however, due to the extremely high voltages and energy levels involved in lightning strikes, these devices may not offer protection to users, the Tieline codec and equipment connected to the codec.

Secondary strikes can occur. These secondary strikes are induced by lightning strikes and also produce dangerously high currents and energy levels. You only need to be near an object struck by lightning to lead to personal injury or damage to equipment. e.g. if located near a lighting tower at a sports facility, water features and drains on golf courses you may be affected by these secondary strikes.

Damage to personnel and Tieline codecs may occur during thunderstorm, even if the codec is turned off but is connected to the system or the power.

ANY DAMAGE TO A TIELINE PRODUCT CAUSED BY LIGHTNING or an ELECTRICAL STORM WILL VOID THE WARRANTY. Use of this product is subject to Tieline's SOFTWARE LICENSE and WARRANTY conditions, which should be viewed at www.tieline.com/support before using this product.

IMPORTANT SAFETY WARNINGS: Bridge-IT XTRA II is powered by dual 240 Volt AC power supplies and Bridge-IT II is powered using a 12V power supply.

SAFETY PRECAUTIONS:
- A readily accessible disconnect device shall be incorporated in the building installation wiring.
- Due to the risks of electrical shock, and energy, mechanical, and fire hazards, any procedures that involve opening panels or changing components must be performed by qualified service personnel only.
- To reduce the risk of fire and electrical shock, disconnect the device from the power line before removing the cover or panels.
- This unit has more than one power supply. Disconnect all power supplies before opening to avoid electric shock. Disconnecting one power supply disconnects only one power supply module. To isolate the unit completely, disconnect all power supplies.

SERVICING WARNINGS:
- Do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so.
- All work should be carried out by suitably qualified personnel.
- This unit has more than one power supply. Disconnect all power cables before maintenance to avoid electric shock.

HIGH VOLTAGE WARNINGS:
- Any adjustment, maintenance, and repair of the opened instrument under voltage must be avoided as much as possible and, when inevitable, must be carried out only by a skilled person who is aware of the hazard involved.
- Capacitors inside the instrument may still be charged even if the instrument has been disconnected from its source of supply.

GROUNDING:
Before connecting Bridge-IT XTRA II to the power line, where required, the protective earth terminal screws of this device must be connected to the protective earth in the building installation.

LINE VOLTAGE:

Before connecting this device to the power line, make sure the voltage of the power source matches the requirements of the device. Refer to the device Specifications for information about the correct power rating for the device.

FUSES:

Make sure that only fuses with the required rated current and of the specified type are used for replacement. The use of repaired fuses and the short-circuiting of fuse holders must be avoided. Whenever it is likely that the protection offered by fuses has been impaired, the instrument must be made inoperative and be secured against any unintended operation.

WARNING:

HIGH LEAKAGE CURRENT. EARTH CONNECTION ESSENTIAL BEFORE CONNECTING SUPPLY.

If the total leakage current exceeds 3.5 mA, or if the leakage current of the connected loads is unknown, connect the supplementary ground terminal to a reliable ground connection in your facility.

SUPPLEMENTARY GROUND CONNECTION:

A supplementary ground terminal is provided on the codec to connect the unit to a ground connection. The ground terminal has an M4 stud with M4 retaining nuts and is compatible with all grounding wires. Remove only the outer **NUT** to connect your ground wire. The ground wire must have a suitable lug. When refitting the outer **NUT** ensure that both **NUTS** are correctly tightened to establish and maintain a proper earth connection.

WARNING: To Reduce the Risk of Electrical Shock and Fire

1. This equipment is designed to permit connection between the earthed conductor of the DC supply circuit and the earthing conductor equipment.
2. Equipment connected to the protective earthing of the building installation through the mains connection or through other equipment with a connection to protective earthing - and to a cable distribution system using coaxial cable, may in some circumstances create a fire hazard. Connection to a cable distribution system must therefore be provided through a device providing electrical isolation below a certain frequency range (galvanic isolator, see EN 60728-11).
3. All servicing must be undertaken only by qualified service personnel. There are not user serviceable parts inside the unit.
4. DO NOT plug in, turn on or attempt to operate an obviously damaged unit.
5. Ensure that the chassis ventilation openings in the unit are NOT BLOCKED.
6. Do not operate the device in a location where the maximum ambient temperature exceeds 0°C to 45°C (32°F to 113°F).
7. Be sure to unplug both power supply cords from the wall socket BEFORE attempting to service the power supply fuse.
8. This product is for indoor use only. Do not expose the device to rain or moisture.

AC units for Denmark, Finland, Norway, Sweden (marked on product):

• Denmark - “Unit is class I - unit to be used with an AC cord set suitable with Denmark deviations. The cord includes an earthing conductor. The Unit is to be plugged into a wall socket outlet which is connected to a protective earth. Socket outlets which are not connected to earth are not to be used!”
• Finland - “Laite on liitettävä suojamaadoituskoskettimilla varustettuun pistorasiaan”
• Norway - “Apparatet må tilkoples jordet stikkontakt”
• Unit is intended for connection to IT power systems for Norway only.
• Sweden - “Apparaten skall anslutas till jordat uttag.”

To connect the power connection:

1. Connect the power cables to the power sockets, located on the rear panel of the device.
2. Connect the power cables to the grounded AC outlets.

WARNING: Risk of electric shock and energy hazard. Disconnecting one power supply disconnects only one power supply module. To isolate the unit completely, disconnect all power supplies.

CHINESE SAFETY WARNINGS:

本设备有两个电源供电，未避免电击危险，操作时需要加倍小心。
只有当这两个电源完全断开时才可以安全操作

Translation of previous Chinese safety warning: This unit has two power supplies, please be very careful when using, disconnect two power supplies before maintenance to avoid electric shock.

此设备仅限于非热带地区使用

This device must only be used in not-tropical climate regions.

此设备只限在海拔高度低于 2000 米处使用。

This device must only be used at altitude not exceeding 2000 metres.

JAPANESE SAFETY WARNINGS:

必ず接地接続を行って下さい

Translation of previous warning: Provide an earthing connection.

接地接続は必ず、電源プラグを電源につなぐ前に行って下さい。又、接地接続を外す場合は、必ず電源プラグを電源から切り離してから行って下さい。

Translation of previous warning: Provide an earthing connection before the mains plug is connected to the mains. And, when disconnecting the earthing connection, be sure to disconnect after pulling out the mains plug from the mains.

Statement for Class A VCCI-certified Equipment:

この装置は，クラスA情報技術装置です。この装置を家庭環境で使用すると電波妨害を引き起こすことがあります。この場合には使用者が適切な対策を講ずるよう要求されることがあります。 VCCI－A

Translation of previous Class A VCCI Statement: This is a Class A product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this equipment is used in a domestic environment, radio disturbance may occur, in which case, the user may be required to take corrective action.

Statement for Class B VCCI-certified Equipment:

この装置は，クラスB情報技術装置です。この装置は，家庭環境で使用することを目的としていますが，この装置がラジオやテレビジョン受信機に近接して使用されると，受信障害を引き起こすことがあります。
取扱説明書に従って正しい取り扱いをして下さい。 VCCI－B

Translation of previous Class B VCCI Statement: This is a Class B product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this is used near a radio or television receiver in a domestic environment, it may cause radio interference. Install and use the equipment according to the instruction manual.

Special Notices for North American Users:
For North American power connection, select a power supply cord that is UL Listed and CSA Certified 3 - conductor, [18 AWG], terminated in a molded on plug cap rated 125 V, [5 A], with a minimum length of 1.5m [six feet] but no longer than 4.5m.

Special Notices for European Users:
For European connection, select a power supply cord that is internationally harmonized and marked “<HAR>”, 3 - conductor, 0,75 mm2 minimum mm2 wire, rated 300 V, with a PVC insulated jacket. The cord must have a molded on plug cap rated 250 V, 3 A.

Installation Codes:

This device must be installed according to country national electrical codes. For North America, equipment must be installed in accordance with the US National Electrical Code, Articles 110 - 16, 110 -17, and 110 -18 and the Canadian Electrical Code, Section 12.

Interconnection Cabling:

Cables for connecting to the unit's RS232 and Ethernet Interfaces must be UL certified type DP-1 or DP-2. (Note: when residing in non-LPS circuit)

Overcurrent Protection:

A readily accessible listed branch-circuit over current protective device rated 15 A must be incorporated in the building wiring for each power input.

Replaceable Batteries:

If equipment is provided with a replaceable battery, and is replaced by an incorrect battery type, then an explosion may occur.

CAUTION: RISK OF EXPLOSION IF BATTERY IS REPLACED BY AN INCORRECT BATTERY TYPE. DISPOSE OF USED BATTERIES ACCORDING TO THE INSTRUCTIONS.

This equipment is provided with a long life replaceable Panasonic CR2032 model 3V manganese dioxide lithium coin battery. Service personnel should only replace this battery with the same brand and type of battery. If this is replaced by an incorrect battery type, then an explosion may occur. Visit http://www.panasonic.com/industrial/includes/pdf/Panasonic_LithiumCR_Info.pdf to view the Material Safety Data Sheet for this battery.

End of Life Statement

Tieline hereby declares that all materials, components and products supplied are in full compliance with RoHS & WEE directives. This product must be disposed of according to local laws and regulations. Because the product contains a battery it must be disposed of separately from household waste. Do not incinerate, but take it to a recycling facility.

Lead-Free

Warranty and Disclaimer

This equipment manufactured by Tieline is warranted by Tieline against defects in material and workmanship from the date of original purchase (Bridge-IT II one year warranty; Bridge-IT XTRA II two year warranty). During the warranty period, we will repair or, at our option, replace at no charge a product that proves to be defective, provided you obtain return authorization from Tieline and return the product, shipping prepaid, to Tieline. For return authorization, contact Tieline's US or Australian office (see www.tieline.com).

This Warranty does not apply if the product has been damaged by accident or misuse or as the result of service or modification performed by anyone other than Tieline. With the exception of the warranties set forth above, Tieline makes no other warranties, expressed or implied or statutory, including but not limited to warranties of merchantability and fitness for a particular purpose, which are hereby expressly disclaimed. Use of this product is subject to Tieline's SOFTWARE LICENSE and WARRANTY conditions, which should be viewed at www.tieline.com before using this product.

In no event will Tieline, its directors, officers, employees, agents, owners, consultants or advisers (its “Affiliates”), or authorized dealers or their respective Affiliates, be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

Whilst every effort has been made to ensure the accuracy of this manual we are not responsible for any errors or omissions within it. The product specifications and descriptions within this manual will be subject to improvements and modifications over time without notice, as changes to software and hardware are implemented. Tieline takes no responsibility for any damage to equipment attached to the codec.

3 Glossary of Terms

AES3	Official term for the audio standard referred to often as AES/EBU
AES67	A technical standard for audio over IP and audio over Ethernet (AoE) interoperability developed by the AES. It is a layer 3 protocol suite facilitating interoperability between IP-based audio networking systems such as WheatNet-IP, RAVENNA, Livewire+, and Dante.
AES/EBU	Digital audio standard used to carry digital audio signals between devices
CCC	Cloud Codec Controller
CSRF	Cross-Site Request Forgery (CSRF) is an attack that forces a user to execute unwanted actions on a web application in which they are currently authenticated.
Destinations	An AoIP Destination in the codec is an AES67 stream received by the codec from an AES67 LAN.
DNS	The Domain Name System (DNS) is used to assign domain names to IP addresses over the World-Wide Web
Domain	A group of computers or devices on a network which are administered with common rules and procedures. Devices sharing a common part of the IP address are said to be in the same domain
DSCP	The Differentiated Services Code Point or Diffserv Value is a field in an IP packet header for prioritizing data when traversing IP networks. This is often used in AES67 streaming.
Ember+	An open standard control protocol developed by Lawo which allows a third party application to gain access to device parameters.
Failover	Method of switching to an alternative backup Audio Stream if the primary connection is lost.
GUI	Graphical User Interface
IFB	Interrupted Foldback/Interruptible Foldback: an intercom circuit consisting of a mix-minus program feed sent to talent, which can be interrupted and replaced by a producer's or director's intercom microphone
IGMP	A communications protocol used by hosts and adjacent routers on IPv4 networks to establish multicast group memberships to IPv4 routers.
IGMP snooping	The process of listening to IGMP network traffic for delivery of IP multicasts. Network switches with IGMP snooping maintain a map of which links need which IP multicast transmission. Multicasts may be filtered to conserve bandwidth on links.
ISP	Internet Service Providers (ISPs) are companies that offer customers access to the internet
IP	Internet Protocol; used for sending data across packet-switched networks
IPv4	Internet Protocol version 4 is the most widely deployed version of the protocol and is widely used on the Internet and on local area networks
IPv6	Internet Protocol version 6 is the most recent revision of the Internet Protocol and is intended to replace IPv4 eventually.
LAN	Local Area Network; a group of computers and associated devices sharing a common communications link
Latency	Delay associated with IP networks and caused by algorithmic, transport and buffering delays
LIO	Logic Input/Output
Livewire+ AES67	Livewire+ is a proprietary protocol from the Telos Alliance which is the second-generation of Livewire. It facilitates AES67 devices connecting directly to Livewire+ AES67 networks to exchange audio streams.
MIB	A management information base (MIB) is a database used for managing the entities in a communications network. This term is associated with the Simple

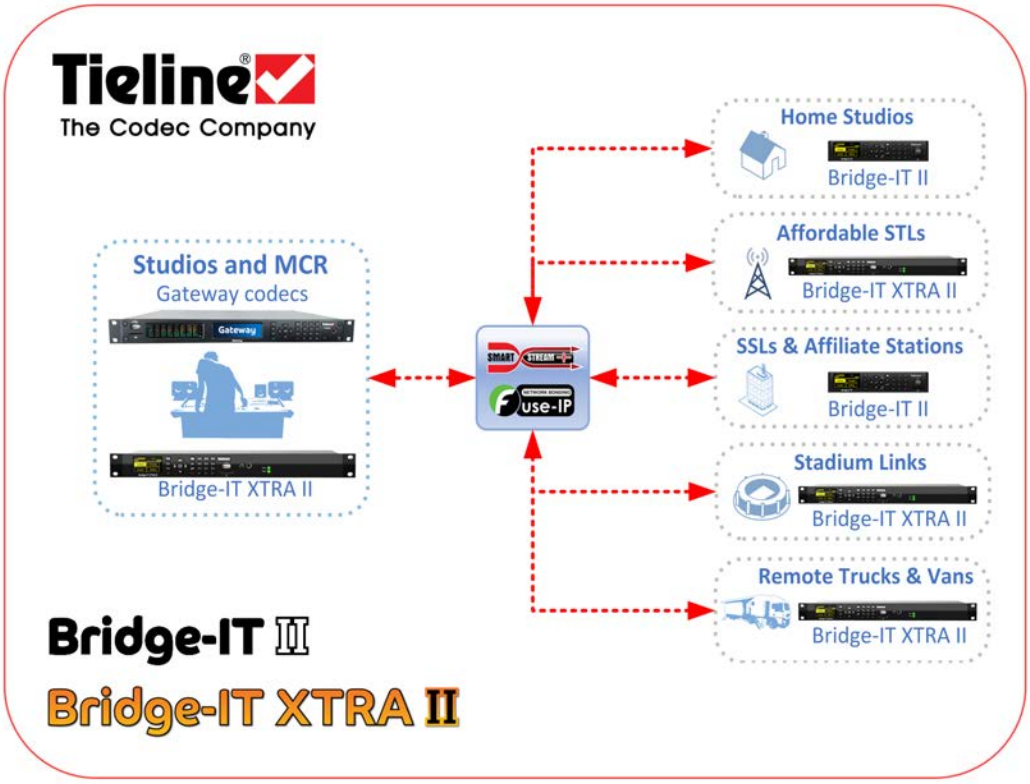
	Network Management Protocol (SNMP).
Multicast	Efficient one to many streaming of IP audio using multicast IP addressing.
Multi-unicast	A multi-unicast program (also known as multiple unicast) can transmit a single audio stream with common connection settings to a number of different destinations.
NAT	Network Address Translation is a system for forwarding data packets to different private IP network addresses that reside behind a single public IP address.
NMOS	Networked Media Open Specifications (NMOS) delivering Discovery and Registration to ensure that parts of a networked media system can find each other. NMOS also provides connection management and audio channel mapping to device I/O channels.
Packet	A formatted unit of data carried over packet-switched networks.
PAT	Port Address Translation is related to NAT; a feature of a network device that allows IP packets to be routed to specific ports of devices communicating between public and private IP networks
POTS	Plain old telephone system: copper phone network infrastructure
Primary Leader Clock	The primary source of synchronization for clock distribution via PTP.
PSTN	Public switched telephone network which is another term for POTS (see previous)
PSU	Power Supply Unit
PTP	The general class clock distribution protocol standardized in IEEE 1588-2002, IEEE 1588-2008 and IEEE 802.1AS-2011. PTP syncing requires a leader clock source (often an installed PTP primary leader clocking device) with clocking replicated on synced devices.
QoS (Quality of Service)	Priority given to different users or data flows across managed IP networks. This generally requires a Service Level Agreement (SLA) with a Telco or ISP to use over WANs.
RTP	A standardized packet format using UDP/IP networking for sending audio and video data streams and ensures consistency in the delivery order of voice data packets.
RTP stream	An RTP stream is a sequence of RTP packets with media data sent at regular interval. A stream may contain multiple channels. There may be multiple media streams per RTP session.
Runtime (edits)	Configuration changes which have not yet been saved, e.g. Matrix Editor edits.
SAP	SAP (Session Announcement Protocol) is used to distribute SDP descriptions to receivers, enabling simplified connection management for multicast streaming.
SDP	SDP defines the type of audio coding used within an RTP media stream. It works with a number of other protocols to establishes a device's location, determines its availability, negotiates call features and participants and adjusts session management features
SIP	SIP is a common protocol which works with a myriad of other protocols to establish connections with other devices to provide interoperability
SIP URI	A SIP URI is a URI used by SIP to identify user agents. SIP URI take the form sip:<user>@<domain> or sips:<user>@<domain>.
SLA	Service Level Agreements (SLAs) a contractual agreement between an ISP and a customer defining expected performance levels over a network
SmartStream PLUS	Tieline implementation of redundant IP streaming.

SNMP	Simple Network Management Protocol: Simple Network Management Protocol: a protocol used mostly in network management systems to monitor devices for conditions that warrant administrative attention.
Sources	An AoIP source in the codec is an AES67 stream sent by the codec to an AES67 LAN.
SSL	Secure Sockets Layer is a security protocol for establishing encrypted links between a web server and a browser for online communication
STL	Studio-to-transmitter link for program audio feeds
STS	Studio-to-studio audio link
STUN	The STUN protocol (Simple Traversal of UDP through NATs) assists devices behind a NAT firewall or router with packet routing. A STUN client generates STUN requests and a STUN server, attached to the public internet, receives STUN requests and sends responses.
TCP	TCP protocol ensures reliable in-order delivery of data packets between a sender and a receiver
TieLink	Traversal Server used to add Tieline codecs to a TieServer Domain and centralize codec contact list management, by providing self-discovery of codecs within call-groups, and NAT traversal to simplify connections.
TieServer	Centralized servers providing domain management facilities for Tieline applications including the TieServer Console, Cloud Codec Controller and TieLink Traversal Server.
TieServer Domain	A high-level group, associated with a particular broadcaster/customer, that is used to securely demarcate their Tieline assets from other broadcasters/customers. It applies to usage and management of Tieline codecs and Report-IT users when using Tieline applications including the TieServer Console, Cloud Codec Controller and TieLink Traversal Server.
TLS	Transport Layer Security is an updated version of SSL.
TTL	Time-to-Live is the setting used in muliticast servers to ensure data packets have a finite life and don't cause congestion over networks. Each time a packet passes through a router it reduces by 1 until it reaches zero, at which point a router will no longer pass the packet.
UDP	User Datagram Protocol: the most commonly used protocol for sending internet audio and video streams. UDP packets include information which allows them to travel independently of previous or future packets in a data stream
Unicast	Broadcasting of a single stream of data between two points
VLAN	Virtual Local Area Network: partitioning of a single layer-2 network to create multiple distinct broadcast domains
WAN	Wide Area Network; a computer network spanning regions and/or countries to connect separate LANs
WheatNet-IP	Network system using Internet Protocol to enable audio to be intelligently distributed to devices across scalable networks

4 Introduction to the Codecs

Tieline's Bridge-IT II and Bridge-IT XTRA II codecs are the ultimate affordable, high-performance, stereo IP audio codec solution for broadcast applications. Capable of dual mono or stereo connections, as well as multi-point connections, the codecs transport audio streams reliably, simply and effectively over IP data networks such as wired and wireless LANs, WANs, the internet and satellite IP networks. Bridge-IT II and Bridge-IT XTRA II codecs are perfect for a large range of broadcast and professional applications that include:

- Home studios
- Studio-to-Transmitter Link (STL) applications
- Interstudio / Studio-to-Studio Links (SSLs)
- Stereo multi-unicast IP audio distribution
- Simple remote broadcast links
- IP multicasts over compatible IP networks
- Multiple codec installations (2 Bridge-IT II codecs fit in 1 x 19” rack unit)



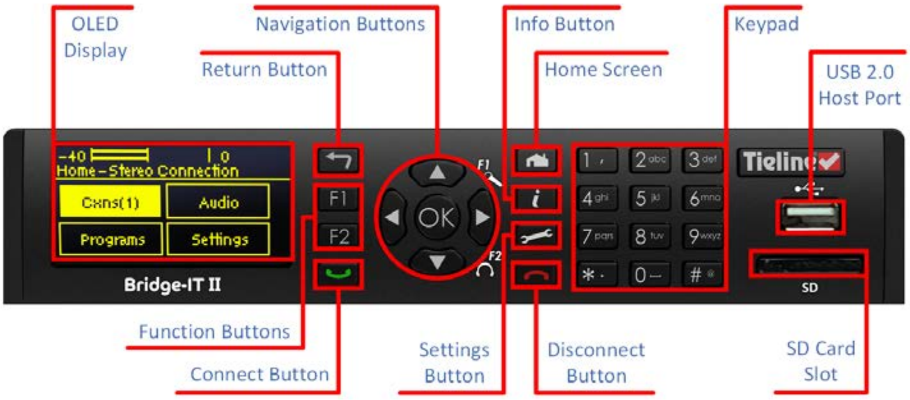
Codec Features

The following table outlines the features available in Bridge-IT II and Bridge-IT XTRA II.

Features	Bridge-IT II	Bridge-IT XTRA II
OLED Menu Screen	Yes	Yes
Analog Inputs	2 (Ch 1 mic/line; Ch2 line)	2 (Ch 1 mic/line; Ch2 line)
Digital AES3/AoIP inputs	2	2
IP Streaming	Mono, Dual Mono, Stereo	Mono, Dual Mono, Stereo
LAN Ports	2 (unless 2nd port used for AoIP)	2
AoIP Ports	LAN 2 switchable to AoIP	2
USB Host Port for Cellular Data/Tethering	Yes	Yes
Native AoIP Protocols	AES67, ST 2110-30, Livewire+, RAVENNA	AES67, ST 2110-30, ST 2022-7, Livewire+, RAVENNA
Power	12VDC	Dual Internal IEC Power Supplies
SmartStream PLUS Redundant Streaming	Included as standard	Included as standard
Fuse-IP Data Bonding	Up to 3 IP interfaces	Up to 3 IP interfaces
Failover Options	Connection, SD card file playback, HTTP streaming	Connection, SD card file playback, HTTP streaming
Multi-unicasting / Multicasting	Yes	Yes
Remote Control	HTML5 Web-GUI and optional Cloud Codec Controller	HTML5 Web-GUI and optional Cloud Codec Controller

5 Bridge-IT Front Panel Controls

Bridge-IT II features a front panel with menu navigation buttons, an **OLED SCREEN** with PPM metering and a dialing keypad. The codec also features a **USB PORT** for cellular and WiFi modems and an SDHC card slot which can be used for automatic file failover.



Navigation Buttons

The codec has four arrow shaped navigation buttons for navigating codec menus and an **OK** button for selecting menu items.

Dialing Keypad

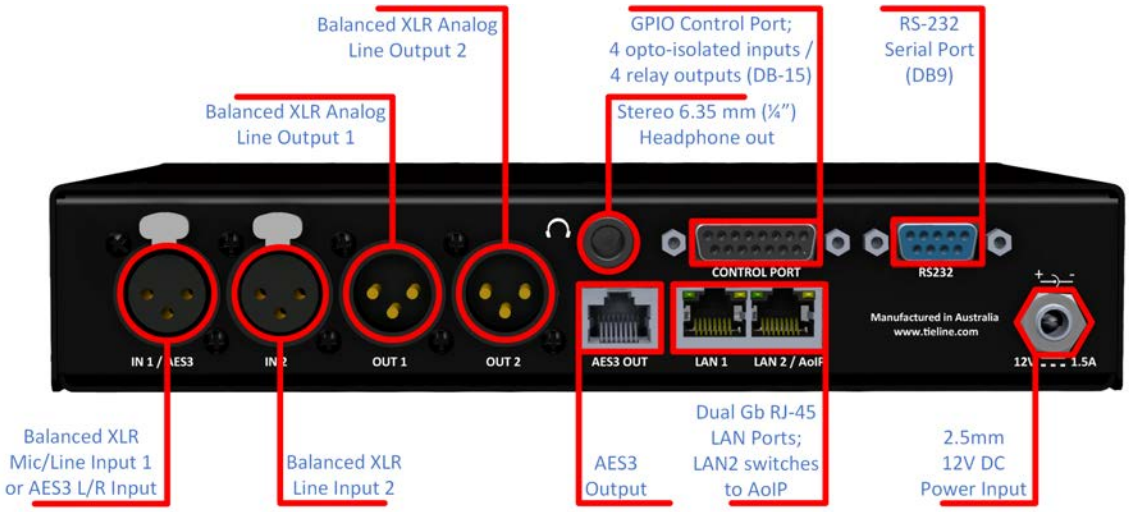
The keypad has alpha-numeric buttons and operation buttons to:

- Launch codec functions
- Navigate menus
- Dial and hang up connections

Operation Button Descriptions

	Features	Operation Button Descriptions
	Return Button	Press to navigate back through menus & delete characters
F1	Function Button 1	Press to launch codec user functions and activate relays
F2	Function Button 2	Press to launch codec user functions
	Connect Button	Press to dial IP connections
	Home Button	Press to return to **Home** screen
i	Information Button	Press to view a help menu on-screen
	Settings Button	Press to configure codec settings
	Disconnect Button	Press to end a call

6 Bridge-IT II Rear Panel Connections



XLR Analog and Digital Inputs

The codec features two XLR microphone inputs. Input 1 is a balanced mic/line input with the ability to connect high, medium and low gain mics, as well as an unbalanced source. It has switchable phantom power of 15 volts that is turned off by default and can also be used as an AES3 (AES/EBU) digital input. This input accepts both mono and stereo digital AES3 signals. Input 2 is a line input only.

XLR Analog Outputs

The codec features two balanced XLR analog audio outputs. Note: Both the XLR analog and RJ-45 digital outputs can be used simultaneously.

RJ-45 AES3 Output

The codec features an RJ-45 digital AES3 (AES/EBU) audio output. The AES3 output supports sending either a mono or stereo signal via the RJ-45 output connector. Note: Both the XLR analog and RJ-45 digital outputs can be used simultaneously.

Stereo Headphone Jack

The codec has a 6.35mm (1/4") stereo headphone output jack for monitoring audio. (See Headphone Monitoring)

Dual Gigabit LAN/AoIP Ports

The codec has two Gigabit (10/100/1000) RJ-45 Ethernet **LAN** ports for IP connections. By default, the codec assumes **LAN1** is the primary LAN connection and **LAN2** is the backup LAN connection when in use. **LAN2/AoIP** is switchable to be either a second LAN port (default setting), or can operate natively over AoIP networks using protocols that include Livewire, RAVENNA, AES67, ST2110-30 and NMOS.

Note: Always use **LAN1** if you are only using one Ethernet port. LED indications allow identification of LAN speeds as follows:

- 1000 Mbps: Green = Blinking or Solid / Orange = Off
- 100 Mbps: Green = Off / Orange = Blinking or Solid

- 10Mbps: Green = Blinking or solid / Orange = Blinking or Solid

Command & Control Interfaces

1. A DB-15 **CONTROL PORT** connector supports 4 relay inputs and 4 opto-isolated outputs for machine control.
2. A nine pin **RS-232** connection for local and remote control of equipment at either end of the link

See Appendix B for pinouts.

DC Power Input

The codec is powered by a 12 volt DC power supply using a standard polarized DC plug.

7 Bridge-IT XTRA II Front Panel Controls

The Bridge-IT XTRA II front panel features an **OLED SCREEN** with PPM metering, menu navigation buttons and a dialing keypad with operation buttons. The codec also features a stereo headphone output, a **USB PORT** for cellular and WiFi modems, and an SDHC card slot which can be used for automatic file failover.

Navigation Buttons

The codec has four arrow shaped navigation buttons for navigating codec menus and an **OK** button for selecting menu items.

Dialing Keypad

The keypad has alpha-numeric buttons and operation buttons to:

- Launch codec functions
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8 Bridge-IT XTRA II Rear Panel Connections

XLR Analog and Digital Inputs

The codec features two XLR microphone inputs. Input 1 is a balanced mic/line input with the ability to connect high, medium and low gain mics, as well as an unbalanced source. It has switchable phantom power of 15 volts that is turned off by default and can also be used as an AES3 (AES/EBU) digital input. This input accepts both mono and stereo digital AES3 signals. Input 2 is a line input only.

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RJ-45 AES3 Output

The codec features an RJ-45 digital AES3 (AES/EBU) audio output. Note: Both the XLR analog and RJ-45 digital outputs can be used simultaneously. The AES3 output supports sending either a mono or stereo signal via the RJ-45 output connector.

Dual Gigabit Ethernet Ports

The codec has two Gigabit (10/100/1000) RJ-45 Ethernet **LAN** ports for IP connections. By default, the codec assumes **LAN1** is the primary LAN connection and **LAN2** is the backup LAN connection when in use. If you are only using one Ethernet port, always use **LAN1**. LED indications allow identification of LAN speeds as follows:

- 1000 Mbps: Green = Blinking or Solid / Orange = Off
- 100 Mbps: Green = Off / Orange = Blinking or Solid
- 10Mbps: Green = Blinking or solid / Orange = Blinking or Solid

Dual Gigabit AoIP Ports

Two Gigabit (10/100/1000) RJ-45 **AoIP** ports provided native support for Livewire+, RAVENNA, AES67, ST2110-30, ST2022-7 and NMOS.

Command & Control Interfaces

1. DB-15 **CONTROL PORT** connector supports 4 relay inputs and 4 opto-isolated outputs for machine control.
2. A nine pin **RS-232** connection for local and remote control of equipment at either end of the link

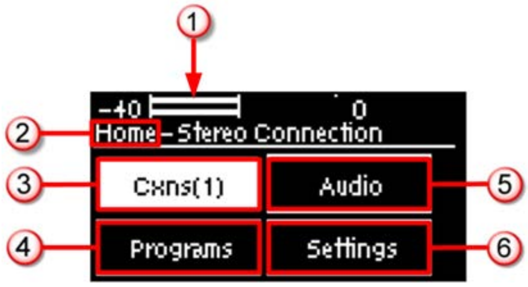
See Appendix B for pinouts.

Dual Redundant AC Power Inputs

The codec is powered by dual 100-240 volt redundant AC power supplies, which use standard IEC connectors.

9 Navigating Codec Menus

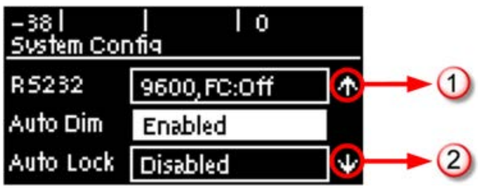
The codec has simple and intuitive menu navigation screens. All main codec menus can be launched from the **Home** screen and audio levels remain visible throughout all menus.



Bridge-IT Home Screen

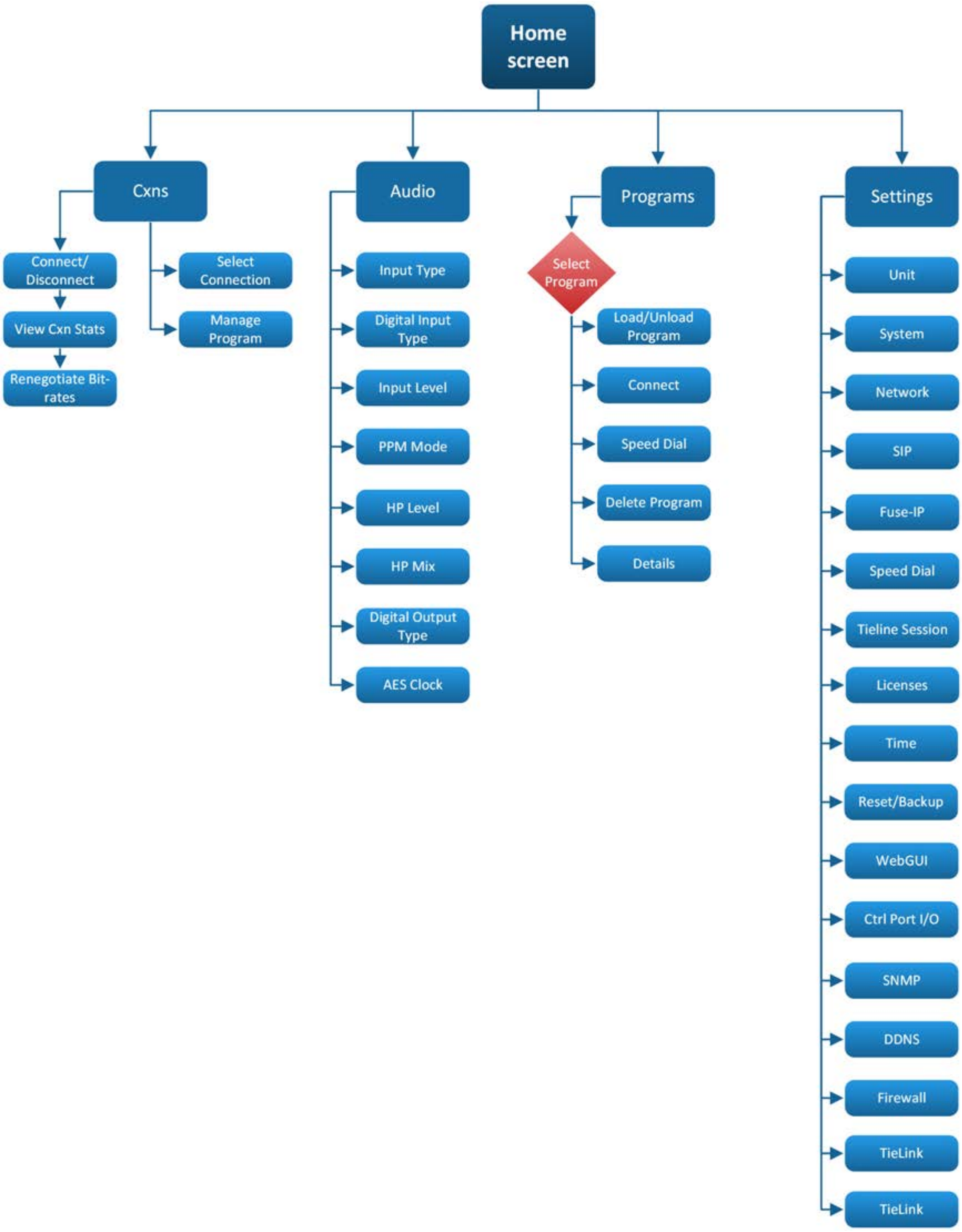
	Features	Codec Home Screen Elements
1	PPM Meters	Left (top) and right channel audio levels
2	Screen Name	The name of the current screen
3	Cxns	Select to configure program and connection settings
4	Programs	View, select, load or delete saved Programs
5	Audio	Select to configure a range of audio settings
6	Settings	Select to configure codec settings

Press the **RETURN** button to navigate backwards through menus, or press the **HOME** button to return to the **Home** screen from any menu. If a full menu cannot be viewed on the codec screen then arrows on the right hand side of the screen indicate that the current menu has items below and/or above the items currently visible. Use the navigation arrows to scroll up and down.

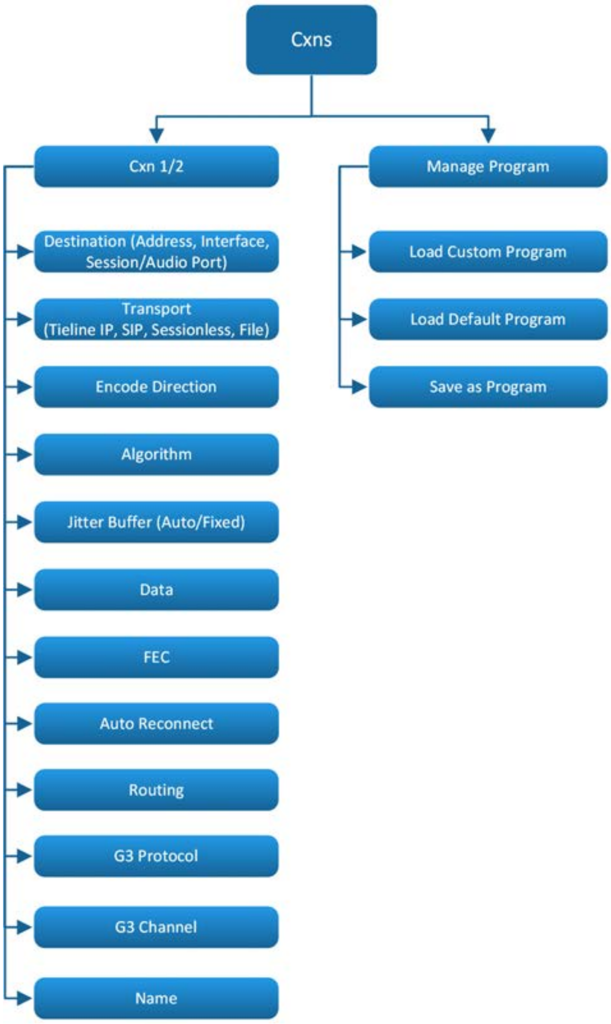


	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

Home Screen Menu Overview



Cxns Menu


Cxns
Cxn 1/2
Manage Program
Destination (Address, Interface, Session/Audio Port)
Transport (Tieline IP, SIP, Sessionless, File)
Encode Direction
Algorithm
Jitter Buffer (Auto/Fixed)
Data
FEC
Auto Reconnect
Routing
G3 Protocol
G3 Channel
Name
Load Custom Program
Load Default Program
Save as Program

10 Input/Output Configuration, Levels and PPMs

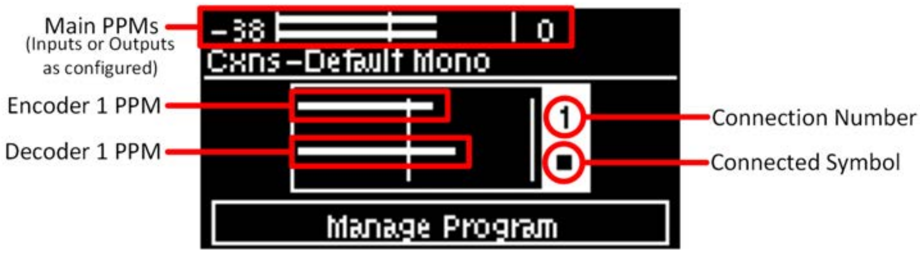
By default, the **PPM METERS** on the front of the codec, or in the HTML5 Toolbox Web-GUI, use dBFS to express nominal operating, headroom and noise floor levels. The PPM meters above the **Home screen** display either input audio or output audio, depending on what has been configured in the **PPM Mode** menu. To adjust this setting, press the **Home** button to return to the home screen and select **Audio > PPM Mode > Input / Output**.

Important Note: See Configuring AES3 audio for more information about AES3 in/outs. Inputs and outputs can also be configured using the Toolbox Web-GUI; see Configuring Input Settings and Configuring Output Settings for more information.

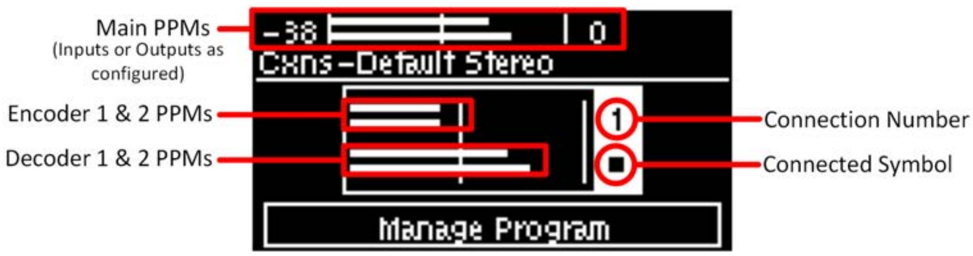
Connections Screen PPMs

Press the **Home** button to return to the home screen and select **Cxns** to display the **Connections** screen. Following are the PPM and connection indications displayed for mono, stereo and dual mono programs.

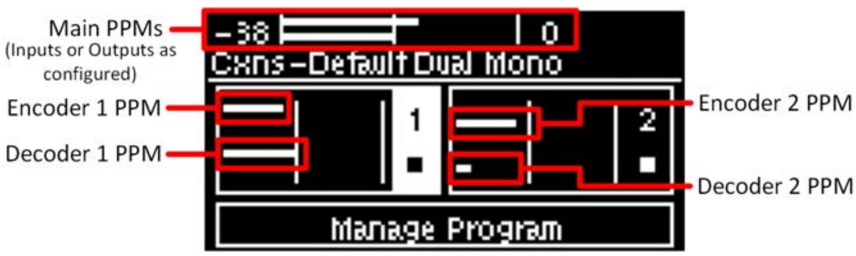
Mono Program



Stereo Program

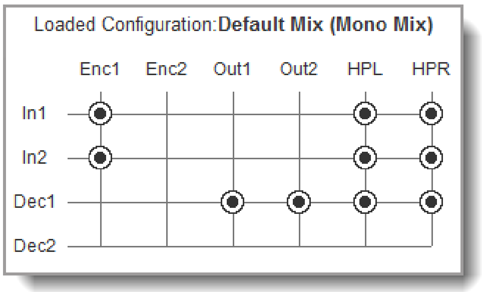


Dual Mono Program

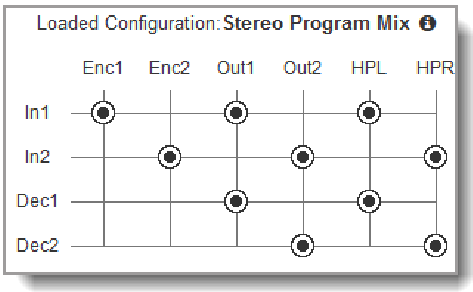


Mono and Stereo Audio Routing

Input and output routing is based on whether a default mono, stereo or dual mono program is loaded. Following is the default routing for mono, stereo and dual mono programs.



Default Mono Matrix Router



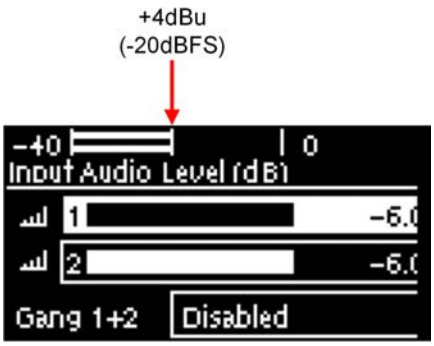
Default Stereo/Dual Mono Matrix

Audio routing can be adjusted using the HTML5 Web-GUI **Matrix Editor panel**. See Matrix Editing. The codec feeds audio to both the analog and digital outputs, whether the audio source is analog or digital. When the codec answers a mono call it feeds mono audio to both the left and right analog XLR outputs, as well as the left and right digital outputs (AES3 or AoIP).

Setting Levels and PPM Meter Reference Scales

The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G6, G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G6** reference scale setting.

The default Tieline G6 audio reference scale displayed on the PPMs when connecting to a Tieline G6 codec is -40dBFS to 0dBFS. Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 20dB of headroom from the nominal 0vu point. The comparison table below outlines the reference scales for G6, G5, G3 codecs and Report-IT in dBFS, as well as the equivalent dBU scale.



	Reference Level	Description	dBu	dBFS
1	Tieline G6 (Gateway, Gateway 4, MPX I, MPX II, Bridge-IT II)	PPM meter low point	-16dBU	-40dBFS
		Nominal 0vu reference level	+4dBU	-20dBFS
		Level at which audio will clip/distort	+24dBu	0dBFS
2	Tieline G5 (Genie, Merlin, Bridge-IT, ViA)	PPM meter low point	-16dBu	-38dBFS
		Nominal 0vu reference level	+4dBu	-18dBFS
		Level at which audio will clip/distort	+22dBu	0dBFS
3	Tieline G3	PPM meter low point	-11dBu	-29dBFS

	(Commandera nd i-Mix)	Nominal 0vu reference level	+4dBu	-14dBFS
		Level at which audio will clip/distort	+18dBu	0dBFS
4	Report-IT	PPM meter low point	-9dBu	-23dBFS
		Nominal 0vu reference level	+4dBu	-10dBFS
		Level at which audio will clip/distort	+14dBu	0dBFS

Important Note: If a codec supports multiple stream programs and the **Auto** (default) reference level is selected, the first codec to connect will configure the reference level used for all subsequent connections. I.e. If a G3 codec connects first then the G3 Audio Reference Level will be configured for all connections.

Configure Tieline G5 Audio Reference Scales

Tieline G6 codecs like Bridge-IT II and Bridge-IT XTRA II have slightly more headroom than Tieline G5 codecs, therefore the audio metering reference scale needs to be adjusted when G6 codecs connect to G5 Merlin, Genie, ViA and Bridge-IT codecs. The G5 metering scale is between -38dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -18dBFS. Audio peaks should not exceed 0dBFS to prevent clipping.

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press OK.
3. Navigate to **Ref Level** and press OK.
4. Select **Tieline G5** and press OK.

Configure Tieline G3 Codec Audio Reference Scales

G5 and G6 codecs have more audio headroom than Tieline G3 audio codecs, therefore the audio metering reference scale needs to be adjusted when these codecs connect to a Commander or i-Mix G3 codec. The G3 metering scale is between -29dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -14dBFS. Audio peaks should not exceed 0dBFS when using analog audio to prevent clipping.

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press OK.
3. Navigate to **Ref Level** and press OK.
4. Select **Tieline G3** and press OK.

Configure Report-IT Audio Reference Scales

The **Report-IT** setting is used for compatibility when connecting using Tieline's Report-IT smartphone application. The Report-IT metering scale is between -23dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -10dBFS. Audio peaks should not exceed 0dBFS when using analog audio to prevent clipping.

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press OK.
3. Navigate to **Ref Level** and press OK.
4. Select **Report-IT** and press OK.

Selecting Analog or Digital Inputs

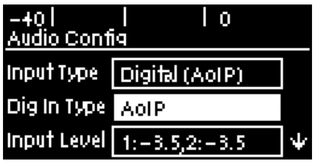
1. Press the **Home** button to return to the home screen.

2. Navigate to **Audio** and press .
3. Navigate to **Input Type** and press .
4. Select **Analog** or **Digital** as required and press . This configures both codec inputs simultaneously.

Selecting the Digital Input Type

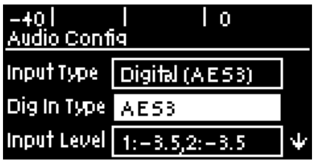
The digital inputs on the codec can be either AES3 or AoIP (AES67, Livewire, RAVENNA etc.). To select the preferred option:

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press .
3. Ensure the **Input Type** selected is **Digital**.
4. Navigate down to **Dig In Type** (digital input type) and press .



5. Select either **AES3** or **AoIP** and press .

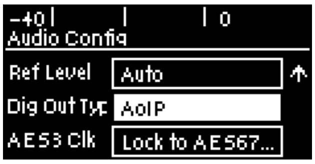
6. Both inputs are now configured for this digital input type.



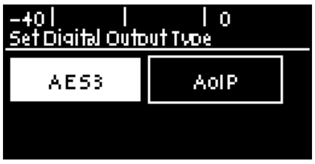
Selecting the Digital Output Type

The codec outputs can be configured to feed either digital AES3 or AoIP audio. To configure the digital output type for either **AES3** or **AoIP**:

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press .
3. Navigate down to **Dig Out Type** (digital output type) and press .



4. Select either **AES3** or **AoIP** and press OK.



Note: The codec simultaneously feeds audio to the analog outputs, as well as either the digital AES3 or AoIP outputs.

Channel 1 Mic/Line Level Settings

The default input level setting in the codec for channel 1 is line level. To adjust this setting for a mic-level or unbalanced source:

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press OK.
3. Navigate to **Input Type** and select **Analog**.

4. Navigate down to highlight and select **Input 1 Gain** and press the OK button.
5. Use the navigation buttons to select the appropriate gain setting and press the OK button to save the setting.

Important Note: 15 volt phantom power is not supplied to input 1 by default. To turn this on:

1. Select **Home** **> Audio** and navigate to **Phantom**.
2. Press the OK button to toggle between **Enabled** and **Disabled**.

Channel 2 is a line input only.

Adjusting Input Levels

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press OK.

3. Select **Input Level** and press .
4. Press **1** on the numeric keypad to toggle channel 1 on and off and press **2** on the numeric keypad to toggle channel 2 on and off.
5. Use the up and down arrow buttons to navigate to the channel you want to adjust. Note: A channel is highlighted when selected. Navigate to **Gang 1+2** and select **Enable** to adjust levels for both channels simultaneously.
6. Use the left-hand and right-hand arrow-shaped navigation buttons to adjust input levels up or down.
7. Press **RETURN** to exit the screen.

	Input Audio Features	Description
1	Channel On Symbol	Symbol indicates a channel is turned on
2	Channel Off Symbol	Symbol indicates a channel is turned off
3	Input 1 Level Control	Input 1 gain indication; adjust in 0.5db increments
4	Input 2 Level Control	Input 2 gain indication; adjust in 0.5db increments
5	Ch1/2 Gang Indication	Indicates whether ganging is enabled or disabled

Important Notes:
- Gain adjustments can be made in 0.5dB increments.
- There is a maximum of 6dB of additional gain available when adjusting a digital input.
- Audio signal processing occurs in the following order: IGC/Limiter (Intelligent Gain Control limiting on the input) > Mixer > Output AGC (Automatic Gain Control limiter).
- Press the **F1** button and the right-hand navigation button to open the **Input Audio Level** adjustment screen.

Ganging Audio Channels

Ganging allows you to adjust the audio level of both inputs simultaneously.

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press .
3. Select **Input Level** and press .
4. Use the up and down navigation buttons to select **Gang 1 + 2**.

5. Press the button to toggle between **Enabled** and **Disabled**.
6. Adjust the levels for both inputs up or down simultaneously.
7. Press the **RETURN** button to exit the screen.

Important Note: When channels 1 and 2 are ganged together:

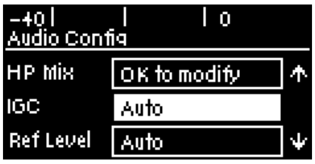
- You can adjust the audio of both channels simultaneously.
- The gain setting for both channels is automatically set to match the gain level of the lowest of the two channels when ganging is configured.
- If one channel is turned on when ganging is configured then the other one will be turned on automatically.

Intelligent Gain Control (IGC)

The codec's inbuilt DSP limiter automatically takes care of any instantaneous audio peaks that occur in demanding broadcast situations. **Input IGC** (Intelligent Gain Control) is enabled by default and is automatically activated at +19 dBu (G6 audio scale), +17 dBu (G5 audio scale) and +13dBu (G3 audio scale) to prevent audio clipping.

There are three settings; **Auto**, **Fixed** and **Off**. If **Auto** is configured the codec will detect when incoming audio levels have reduced sufficiently and automatically return input levels to the gain setting prior to IGC being activated. The codec takes just 250 milliseconds to detect audio levels have returned to normal (after **IGC Level** has been initiated) and will return levels to the previous setting within half a second. This response is linear. If the setting is **Fixed** then audio levels will remain lower and not return to the original setting.

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press OK.
3. Navigate to **IGC** and press OK.



4. Select the preferred setting and press OK.

11 Configuring AES3 Settings

If your input source is AES3 (AES/EBU) format use the **IN1/AES3** input on the rear panel of the codec. This is a balanced 110 ohm female XLR input and can operate effectively over distances of up to 100 meters. The AES3 XLR input accepts both mono and stereo AES3 signals. The AES3 output also supports sending either a mono or stereo signal via the RJ-45 output connector.

To configure the codec to use AES3 requires the codec to be configured for AES3 input and output audio, instead of digital AoIP audio. Configure the following settings:

Selecting Analog or Digital Inputs

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio > Input Type** > **Digital** and press . This configures both codec inputs simultaneously.

Selecting the Digital Input Type

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio > Dig In Type** > **AES3** and press . This configures both codec inputs simultaneously.

Selecting the Digital Output Type

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio > Dig Out Type** > **AES3** and press . This configures both codec outputs simultaneously.

Important Notes:

- There is a maximum of 6dB of additional gain available when adjusting an AES3 digital input.
- If you switch back to the analog input setting after selecting AES3, the previous analog settings will be recovered.
- AES3 audio is sent to the outputs, as well as AoIP audio, regardless of the Digital Output Type setting. However, there is an important point to note. When both the Digital Input Type and Digital Output Type on the codec is set to AES3, the AES Output Clock can be changed. However, when either the Digital Input Type or Digital Output Type is set to AoIP, the AES Output Clock is locked to the AoIP clock (e.g. PTP in AES67, WNIP clock in WNIP mode etc). Since the AES Output Clock also sets the synchronization of the AES67 sources, it is important that the AoIP clock is selected when using AES67 output. If it is not set to use the AoIP Clock, then the AES67 output will usually appear to be ok initially, but the timestamps will eventually deviate from the master clock reference (e.g. PTP) and some devices may stop receiving the streams when the timestamp deviates too far. So, when using AoIP outputs, it's always best to set the Digital Output Type to AoIP. If AES3 outputs are also being used, then the AES3 output sampling clock will be locked to the AoIP reference clock.

AES3 Out Clock Source

The codec contains two sample rate converters.

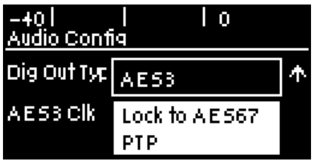
Input Sample Rate Converter

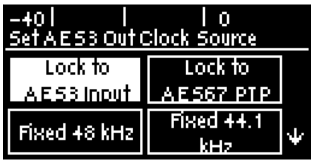
The codec implements an Asynchronous Sample Rate Converter (ASRC) to convert the sample rate of an AES3 input to the sample rate set in the codec. The codec sample rate is determined by the selected algorithm. For example, if you select the Music algorithm, the sample rate will be set to 32kHz. By default the codec will up-sample all AES3 input sources to 96kHz sampling and then convert to match the AES output sample rate setting.

Important Note: See Appendix A for AES3 pin-outs.

Output Sample Rate Converter

The sample rate of the AES3 output is configured using the clock source setting via **Audio > AES3 Clk** (AES3 Output Clock Source). This configures the sample rate frequency of all AES3 output signals.





Lock to AES3 Input
With this setting the codec uses AES3 input sync information to set the codec output sample rate (Note: this is the same as the **AES Rx Clock** setting in Tieline G3 codecs). Supported sample rates include 32 kHz, 44.1 kHz, 48 kHz and 96 kHz. Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Lock to AES67 PTP
AES67/ST2110-30/Livewire derives the sampling clock from a primary leader clock over the network or a Livewire clock. This clock can be selected as the AES3 Output Clock Source. The AES3 Output Clock Source uses the **Lock to AES67 PTP** setting if either the **Digital Input Type** or **Digital Output Type** is set to **AoIP**.

Fixed Sample Clock
Select from a range of fixed output sample rates including:

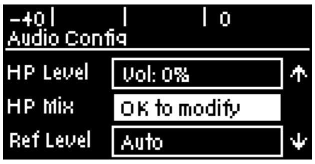
1. 32 kHz
2. 44.1 kHz
3. 48 kHz
4. 88.2 kHz
5. 96 kHz

Note: The reference clock must be within +/- 50ppm of the listed sample rates.

12 Headphone Monitoring

The 6.35mm (1/4") stereo headphone output on the codec can be used for monitoring audio inputs 1 and 2 and return decoder audio. The headphone monitoring mix can be adjusted on the codec as follows:

1. Press the **Home** button to return to the home screen.
2. Navigate to **Audio** and press OK.
3. Navigate to **HP Mix** and press OK.



4. Use the navigation keys to select inputs and decoders to monitor from the headphone output. Navigate to a crosspoint and then press OK to select and deselect a crosspoint. Routing edits are also reflected automatically in the **Matrix Editor panel** and the **Headphones panel** in the HTML Toolbox Web-GUI. For more information see Matrix Editing and Headphone Monitoring.

Important Notes:

- Edits to crosspoints on the **Headphone Mix** screen are saved in current runtime.
- Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.
- To save edits to a program use the **Matrix Editor panel** to save a custom matrix and attach this matrix to the program using the HTML5 Toolbox Web-GUI **Program Manager panel**.

Adjusting the Send / Return Balance Slider

The send/return balance between incoming and outgoing audio sources can also be adjusted.

1. Navigate down to the **Send/Return** slider at the bottom of the **Headphone Mix** screen.

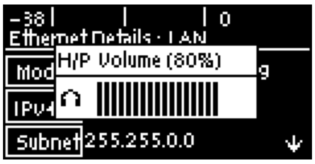
2. Use the left-hand and right-hand navigation keys to adjust the **Send/Return** slider balance between incoming and outgoing audio sources.

Important Note: Press and hold the **F2** button and then press the right-hand arrow button to display the **H/P Volume** adjustment screen.

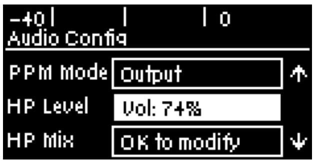
Adjusting Headphone Output Levels

To adjust the audio level of the headphone output.

1. Press and hold the **F2** button and then press the right-hand arrow button to display the **H/P Volume** adjustment screen.
2. Use the left or right navigation buttons to adjust the volume levels up or down. The screen displays level adjustments in real-time.
3. Press OK when you have finished.



Headphone levels can also be adjusted by navigating to **SETTINGS** **> Audio** and use the down navigation button to navigate to **HP Level** and press OK.

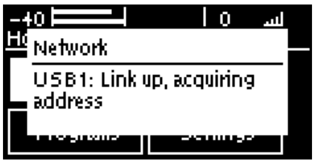


13 USB Cellular Connections

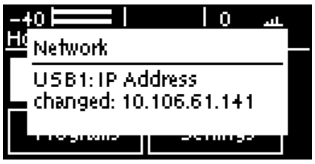
Connecting with Cellular USB Modems / Air Cards

To get started, attach a supported USB Modem to the **USB PORT** on the front of the codec.

1. When the modem is detected by the codec, the cellular signal strength indicator at the top of the **OLED SCREEN** flashes and an onscreen dialog indicates the codec is acquiring an address.



2. Eventually the codec should acquire an address and this is displayed in a dialog on the **OLED SCREEN**. Press **RETURN**  to remove the dialog from the screen.



3. The signal strength indicator is displayed at the top right-hand side of the **OLED SCREEN**. Note: It may take up to 90 seconds for the modem to be detected by the codec and connect to the network.

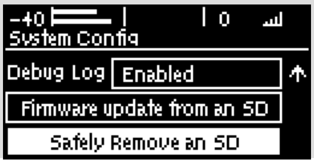
4. From the **Home** screen press the **Settings**  button and navigate to **Network > USB** to view cellular configuration menus and cellular modem details.

5. The USB menu displays a range of cellular related info to assist with troubleshooting connections. The signal strength is measured out of 4. In the following example the modem is connected to the **LTE** network and signal strength is **4/4** and is therefore excellent.

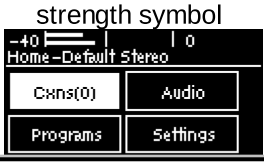
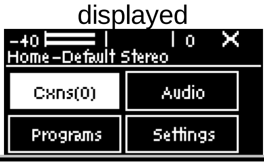
6. Other info and settings are also displayed as follows.

Important Notes:

- Sometimes a cellular network will be detected and a USB modem will connect automatically after it is inserted. Tieline recommends always enabling an APN (Access Point Number) and entering the correct APN for your Telco. This ensures use of the correct data APN. It is usually simple to search the internet for the correct APNs used by each Telco. See Adding Access Points and a SIM PIN for more details.
- It may be necessary to enter a SIM PIN if the codec cannot connect automatically to the network.
- Visit https://tieline.com/via-usb-modem-compatibility-list/ to view a list of compatible USB modems.
- To safely remove a USB modem press the **HOME** button to return to the **Home** screen, then tap **Settings** and navigate down to **Safely Remove an SD** and press .



USB Cellular Modem Indications

	Symbol	Description of Status
1	No signal strength symbol 	• A cellular modem is not attached to the codec. • A modem is attached but not ready yet; it can take up to a minute for the modem to be detected. • The modem is not supported by the codec.
2	Cross symbol displayed 	A cellular modem is attached to the codec but there is an error. Check if: • The modem has a SIM card issue (locked or not present). • The SIM card does not have a PIN code enabled. • The modem is compatible with the network to which you are attempting to connect.
	Flashing signal strength symbol	If the modem does not have data enabled the signal strength indicator will continue to flash.
3		The cellular modem is connected to the network and signal strength is displayed.

Troubleshooting Cellular Modem Data Issues

If a good solid signal strength is displayed but connections are not possible, please check for the following potential issues:

- No APN is selected and the modem doesn't contain a correct APN setting from a factory default or previous connection.
- An incorrect APN is selected. When using different SIMs they may require different APNs, even if they are from the same Telco.

- The correct APN is selected but data has run out.
- A custom APN has been added but either the APN info, or the Authentication Type, is incorrect.

Tethering a Smartphone

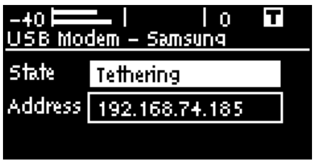
Important Note: Only tethering to an Android phone is supported.

It is possible to tether a cellphone to the codec with a USB cable and use cellular or Wi-Fi data from the connected device.

1. Unlock your screen and enable USB tethering in the internet-connected smartphone to allow sharing of the phone's internet connection. Note: This will probably need to be enabled each time the phone is connected to the codec.
2. Attach a USB cable to the phone and then attach the USB cable to the **USB PORT** on the codec front panel.
3. The codec should detect the smartphone USB connection and start to connect and acquire an IP address from the phone.

4. When it connects successfully the codec should acquire and display the IP address in the menu via **Settings > Network > USB** and the **Tethering** symbol is displayed at the top of the screen.

5. The **USB** menu summary displays as follows with the device name at the top, e.g. Samsung in this example.



6. When the codec is connected the screen continues to display the **Tethering** symbol to identify this interface is connected.

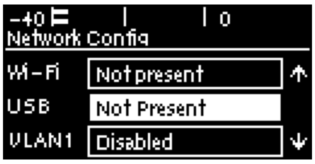
14 Adding Access Points and a SIM PIN

It may be necessary to add a custom access point number (APN) profile if a USB modem does not connect to a cellular network. Often this is necessary when you are using different SIM cards in an unlocked USB modem. E.g. When using SIM cards from various carriers in different countries. As a rule of thumb, Tieline recommends always enabling an APN in the USB menu and entering the correct APN for your Telco. This ensures use of the correct data APN. It may also be necessary to enter a SIM PIN unlock code.

Up to 10 custom access point profiles can be added to the codec. Before configuring the custom access point you need to obtain the access point details from the cellular network provider; this is normally found on a Telco's website. Usually a Telco will list internet and MMS APN information and you need to enter the internet APN details in the codec, as well as the correct authentication type.

Entering a SIM PIN Unlock Code

1. Press the **Home** button to return to the home screen, then select **Settings**.
2. Select **Network** and press and then navigate to **USB**.



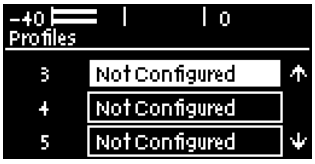
3. Press and navigate down to **SIM PIN**.

4. Enter the **SIM PIN** and press to apply the new PIN setting.

Add a Custom APN Profile

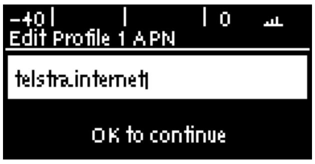
Up to 10 custom access point profiles for different networks can be added to the codec.

1. Press the **Home** button to return to the home screen, then select **Settings**.
2. Select **Network** and press and then navigate to **USB** and press .
3. Select **Data** and press the button to disable this setting.
4. Navigate to **Profile** and press the button to add a new APN profile.
5. Select a profile number that has not been configured and press the button.

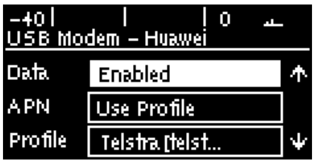


6. Select **Edit** and press the button.

7. Name the profile and enter custom APN settings, then press the OK button.



8. Press the **Return** button to navigate back to the **USB Modem** screen and enable **Data**. Ensure **Use Profile** is selected for the **APN** setting and the correct custom **Profile** is loaded.



Note: To configure this setting in the HTML5 Toolbox Web-GUI see Configure Cellular Modems.

Select a Custom APN Profile

1. Press the **Home** button to return to the home screen, then select **Settings** .
2. Select **Network** and press OK and then navigate to **USB** and press OK.
3. Navigate to **Profile** and press the OK button view existing APN profiles. The currently selected APN profile has a tick next to it.

4. Navigate to a preferred APN profile and press the OK button and then **Select** to load it. The newly selected APN profile will have the tick next to it.

5. Press the **Return** button to navigate back to the previous screen and navigate up to **Data** and press the OK button to enable this setting. The USB modem should connect to the network as displayed in the following image.


-40 0
USB Modem - Sierra
Data Enabled
APN Enabled
Profile telstra (telst...

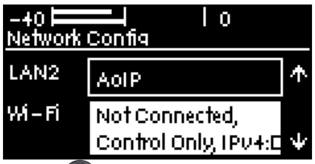
15 Connect Wi-Fi for Control

Important Note: Wi-Fi is only used for codec control and not for IP streaming. E.g. Dual LAN ports can be used for redundant IP streaming exclusively and a Wi-Fi network can be used for codec control.

Connecting a USB Wi-Fi Dongle to the Codec

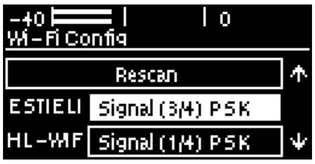
1. Attach a supported USB Wi-Fi dongle to the **USB PORT** on the front of the codec.
2. When the Wi-Fi dongle is detected by the codec, the signal strength indicator at the top of the **OLED SCREEN** flashes and an onscreen dialog indicates the dongle has been attached.

3. Press the **Home** button to return to the home screen, then select **Settings** .
4. Select **Network** and press and then navigate to **Wi-Fi**.



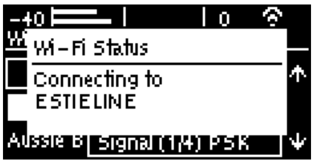
5. Navigate down to **Rescan** and press to scan for available Wi-Fi access points.

6. Navigate down to the preferred access point and press to select it. Note: signal strength out of 4 is indicated for each available network.

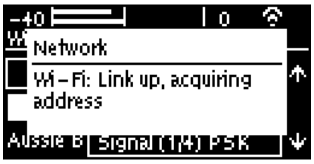


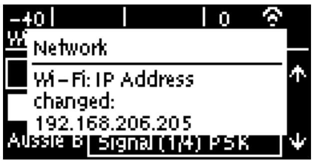
7. Select the correct security authentication mode and enter the **Password** for the network.

8. Navigate up to **Save and add Wi-Fi Network** and press to add the Wi-Fi access point.



9. The codec will then connect to the Wi-Fi network and acquire an IP address.





10. When the codec has successfully connected the Wi-Fi symbol is displayed on the screen and the following image displays full signal strength.

Important Notes:

- It is also possible to configure Wi-Fi settings remotely using the Toolbox web-GUI and Cloud Codec Controller if required using the **Network panel**. For more details see Connecting Wi-Fi.
- It is also possible to manually add a Wi-Fi network via **Settings > Network > Wi-Fi > Add Wi-Fi Network**. Enter the SSID, authentication method and password credentials.
- If signal strength is very low the Wi-Fi symbol may not be displayed at all, or only the bottom section of the Wi-Fi symbol will be displayed. You can verify signal strength in the **Wi-Fi** menu via **Settings > Network > Wi-Fi**.

16 Language Selection

English is the default language in the codec. To adjust this setting:

1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Use the navigation buttons to select **Language** and press .
4. Select a language and press .

17 IP Streaming, Programs and TieLink

Tieline codecs support high quality IP audio streaming connections using the following protocols:

1. Tieline Session Data: Proprietary session data sent when Tieline codecs connect to each other in order to establish, manage and terminate connections.
2. SIP (Session Initiation Protocol): EBU N/ACIP Tech 3326 compliant connections using SDP (Session Description Protocol) when connecting to other brands of IP codecs.
3. Sessionless: The codec does not send session data when attempting to connect. Requires the "send" audio port and "return" audio port to be configured.

Tieline Session Data

When a connection between two codecs is established:

1. The dialing codec sends session data including information about how the codec receiving the call should be configured.
2. Once the answering codec receives session data it sends an acknowledgment to the dialing codec and streaming can commence.

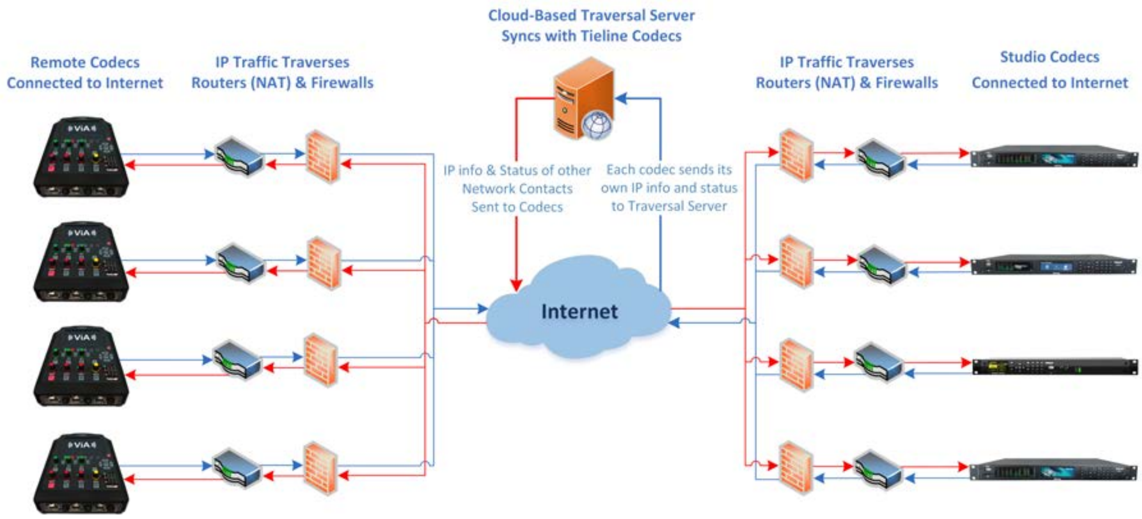
For example, if you configure a stereo program on the dialing codec using Opus encoding at 128kbps with specific jitter settings, this will be configured on the answering codec when it connects.

SIP

SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices. The codec is fully EBU N/ACIP Tech 3326 compliant for connecting to any codec supporting SIP connections. For more information about SIP connections see About SIP.

TieLink Traversal Server Network

TieLink is a secure, independently hosted global server network, with multiple global backups. Linked to a TieServer domain, it centralizes Tieline codec contact list management and provides self-discovery of codecs within customized ‘call-groups’. It also provides NAT traversal to simplify connections. This service is free to all Tieline customers but is not supported in first generation Bridge-IT or G3 codecs.



How does it work?

Using a simple HTML5 web-browser interface, engineers can log in and register their codecs with TieLink. Simply name your codec (e.g. BAYFM Studio A) and complete the registration details. Once registered, a codec will automatically connect to TieLink each time it connects to the internet:

1. TieLink automatically detects a codec's IP address and NAT routing information. Seconds later the codec address book populates with all codecs in the network, or a subset of codecs in an authorized call-group.
2. The registered codec will appear in the address books of all other codecs registered with TieLink and connected to the internet. Call-groups can be created to customize the number of codecs visible to users and engineers within large networks.
3. Users can easily view the online or offline status of all codecs in a group and whether it is connected or disconnected.

Manage Call Groups Easily

Whether you are running a large network with hundreds of IP codecs, or a small station with just a few, the TieLink Traversal Server and Toolbox software makes connecting over IP networks a simple, no-fuss task for non-technical broadcast personnel.

For networks with large numbers of devices, it is easy to create sub-groups and configure a codec to follow a specific group, e.g. news or sports. All of the codecs in your network are secure and cannot be seen by other users unless they have permission to join the network, or a specific codec call-group. See Configuring TieLink Settings for more information.

Programs

A **Program** configures a Tieline codec to send or receive one or more **Audio Streams** based upon the particular application the codec is being used for at any given time. The attributes of the audio stream and associated connections are embodied within a program when it is created, including the configuration, dialing and answering parameters. By default, Tieline codecs send proprietary session data when connecting to each other in order to establish, manage and terminate connections. When a connection between two codecs is established:

1. The dialing codec sends session data including information about how the codec receiving the call should be configured.
2. Once the answering codec receives session data it sends an acknowledgment to the dialing codec and streaming can commence.

It is also possible to lock a loaded program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with different program settings. For example, if your routing requirements require the codec at the studio to always connect in mono, simply load and lock a mono program in the codec. Generally programs will be up or down-mixed by the answering codec to match the loaded program type where possible. In some situations incompatible program types will be rejected.

Program Types

The HTML5 Toolbox Web-GUI contains a **Program Manager panel** with a wizard for configuring program settings and backup connections. Choose from a wide selection of preconfigured program templates like mono, dual mono and stereo connections, as well as multicasting and multi-unicasting.

Defining Audio Streams within Programs

Each audio stream within a program can be defined separately and contain a variety of settings relating to the number of connections (e.g. primary and backup) and the number of destinations to which each audio stream is distributed. Each audio stream is capable of being configured to include dial and answer connections, dial connections only, or answer connections only. Each audio stream includes:

- Name.
- Connection, Transport, and Destination settings.
- Backup configuration options.

Multi-Unicast Programs

A multi-unicast program (also known as multiple unicast) can transmit a single audio stream with common connection settings to a number of different destinations. All connections that are part of a multi-unicast connection will automatically attempt to reconnect if they are terminated remotely.

Multi-unicasts are useful for distributing audio to several studios and can only be created using the **Program Manager panel** in the HTML5 Toolbox Web-GUI. Once multi-unicast connections have been created and loaded you can press **CONNECT** on the codec keypad to connect without using the HTML5 Toolbox Web-GUI. See Configure Multi-unicast Dialing Programs for more info.

Multicast Programs

A multicast audio stream can be used to broadcast to unlimited numbers of 'subscriber' (client codecs) wanting to receive a particular audio transmission. Multicast transmissions are broadcast by a multicast server codec using a dedicated IP multicast address, which looks similar to a regular IP address, and multicast subscribers request transmissions from this address. See Configure Multicast Server Programs or Configure Multicast Client Programs for more info.

18 Getting Connected Quickly

Preparing to Connect

Before attempting a new connection please do the following:

1. Attach power to the codec.
2. Attach an RJ45 Ethernet cable to the **LAN1** port on the rear panel of the codec.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the codec.
4. Check that the correct country is selected in the codec to ensure it complies with applicable G.711 encoding in your region, as well as WiFi channel regulatory requirements:
   i. Press the **SETTINGS** button.
   ii. Navigate to **System** and press the OK button.
   iii. Navigate to **Country** and press the OK button.
   iv. Use the navigation buttons to select your country of operation.
5. Note the IP address of the codec that will be dialed, or ensure programs have been created using the codec **OLED SCREEN** or HTML5 Toolbox web-GUI to dial or answer connections.

Important Note: The country setting affects whether G.711 µ-Law (North America/Japan) or A-Law (Europe/Australasia) coding is used over IP and SIP connections.

18.1 Steps to Connect

The codec supports a range of different connection options. The following procedures are valid for connecting a:

- Mono Program
- Stereo Program
- Dual Mono Program

Important Notes:
- See Input/Output Configuration, Levels and PPMs to adjust input levels before connecting.
- 15 volt phantom power is not supplied to input 1 by default. To adjust this setting select **Audio** , select **Analog** as the **Input Type** and then **Phantom**. Press the **OK** button to toggle between **Disabled** and **Enabled**.
- See Headphone/Output Monitoring for information on monitoring incoming and outgoing audio sources.
- See HTML5 Toolbox Web-GUI Configuration for details on configuring connections using a computer.
- To configure SmartStream PLUS redundant streaming it is necessary to configure a program using the HTML5 Toolbox Web-GUI.
- See Installing the Codec at the Studio for valuable information about installing your codec, negotiating firewalls and port forwarding.
- See Tips for Creating Reliable IP Connections for a range of IP information to assist with setting up IP services for your codecs.
- See Testing IP Network Connections to learn how you can test your IP connection.

Configure a New Connection

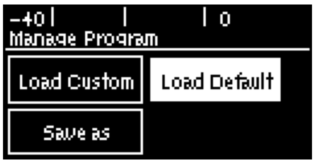
Configure the audio inputs and set levels and then use the following procedures to connect:

1. Press the **HOME** button to return to the **Home** screen.

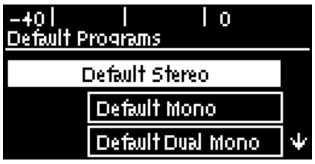
2. Navigate to **Cxns** and press the button.

3. To change the program type navigate to **Manage Programs** and press the button.

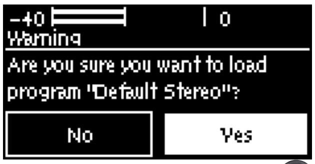
4. In this example we will navigate to **Load Default** and press the button to load a default program. Note: Select **Load Custom** to choose and load a previously saved custom program.



5. Select the preferred program type and press the button.

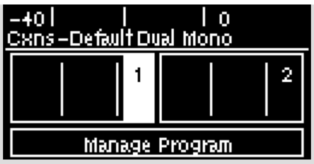


6. Confirm program selection.

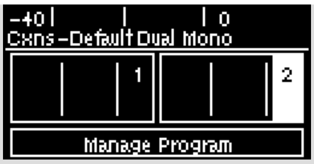


7. Navigate back to select connection **1** and press the button.

Important Note: When a **Default Dual Mono** program has been selected, both connections are displayed on the **Cxns** screen. Select connection **1** and press the button to configure connection **1** settings. Select connection **2** and press the button to configure connection **2** settings.

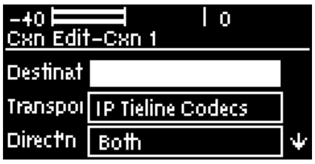


Connection 1 Selected



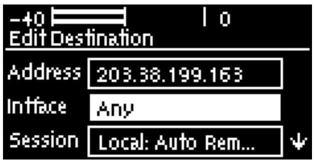
Connection 2 Selected

8. Select **Destination** and press the OK button.

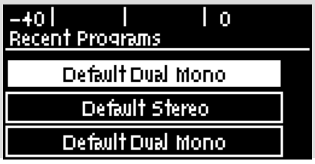


9. Select **Address** and press the OK button. Use the numeric **KEYPAD** to enter the IP address of the codec you want to dial, using the or buttons to enter the periods in the IP address. Use the **RETURN** button to delete characters already entered. Press the OK button to confirm it.

10. Select **Interface** to specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN1**) or **Secondary** (**LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.



Important Note: The codec remembers recently dialed addresses and programs like a cell-phone remembers recent calls. To view recent calls press the **CONNECT** button from the **Home** screen. The most recently dialed programs are listed first and you can select a program and press the OK button to reconnect it.



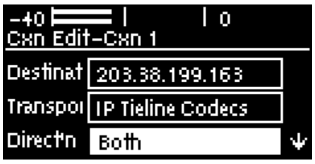
11. Navigate down to **Session** to adjust the session port setting or select **Audio** to adjust the audio port setting if required.

12. Press the **RETURN** button to return to the **Cxn Edit** screen and navigate down to adjust the **Algorithm** setting (default is Tieline Music and select your preferred sample rate

if displayed, and bit rate), **Jitter** setting (default is Auto, Best Compromise), and **Auto Reconnect** (default disabled) as required.

- See Selecting an Algorithm for recommended encoding settings.
- See Configuring the Jitter Buffer for information on configuring jitter settings. The default **Auto, Best Compromise** setting is a good starting point for most internet connections. Alternatively you can enter a fixed jitter buffer value in milliseconds (maximum 5000 ms).

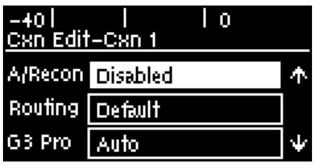
13. It is also possible to restrict the codec to encode or decode audio only to save bandwidth. Select **Direction** and press the button to adjust this setting.



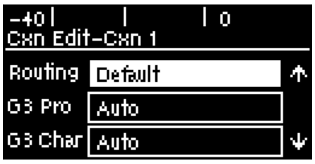
14. Navigate to **Data** to enable out-of-band RS232 data and activate rules employing relay reflection over a connection. This allows the codec to connect to external devices and send RS232-compatible data via the serial port on the codec's rear panel.

15. Navigate to **FEC** and press to configure local and remote FEC settings. Use the navigation buttons to select a preferred FEC percentage and press .

16. Select **A/Recon** to enable auto reconnect and press to toggle between **Enabled** and **Disabled**.

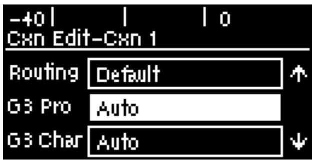


17. Navigate to **Routing** to configure routing options as per the table below:



Routing Type Options:	
Default IP	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic IP	Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed.
Line Hunt IP	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

18. Navigate to **G3 Profile** to configure profile settings when dialing a Tieline G3 Codec.



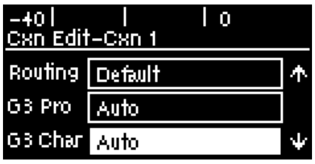
Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 or G6 codec.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. Note: This is overridden in a ViA codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. Note: When connecting in dual mono (2 x Mono Peer-to-Peer) mode to a G3 codec over IP both audio streams must encode using the same algorithm and sample rate or the G3 codec will not connect.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 or G6 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

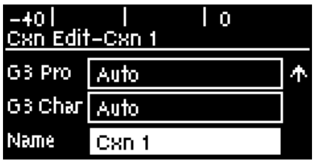
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

19. Navigate to **G3 Channel** when connecting to a G3 codec in dual mono mode. This setting lets you configure which G3 channel (encoder) is used when the G3 codec receives a call from the codec. E.g. **Channel 1** will route the incoming stream to Encoder 1 on the G3 codec and **Channel 2** will route the incoming stream to Encoder 2 on the G3 codec.

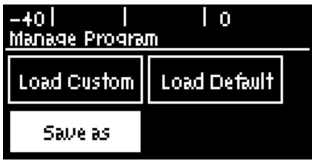


20. It is also possible to name each connection. Select **Name** and press to edit the default connection name.

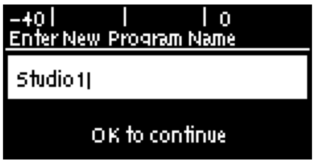


21. To save program settings navigate to the **Cxns** screen, select **Manage Program**, and press .

22. Select **Save as** and press .



23. Enter a program name using the **KEYPAD** and press .



24. To dial navigate to the **Cxns** screen, select a connection, and press the **CONNECT** button to dial an individual connection.

Select Connection and press CONNECT

Connection displayed when Connected

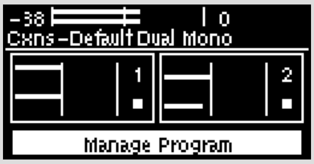
25. Press the **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup a connection.

Important Note: If a dual mono program supporting two audio streams is loaded, it is possible to dial and hangup each connection either individually, or simultaneously. To dial simultaneously from the **Cxns** screen:

- Select **Manage Program** and press the **CONNECT** button to dial both connections simultaneously.
- Select a connection and select **Connect All** to dial all connections.

Press the **DISCONNECT** button to disconnect both connections simultaneously when **Manage Program** is selected.

Press CONNECT to Dial both Connections

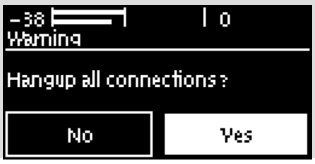


Press DISCONNECT to hangup 2 connections

It is also possible to redial all connections using the "Recent Programs" feature from the **Home** screen. Press the **CONNECT** button from the **Home** screen to display recently dialed programs.

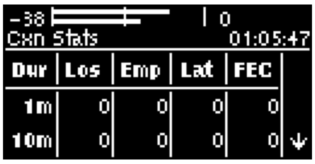
Select a program and then press the **CONNECT** button to dial both connections simultaneously. After connecting, the number of streams and connections is displayed on the **Home** screen.

Press the **DISCONNECT** button when on the **Home** screen to hangup both connections simultaneously.



26. After successfully connecting, navigate to a connection and press to display connection details. Use the down navigation button to view details like link quality, as well as bit-rate and jitter information. To negotiate higher bit-rates press F2 then **3** on the numeric **KEYPAD**; for lower bit-rates press F2 then **9**.

27. Navigate to **Status** and press to view connection statistics for IP packets sent over the connection.

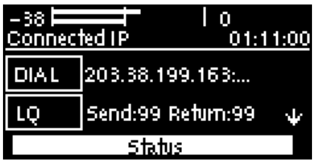


18.2 Monitoring IP Connections

To monitor IP connections:

1. Press the **HOME** button to return to the **Home** screen.
2. Navigate to **Cxns** and press the button.
3. Navigate to a connection and press to display connection details.

The IP address dialed and the **LQ** (link quality) is displayed on the screen and you can use the down navigation button to view the algorithm configured, the connection bit-rate, total bytes used and the jitter buffer latency.



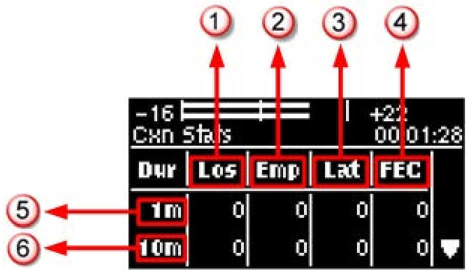
Link Quality (LQ) Readings

Send and return LQ numbers can help to determine whether a problem is occurring at one end of a connection or the other. For example, on an IP connection the **Return** reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the **Send** link quality reading represents the audio data being sent by the local codec (i.e. being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** LQ reading.

Important Note:
- The **Return** link quality reading is the same as the Local (**L**) setting displayed on a G3 codec.
- The **Send** link quality reading is the same as the Remote (**R**) setting displayed on a G3 codec.

Viewing Connection Statistics

Navigate to **Status** in the **Connected IP** screen and press the OK button to display the **Cxn Stats** (connection statistics) screen. This displays the performance of the codec while sending IP audio packets across the network. Analysis is historic and assessed over 60 seconds and 10 minutes of connection time.



	Feature	Description
1	Lost Packets	Packets sent that failed to arrive
2	Empty (Jitter Buffer)	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio
3	Late Packets	The number of packets that arrive late, i.e. after audio play out
4	FEC Packets	Indicates the number of forward error correction (FEC) packets that have been replaced when enabled in the codec
5	1 minute	Statistics listed for the last minute of network activity
6	10 minutes	Statistics for the last 10 minutes of network activity

Important Note: If the jitter buffer, FEC or the connection bit rate is changed, we recommend assessing a minute of recent connection performance in preference to 10 minutes of historical connection performance. 10 minutes of data will include connection settings which may no longer be relevant. 'Packet arrival history' is cleared when you hang up a connection.

Following is an analysis of possible causes and solutions for the packet analysis statistics displayed on the screen.

Packet Analysis	Displays	Possible Causes	Possible Solutions
Loss	Packets failed to arrive.	• LAN/WAN congestion • Unreliable ISPs • Unreliable networks • Unreliable IP hardware	• Renegotiate connection bit rate downwards • If link quality good add or increase FEC as required • Assess ISP's QOS if very bad performance
Empty	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio.	• High number of packets being lost or arriving late • Signal drop-outs using cell-phone networks • Renegotiation causes the jitter buffer reservoir to empty	• Once could be an anomaly – assess lost & late packets • If many lost packets network is unreliable – renegotiate bit rate and /or FEC down • If many late packets increase jitter buffer
Late	The number of packets that	• Network congestion	• Auto-jitter buffer will adjust automatically

	arrive late and after audio play out.	• Jitter Buffer depth is too low	• For manual jitter buffer settings increase jitter buffer depth 50-100 ms & reassess (if only a few packets arrive late over time, audio repairs will be automatic and may not require buffer changes).
FECd	Indicates the number of FEC repaired packets if FEC active.	• Packets have been lost or corrupted over the network	• Assess audio quality & the number of FEC repairs – if many packets are being 'lost' perhaps reduce FEC &/or renegotiate bit rate down.

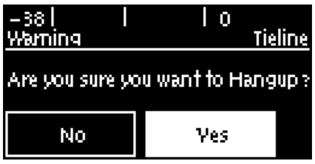
18.3 Load and Dial Custom Programs

Custom programs stored on the codec are simple to load and dial from the codec front panel.

1. Press the **HOME** button to return to the **Home** screen.
2. Select **Programs** and press the button.
3. Select and **Load** a program and then press the **CONNECT** button to dial.

18.4 Disconnecting a Connection

1. Press the **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup a connection.
2. Use the navigation buttons to select **Yes** and press the **DISCONNECT** button or the button to confirm the connection hangup.



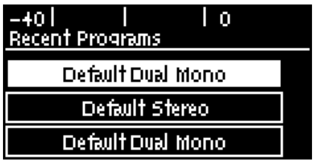
18.5 Redialing a Connection

After configuring all settings, navigate to the **Cxns** screen, select a connection, and press the **CONNECT** button to dial an individual connection.

Select Connection and press CONNECT

Connection displayed when Connected

The codec remembers recently dialed addresses and programs like a cell-phone remembers recent calls. To view these addresses press the **CONNECT** button from the **Home** screen. The most recently dialed programs are listed first and you can select a program and press the button to reconnect it.



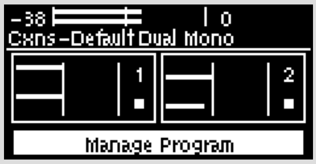
Press the **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup a connection.

Important Note: If a dual mono program supporting two audio streams is loaded, it is possible to dial and hangup each connection either individually, or simultaneously. To dial simultaneously from the **Cxns** screen:

- Select **Manage Program** and press the **CONNECT** button to dial both connections simultaneously.
- Select a connection and select **Connect All** to dial all connections.

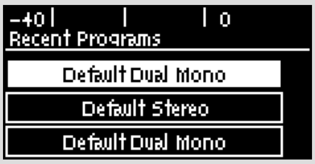
Press the **DISCONNECT** button to hangup both connections simultaneously when **Manage Program** is selected.

Press CONNECT to Dial both Connections



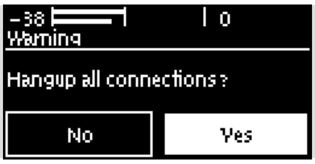
Press DISCONNECT to hangup 2 connections

It is also possible to redial all connections using the "Recent Programs" feature from the **Home** screen. Press the **CONNECT** button from the **Home** screen to display recently dialed programs.



Select a program and then press the **CONNECT** button to dial both connections simultaneously. After connecting, the number of streams and connections is displayed on the **Home** screen.

Press the **DISCONNECT** button when on the **Home** screen to hangup both connections simultaneously.



18.6 Configuring Auto Reconnect

Auto Reconnect is disabled by default. When enabled, the dialing codec attempts to reconnect if data packets are temporarily lost over an IP connection. To adjust the setting:

1. Press the **HOME** button to return to the **Home** screen.
2. Select **Cxns** and press the button.
3. Select a connection and press the button
4. Navigate to **A/Recon** and press to toggle between **Enabled** and **Disabled**.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the web-GUI. This setting should only be configured on the dialing codec.

18.7 Speed Dialing Connections

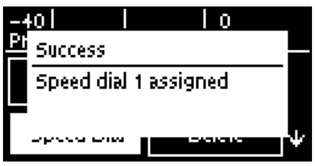
Assigning Speed Dial Numbers

There are two methods of assigning speed dial numbers to saved programs:

Assigning Speed Dials via the Programs Menu

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.
3. Navigate to the program to which you are assigning a speed dial number, then press the button.
4. Navigate to **Speed Dial** and press the button.

5. Navigate to the speed dial number you want to assign to the selected program and press the button.
6. A confirmation message is displayed.

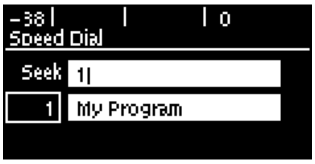


Assigning and Viewing Speed Dials via the Speed Dial Menu

New speed dials can be created and existing speed dials can be viewed in the **Speed Dial** menu. To add a new speed dial:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Speed Dial** and press the button.

3. Navigate to an available speed dial number and press the button.
4. Navigate to a program in the **Program List** and press the button to select and assign a program.



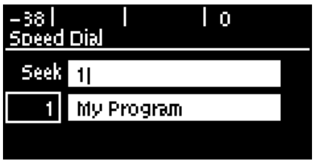
Speed Dialing

There are two ways to speed dial.

Option 1: Press and hold a speed dial number on the **KEYPAD** for two seconds and then the codec will automatically dial when you release your finger from the keypad. Note: this is only available for buttons 0-9. Use the following procedure to dial speed dial number 10 or higher.

Option 2:

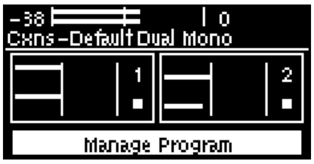
1. Press the **HOME** button to return to the **Home** screen.
2. Use the numeric **KEYPAD** to enter the speed dial number.
3. When the **Speed Dial** screen appears, press the button or the **CONNECT** button to connect.



18.8 Dial/Disconnect Multiple Connections

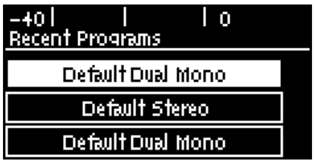
If a dual mono program supporting two audio streams is loaded, it is possible to dial and hangup each connection either individually, or simultaneously. To dial simultaneously from the **Cxns** screen select **Manage Program** and press the **CONNECT** button to dial both connections simultaneously. Press the **DISCONNECT** button to hangup both connections simultaneously when **Manage Program** is selected.

Press CONNECT to Dial both Connections



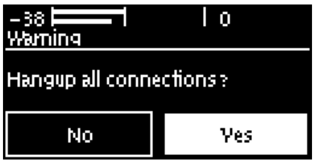
Press DISCONNECT to hangup 2 connections

It is also possible to redial all connections using the "Recent Programs" feature from the **Home** screen. Press the **CONNECT** button from the **Home** screen to display recently dialed programs.



Select a program and then press the **CONNECT** button to dial both connections simultaneously. After connecting, the number of streams and connections is displayed on the **Home** screen.

Press the **DISCONNECT** button to hangup both connections simultaneously.



Multi-unicast Programs

Multi-unicast programs allow you to simultaneously transmit a mono or stereo audio stream to up to 6 destination codecs (Bridge-IT II), or up to 10 destinations (Bridge-IT XTRA II). Multi-unicast programs can only be created using the HTML5 Toolbox web-GUI. There are two ways to simultaneously dial multiple IP audio stream connections multi-unicast programs:

1. Load the program into the codec via the front panel and press the **CONNECT** button to dial.
2. Connect to the codec using the Toolbox HTML web-GUI and use the **Program Manager** or **Connections panel** to load the program and then use the **Connections panel** to connect.

Dialing Multi-unicast Connections via the Keypad

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
3. Use the up and down navigation buttons to select the multi-unicast program you want to load and then press the **CONNECT** button to make a connection. Note: It is also possible to redial the connection.

Disconnect All Multi-unicast Connections

1. Press the red **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup all connections.
2. Use the right navigation button to select **Yes** and press the **DISCONNECT** button or the button to confirm the disconnection.

Disconnecting Individual Multi-unicast Connections

It is only possible to disconnect individual connections via the **Connections panel** in the dialing codec's HTML5 Toolbox web-GUI.

18.9 Creating a Multicast Server Program

Two different types of multicast programs need to be created when multicasting:

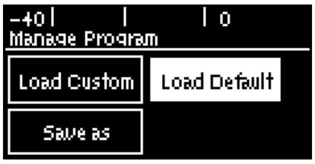
- A multicast server program is used by the broadcasting codec to send multicast IP packets to multicast routers on a network.
- A multicast client program is used by codecs to receive multicast IP audio packets.

Important Notes:
- You cannot edit a program when it is currently loaded in the codec.
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not be able to join multicast streaming sessions.
- There is no jitter buffer setting in a multicast server program because it is an encode only program and never receives audio packets.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
- It is not possible for a G3 codec to receive multicast IP audio streams.
- See HTML5 Toolbox web-GUI documentation for more detailed information about Configuring Multicast Server Programs or Configuring Multicast Client Programs.

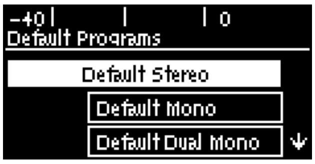
1. Press the **HOME** button to return to the **Home** screen.
2. Navigate to **Cxns** and press the button.

3. To change the program type navigate to **Manage Programs** and press the button.

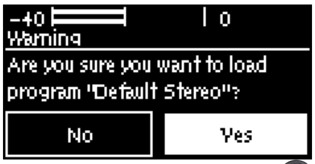
4. In this example we will navigate to **Load Default** and press the OK button.



5. Select the preferred program type and press the OK button.

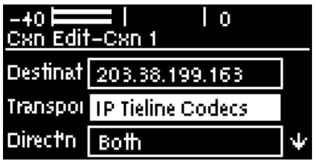


6. Confirm program selection.



7. Navigate back to select connection **1** and press the OK button.

8. Select **Transport** and press the OK button.



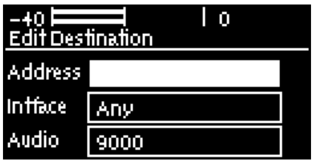
9. Select **IP** and press the OK button.

10. Select **Sessionless** and press the OK button.

11. Select **Mcast-S** and press the OK button.

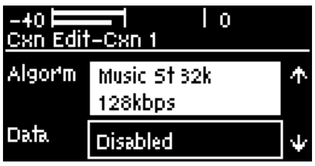
12. Navigate to **Destination** and press the button.

13. Select address and press the button. Use the **RETURN** button to delete any numbers already entered, then use the numeric **KEYPAD** to enter the multicast IP address you want to dial, using the or buttons to enter the periods in the IP address. Then press the button. Note: The same multicast address and audio port must be used for both the server and client programs.

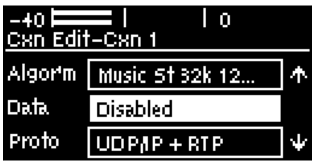


14. Select **Audio** to adjust the remote audio port if required, and select **Interface** to select a preferred transport interface if required. Then press the **RETURN** button to load the **Cxn Edit** screen.

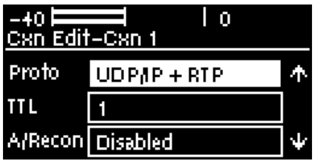
15. Navigate to **Algor'm** (algorithm) and press to configure encoding settings.



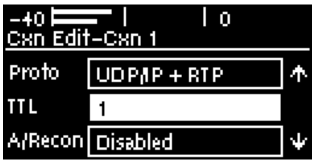
16. Navigate to **Data** and press to enable or disable auxiliary data. If auxiliary data is enabled the audio stream will not be RFC-compliant.



17. Navigate to **Proto** to select the audio protocol. Select **UDP/IP +RTP** for RFC compliant IP streaming.

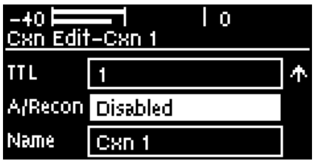


18. Navigate to **TTL** to enter the IP Time-To-Live value and press .



Important Note: Time-to-Live (TTL) is a value you can configure to set a finite life for data packets sent by the codec and the value to use is dependent upon your network infrastructure. Please consult your network administrator if you are unsure about how to configure this setting. Setting a TTL value avoids situations where packets can keep circulating through routers causing network congestion. It sets the maximum number of router hops allowable for multicast data packets. In most situations the default value of **1** is used, to ensure packets are sent through a single LAN router and not over multiple router hops and networks.

19. Select **A/Recon** to enable auto reconnect and press to toggle between **Enabled** and **Disabled**.



20. Use the **RETURN** button to navigate back to the **Cxns** screen and press the **CONNECT** button to dial the connection.

Important Note: To save the program from the **Cxns** screen select **Manage Program** and press . Then select **Save as** and press . Enter a program name using the **KEYPAD** and press .

Connecting a Multicast Program

After you have created multicast server and client programs you are ready to dial multicast connections. First select, load and dial the multicast server program on the server codec:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.
3. Select a program and press the button.
4. Navigate to **Load** and press the button to load the custom multicast program.
5. Press the **CONNECT** button to dial the connection.
6. Select and load the multicast client program on each of the multicast client codecs and dial the multicast IP address to begin receiving multicast audio packets.

18.10 Creating a Multicast Client Program

Use the procedure which follows to configure a multicast client program and allow the codec to receive multicast IP audio packets.

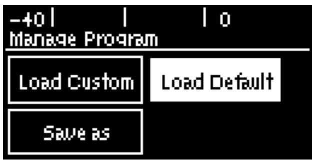
Important Notes:

- You cannot edit a program when it is currently loaded in the codec.
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not be able to join multicast streaming sessions.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- To copy multicast client programs onto multiple codecs see Backup and Restore Functions.
- See HTML5 Toolbox web-GUI documentation for more detailed information about Configuring Multicast Server Programs or Configuring Multicast Client Programs.

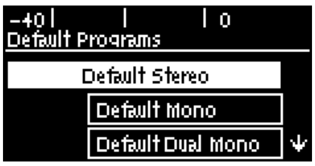
1. Press the **HOME** button to return to the **Home** screen.
2. Navigate to **Cxns** and press the button.

3. To change the program type navigate to **Manage Programs** and press the button.

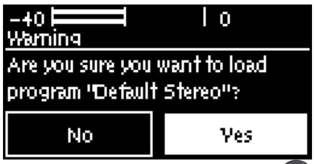
4. In this example we will navigate to **Load Default** and press the button.



5. Select the preferred program type and press the button.

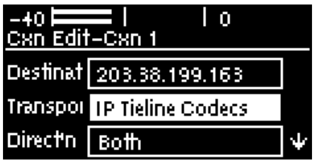


6. Confirm program selection.



7. Navigate back to select connection **1** and press the button.

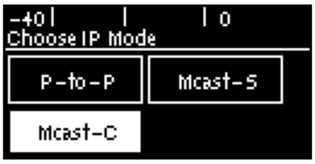
8. Select **Transport** and press the button.



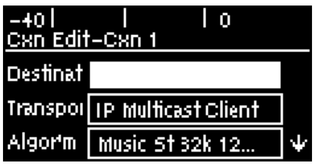
9. Select **IP** and press the button.

10. Select **Sessionless** and press the button.

11. Select **Mcast-C** and press the button.

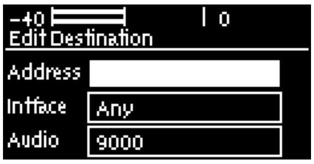


12. Navigate to **Destination** and press the button.



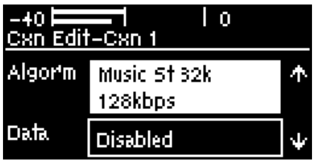
13. Select address and press the button. Use the **RETURN** button to delete any numbers already entered, then use the numeric **KEYPAD** to enter the multicast IP address

you want to dial, using the *· or #• buttons to enter the periods in the IP address. Then press the OK button. Note: The same multicast address and audio port must be used for both the server and client programs.



14. Select **Audio** to adjust the remote audio port if required, and select **Interface** to select a preferred transport interface if required. Then press the **RETURN** button to load the **Cxn Edit** screen.

15. Navigate to **Algor'm** (algorithm) and press OK to configure encoding settings.

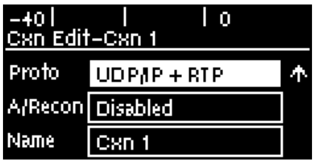


16. Click to select **Jitter** and configure the jitter buffer. Select either **Auto Jitter Adapt** or **Fixed Buffer Level** , then and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select, then press the OK button.

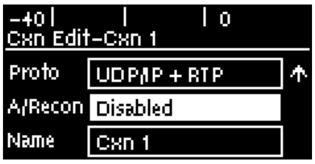
Important Notes: Automatic or fixed jitter buffer settings can be adjusted on individual client codecs. There is no jitter buffer setting on the server codec because it never receives audio packets.

17. Navigate to **Data** and press OK to enable or disable auxiliary data. If auxiliary data is enabled the audio stream will not be RFC-compliant.

18. Navigate to **Proto** to select the audio protocol. Select **UDP/IP +RTP** for RFC compliant IP streaming.



19. Select **A/Recon** to enable auto reconnect and press OK to toggle between **Enabled** and **Disabled**.



20. Use the **RETURN** button to navigate back to the **Cxns** screen and press the **CONNECT** button to dial the connection.

Important Note: To save the program from the **Cxns** screen select **Manage Program** and press OK. Then select **Save as** and press OK. Enter a program name using the **KEYPAD** and press OK.

Connecting a Multicast Client Program

After you have created multicast server and client programs you are ready to dial multicast connections. First, select the multicast server program on the server codec and dial to connect. Select and load the multicast client program on each of the multicast client codecs and dial the multicast IP address to begin receiving multicast audio packets.

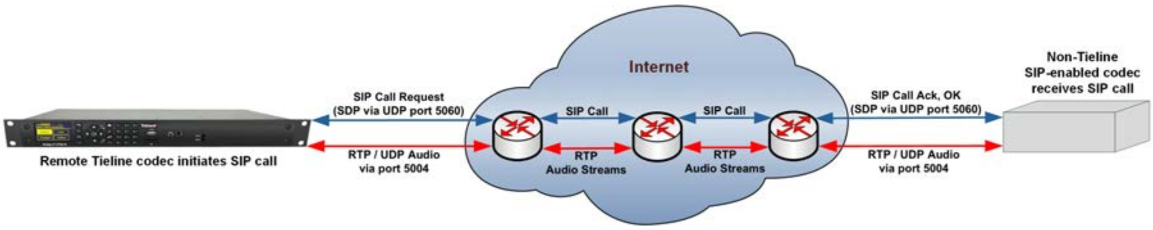
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
3. Select a program and press the OK button.
4. Navigate to **Load** and press the OK button to load the custom multicast client program.
5. Press the **CONNECT** button to dial the connection.

18.11 About SIP

SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices. The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs. SIP is also a useful way of dialing another device and locating it easily. This task is usually performed by SIP servers, which communicate between SIP-compliant devices to set up a call. SIP connections can be made in two ways; registered or unregistered.

Unregistered Peer-to-Peer SIP Connections

Codecs don't need to be registered to a SIP server to dial peer-to-peer SIP connections. An unregistered SIP peer-to-peer connection involves two codecs connecting to each other directly using an IP address, as you would for a standard Tieline IP call. The difference is that a Tieline IP call uses proprietary Tieline session data to negotiate call parameters (e.g. algorithm and bit rate) when a call is established, whereas a peer-to-peer SIP connection uses Session Description Protocol (SDP) for this purpose. SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting dissimilar devices and is used when connecting Tieline codecs to non-Tieline devices.



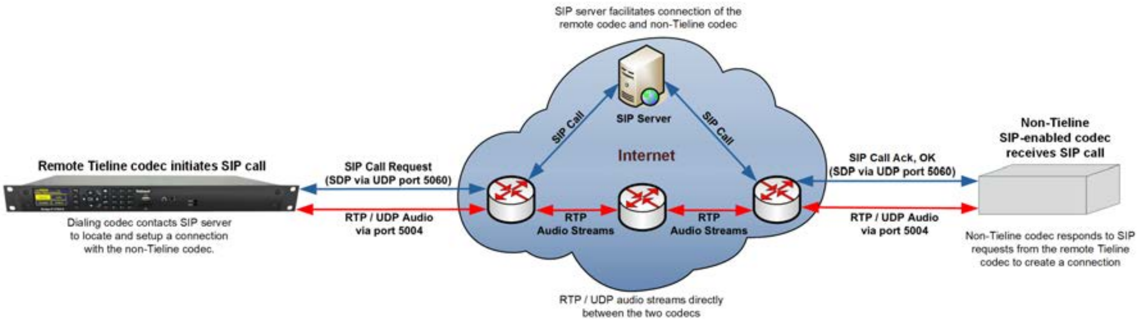
There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP and SDP is used for. The second stage is when data transfer occurs and this is left to the other protocols such as RTP/UDP to stream audio data. SDP works with a number of other protocols, to deliver the following functions when connecting devices over SIP:

- Establish a codec's location.
- Determine the availability of a codec.
- Negotiate the features to be used during a call, e.g. the algorithm and bit rate.
- Provide call management of participants.
- Adjust session management features while a call is in progress (e.g. termination and transfer of calls).

All the mandatory EBU N/ACIP 3326 algorithms are supported in the codec, including G.711, G.722, MPEG-1 Layer 2 and 16 bit PCM, as well as optional algorithms including Opus, LC-AAC, AAC-LD, HE-AACv2 and aptX Enhanced.

SIP Server Connections

The benefit of using a SIP server to connect is that any device can be 'discovered' via its SIP server registration. This is particularly useful if a codec is being used in multiple locations with IP addresses that are DHCP assigned. These DHCP addresses are unreliable and are not recommended for live broadcast connections. As long as your codec and the device you are dialing are both registered to a SIP server you can connect by simply dialing the destination SIP address.



Some SIP servers route RTP audio through the SIP server as well and Tieline recommends avoiding this type of server whenever possible. Otherwise you will be reliant on the SIP server for streaming broadcast audio packets and most servers are not designed for mission critical packet streaming. To dial a codec via a SIP server requires:

1. Both devices to be registered with separate SIP accounts.
2. Both codecs configured to operate in SIP mode.
3. The IP address of the SIP server.
4. An IT administrator to open UDP port 5060 to enable SIP traffic, as well as UDP audio port 5004.

A SIP server administrator should be able to provide the following details to enable SIP registration of a device:

- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port

Getting Started with SIP

To dial over SIP peer-to-peer without using a SIP server see Dialing SIP Connections. To dial over SIP using a SIP Server you will need to:

1. Register the codec to a SIP server using SIP account credentials.
2. Configure a SIP interface in the codec. Note: The **SIP1** or **SIP2** interface will include the proxy and port settings, as well as the selected IP interface used to make the connection, e.g. **LAN1**.

Important Notes:

- The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**.
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.
- Tieline G3 codecs do not support connections using algorithms like AAC, aptX Enhanced and Opus and will default to MPEG Layer 2 if an incoming call is configured to use these algorithms.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level in the codec. To do this press the **HOME** button to return to the **Home** screen. Then select **Audio > Ref Level > Tieline G3**. In addition, configure the following on the G3 codec prior to dialing:
    - Select either a mono or stereo profile
    - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
    - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

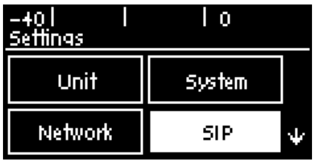
18.11.1 Configuring SIP Interfaces

The codec supports dialing over two SIP interfaces simultaneously.

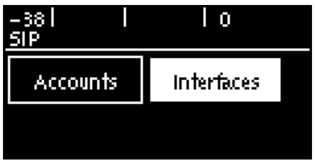
Important Notes:
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN1** by default, which is mapped to the **Primary** interface by default.
3. **SIP2** is configured to use **LAN2** by default, which is mapped to the **Secondary** interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN1** or **LAN2**.
5. **SIP1** and **SIP2** can however each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface is enabled.
7. Enter a public IP address in the **Public IP Address** text box if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP Address** if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.

To configure **SIP1** or **SIP2**:

1. Press the the **SETTINGS** button, then navigate to **SIP** and press the button.



2. Navigate to **Interfaces** and press the button.

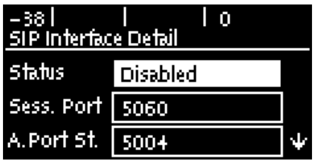


3. Select either **SIP1** or **SIP2** and press to view and edit settings. Note: A SIP interface must be disabled before settings can be edited.

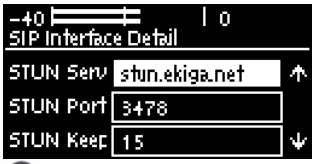
4. Select **Disable** to disable the interface before editing settings.

5. Navigate to **Edit** and press the OK button. Note: The SIP interface **Wizard** can also be used to navigate through each screen in turn automatically.

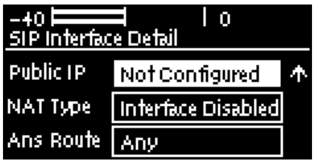
6. Navigate to each field in turn and press the OK button to edit SIP interface settings as required.



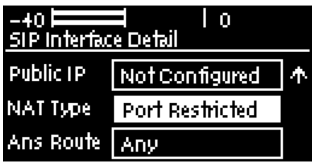
7. Select **STUN Server** and press the OK button to enter server details if using STUN (Session Traversal of User Datagram Protocol). Note: UDP Port 3478 is the default port assigned.



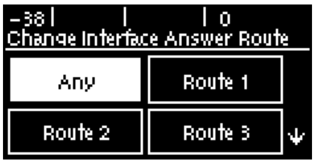
8. Select **Public IP** and press the OK button if you want to dial over SIP from behind a firewall. Enter the public IP address and then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.



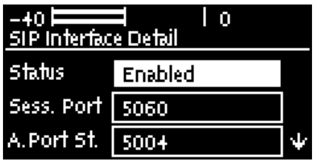
9. The **NAT Type** used by the firewall is updated if STUN has been enabled and the **NAT Type** can be determined. Note: the SIP interface must be enabled to view this.



10. Select **Ans Route** and use the **Change Interface Answer Route** screen to route calls using this SIP interface to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded program. If the default value **Any** is used then a call will be routed to an audio stream on a first-come-first-served basis in a multi-stream program.



11. Navigate back up to **Status** and press the button to **Enable** the interface.



12. It is also possible to **Enable** the interface in the **Modify SIP Interface** menu. Press the button to enable the interface so that it can be used to make a SIP call.

18.11.2 Configuring SIP Accounts

Getting Started

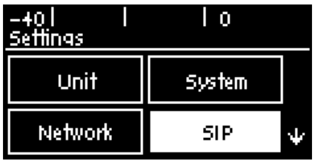
Up to 6 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, select the SIP server to which you will register your codec. On a LAN this may be your own server, or it could be one of the many internet servers available. When you register an account with a SIP server you will be provided with:

- The SIP registrar and Proxy Server details.
- An authorized user or username (often the same as a SIP number).
- A password.
- Domain details.
- Realm details (in most cases leave blank).
- Registration Timeout (this shouldn't need to be adjusted from the default setting).

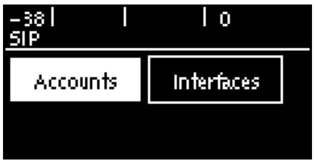
Important Notes:
- In most situations it is best to configure a SIP account when the codec is configured with a public IP address.
- Each SIP account can only be mapped to a single SIP interface, i.e. **SIP1** (**LAN1**) or **SIP 2** (**LAN2**).
- Up to 6 SIP accounts can be added to the codec.
- It is also possible to add and register a SIP account to your codec using the HTML5 Toolbox Web-GUI.

Adding a SIP Account

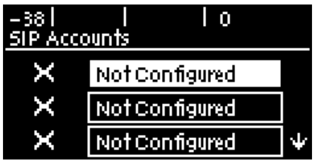
1. Press the the **SETTINGS** button, then navigate to **SIP** and press the button.



2. Navigate to **Accounts** and press the button.

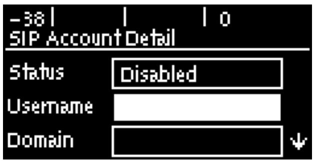


3. Select an account which is **Not Configured** and press .

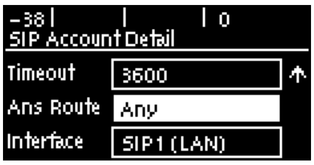


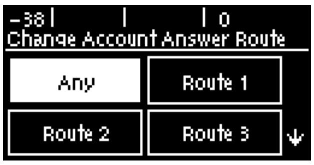
4. Select **Edit** to enter SIP server account details, then press . Note: The SIP accounts **Wizard** can also be used to navigate through each screen in turn automatically.

5. Navigate to each field in turn and press the button to enter SIP account credentials.



6. Select **Ans Route** and use the **Change Account Answer Route** screen to route incoming calls to this SIP account to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded program. If the default value **Any** is used then a call will be routed to an audio stream on a first-come-first-served basis in a multi-stream program.





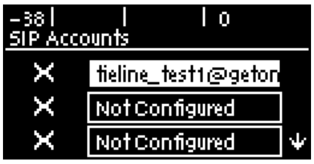
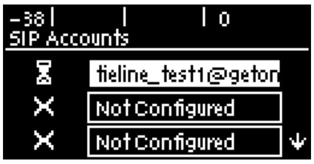
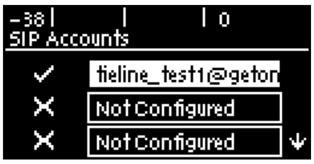
7. After completing account configuration, navigate back up to **Status** and press the button to **Enable** the account. You can also select **Enable** in the **Modify SIP Account** menu and press the button to enable the SIP account and register it with the SIP server.

8. To confirm the account has been registered successfully, verify the account has a "tick" next to it in the **SIP Accounts** menu.

Important Note: Once enabled, the SIP account can be used when creating a new SIP connection.

Confirming Account Registration

There are three symbols displayed on the screen next to an account which indicate SIP account registration status.

Symbol	Description
	Cross symbol indicates the account is not yet registered.
	Hourglass symbol indicates account registration is currently being attempted.
	Tick symbol indicates the account is registered to a SIP server.

Troubleshooting SIP Registration

If a SIP account is not being registered please check the following:

1. Confirm all account registration information has been entered correctly.
2. Confirm the SIP **Interface** (**SIP1** or **SIP2**) configured in the account is enabled.
3. Verify that the interface selected in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **LAN1** or **LAN2**.

18.12 Dialing SIP Connections

Dialing SIP Connections

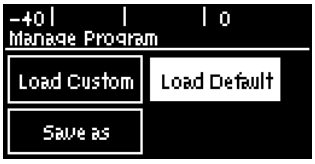
SIP can be used to make direct peer-to-peer calls to different brands of IP codecs with public IP addresses, or between two codecs over a LAN which do not pass through firewalls. Peer-to-peer SIP calls are often used to connect to other brands of codecs and perform call and session management tasks. Peer-to-peer SIP calls between two codecs are detected automatically and require no special pre-programming.

To make a peer-to-peer SIP call between codecs we recommend both codecs use public IP addresses. Find out the IP address of the codec being dialed and configure each codec with a compatible algorithm and sample rate etc. If the remote codec has a private IP address then it should be configured for port forwarding and should dial the public IP address at the studio.

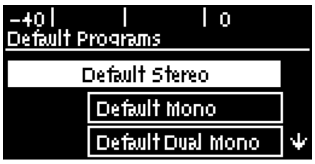
1. Press the **HOME** button to return to the **Home** screen.
2. Navigate to **Cxns** and press the button.

3. To change the program type navigate to **Manage Programs** and press the button.

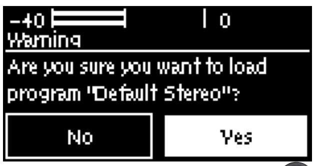
4. In this example we will navigate to **Load Default** and press the button.



5. Select the preferred program type and press the button.

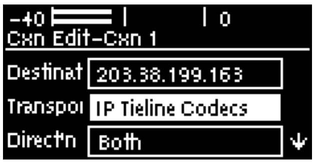


6. Confirm program selection.



7. Navigate back to select connection **1** and press the button.

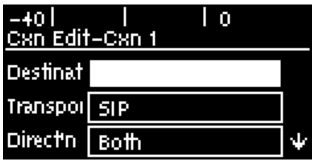
8. Select **Transport** and press the button.



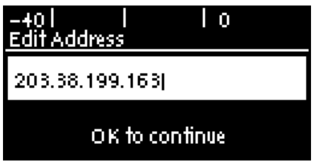
9. Select **IP** and press the button.

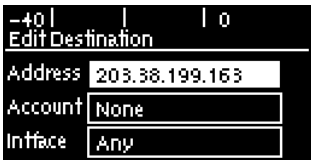
10. Select **SIP** and press the button.

11. Navigate to **Destination** and press the button.



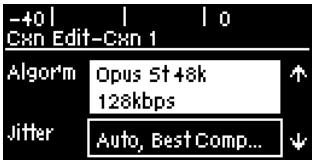
12. Select **Address** and press the button. Use the **RETURN** button to delete any numbers already entered, then use the numeric **KEYPAD** to enter the IP address you want to dial, using the or buttons to enter the periods in the IP address. Then press the button to continue.





13. Select **Interface** to select a preferred transport interface if required. Note: If SIP 1 or SIP2 are not enabled these interfaces will not be displayed. To enable these interfaces select **Settings > SIP > Interfaces > SIP1/SIP2** and press the button to enable these interfaces.

14. Press the **RETURN** button to load the **Cxn Edit** screen. Navigate to **Algor'm** (algorithm) and press to configure encoding settings. The default setting is Opus (select a preferred encoding option, sample rate - if displayed - and bit rate), **Jitter** setting (default is Auto, Best Compromise), and **Auto Reconnect** (default disabled) as required.
    - See Selecting an Algorithm for recommended encoding settings.
    - See Configuring the Jitter Buffer for information on configuring jitter settings. The default **Auto, Best Compromise** setting is a good starting point for most internet connections. Alternatively you can enter a fixed jitter buffer value in milliseconds (maximum 5000 ms).



15. Use the **RETURN** button to navigate back to the **Cxns** screen and press the **CONNECT** button to dial the connection.

Important Note: To save the program from the **Cxns** screen select **Manage Program** and press . Then select **Save as** and press . Enter a program name using the **KEYPAD** and press .

Dialing SIP Addresses

Use the **KEYPAD** to enter any combination of alpha-numeric characters in the SIP address of the codec you want to dial. Use the *. or # buttons to enter the periods in the SIP address and use the **RETURN** button to delete any numbers or characters already entered.

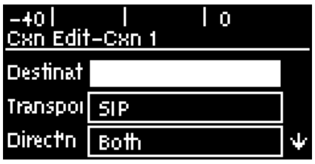
Dialing Using a SIP Account

To dial using a SIP account you first need to add a SIP account and register it to the codec. See Configuring SIP Accounts for more info.

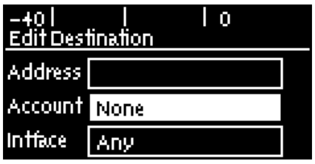
1. Press the **HOME** button to return to the **Home** screen.
2. Navigate to **Cxns** and press the button.

3. Select connection **1** and press the button.

4. Navigate to **Destination** and press the OK button.



5. Navigate to **Account** and press the OK button.



6. Navigate to a registered SIP account and press the OK button.

7. The SIP **Account** is now selected and the codec will attempt to establish the connection via a SIP server.

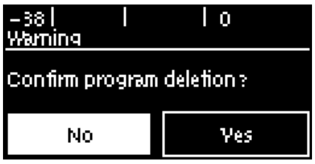
Important Notes:
- See Configuring SIP Accounts for instructions on adding SIP accounts.

It is also possible to configure SIP programs using the Toolbox web-GUI.

18.13 Deleting Programs

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
3. Navigate to the program you want to delete and press the OK button.
4. Navigate to **Delete** and press the OK button.

5. Confirm program deletion and press the OK button.



18.14 Backup Options

Tieline Audio Codec Backup Features			
Backup Option	**Transport: IP**	**Time Required to Respond**	**How to Enable**
SmartStream PLUS	Concurrent packet stream sent; codec detects IP packet loss or delayed packets	No time delay - simultaneous redundant streaming	Enabled in dialing codec program; configures local decoding, or remote decoding via session data
Fuse-IP	Multiple IP interfaces are bonded and data is shared across interfaces	No time delay if interface is lost; immediately adjusts to existing network data capacity	Configure multiple IP interfaces and create a Fuse-IP "Tunnel" prior to dialing destination codec
On-demand Failover	Codec detects loss of data or connection and redials the backup connection	User configurable detection parameters during program configuration. Delay is equal to detection time plus the time required to dial the alternative connection	Dialing codec program monitors streaming and manages failover
Output Audio Source: File Playback	All transports (Note: enabled by loss of decodable audio or connection problem)	User configurable detection parameters during program configuration. Delay is equal to detection time and audio threshold specified.	Enabled in a codec program.
Output Audio Source: HTTP	Connect to an Icecast media server streaming from a specified URL	User configurable detection parameters during program configuration. Delay is equal to detection time and audio threshold specified.	Enabled in a codec program.
Output Audio Source: Input	Input audio is looped to the physical codec outputs.	User configurable detection parameters during program configuration. Delay is	Enabled in a codec program.

		equal to detection time and audio threshold specified.	
FEC (Tieline FEC or RFC compliant FEC)	Decoding codec detects IP packet loss or delayed packets. Note: Only Tieline Music and Music PLUS can be used for Tieline FEC.	No failover time delay as packet replacement occurs in real-time. Note: RFC compliant FEC can be configured to send 100% FEC at a specified delay if desired.	• Dialing codec using Tieline session data configures local and remote FEC settings via session data transfer when connecting, or • In sessionless mode FEC configuration is as per RFC compliant settings configured in the local and remote codecs.
Auto Reconnect	All transports; codec will redial continuously to try and reconnect	Immediately redials after loss of IP stream detected	Enabled in dialing codec program

Important Note: Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.

SmartStream PLUS Redundant IP Streaming

Tieline's proprietary SmartStream PLUS IP technology ensures you're always on the air. There are three levels to SmartStream PLUS IP streaming.

1. The codec can stream simultaneous redundant data streams and deliver hitless redundancy by switching back and forth, without loss of audio, from the nominated primary data link to the backup link if one fails and then subsequently recovers. Use IP links from different IP network providers for optimal redundancy over mission critical connections.
2. Second, when multiple redundant audio streams are sent, the decoding codec automatically reconstructs audio into a single stream on a first packet arrived basis, to minimize program latency and ensure audio integrity.
3. Third, SmartStream features automated jitter buffer management and Forward Error Correction (FEC) and these advanced network management tools deliver uncompromising audio quality, while dynamically responding to variable conditions over unmanaged IP networks like the internet.

These combined measures ensure Tieline is capable of offering a rock solid IP audio solution for distributing IP audio economically and efficiently across broadcast networks. See the procedures for configuring different programs for more configuration details.

Fuse-IP

Tieline's proprietary Fuse-IP data aggregation technology uses a point-to-point tunnel between two codecs to bond multiple IP interfaces (peers). See Configure Fuse-IP Bonding for more info.

On-Demand Failover

On-demand failover requires configuration of a primary connection and an on-demand backup connection. On-demand failover is activated when the dialing codec program detects the loss of the primary connection, or if audio streaming ceases. The backup connection is then dialed to replace the primary connection. For details on configuring backup connections using failover see Configure Mono or Stereo Peer-to-Peer Programs.

Forward Error Correction (FEC)

There are two modes of Forward Error Correction (FEC) available in the codec:

1. Tieline FEC.
2. RFC compliant FEC (Sessionless connections only).

FEC transmits a secondary stream of audio data packets over a single connection. If packets are lost or corrupted over the connection then replacement FEC data packets can be substituted to replace them. Note: FEC should not be confused with SmartStream PLUS. FEC packets are sent over a single data stream connection, whereas SmartStream PLUS redundant streaming transmits completely redundant audio data streams. FEC is also a subset of features within SmartStream PLUS, which means you can configure SmartStream PLUS redundant data streams and also configure FEC on each of these data streams. For more info on FEC see Configuring Forward Error Correction.

Auto Reconnect

Auto reconnect can be enabled when configuring a codec program and is the simplest form of connection backup whereby the codec will redial a lost connection continuously until it is either:

- Re-established, or
- Dialing is manually stopped.

SD Card File Backup Connections

The codec features an SD/SDHC card slot for automatic backup to MP2 or MP3 recordings if an IP connection is interrupted. Backup connections are configured using the web-GUI and this is outlined in Configure Mono or Stereo Peer-to-Peer Programs. File backup is automatic and occurs:

1. According to the silence threshold parameters configured for audio file backup, or
2. Immediately if a connection to another codec is lost.

After SD/SDHC file backup is activated the audio file plays continuously in loop mode until a backup connection is dialed and connects, or the primary connection is restored. The codec **Home** screen indicates failover to the backup SD/SDHC card has occurred by displaying **(F)** in the **Cxns** display. Playback continues during reconnection attempts and ceases when a connection is restored. See SDHC Card File Playback for more details.

Important Notes for File Playback:

- File playback will occur automatically if the silence threshold parameters are breached; if the codec is not connected for any reason file playback will commence. To stop file playback open the **Connections panel** in the web-GUI, click to select the file playback connection, then click **Disconnect**.
- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB. Note: The Windows built-in formatting tool cannot format a SD card larger than 32GB with the FAT32 file system.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.

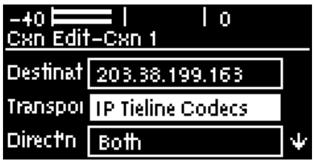
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
- The card can be inserted or removed at any time as long as the codec is not already playing audio in failover mode. Avoid removing the card while audio is playing or it will result in poor audio quality. If it is removed accidentally you must reboot the codec to ensure backup audio will continue to operate reliably.

18.15 SDHC Card File Playback

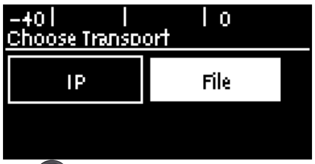
SDHC card files can be played back using the codec front panel controls.

1. Press the **HOME** button to return to the **Home** screen.
2. Select **Cxns** and press the button.
3. Select connection **1** and press the button.

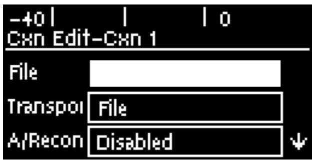
4. Select **Transport** and press the button.



5. Select **File** and press the button.



6. Navigate to **File** and press the button.



7. Use the navigation buttons to select a file and press the button.

8. Use the **RETURN** button to navigate back to the **Cxns** screen and press the **CONNECT** button to play the selected file.
9. Press the red **DISCONNECT** button on the numeric **KEYPAD** to stop file playback.

Important Notes for SDHC Card File Playback:

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB. Note: The Windows built-in formatting tool cannot format a SD card larger than 32GB with the FAT32 file system.

- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
- File playback will occur automatically if the silence threshold parameters are breached. To manage file playback open the Connections panel in the web-GUI.

18.16 Lock or Unlock a Program in the Codec

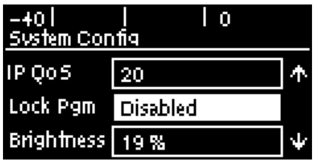
By default Tieline codecs will attempt to answer a call from another codec whenever possible. For example, if a mono program is loaded in the codec and a stereo incoming call is detected, the codec will adjust and load a compatible answering program.

It is also possible to lock a loaded custom program in a codec to ensure a program with your preferred settings is not unloaded when a codec dials in. Incoming calls are generally down or up sampled to accommodate a locked program where possible. Scenarios in which you may wish to lock a program in the codec include:

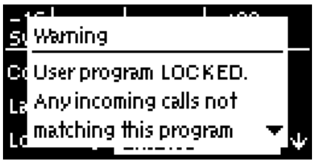
1. Locking a dialing program to ensure the codec only dials and never answers an incoming call.
2. Locking an answering program to ensure an incoming codec call is not allowed to:
   - Unload the current codec program, e.g. mono or stereo.
   - Change the preferred local site settings like the jitter buffer and FEC configuration etc.

Incoming calls to an answering codec with a locked program can still specify different connection parameters such as algorithm preferences and bit rates via session data.

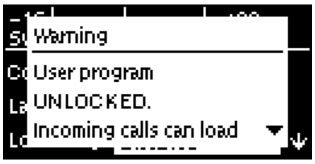
1. Press the **HOME** button to return to the **Home** screen.
2. Select **Settings** and press .
3. Navigate to **System** and press .
4. Navigate to **Lock Pgm** (lock program) and press to toggle between **Enabled** and **Disabled**.



5. When program lock is **Enabled** a warning message confirms program status.



6. When program lock is **Disabled** a warning message confirms incoming calls may load any supported factory program.



7. Press the **RETURN** button to exit the warning message.

Important Note: It is only possible to lock custom programs in a codec, not default programs. If **Lock Program** is enabled and you load a new custom program in the codec, **Lock Program** remains enabled and locks the most recently loaded custom program.

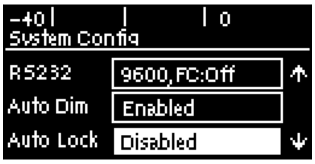
18.17 Locking the Front Panel

The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default. There are two levels of panel lock and each requires a user to enter a PIN to access different features:

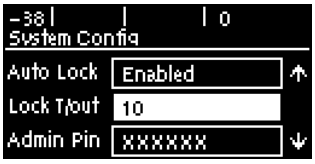
1. **Admin PIN**: Required to change codec connection or configuration settings accessed via the **SETTINGS** button (Default PIN is: 456789)
2. **User PIN**: Required to use the codec front panel buttons and dial/hangup a connection (Default PIN is: 123456)

Enabling the Front Panel Lock Feature

1. Press the **SETTINGS** button.
2. Navigate to **System** and press .
3. Navigate to **Auto Lock** and press to toggle from **Disabled** to **Enabled**.



4. Navigate down to the panel **Lock Timeout** field and press to enter the desired time-out period in seconds. Note: The time-out period is the time in seconds before the codec front panel is relocked after being used.



5. If you want to change the default **Admin PIN** or **User PIN**, navigate to each in turn and press to enter a new PIN.

18.18 Front Panel Alarm Indications

If an alarm is active in the codec the warning symbol will flash in the top right-hand corner of the **OLED SCREEN**. An alarm dialog is also displayed providing details about the alarm.

To diagnose the cause of an active alarm use the Toolbox web-GUI **Alarm panel**. See Managing Alarms for more information.

19 Connecting to the ToolBox Web-GUI

There are two graphical user interface (GUI) options for configuring and connecting Tieline G5 and G6 codecs:

About the HTML5 Toolbox Web-GUI

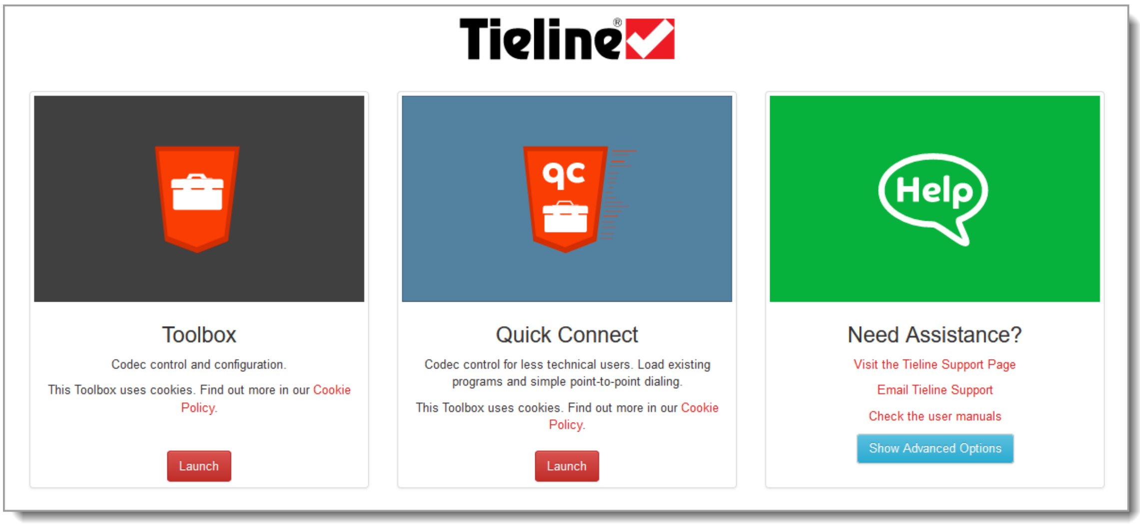
The HTML5 Toolbox Web-GUI improves the user experience with command and control and runs seamlessly on modern browsers. Fully configure codec settings, create dialing programs and dial, hangup and monitor connections. It will run on Mac, Windows and Linux computers.

About the HTML5 Toolbox Quick Connect

The HTML5 Toolbox Quick Connect Web-GUI has a reduced feature-set and allows non-technical users to load existing programs and dial via the **Quick Connect panel**. Users can dial a simple peer-to-peer connection over IP. To enable the **Quick Connect panel** press the **SETTINGS** button, then navigate to **WebGUI > Quick Connect > Enabled**.

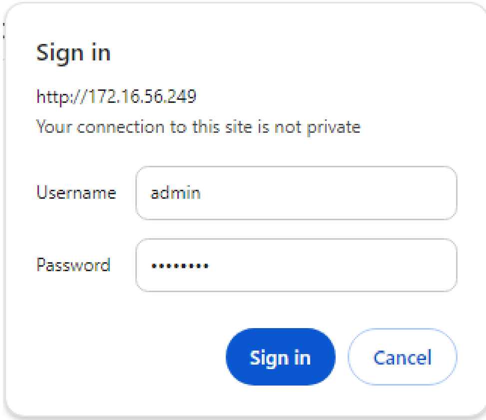
19.1 Opening the HTML5 Web-GUI & Login

1. Attach an Ethernet cable to the **LAN1** port on the codec.
2. Press the **SETTINGS** button and select **Unit** to display the IP address programmed into your codec.
3. Ensure your PC is connected to the same LAN.
4. Open your web browser and type the IP address of your codec into the address bar of your browser, e.g. **http://192.168.0.xxx** (the last digits are the private address details unique to your codec over a private LAN).
5. Refresh the browser and the Web-GUI landing page will display the various command and control options.



Launching the HTML5 Toolbox Web-GUI

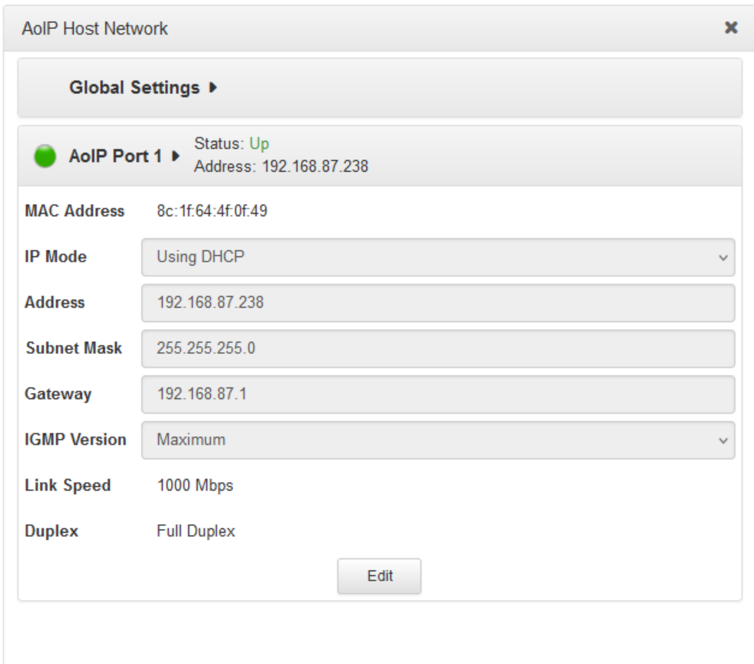
1. Click to launch the **HTML5 Toolbox Web-GUI**.
2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the default user name "**admin**" and password "**password**" to login. Tieline **highly recommends** changing the password (see Changing the Default Password). This provides better network security to maintain reliability during live broadcasts.



Launching the AoIP Web-GUI

To access the AoIP Web-GUI type the IP address of the AoIP interface into a browser. IP address details can be found in the codec via the **AoIP Host Network panel** accessed via the HTML5 Toolbox Web-GUI **AoIP** menu. Note: This can only be accessed when a computer is attached to the same AoIP/AES67 LAN as the Tieline codec.

Important Note: Bridge-IT XTRA II codecs have dual AoIP ports, whereas LAN2 on the Bridge-IT II codec is switchable to AoIP mode if required.



AoIP Host Network panel in the Bridge-IT II codec Web-GUI

Alternatively, press the **SETTINGS** button on the codec front panel and use the navigation buttons to select **Unit > AoIP** and press the button to discover the AoIP port address details.

Port Selection

By default port 80 is used by your PC to communicate with the codec and launch the Web-GUI. If port 80 cannot be used across your network for some reason, type the IP address of your codec into your browser with a full colon and the port number 8080. E.g. 192.168.0.176:8080

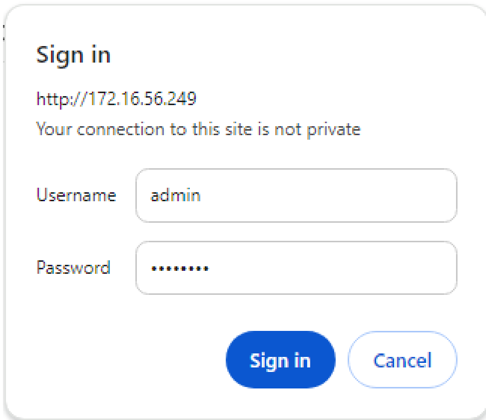
It is also possible to specify a different port for connecting the Toolbox web-GUI to your codec.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to navigate down to **WebGUI** and press the button.
3. Select **Alternate Port** and press .
4. Use the **KEYPAD** to enter a new port number and press the button to save the new setting.
5. Type the codec IP address into your browser with a full colon and then the new port number.

Note: Any new port specified must be within the range 2000 to 65535 inclusive.

Launching the HTML5 Toolbox Quick Connect

1. Click to launch the **HTML5 Toolbox Quick Connect** Web-GUI.
2. When you launch Toolbox Quick Connect for the first time an authentication dialog prompts you to enter the default user name "**admin**" and password "**password**" to login. Tieline **highly recommends** changing the password (see Changing the Default Password). This provides better network security to maintain reliability during live broadcasts.



Using the Web-GUI over the Internet

If your codec is connected over the internet via a public static IP address it is possible to connect and configure it from any PC which is also connected to the internet. If you have multiple browsers open on a PC for different codecs it is possible to customize the browser title for simple identification. To configure this:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **WebGUI** and press the button.
3. Navigate to **Browser** to enter the custom title.
4. Use the keypad to enter the title, then press the button to save the setting.

To configure this setting using the HTML5 Toolbox Web-GUI, click **Settings** at the top of the screen, then click **Options** to display the **Options panel**. Enter the **Browser Title** and then click **Save**.

19.2 Security and Changing the Default Password

Codecs connected to the internet can be accessed by anyone with knowledge of the codec's public IP address. In addition, search engines are widely available which can discover and expose unsecured 'internet connected devices'. Tieline recommends the following IP codec security precautions are followed as a bare minimum, to ensure your codec connections remain secure.

Maintaining Codec Network Security

Adequate security is a major factor in ensuring your codecs and your broadcast network remain secure. There are several layers of security available in Tieline codecs to maintain secure connections. These include:

1. Immediately change the default password when you commission and install your codecs (see instructions which follow). Create a strong password which includes both capital and lower case letters, symbols and numbers (up to 15 characters can be entered). Password managers can be useful when managing multiple passwords within organizations.
2. Ensure your codec is behind a firewall and only open the TCP and UDP ports required to transmit session and audio data between your codecs. Using non-standard ports instead of Tieline default ports can also ensure the codec is more difficult to discover by external parties.
3. Ports 80 and 8080 are commonly used to access the Tieline codec web server. You can add an additional layer of security by translating these ports on the WAN side of your network into non-standard port numbers. Adjust ports using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Configuring TCP/UDP Ports.
4. By default SIP interfaces are disabled to avoid unwanted traffic. The **SIP Filter Lists panel** in the Toolbox HTML5 Web-GUI allows filtering of SIP URIs and User Agents to provide greater security when using SIP. See Configure SIP Allow and Block lists for more information.
5. An SSL security certificate can be installed on each codec in your network to ensure it is a trusted device within your network. See Installing a Security Certificate for more information.
6. Firewall settings facilitate enabling or disabling a range of firewall-related network services, or limit ping to only work in a local subnet. Tieline also recommends SNMP is disabled if a codec is connected to a public network like the internet. Adjust settings using the Toolbox HTML5 Web-GUI **Options panel** in the **Firewall** tab, or see Firewall Configuration.
7. Implementation of CSRF protection (Cross-Site Request Forgery). Enable and disable this setting using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Enabling CSRF Security for more info.

Be sure to document any port changes because this information will be required if you need to contact Tieline or other online support services.

Creating a New Password

The authentication login password can be changed at any time using the codec keypad and OLED screen. Note that passwords are case sensitive:

1. Press the **SETTINGS**  button.
2. Use the navigation buttons to select **WebGUI** and press the  button.
3. Select **Password** and press .
4. Use the **KEYPAD** to enter a new password and press the  button to save the new setting (Note: there is no character limit for passwords).

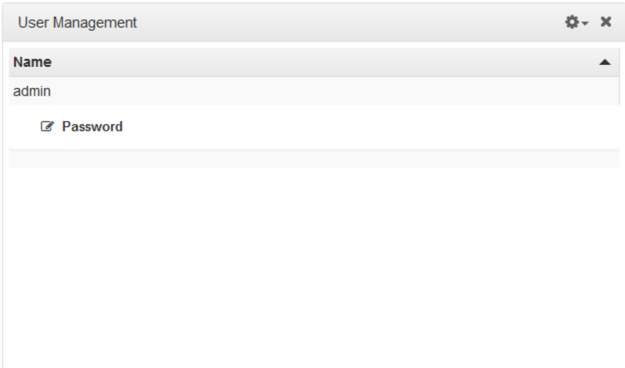
If you forget the password for the Toolbox web-GUI then you can always press the **SETTINGS** button on the codec and navigate to **WebGUI** to view the current password and change it if required.

Important Note: The **Username** in the codec menu is permanently set to **admin** and cannot be changed; only the **Password** can be changed.

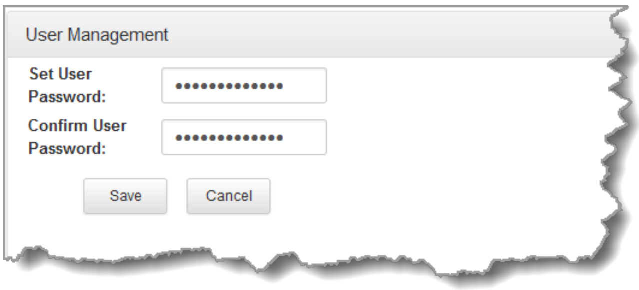
Changing the Password Using the Web-GUI

The codec password can be changed with the **User Management panel** in the Toolbox Web-GUI. This feature is disabled by default. Use the codec front panel to enable this feature by navigating to **Settings > Web GUI > Change Remotely** and then press the OK button.

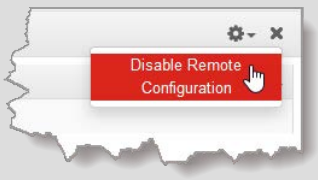
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **User Management** to display the **User Management panel**.



2. Click **Password** and then enter and confirm the new password, then click **Save**.

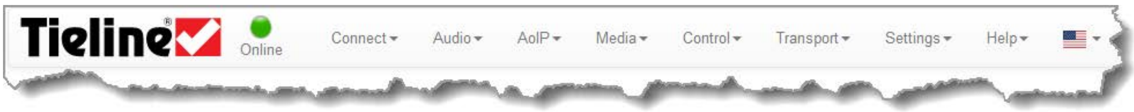


Important Note: This remote password change feature can only be enabled using the codec front panel. However, it can be disabled in the **User Management panel** by selecting the **Options symbol** and then **Disable Remote Configuration**.



20 HTML5 Toolbox Web-GUI Configuration

The following sections provide an overview of the different configuration panels available within the codec's HTML5 Toolbox Web-GUI. Navigate with the mouse pointer to the **Menu bar** at the top of the Web-GUI screen and click to select and open each panel in turn.



HTML GUI Menu Bar for Opening Panels

When you first open the HTML5 Toolbox Web-GUI the **Program Manager panel**, **Connections panel** and **PPMs panel** are loaded by default. If you retain cookies in your browser, any panels opened previously in the Web-GUI are automatically populated when you log in next. The default panel view is displayed on login if cookies have been cleared. The green **Online** indication in the top left-hand corner of the Toolbox Web-GUI indicates it is online and can be used for codec control. A red **Offline** indication is displayed when the codec is unavailable. The **Upgrade** symbol is displayed when a new firmware version is available for the codec. Open the **Firmware panel** in the **Settings** menu to upgrade the codec with new firmware.

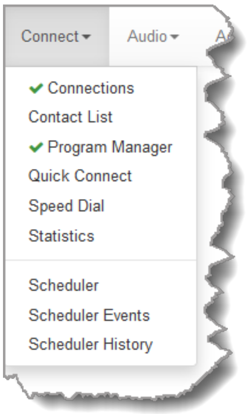
Important Note: The AoIP menu and associated panels are for managing AoIP streaming (e.g. AES67, ST 2110-30, ST 2022-7, Livewire+, RAVENNA). Please see the AoIP Streaming Manual for configuration and streaming information.

Adjusting the Theme

To adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI, navigate to the **Menu bar** at the top of the screen and click **Settings**, then click to select your preferred option. Note: this manual uses the **White** theme for most images.

Opening a Panel & Adjusting Screen Position

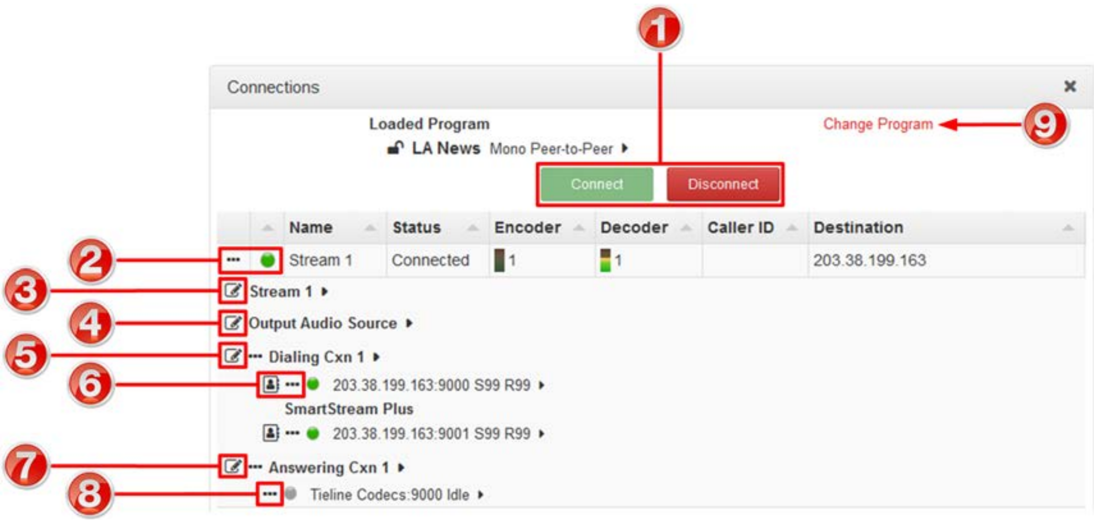
Click an item in the **Menu bar** to display available panel options, then click to select and open a panel. New panels automatically open in the top left of the screen. A green **Tick** adjacent to a panel name in the menu signifies it is already open in the web-GUI.



Position the mouse pointer over a panel's **Title bar** and click and drag to move a panel and reposition it in a preferred screen position.

Connect Panels: Load & Connect Programs & Manage Audio Streams

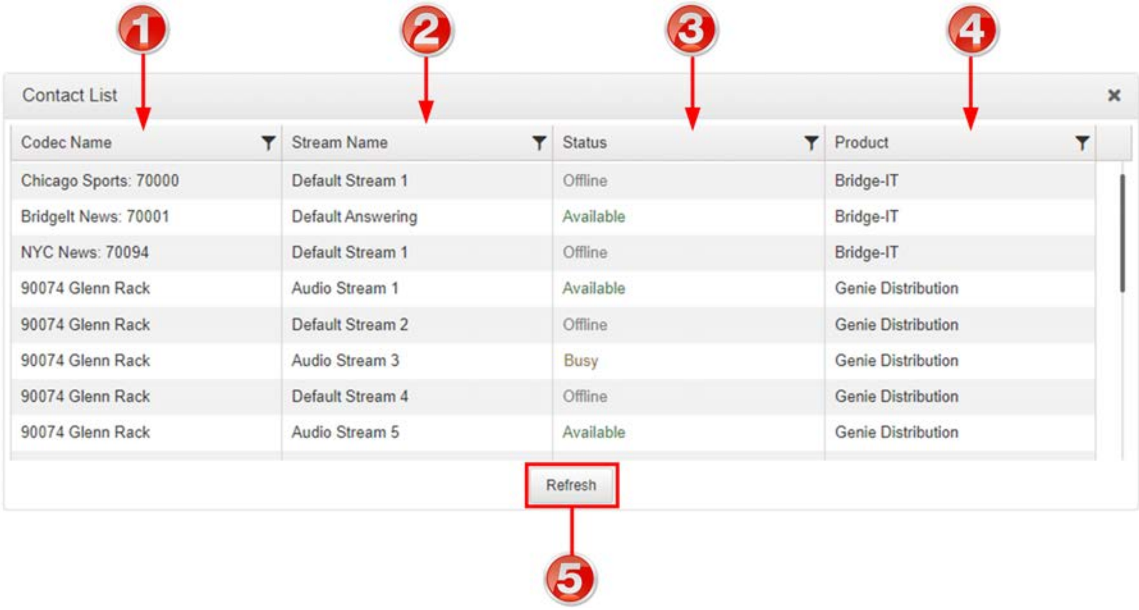
Connections Panel



	Feature	Description
1	Program **Connect / Disconnect** buttons	Click to connect or disconnect all audio streams in a program.
2	**Connect / Disconnect** button and **Connection State LED**	Click over stream details to expand stream information. Click again to collapse stream details. Click the **3 dot** ··· icon to connect or disconnect all connections in an audio stream. The **Connection State LED** displays status of an audio stream connection as follows: • Gray: Not in use • Green: Connected and stable (LQ greater than 70) • Yellow: Connected but connection quality is not stable (LQ 50 to 70) • Red: Connection establishing or problem with connection (LQ less than 50). LED flashes when establishing connection.
3	Audio Stream **Edit**	Click the **Edit** button to edit audio stream settings.

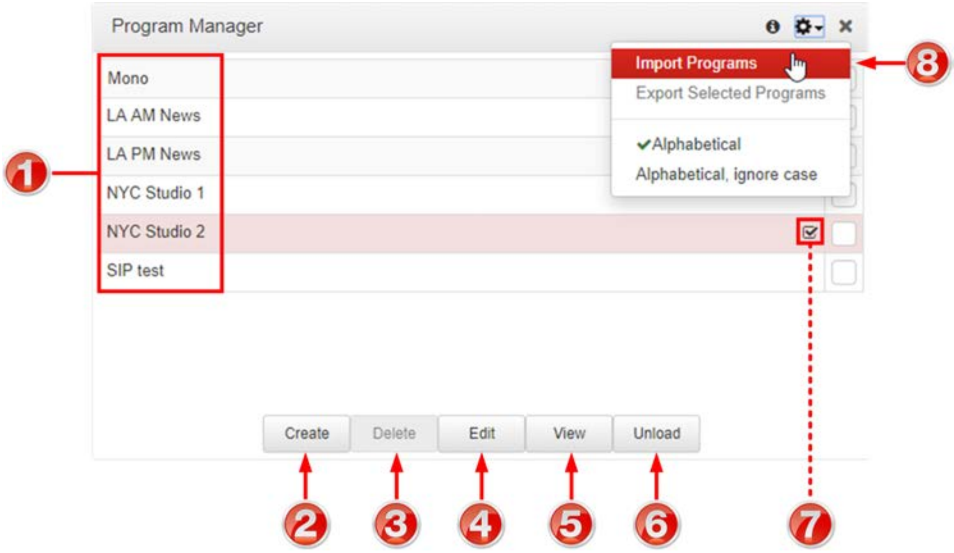
4	Output Audio Source **Edit**	Click the **Edit** button to edit backup audio settings to maintain program audio.
5	Connection **Edit** button (dialing connection)	Click the **Edit** symbol to edit audio stream settings, including the IP address, or select a contact if a TieLink Traversal Server contact list is configured.
6	TieLink **Contact List** and Connection **Connect / Disconnect** button	Click the **3 dot** ··· icon to connect/disconnect an individual connection and adjust the bit-rate when a connection is active. Click the **Address Book** button to select a contact if a TieLink Traversal Server contact list is configured.
7	Answering connection **Edit** button	Click to edit answering connection settings
8	Answering Connection **Disconnect** button	Click to disconnect an answering connection
9	**Change Program**	Click to unload the current program and load a new program.

Contact List



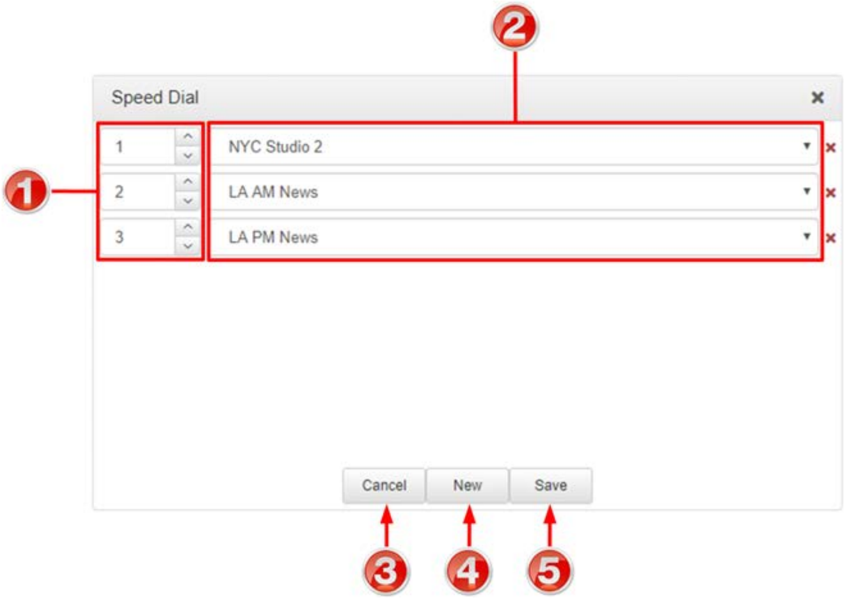
	Feature	Description
1	Codec Name	The name given to a codec to identify it when a program is created
2	Stream Name	The name given to a stream to identify it when a program is created
3	Status	Displays the status of an audio stream, which can be **Available**, **Busy** or **Offline**
4	Product	Displays the model of codec
5	Refresh	Click Refresh to refresh the codec and stream list

Program Manager Panel



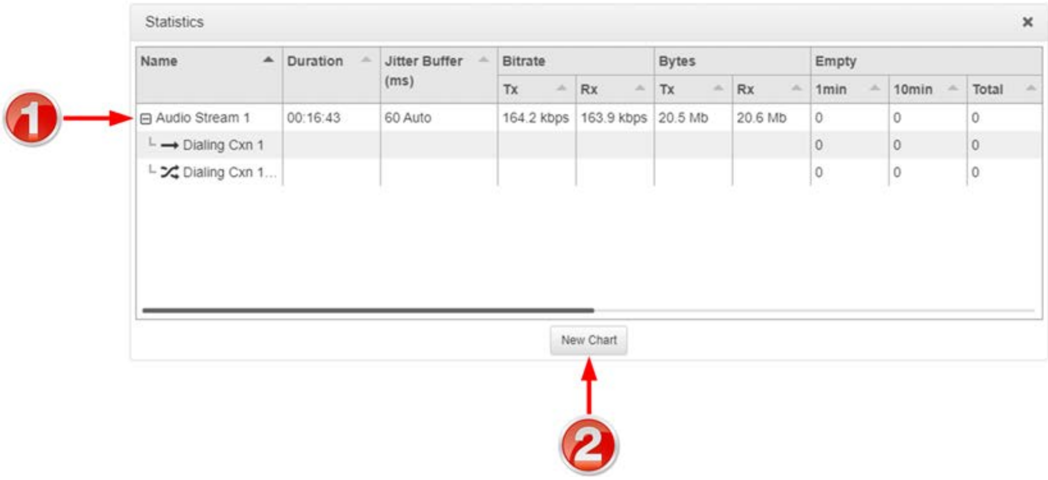
	Feature	Description
1	Program list	The list of saved programs in the codec
2	Create New Program button	Click to create a new program using the program wizard
3	Delete Selected Programs button	Click to delete all selected programs
4	Edit Selected Program button	Click to edit the selected program
5	View Selected Program button	Click to view configuration settings for a selected program
6	Unload/Load program button	Click to load or unload a program
7	Loaded program symbol	Symbol identifies the currently loaded program
8	Import Programs and Export Selected Programs	Click the **Options** menu to select and import previously saved programs or export selected programs as a .zip file

Speed Dial Panel



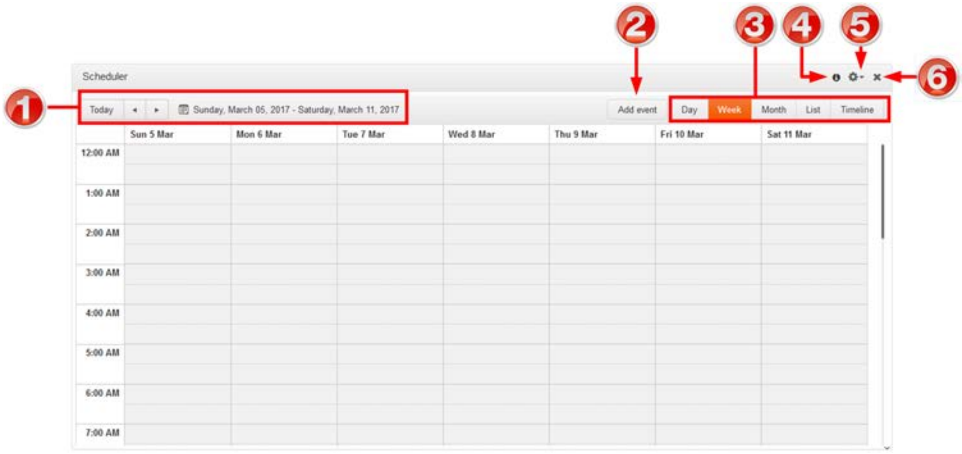
	Feature	Description
1	Speed dial numbers	The number allocated for each speed dial configured
2	Speed dial program	Select the program to associate with a speed dial number
3	**Cancel** button	Click to cancel pending changes
4	**New** button	Click to create a new speed dial
5	**Save** button	Click to save configuration changes

Statistics Panel



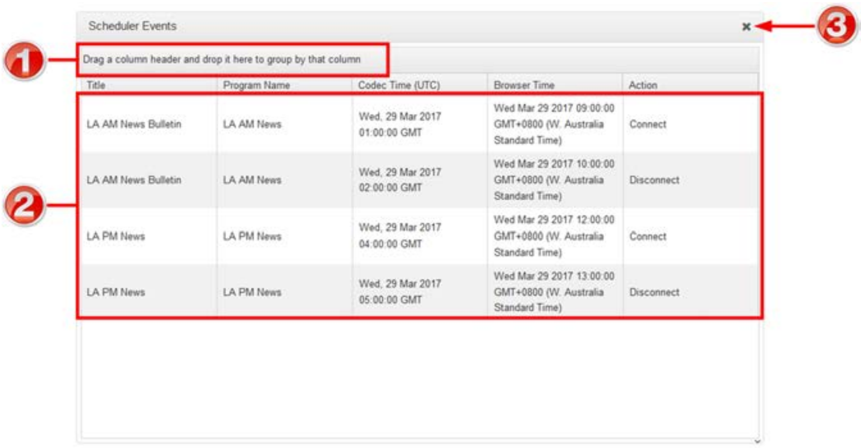
	Feature	Description
1	Expand/Collapse	Click to show/hide audio stream statistics, including packet arrival data info.
2	**New Chart** button	Click the **New Chart** button to select a **Data Series** and create a customized **Statistics panel**.

Scheduler Panel



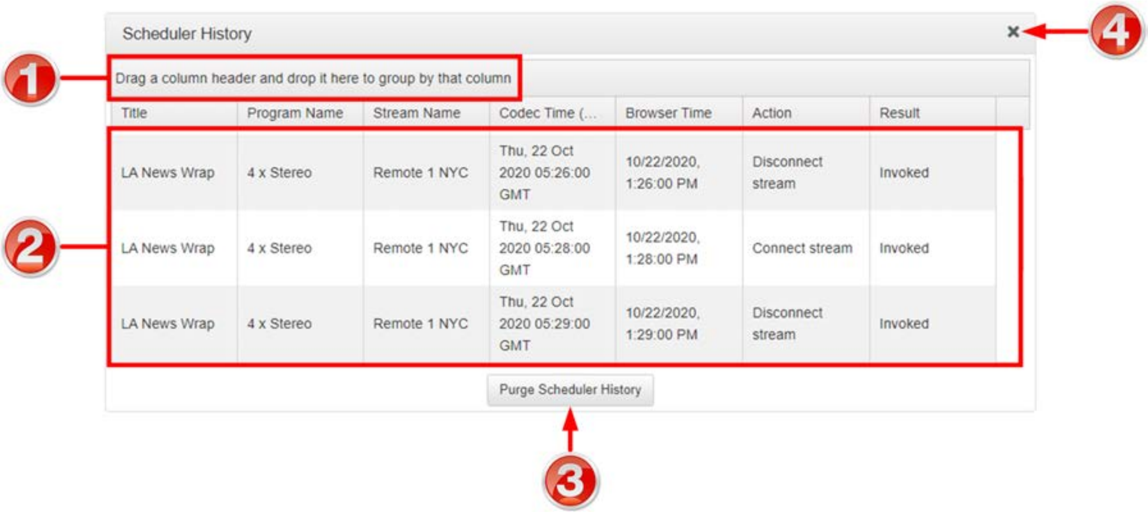
	Feature	Description
1	Date selection	Select the days you wish to view in the scheduler
2	Add event button	Click to create a new scheduled event
3	View type	Click to select the timeframe or timeline view of scheduled events
4	**Information symbol**	Hover over the **Information symbol** to view information about the **Scheduler panel**
5	**Options** menu	Click the **Options symbol** to view timezone options and generate a PDF view, or enable/disable the scheduler
6	**Close** button	Click to close the panel

Scheduler Events Panel



	Feature	Description
1	Grouping	Drag and drop a column header to group scheduled events by that column, e.g. **Codec Time**
2	List of scheduled events	View scheduled events in a list view
3	**Close** button	Click to close the panel

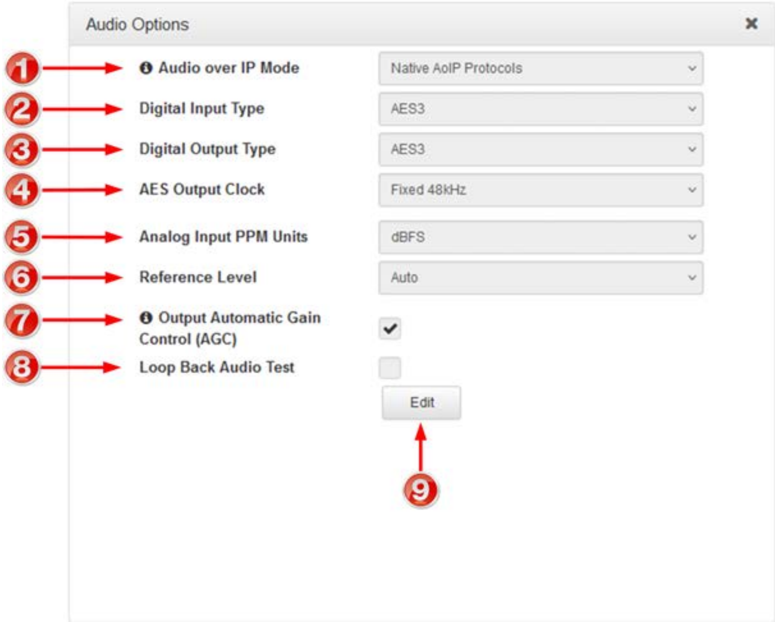
Scheduler History



	Feature	Description
1	Grouping	Drag and drop a column header to group event history by that column, e.g. **Codec Time**
2	Event History List	View the codec's history of scheduled events
3	**Purge Scheduler History** button	Click to clear all event history displayed
4	**Close** button	Click to close the panel

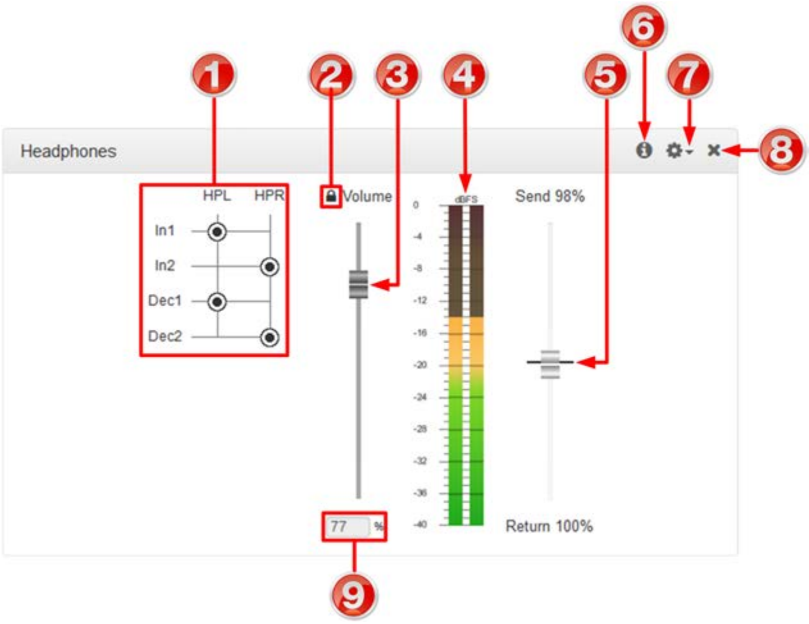
Audio Menu Panels

Audio Options



	Feature	Description
1	Audio over IP Mode	Select **Native AoIP Protocols** as the AoIP mode when streaming with natively supported protocols, e.g. AES67, ST 2110-30, Livewire+ and RAVENNA.
2	Digital Input Type	Select either **AES3** or **Audio over IP** as the **Digital Input Type** for the codec. This setting is global for all digital inputs.
3	Digital Output Type	Select either **AES3** or **Audio over IP** as the **Digital Output Type** for the codec. This setting is global for all digital outputs.
4	AES Output Clock	The sample rate of the AES3 output is configured using the clock source which sets the sample rate frequency of all AES3 output signals. AES67/ST2110-30/Livewire derives the sampling clock from a primary leader clock or Livewire clock over the network and the **AES3 Output Clock** source uses the **Lock to AoIP Clock Source** setting if either the **Digital Input Type** or **Digital Output Type** is set to **AoIP**.
5	Analog Input PPM Units	Use this setting to change the analog input PPM meter unit of measurement from dBFS (default) to dBU
6	Reference Level	The codec will automatically adjust the reference level to suit G6, G5 and G3 codecs, or Report-IT. The reference level can also be preconfigured using this setting.
7	Output Automatic Gain Control (AGC)	Check-box to enable/disable AGC. AGC is independent of IGC (Intelligent Gain Control) on each input.
8	Loop Back Audio Test	Check-box to enable/disable an input/output loopback test of audio. E.g. **Input 1** is routed to analog **Output 1**, **Input 2** is routed to **Output 2**.
9	Edit button	Click to edit settings in the **Audio Options panel**.

Headphones Panel

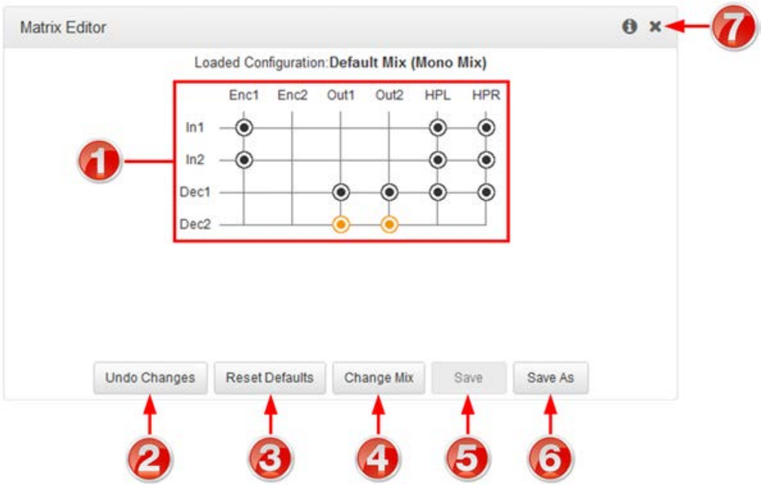


	Feature	Description
1	Headphone Matrix	Select inputs and decoders to monitor outgoing and incoming audio
2	**Headphone Adjust Lock**	Click the **Headphone Adjust Lock symbol** to enable adjustment of headphone output volume. **WARNING: This may adjust the headphone volume of someone listening to a headphone output at a remote location. Extreme caution is recommended when this setting is enabled. High headphone levels can cause hearing loss!**
3	Headphone Volume Slider	Slider used to increase and decrease headphone output volume. **WARNING: This may adjust the headphone volume of someone listening to a headphone output at a remote location. Extreme caution is recommended when this setting is enabled. High headphone levels can cause hearing loss!**
4	Headphone Level PPMs	View PPM levels of input/decoder audio fed to the headphone output
5	Send/Return slider	Use the **Send/Return** slider to adjust the balance between the mix of incoming (return/decoder) audio and outgoing (send/input) audio fed to the headphone output
6	**Information symbol**	Hover over the **Information symbol** to view program matrix information
7	**Options** menu	Click the **Options symbol** to enable or disable the headphone volume limiter
8	**Close** button	Click to close the panel
9	Headphone Gain %	Headphone gain expressed as a percentage

Inputs Panel

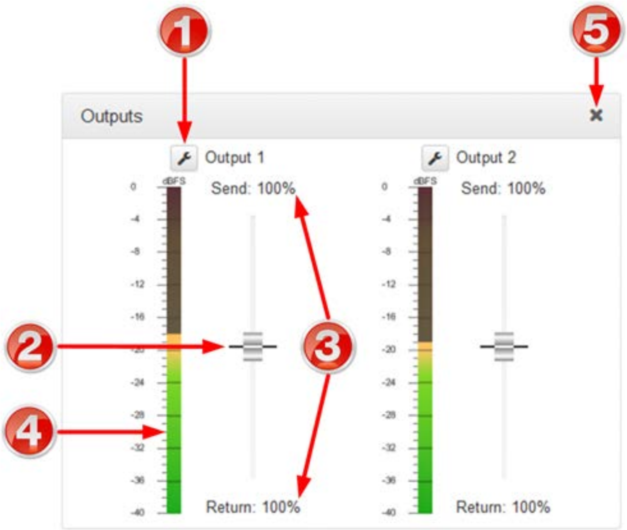
	Feature	Description
1	**Settings** button	Click to adjust input **Type**, **Gain, IGC** and **Polarity** settings, and configure the input **Name**.
2	**On/Off** button	Click to toggle an input on or off.
3	**Close** button	Click to close the panel.
4	Input Sliders/Faders	Input gain control sliders/faders.
5	PPM meter	Input PPM meter.
6	Input gain adjustment	Click to adjust input gain in 0.5dB increments

Matrix Editor Panel



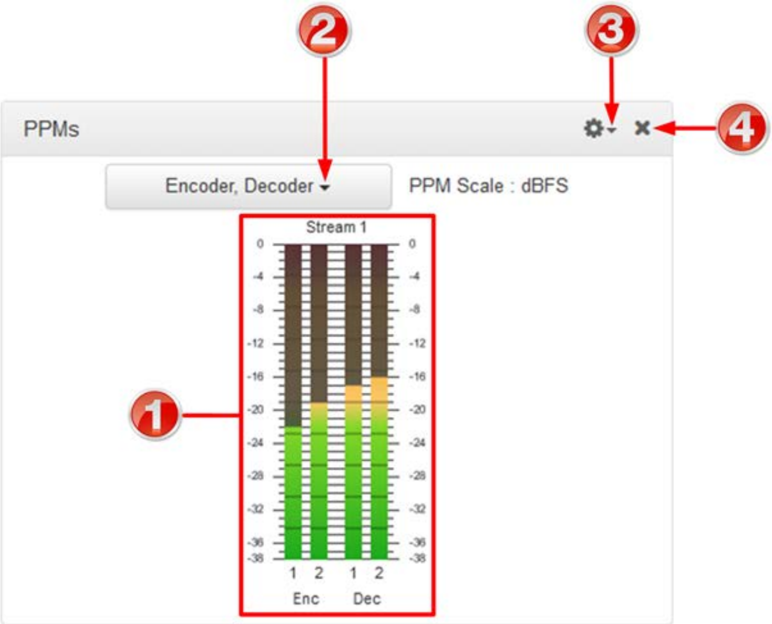
	Feature	Description
1	Matrix Editor crosspoint matrix	Click to select and deselect audio crosspoints to customize routing.
2	**Undo Changes**	Click to undo any changes
3	**Reset Defaults**	Click to reset to the factory default matrix mix for the program
4	**Change Mix**	Click to select the default or Custom matrix mix and view and edit. Options include **Load**, **Delete** and **Rename** a mix
5	**Save**	Click to save changes to a custom mix
6	**Save as**	Click to save as a new custom matrix mix
7	**Close** button	Click to close the panel.

Outputs Panel



	Feature	Description
1	Settings	Name the output if required
2	**Send/Return slider**	Use the **Send/Return** slider to adjust the balance between the mix of incoming (return/decoder) audio and outgoing (send/input) audio fed to each codec output
3	**Send/Return** indications	Balance of send and return audio included in the mix for each output (expressed as a percentage)
4	Output PPM meter	Output PPM meter
5	**Close** button	Click to close the panel

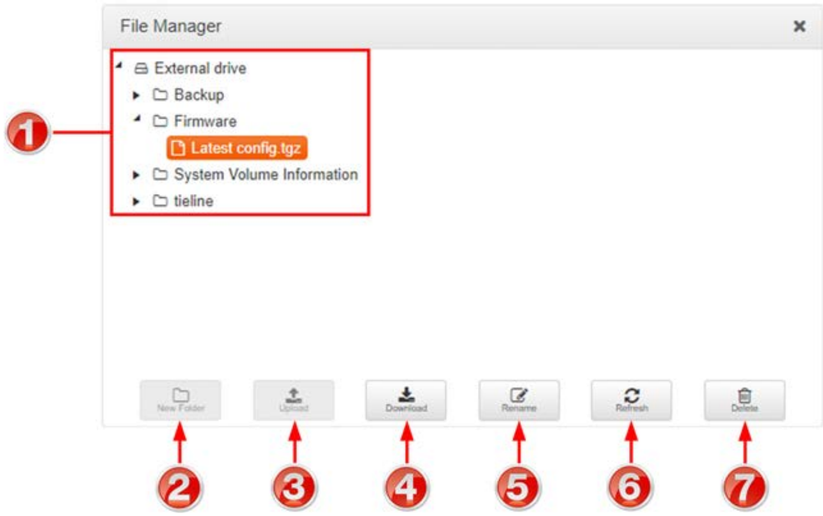
PPMs Panel



	Feature	Description
1	**PPMs**	Audio stream PPMs grouped
2	**PPM Select** menu	Selectable audio stream PPM metering options using dBFS audio scale
3	**Options menu**	Options menu allows users to group PPMs by stream, or by type, e.g. input, output, encoder, decoder and headphone.
4	**Close** button	Click to close the panel

Media Menu

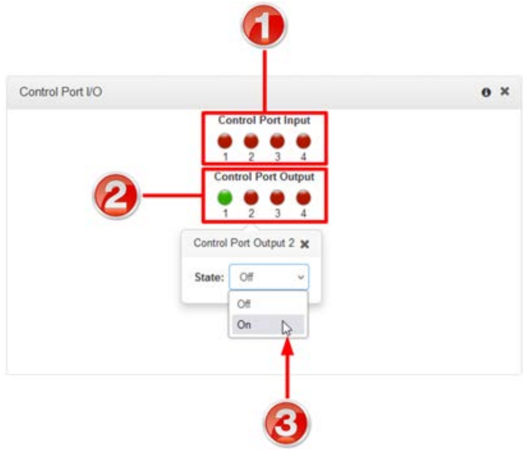
File Manager Panel



	Feature	Description
1	Folder and File View	View all folders and files on an external drive / removable media
2	**New Folder**	Select the drive or a folder and then click to create a new sub-folder
3	**Upload**	Click to select and upload a new file onto removable media
4	**Download**	Click to download a file onto removable media
5	**Rename**	Click to rename a selected file or folder
6	**Refresh**	Click to refresh the panel and view all files and folders
7	**Delete**	Click to delete a selected file or folder

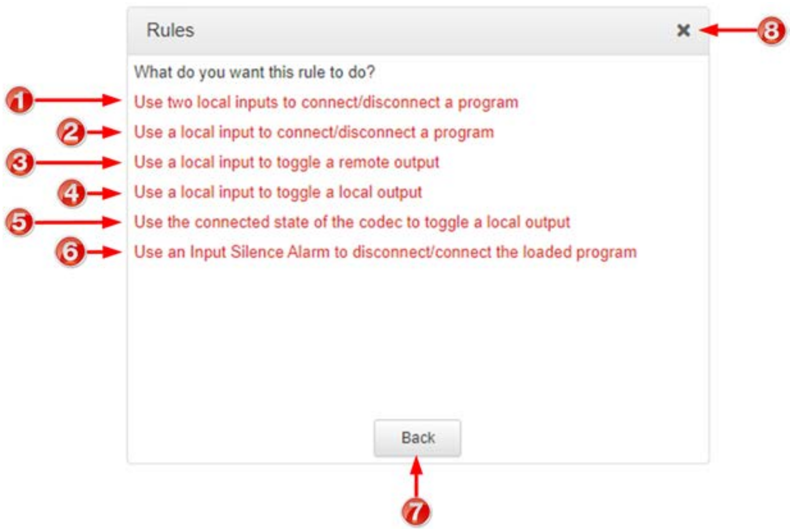
Control Menu

Control Port I/O



	Feature	Description
1	Control Port Input state	Displays the state of a control port input
2	Control Port Output	Displays the state of a control port output
3	**State**	Click a **Control Port Output** to change the **On/Off** state

Rules Panel

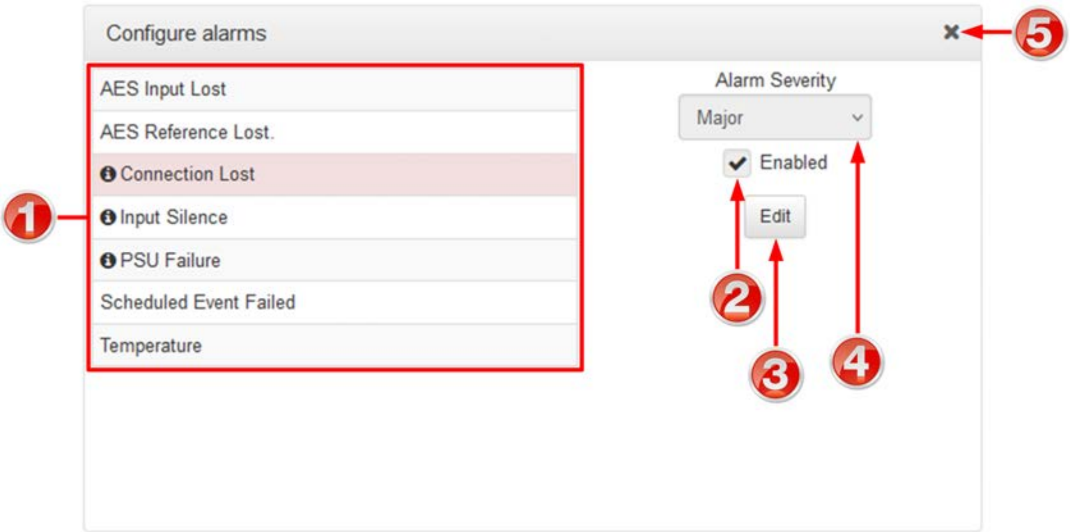


	Rule	Description
1	Use two local inputs to connect/disconnect a program	Click to configure connection and disconnection after different relay inputs are switched **ON**
2	Use a local input to connect/disconnect a program	Click to configure connection and disconnection by toggling an input
3	Use a local input to toggle a remote output	Click to configure a local relay input to synchronize with the state of a remote relay output
4	Use a local input to toggle a local output	Click to configure a local relay input to synchronize with the state of a local relay output
5	Use the connected state of the codec to toggle a local output	Click to configure a relay to toggle based on connection status

6	Use an Input Silence Alarm to disconnect/connect the loaded program	If an Input Silence Alarm is triggered, disconnect the loaded program; if the alarm is cleared, connect the loaded program
7	**Back** / **Add New Rule** button	Click to add a new rule, or exit the rule creation function
8	**Close** button	Click to close the panel

Alarms Panels: Configure & Monitor Alarms

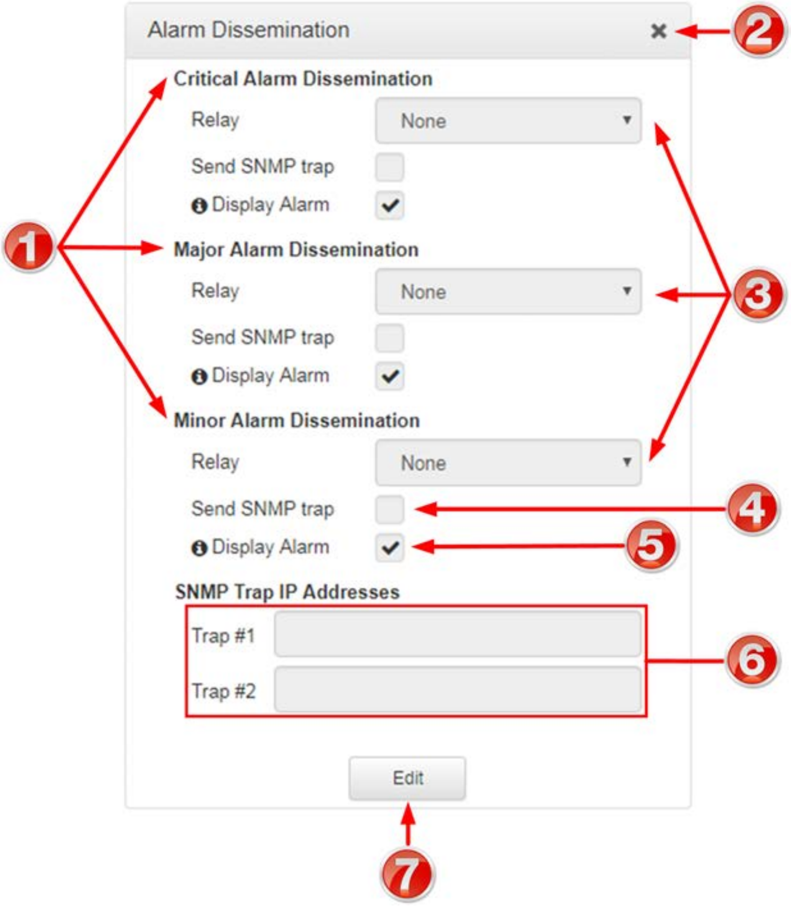
Configure Alarms Panel



	Feature	Description
1	List of alarm types	Click to select an alarm type to configure. Note: Some alarms may not be available in all codec variants, e.g. PSU alarms in Bridge-IT II.
2	Enable Alarm check-box	Click the **Enabled** check-box to enable the currently selected alarm
3	**Edit** / **Save** button	Click to edit an alarm, or save configured alarm settings when in edit mode
4	Alarm Severity Setting	Click the drop-down arrow to select an alarm severity setting
5	**Close** button	Click to close the panel

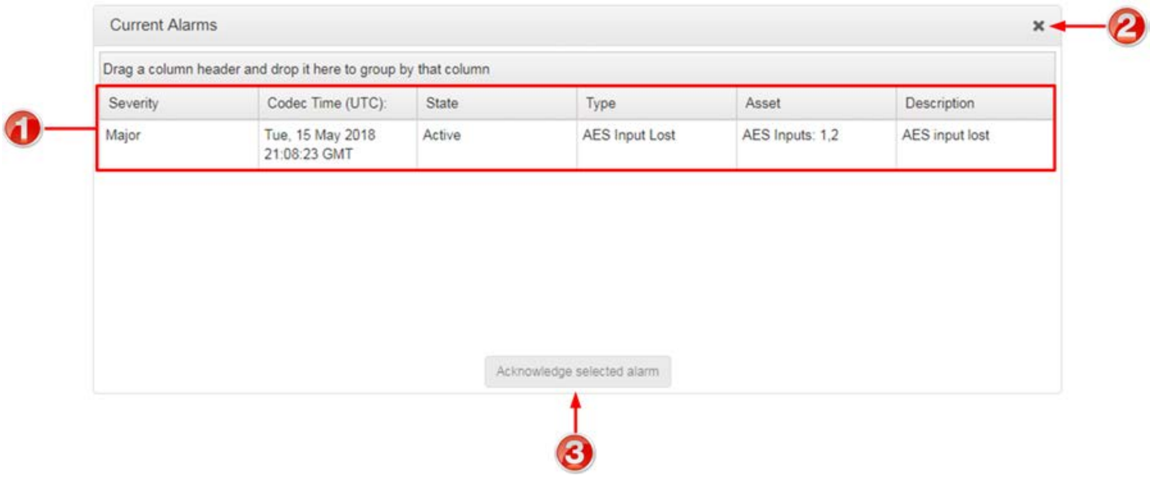
Alarm Dissemination Panel

Alerts for each alarm severity level are configured using the **Alarm Dissemination panel**.



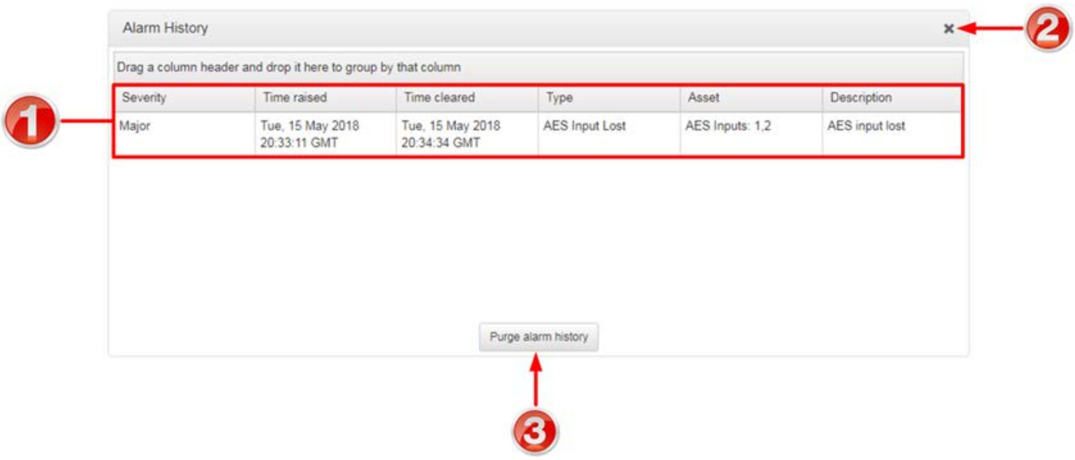
	Feature	Description
1	Alarm severity levels	Click to select an alarm severity level to configure it.
2	**Close** button	Click to close the panel.
3	**Relay** drop-down selection	Click the drop-down arrow to select a relay to open when an alarm is activated.
4	**Send SNMP trap** check-box	Select the check-box to enable SNMP traps to be sent for alarms configured at the selected severity level.
5	**Display Alarm** check-box	Select the check-box (default enabled) to deliver front panel **ALARM LED** notifications and HTML5 Toolbox Web-GUI alarm notifications.
6	SNMP Trap Target text-boxes	Click in the text boxes in edit mode to enter the SNMP trap targets for alarms using the currently selected severity level.
7	**Edit** / **Save** button	Click to edit alarm dissemination settings, or save configured settings when in edit mode.

Current Alarms



	Feature	Description
1	Current alarm description	View a list of active alarms in the codec
2	**Close** button	Click to close the panel
3	**Acknowledge selected alarm** button	Click to acknowledge a selected alarm.

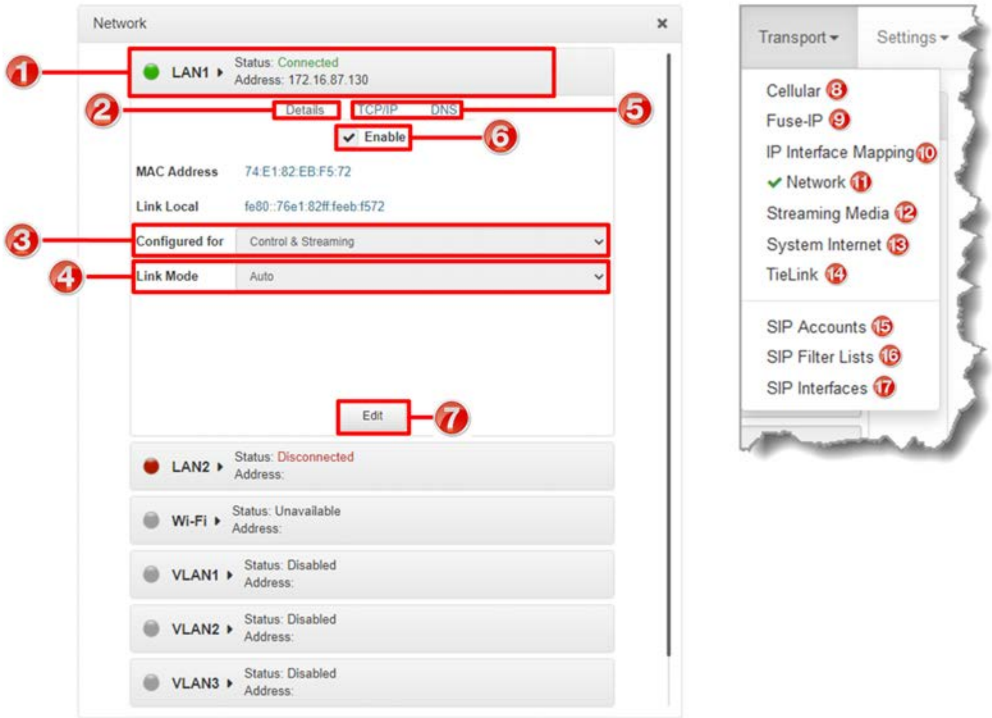
Alarm History



	Feature	Description
1	Alarm history description	View the history of previous alarms in the codec.
2	**Close** button	Click to close the **Alarms** panel.
3	**Purge History** button	Click to purge alarm history.

Transport Panels

There are several **Transport** panels which can be opened in the Web-GUI. Each panel provides specific transport-related configuration settings and options. Click to select and open each panel. As an example, the **Network panel** is displayed with network interface configuration options. A brief description of the other panel options is also provided.



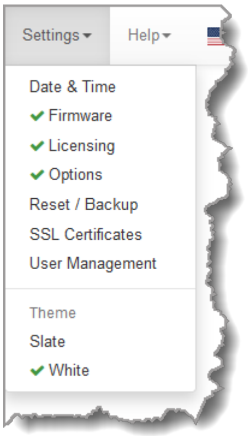
	Feature	Description
1	Network	Click to select and edit, or view network configuration settings for the LAN, VLAN and USB interfaces.
2	Details tab	Display configuration options for a selected network interface, plus other device details.
3	Control/Streaming	Select **Control** and/or **Streaming** options for the selected interface.
4	**Link Mode**	Configure the Ethernet or VLAN link speed (10/100/1000/Auto) and whether an interface will operate in Full-Duplex or Half-Duplex mode.
5	**TCP/IP** and **DNS** tabs	Select the **TCP/IP** tab to configure **IPv4/IPv6** address details. Select the **DNS** tab to specify DNS addresses and domains to search.
6	**Enable check-box**	Select the check-box to enable an interface.
7	**Edit/Save/Cancel** button	Click **Edit** to edit settings, click **Save** to store settings, or click **Cancel** to revert to previously configured settings.
8	**Cellular**	Click to open the **Cellular panel** and configure a connected USB air card.
9	**Fuse-IP**	Click to open the **Fuse-IP panel** and configure Fuse-IP bonding.
10	**IP Interface Mapping**	Configure default **Primary**, **Secondary**, **Tertiary** and **Quaternary** interfaces.
11	**Network**	Click to open the **Network panel** and configure network settings.

12	**Streaming Media**	Click to configure HTTP streaming media settings for a local Icecast server running on the codec. Enable a local Icecast server and specify the Icecast Server Port.
13	**System Internet**	Click to select the preference for interfaces used to connect to the internet.
14	**TieLink**	Click to view the TieLink panel to enable, disable and prioritize interfaces, configure ports and Fuse-IP interfaces, and configure STUN server settings.
15	**SIP Accounts**	Click to open the panel and edit SIP account settings. Up to 6 SIP accounts are supported.
16	**SIP Filter Lists**	Add trusted network codecs to the **URI Allow List**. Add SIP URIs to the **URI Block List** and add user agents to the **User Agent Block List** to deny them access.
17	**SIP Interfaces**	Click to open the panel and configure port, proxy and **Interface** settings for the **SIP1** and **SIP2** interfaces. The codec supports dialing over these SIP interfaces simultaneously.

Settings Panels

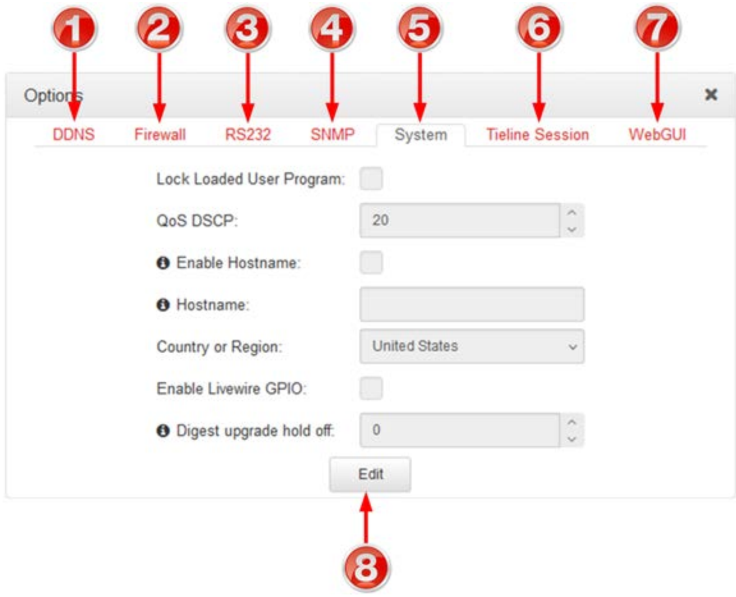
There are several **Settings** panels which can be opened in the Web-GUI. Each panel provides specific codec configuration settings and options. Click to select and open each panel. It is also possible to change the HTML5 Web-GUI theme.

As an example, the **Options panel** is displayed with a brief description of the other panels available.



	Feature	Description
1	**Date and Time**	Click to open the panel view and sync the codec to NTP time.
2	**Firmware tab**	Click to open the panel; view software versions, download firmware and perform an upgrade.
3	**Licensing tab**	Click to open the panel; select a license file and install it in the codec.
4	**Options tab**	Click to open the panel and adjust a wide range of codec audio, DDNS, firewall, power management, RS232, SNMP, system, session data and Web-GUI settings.
5	**Reset / Backup**	Click to open the panel; reset codec default settings and perform backup/restore of codec programs and settings.
6	**SSL Certificates**	Click to open the panel and update digital SSL certificates remotely. Note: The first certificate must be installed using the front panel. Certificates cannot be updated if SSL is not enabled.
7	**User Management**	Click to open the panel and change the Web-GUI password. Note: This feature must first be enabled in the front panel via **Settings > IP Options > Web GUI > Allow Password Change Remotely**.
8	**Theme**	Adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI; options include **White** or **Slate**.

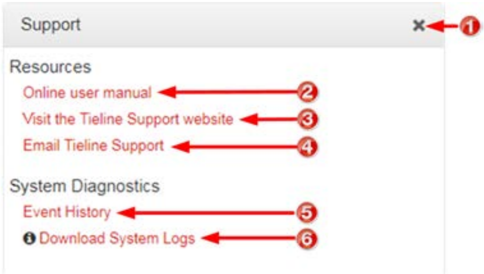
Options panel



	Feature	Description
1	**DDNS** Settings	Configure Dynamic DNS to use a hostname to connect to a codec when using a dynamic IP address.
2	**Firewall** Settings	Enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Ember+ can also be enabled.
3	**RS232** Settings	Click **Baud rate** to adjust the baud rate used by the RS-232 serial port on the codec. Select the check-box to **Enable Flow Control**.
4	**SNMP** Settings	Configure SNMP settings in the codec.
5	**System** Settings	Configure various system settings, including: Country setting; select the **Lock Loaded User Program** check-box to lock the currently loaded program in the codec; enable and assign a hostname to the codec to provide a flexible way of identifying the codec on a network; configure IP audio data packets for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks.
6	**Tieline Session**	Edit the Tieline session and alternative session port used by the codec.
7	**WebGUI**	Includes settings such as **Quick Connect Enabled**, **CCC Enabled** (select this option to enable Cloud Codec Controller use with the codec), enable **CSRF**, enter a **Browser Title**, adjust the **SSL Port**.
8	**Edit button**	Press to edit settings in the **Options panel**.

Help Panels

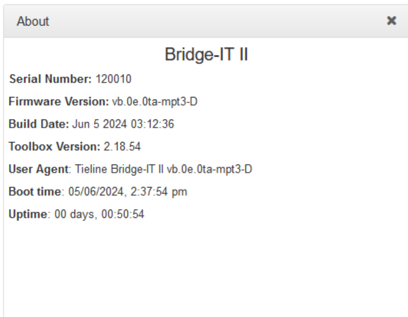
Support Panel



	Feature	Description
1	**Close** button	Click to close the panel.
2	User manual link	Click to open the codec user manual in a new browser, or view support information.
3	Support website link	Click to visit the support page on the Tieline website.
4	Email Tieline Support	Click to email Tieline support.
5	Event History	Click to download user-viewable event logs
6	Download System Logs	Click to download diagnostic information that can be sent to Tieline support

About Panel

The **About panel** provides details of the codec Toolbox and firmware version, as well as the codec serial number. Note: the codec name displayed will vary by product type.



Language Selection

The HTML5 Toolbox Web-GUI offers language support for several languages.

1. Click the **Language** drop-down arrow in the top right-hand corner of the Web-GUI page.
2. Select the preferred language to display.

20.1 Using the HTML5 Toolbox Quick Connect Web-GUI

The HTML5 Quick Connect Web-GUI is designed for simple peer-to-peer connections and non-technical users. It has a reduced feature-set and allows users to:

1. Load existing programs in a codec via the **Program Loader panel** and then dial via the **Quick Connect panel**.
2. Use the **Quick Connect panel** to create and dial a simple peer-to-peer connection using IP/SIP.

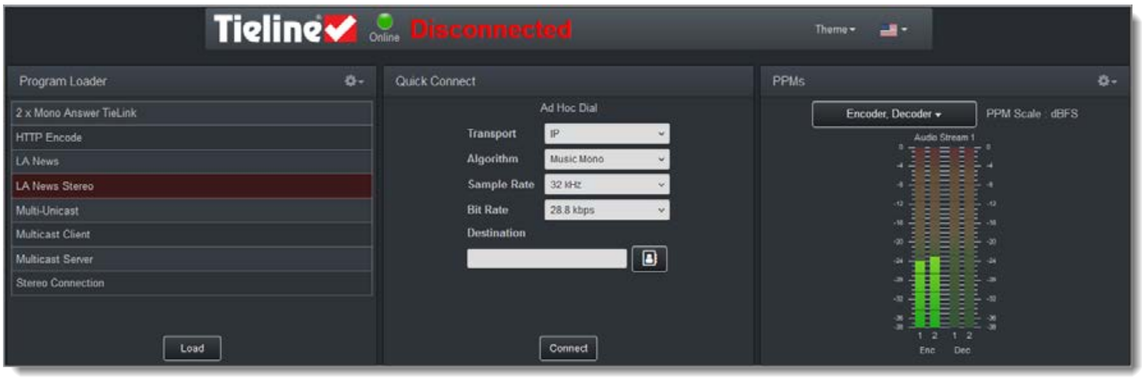
See Opening the HTML5 Web-GUI and Login for details about launching the standalone HTML5 Quick Connect Web-GUI. The **Quick Connect panel** can also be launched from the **Connect** menu in the HTML5 Toolbox Web-GUI.

Important Note:
- Simple peer-to-peer connections are not saved as programs with unique names.
- Details of the last ad hoc dial are retained in the **Quick Connect panel**, even after a program is loaded and unloaded using the **Program Loader panel**.

To enable the **Quick Connect panel** press the **SETTINGS** button, then navigate to **WebGUI > Quick Connect (Q Cnct) > Enabled**.

Launching the HTML5 Quick Connect Web-GUI

1. Type the codec IP address into your web-browser.
2. Click to launch the HTML5 Toolbox Quick Connect Web-GUI.
3. Enter the authentication **Password** for the codec and click **OK**.
4. The panels in the Quick Connect Web-GUI will automatically be displayed.

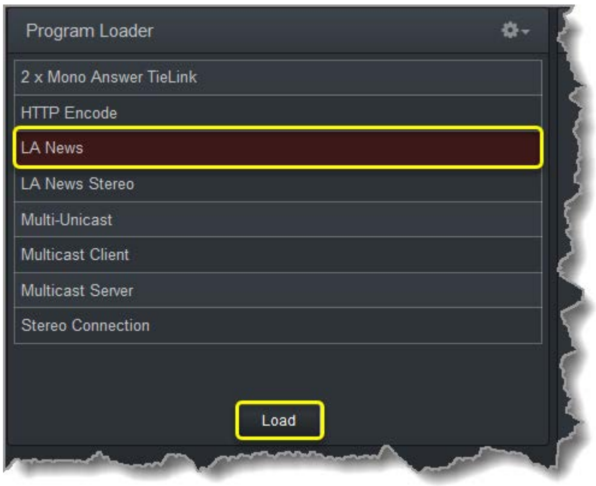


Important Note: To change the password using the codec front panel navigate to **Settings > WebGUI > Password** and press the OK button. Use the keypad to enter a new password and press the OK button to save the new setting.

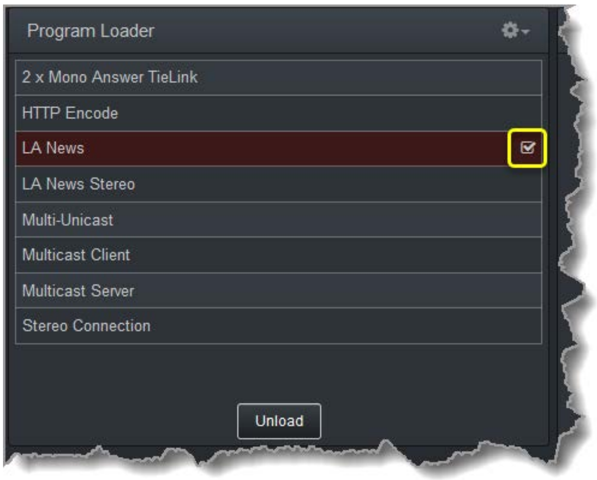
Loading and Unloading an Existing Program

If programs are saved in the codec they are displayed in the **Program Loader panel**.

1. Click to select a program in the **Program Loader panel** and click **Load** to load it in the codec.



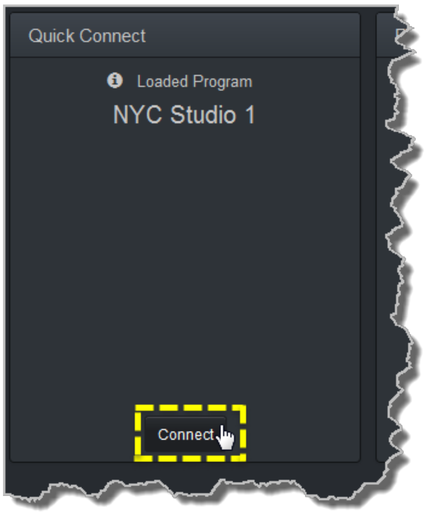
2. The **Check-box symbol** appears next to the program name to confirm it has been loaded and then **Load** changes to **Unload**.



3. To unload a program click **Unload**.

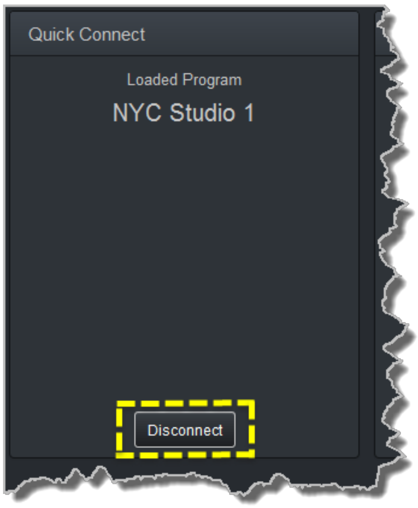
Dial a Loaded Program

1. Click the **Connect button** in the **Quick Connect panel** to dial a loaded program. Note: After connecting, the **Connect button** changes to a **Disconnect button**.

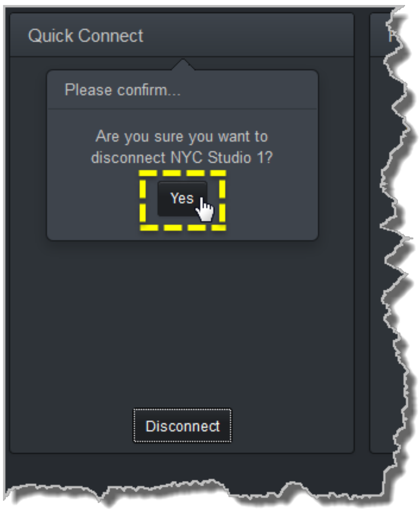


Disconnect a Loaded Program

1. Click **Disconnect** in the **Quick Connect panel**.



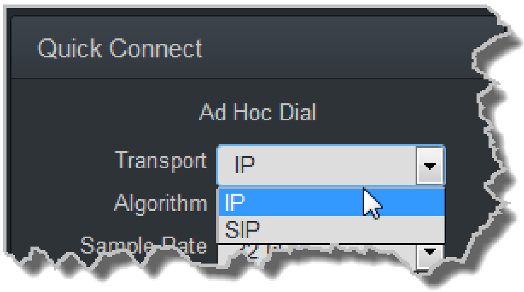
2. Click **Yes** in the confirmation dialog to disconnect the connection.



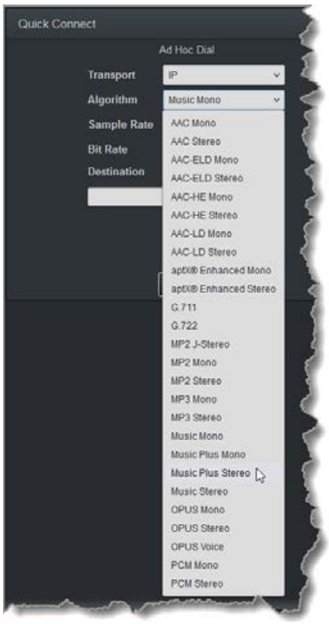
Dial Peer-to-Peer over IP with Quick Connect

Important Note: Click **Unload** in the **Program Loader panel** if a program is currently loaded.

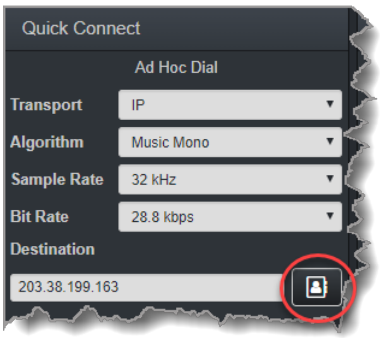
1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **IP**.



2. Click the drop-down **Algorithm** menu and select an algorithm.



3. Select the appropriate **Sample Rate** and **Bit Rate** for the connection. Note: If only one sample rate is available this will be automatically selected.
4. Click in the **Destination** text box and enter the IP address of the destination codec, or click the **Address Book button** to select a preconfigured TieLink contact.



5. Click **Connect** to dial the connection.

Monitoring PPMs

See Monitoring PPMs for details of how to monitor audio streams using the **PPMs panel**.

20.2 Configure Programs with the Connections Panel

The **Connections panel** allows:

- The configuration of new programs.
- Editing of existing programs.
- Loading and unloading of programs.

This panel delivers the ability to edit destination settings for an audio stream without unloading a program.

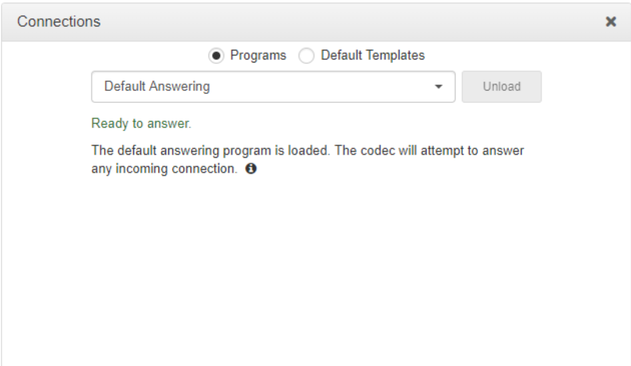
Configuring New Programs

Important Note: There are some limitations when creating and editing programs using the **Connections panel** in comparison to using the **Program Manager panel**. In the **Connections panel** is not possible to:

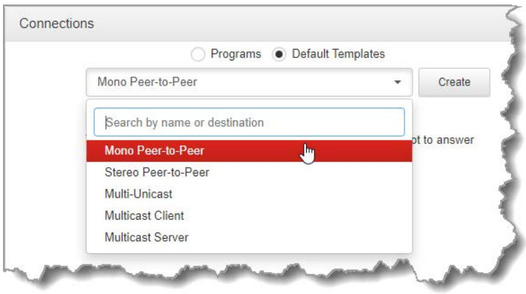
- Name or rename a program.
- Manage Program level Rules.
- Create a program from a copy of an existing program.

To configure new programs using the **Connections panel**:

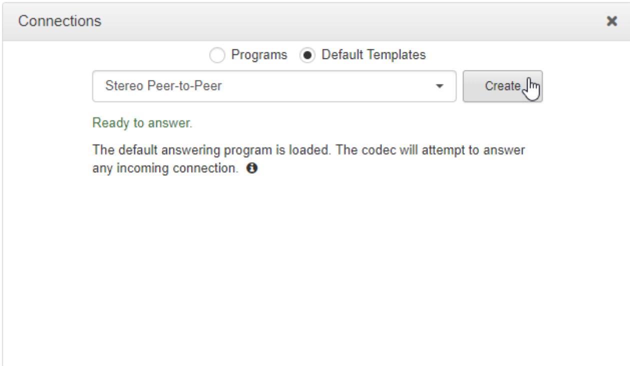
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**.



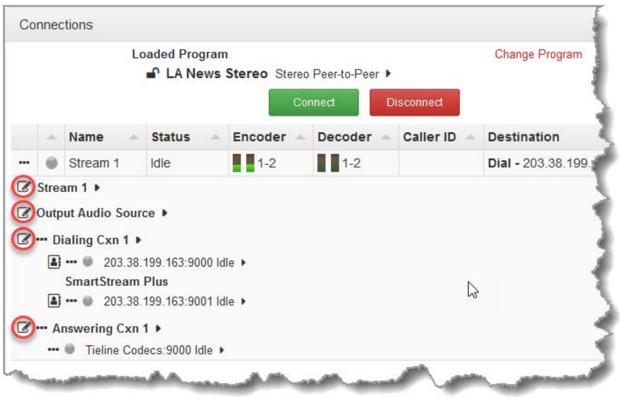
2. Select **Default Templates** and then click the drop-down arrow to select one of the supported default templates in the codec. Note: The default templates available will vary from codec-to-codec.



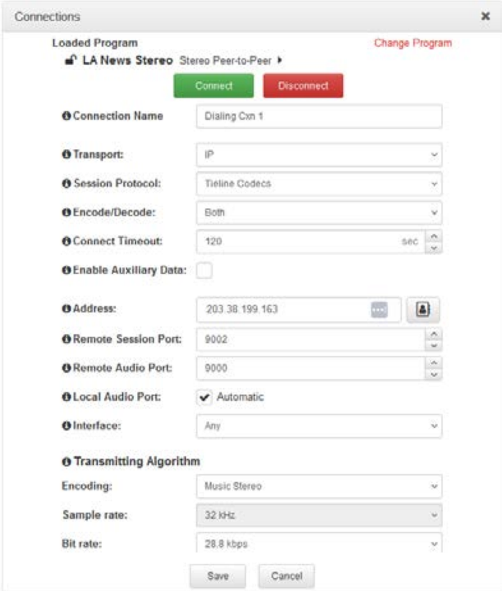
3. Click **Create** to add the new program.



4. To configure new connection settings, or edit existing connection settings, click the **Edit symbol** .



5. The **Edit symbol** expands each audio stream and dialing and answering connection to reveal configuration settings. Hover over the **Information symbol** to view details of each configuration setting. Click **Save** to confirm new settings.


Connections
Loaded Program
Change Program
LA News Stereo Stereo Peer-to-Peer
Connect
Disconnect
Connection Name
Dialing Cxn 1
Transport:
IP
Session Protocol:
Tieline Codecs
Encode/Decode:
Both
Connect Timeout:
120
sec
Enable Auxiliary Data:
Address:
203.38.199.163
Remote Session Port:
9002
Remote Audio Port:
9000
Local Audio Port:
Automatic
Interface:
Any
Transmitting Algorithm
Encoding:
Music Stereo
Sample rate:
32 kHz
Bit rate:
28.8 kbps
Save
Cancel

20.3 Configuring IP Settings

Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **Network** to view and configure Ethernet and VLAN interface settings in the Web-GUI.

Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

IPv4 versus IPv6

An IP address is a unique address to identify a device on a TCP/IP network. Your codec uses dual IP protocol stacks to allow it to work on both IPv4 and IPv6 networks. Tieline codecs support both DHCP (default) IP addressing and static IP addresses for dialing IPv4 connection endpoints.

To dial a codec with a public IP address you simply dial the IP address to connect.To dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN. Usually this will be set up at the studio end so you can dial into the studio from a remote codec.

Support for IPv6 connections allows you to use IPv6 infrastructure to connect to other codecs globally.

Configuring Ethernet Ports and VLANs

Bridge-IT II and Bridge-IT XTRA II codecs feature two physical Ethernet ports and up to four additional VLAN interfaces.

Important Note:
- Bridge-IT XTRA II codecs have two dedicated Ethernet ports and two dedicated AoIP ports for connecting to networks using protocols like Livewire, RAVENNA, AES67, ST2110-30, ST2022-7 and NMOS.
- Bridge-IT II codecs have a dedicated **LAN1** Ethernet port and **LAN2/AoIP** is a second LAN port (default setting), or switchable to operate over AoIP networks using protocols like Livewire, RAVENNA, AES67, ST2110-30 and NMOS.
- When **LAN2/AoIP** operates in AoIP mode, **LAN2** in the **Network** panel appears as **Disconnected** and **AoIP Port 1** is active in the **AoIP Host Network** panel.

- Download the AoIP user manual from www.tieline.com for more information about how to configure sources and destinations and AoIP streaming generally.

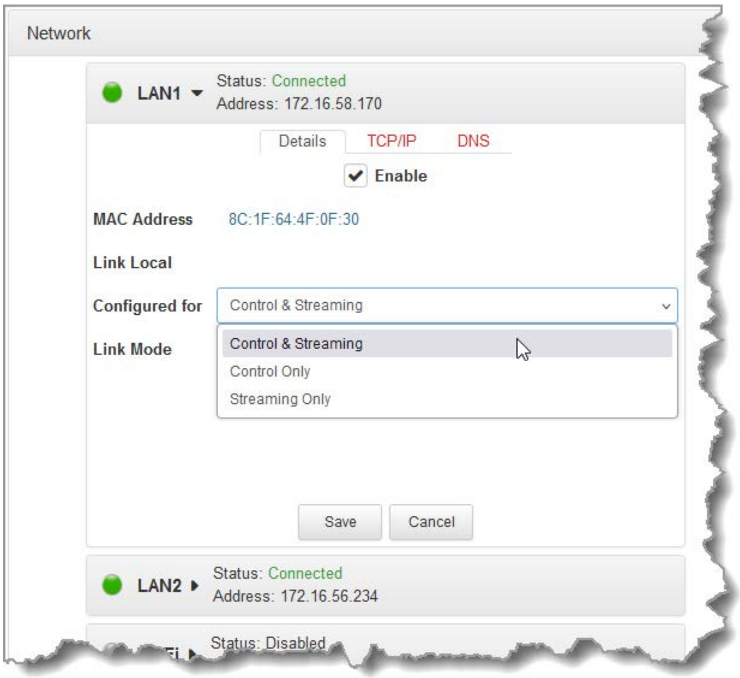
VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec. As an example, VLANs can be used to separate codec Control and Streaming functions if required. Ethernet (**LAN**) and VLAN interfaces can be configured for:

- Control Only: codec control and command only from the Ethernet port.
- Control and Streaming: stream audio and control and command the codec via the Ethernet port.

- Streaming audio: stream audio only from an Ethernet port (**VLANs** only).
- Nothing: Disable the Ethernet port from streaming audio and codec command and control (**VLANs** only).

To edit control and streaming settings:

1. Click an interface to expand the details for the selected interface.
2. In the **Details** tab click the **Edit** button.
3. Select the **Details** tab and configure control and streaming as required.
4. Click **Save**.



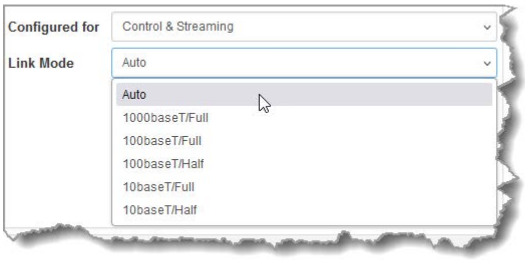
To enable or disable an interface select the **Enable** check-box when in **Edit** mode, then click **Save**. **LAN1** and **LAN2** are enabled by default. Please note that the **Status LED** color indicates the status of each network interface.

1. Any interface with a grey **Status LED** indication is disabled.
2. Any interface with a green **Status LED** indication is enabled and active.
3. Any interface with an amber **Status LED** indication is enabled and attempting to connect.
4. Any interface with a red **Status LED** indication is enabled and unavailable for some reason, e.g:
   - Ethernet cable is not connected.
   - Incorrect configuration, e.g. VLAN Identifier not correct.
   - Wi-Fi access point is not selected.

Configure Link Mode

It is possible to configure the Ethernet link speed (10/100/1000/Auto) and whether each available interface operates in full-duplex or half-duplex modes.

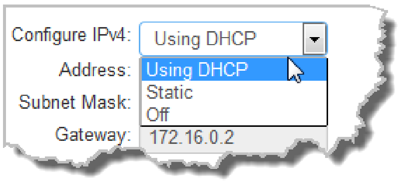
1. Click the drop-down **Link Mode** arrow and select the preferred setting.



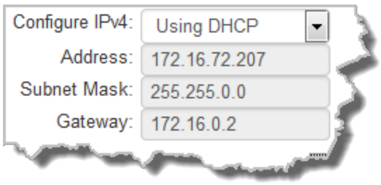
2. Click **Save** to store the new setting.

IPv4 Address Configuration

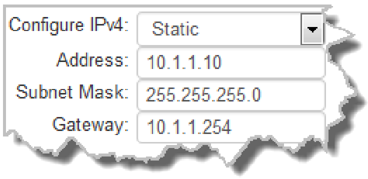
Click to select the **TCP/IP** tab in the **Network panel** to configure settings. The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the drop-down **Configure IPv4** menu. If you want to ignore IPv4 settings select **Off**.



DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider, or to your own local area network (LAN). By default the codec is configured for DHCP-assigned IP addresses.



Static IP addresses are fixed addresses that are recommended for studio installations, so that IP address dialing remains the same over time for incoming codec connections. Note: DNS settings must be specified when a static IP address is configured.



Click **Save** to store all configuration settings.

Note: The **Subnet Mask** is used by the TCP/IP protocol to determine whether a host is on the local subnet or on a remote network. The default **Gateway** is the router linking the codec's subnet to other networks. See your IT administrator for more details.

IPv6 Address Configuration

An IPv6 address is represented by 8 groups of 16-bit hexadecimal values separated by colons (:). The drop-down **Configure IPv6** menu provides three address configuration options:

1. **Automatically**: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. **Manually**: Select to enter static IPv6 address details.

3. **Off**: Select to ignore IPv6 address details.

Important Note: Select **Off** in the drop-down **Configure IPv6** menu if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

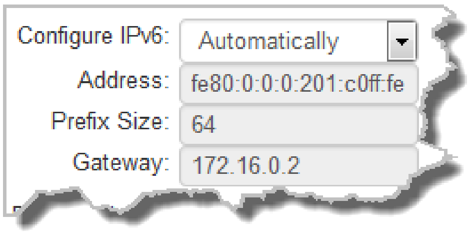
Types of IPv6 Addresses

There are two types of addresses displayed in the IPv6 section:

1. IPv6 address (normally global): A router-allocated IP address with 'global' visibility, details of which are displayed in the **Address, Prefix size** and **Gateway** text boxes.
2. Link Local: A local address which can only be used to connect to another device directly over a LAN. This address is allocated by the codec internally based on MAC address details.

Auto Address Assignment

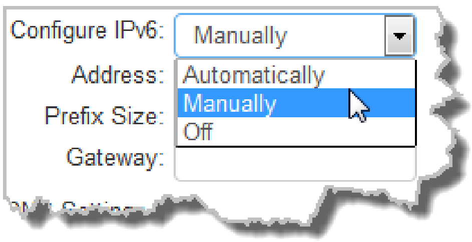
1. Click the **Edit** button in the **Network** panel to configure settings.
2. By default the codec is configured to connect to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.



3. Click **Save** to store all configuration settings.

Manual IPv6 Address Assignment

1. Click the **Edit** button in the **Network** panel to configure settings.
2. To configure IPv6 address details into the codec manually, select **Manually** and enter details into the **Address**, **Prefix** and **Gateway** text boxes.

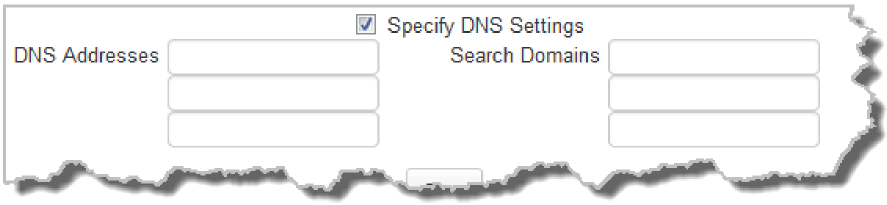


3. Click **Save** to store all configuration settings.

Specifying DNS Settings

It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains**. Note: When installing codecs at the studio it is important to ensure that DNS settings are configured correctly. This is particularly important when using tools like the Cloud Codec Controller. Correct DNS settings are also required for NTP, DDNS, use of host names, and if the **Download and Install** option is used to upgrade codec firmware.

1. Select the **DNS** tab in the **Network panel** to configure settings.



2. Click **Save** to store all configuration settings.

IP Interface Mapping

When dialing over IP you can select the preferred interface to use when establishing a connection. By default **Any** is selected, which means the first available interface is used to dial a connection. The default interfaces in order of use when available are:

1. **LAN1** Ethernet port (default **Primary** interface)
2. **LAN2** Ethernet port (default **Secondary** interface)
3. **USB** (default **Tertiary** interface)
4. **VLAN1** (default **Quaternary** interface)

If none of the primary, secondary, tertiary or quaternary interfaces are available, the codec will look for the first available interface in the following order:

1. LAN1
2. LAN2
3. USB
4. VLANs

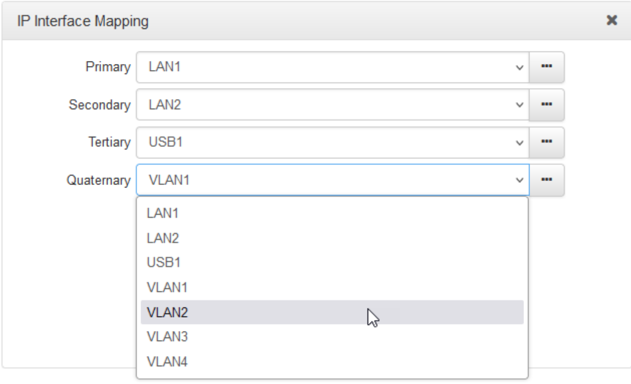
This is determined at the time of connection. For example, if **Any** is selected and VLAN1 is unavailable but VLAN2 is available, then VLAN2 will be used to connect. If a subsequent connection is dialed and VLAN1 becomes available then VLAN1 will be used for the second connection as it has higher priority.

Important Note: VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.

Reconfigure Default Interfaces

It is possible to reconfigure the default **Primary** (Ethernet 1), **Secondary** (Ethernet 2), **Tertiary** (USB) and **Quaternary** (VLAN1) IP interfaces in the codec. As an example, you may want to select **Primary** as the dialing interface in a program and then copy this program onto multiple codecs. However, the actual primary interface used at each location can vary for each codec. For one codec it may be an Ethernet port and for another it may be a Wi-Fi interface. This allows you to configure site-specific settings to suit available network interfaces at different remote locations.

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **IP Interface Mapping** to open this panel.
2. Click the drop-down arrow for each interface to select the preferred default setting.



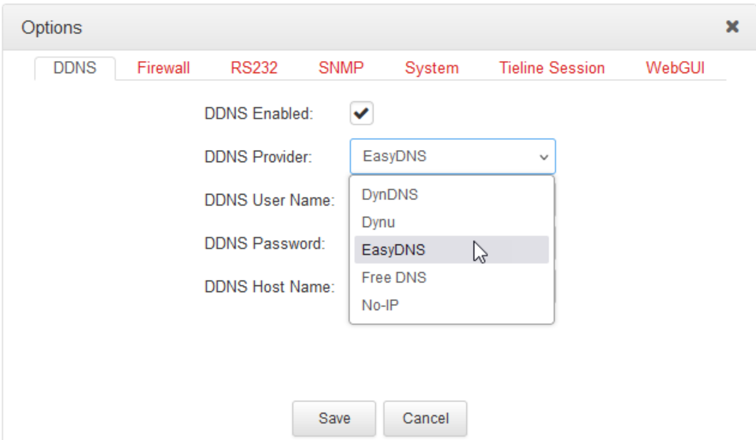
3. Click **Save** to store the configuration.

Important Note: Fuse-IP cannot be configured as a default Primary, Secondary, Tertiary or Quaternary interface.

Configure Dynamic DNS (DDNS)

Dynamic DNS can be configured in the codec to allow the use of a host name to connect when using a dynamic IP address. To facilitate this a DDNS service is used to register a device's IP address to a host name, allowing a remote codec to dial to a host name. The host name will periodically be updated automatically as required, or when the dynamic IP address changes.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **DDNS** and then click **Edit** to adjust settings.
3. Click **Save** to store the new configuration settings.

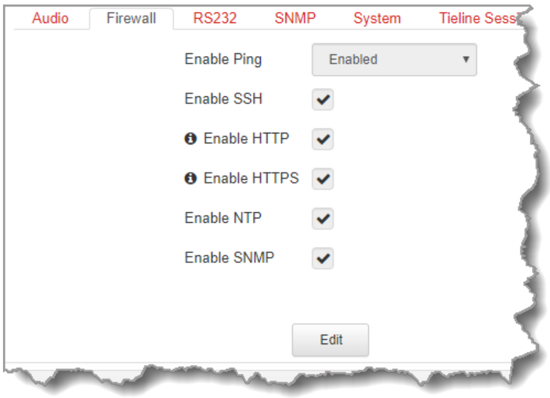


Important Note:
- Supported DDNS providers are listed in the **DDNS Provider** drop-down menu.
- Devices should be DDNS registered to public IP addresses.
- The codec will utilize the **System Internet** interface order when contacting DDNS service providers.
- No connection support to remote G3 codecs.
- DDNS Host Name settings are unrelated to the **Hostname** setting accessed via the **Options panel** under **Settings**.
- Codec DNS settings must be specified when configuring DDNS.

Configure Firewall Settings

The **Firewall** menu can be used to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.

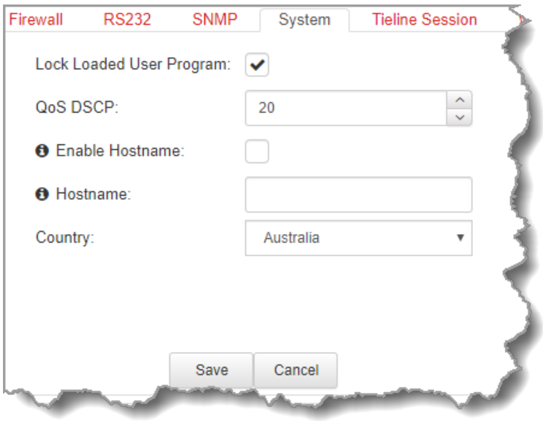
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **Firewall** and then click **Edit** to adjust settings.
3. Click **Save** to store the new configuration settings.



Configure Cross-Site Request Forgery

CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **WebGUI**, then click **Edit** and select the **CSRF Enabled** check-box to enable this feature.
3. Click **Save** to store new configuration settings.

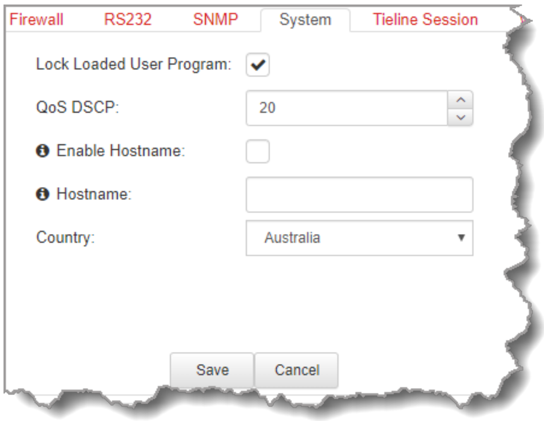


Configuring QoS

The codec can be configured to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted over the network.

1. Open the HTML5 Toolbox web-GUI in a browser on your PC.

2. Click **Settings** at the top of the screen and then click **Options** to display the **Options panel**.
3. Select **System** and then click **Edit**. Click in the **QoS DSCP** text box and enter the preferred value.
4. Click **Save** to store configuration settings.



5. Click the **Save** button at the bottom of the panel to save the new setting.

Important Note: Check with your IT administrator before changing this setting. By default the codec is programmed for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp. For more information on configuring QoS see Configuring QoS in this manual.

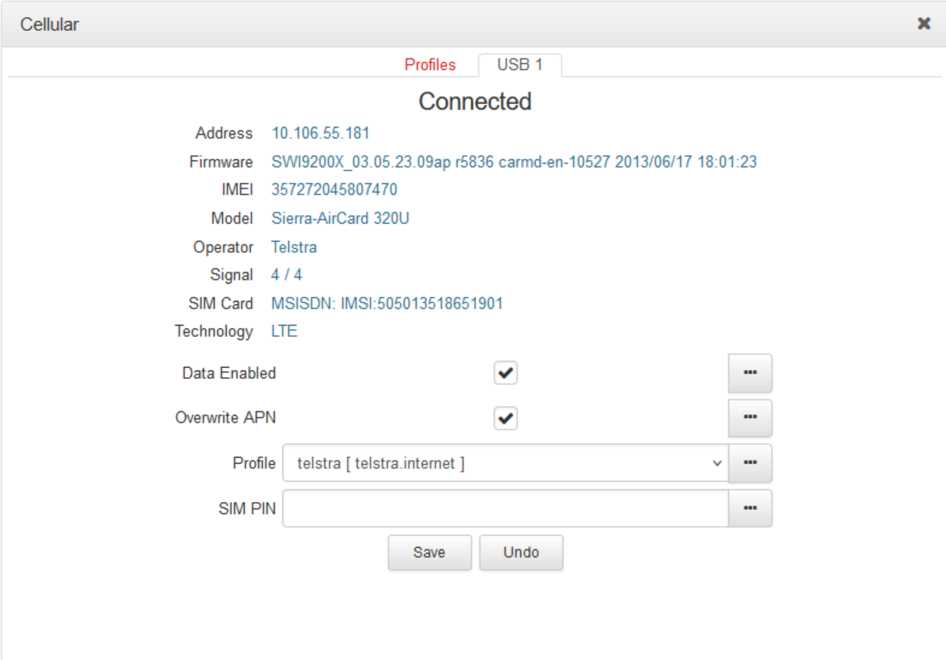
20.4 Configure Cellular Modems

It may be necessary to add a custom access point number (APN) profile if a USB modem or cellular module does not connect to a cellular network. Often this is necessary when you are using different SIM cards in an unlocked USB modem. E.g. When using SIM cards from various carriers in different countries. As a rule of thumb, Tieline recommends always enabling **Overwrite APN** in the **Cellular panel** and entering the correct APN for your Telco. This ensures use of the correct data APN. It may also be necessary to enter a SIM PIN unlock code.

Up to 10 custom access point profiles can be added to the codec. Before configuring the custom access point you need to obtain the access point details from the cellular network provider; this is normally found on a Telco's website. Usually a Telco will list internet and MMS APN information and you need to enter the internet APN details in the codec, as well as the correct authentication type. See Adding Cellular Access Points and a SIM PIN for more details on configuration via the codec **TOUCH SCREEN**.

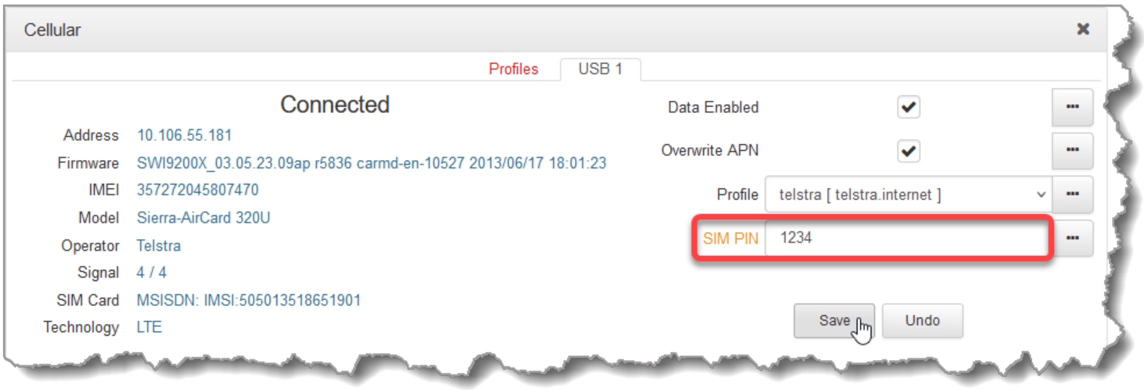
Cellular Configuration

Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Cellular** to view and configure a cellular modem attached to the codec. Data is enabled by default.



Entering a SIM PIN Unlock Code

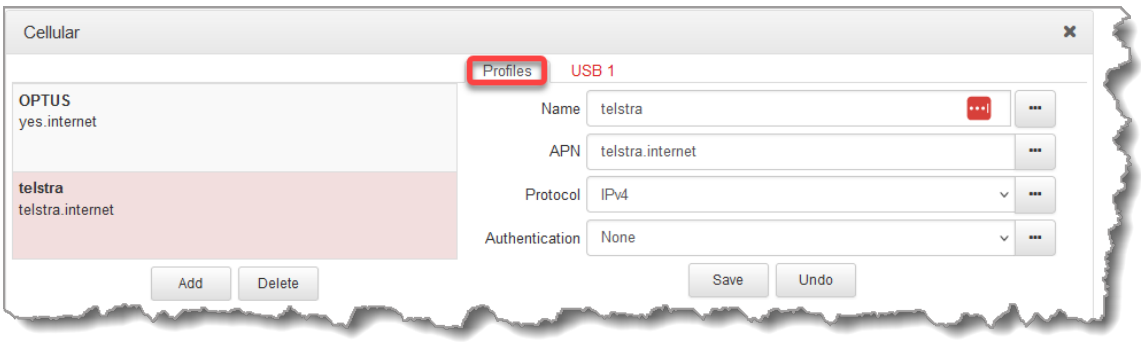
1. Enter the **SIM PIN** in the text box to unlock a SIM card using this feature in a USB modem if it is locked.



2. Click the **Save** button.

Adding a Custom APN

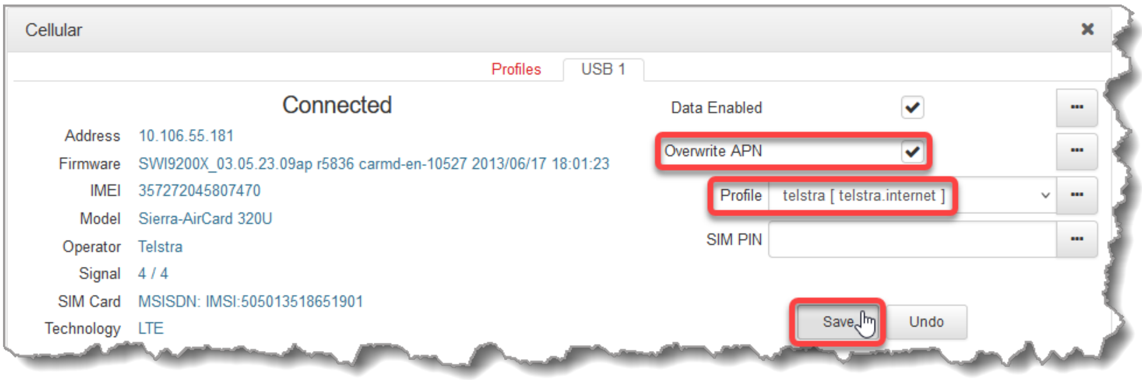
1. Click to select the **Profiles** tab to add a custom APN profile.



2. Enter the APN details and click the **Save** button.

Important Note: Up to 10 custom access point profiles can be added to the codec.

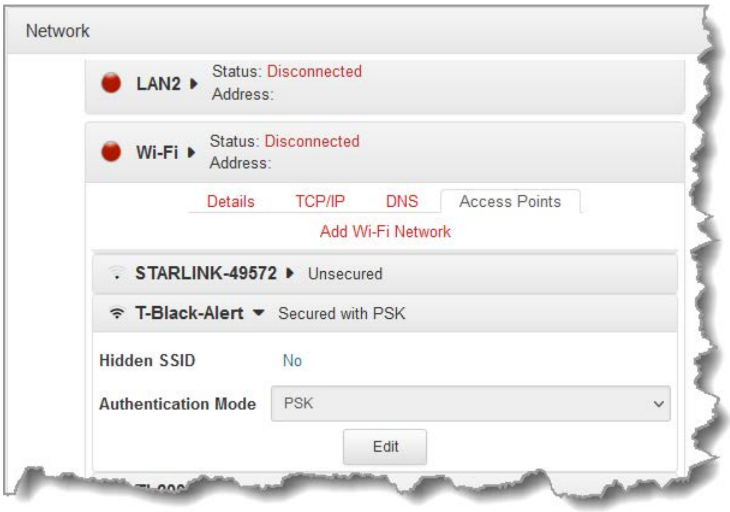
3. Click the **USB tab** and select the **Overwrite APN** checkbox, then choose the correct profile in the **Profile** drop-down menu and click **Save.**



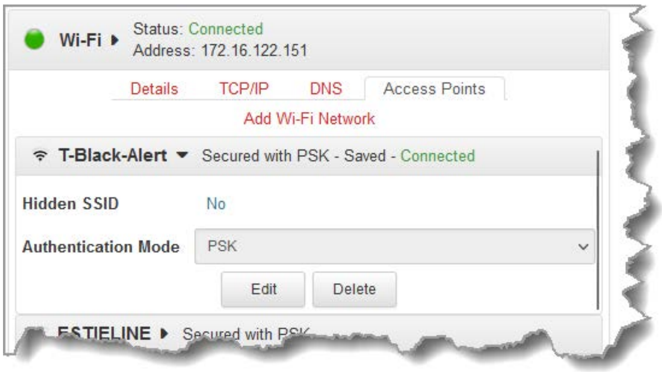
20.5 Connecting Wi-Fi

1. Ensure a Wi-Fi dongle is attached to the **USB PORT** on the front panel of the codec.
2. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **Network** to view and configure Wi-Fi settings using the Web-GUI.
3. Click **Wi-Fi** to display Wi-Fi access points and configuration settings.
4. Click the **Access Points tab** to display available access points. Note: Click the **Rescan Wi-Fi Networks** button to refresh and find access points.

5. Click to select an access point and then click **Edit** to enter a **Password**.



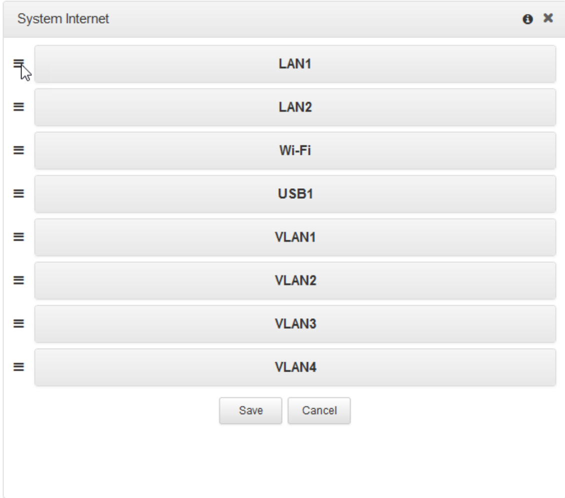
6. Click **Save** to connect to the Wi-Fi access point.



20.6 Configure System Internet Settings

In some situations when multiple interfaces are connected to the codec it may be necessary to select the preference for interfaces used to connect to the internet. For example, when using DDNS services. Or perhaps where the default setting is LAN1, however this may be connected to a closed network and LAN2 is the interface connected to a WAN, i.e. the internet. To adjust the default settings:

1. Open the HTML5 Toolbox Web-GUI and click **Transport** at the top of the screen, then click **System Internet** to display the **System Internet panel**.
2. Click **Edit** and drag the handles on the left-hand side to adjust the order of the listed interfaces.



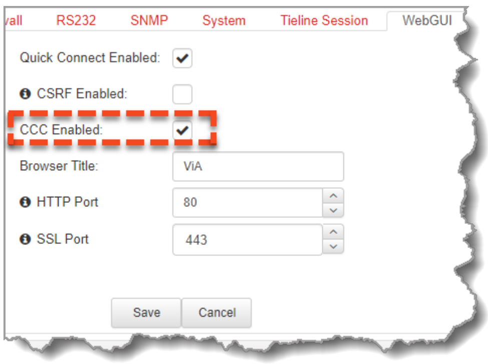
3. Click **Save** to store the new configuration.

Important Notes:
- The **System Internet** setting is only related to internet connectivity. It is not related to the IP Interface Setup for IP streaming interfaces.
- If interface 1 is not connected to the internet, the codec will attempt to use interface 2 and other interfaces listed thereafter in order to access the internet.
- If the codec is connecting to the Cloud Codec Controller and has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. See Enabling the Cloud Codec Controller for more details.

20.7 Enabling the Cloud Codec Controller

To allow the codec to be configured and managed by Tieline's Cloud Codec Controller over the public internet it needs to be enabled for CCC management:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **WebGUI**, then click **Edit** and select the **CCC Enabled** check-box to enable this feature.



3. Click **Save** to store the new configuration.

Important Notes:

- Ensure **CSRF** is disabled in the codec or it will not be able to connect to the CCC. This setting is **[OFF]** by default and is also available in the codec menu via **Settings > WebGUI**, and in the **Options** panel in Toolbox.
- Locally Defined Codecs over a private network do not need to be enabled for CCC operation. Only codecs that require internet access need to be enabled.
- The CCC needs to continually send and receive data between codecs to update information displayed. If the CCC is left open on a computer and is not used for more than 4 hours, the Codec Viewer is placed in 'sleep' mode to save on data use.

DNS and the Cloud Codec Controller (CCC)

If a codec has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. When using the Toolbox HTML5 Web-GUI to configure DNS network settings:

1. Select **Transport** and then **Network** to open the **Network panel**.
2. Select an interface and then select the **DNS** tab to enter details into the **DNS Addresses** fields as required.
3. Ensure that the **Specify DNS Settings** check-box is selected. Note: This will ensure the DNS servers are used.

Note: These settings can also be configured using the front panel of the codec.

Where can I find my DNS settings?

DNS settings for static public IP addresses can be sourced from your internet provider/Telco. Sometimes the Gateway setting in the TCP/IP tab in the Network panel will work as the DNS Address to enter. Another common scenario encountered is that static DNS settings are configured to override DHCP settings when they are not compatible with the network being used. When using DHCP IP settings ensure the **Specify DNS Settings** check-box is disabled in the **Network panel**.

20.8 Configuring TieLink Settings

The TieLink Traversal server centralizes Tieline codec contact list management and provides self-discovery of codecs within customized 'call-groups'. It also provides NAT traversal to simplify connections. Note: The codec serial number is required to create a license for each codec. The following needs to be configured before TieLink connections can be created.

Getting Started with TieLink

1	Register a TieServer Domain	Visit www.tieline.com/register to register a TieServer Domain if you don't have Report-IT Enterprise.
2	Purchase Licenses	If using Bridge-IT codecs purchase TieLink licenses and install them using the 'Licensing panel' in the Toolbox web-GUI.
3	Configure TieLink Traversal Server	1. Log in to TieLink at www.tieserver.com/tielink and add codecs to TieServer. 2. Create a contact list. 3. Set the answering codec as a member of the contact list. 4. Set the dialing codec as a follower of the contact list.
4	Enable TieLink and Connection Interfaces	Enable TieLink and enable interfaces for use with TieLink ('TieLink panel' in Toolbox, or TieLink screen in the codec).
5	Adjust TieLink Interface Priority	Adjust TieLink interface settings in the Toolbox web-GUI 'TieLink panel', e.g. priority of use.
6	Configure Answering Codecs	1. A default program will automatically be 'available' in TieLink and answer calls, or 2. Create a program with "Advertise on TieLink" selected, and 3. Lock this program in the codec.
7	Configure Dialing Codecs	On the dialing codec: 1. Refresh the 'Contact List panel' in the Toolbox web-GUI. 2. Create a program and select a TieLink contact to dial.

To learn how to obtain a domain or add a codec to a TieServer Domain, please visit www.tieline.com and download the TieLink user manual. Please note that there are two ways of connecting using the TieLink Traversal Server:

1. Use the front panel of a codec to select a contact and dial without creating and saving a program.
2. Create a Program with TieLink configuration parameters specified.

TieLink Settings in Toolbox

TieLink needs to be enabled in the codec before it can communicate with the TieLink Traversal Server. To facilitate connections the **TieLink panel** is used to configure interfaces and options like STUN. To view the panel open the HTML5 Toolbox Web-GUI and click **Transport** and then **TieLink** to open the **TieLink panel**. Note: Ensure that the DNS Server settings are also configured because the codec must be able to resolve with Tieline's various TieLink servers in order to be able to contact them.

1. Enable TieLink

Click **Enable TieLink** to enable TieLink connectivity.

2. Enable Interfaces

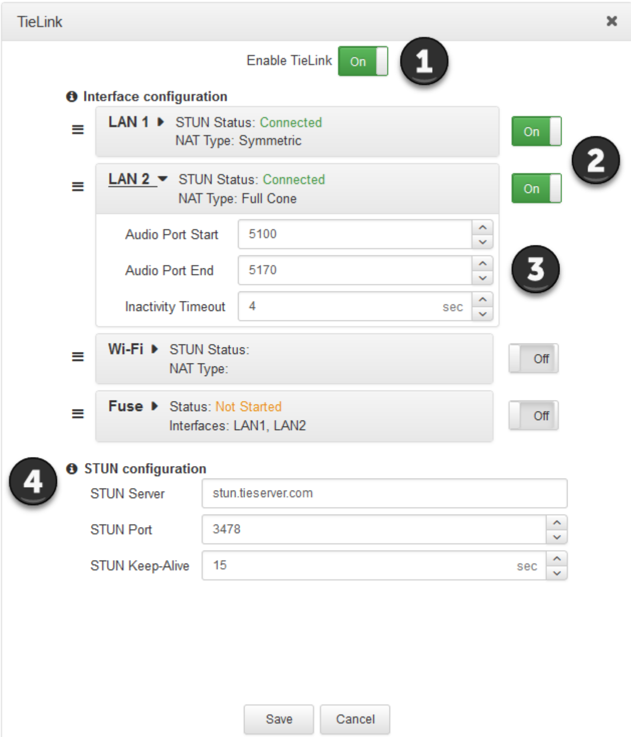
Click the **ON/OFF** button to enable each individual interface and/or select the interfaces used for **Fuse-IP** connections (Note: This is a simpler way to configure Fuse-IP. TieLink supports one connection to a single codec using Fuse-IP).

3. Adjust Tieline Interface Settings

Adjust TieLink interface ports, inactivity timeouts and Fuse-IP interface selections.

4. STUN Configuration

A STUN server is required to use TieLink. Adjust TieLink **STUN configuration** and use a different STUN server if required, e.g. to use a different port.



Important Notes:
UDP ports can be used for both the outgoing and incoming ports when using TieLink. The codec will treat the port range configuration as a pool of ports to be used. The actual allocation can appear to be in order, but this is not guaranteed. Users should assume that the allocation may be random.

STUN messages are sent via UDP and STUN port 3478 refers to the STUN server configuration and not the local system (codec). In a general sense, the way STUN works is that the codec will pick a local port (e.g. any port in the range 5100-5170), and then send a UDP message to the server's port 3478. In compatible NAT, this will create a mapping between the local address/port and the external address/port. The server will receive the message as coming from the external address/port and send a message back through the same address/port to let the codec know that the mapping has been successful (i.e. a hole/path was formed). The codec can now advertise this external address/port mapping to reach the codec through NAT. TCP is not used during this process. Note: The codec will also need to send/receive TCP packets to codeclounge.com port 443. This is used to coordinate connection establishment.

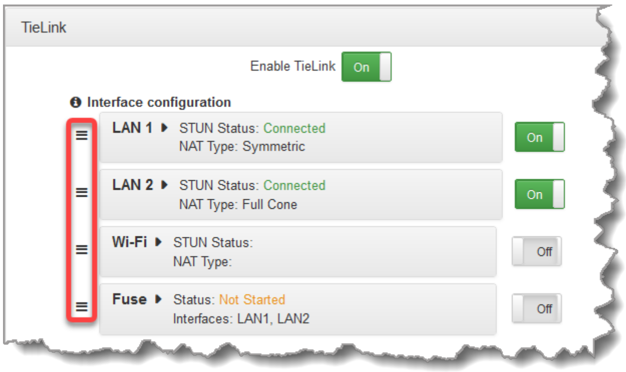
When using TieLink and SmartStream PLUS, if you want to configure redundancy between say LAN1 and LAN2, the port range can be the same or different.

A green **TieLink Symbol** is displayed at the top of the Toolbox HTML5 Web-GUI when TieLink has been enabled successfully. Note: the symbol is orange if STUN is not successfully enabled on any interface.

Reorder TieLink Interface Priority

When the codec attempts to connect to a TieLink contact, local interfaces are used in the order in which they are listed. For example, if both **LAN1** and **Wi-Fi** are configured for use when dialing a contact, TieLink will attempt to use these interfaces in the order in which they are listed in the panel. It is possible to reorder local TieLink interface priority in the **TieLink panel** in Toolbox.

1. Click **Edit** in the TieLink panel.
2. Click the **Drag Handle** to the left of an interface to drag and reorder it.



3. Click **Save** to save the new settings.

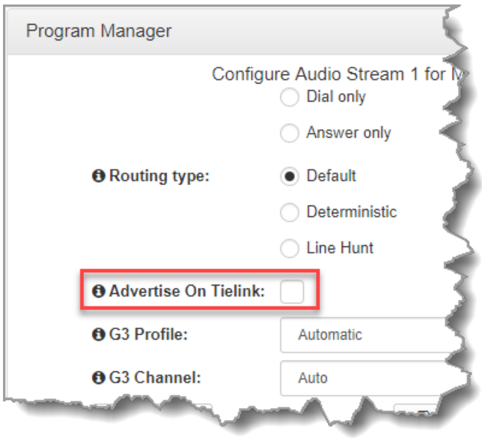
Configuring TieLink Connections in the Program Manager Panel

Important Note: The following settings can also be edited in the **Connections panel**.

Include a Codec's Answering Audio Streams in TieLink Contact Lists

A default program will automatically display audio streams as ‘available’ in TieLink and allow answering of calls from codec 'followers' in a contact list. When creating custom programs, from an answering perspective the **Program Manager panel** is used to select which answering streams are "Stream Members" visible within a TieLink contact list. I.e if an answering stream is 'advertised' on TieLink then is becomes an available endpoint for other codecs to dial.

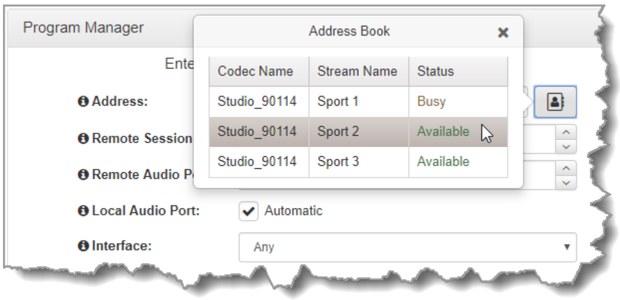
When creating a program select the **Advertise On TieLink** check-box to register answering audio streams on a loaded program with the TieLink Traversal server. These audio streams will be visible in TieLink contact lists which include this codec's audio streams. In other words, the answering audio streams will appear as available to dial (or "Busy" when connected) by other codec 'followers' in the same contact list.



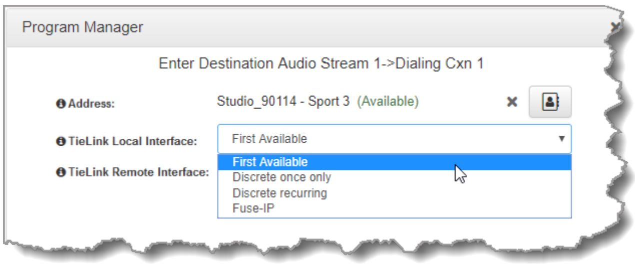
Select a Contact to Dial using TieLink

From a dialing perspective, the **Program Manager panel** is used to select a contact to dial from within a TieLink contact list that has been shared with a codec. When creating a program:

1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.

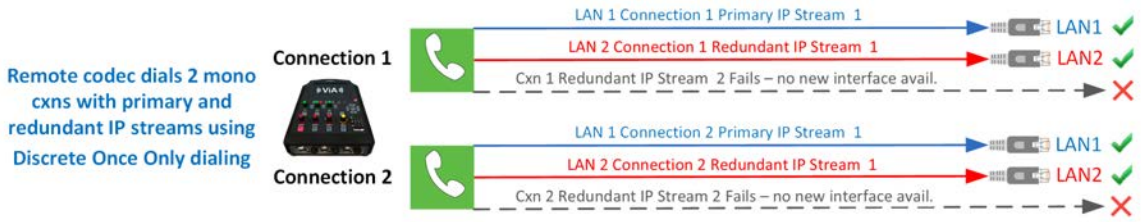


2. Select the interface options to use on local and remote codecs when dialing TieLink connections.

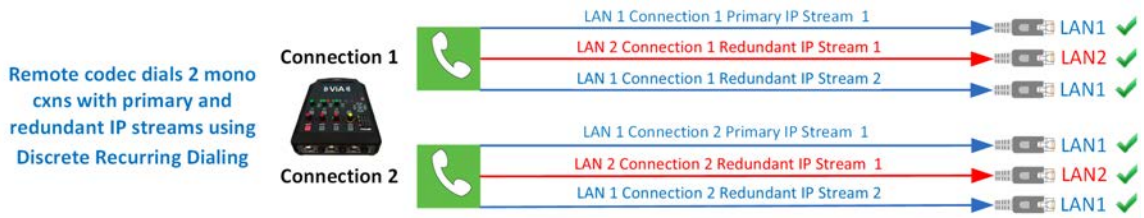


i. **First Available**: The first available interface that can successfully connect will be used to dial. If you dial two connections the same interface will be used for both connections, e.g. LAN1, even if another interface is available.

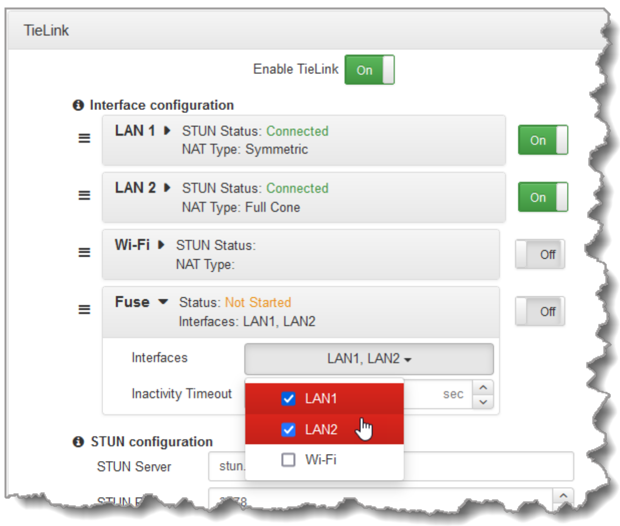
ii. **Discrete once only**: The first interface not being used by another connection will be used to dial. In the following example using two LAN interfaces, the codec connects a primary and redundant IP stream using different interfaces for each connection. A second redundant IP stream configured on each connection fails because both available interfaces are already being used for that connection.



iii. **Discrete recurring:** Uses the first interface that is not being used by another connection (within the same audio stream). After all interfaces have been used, each audio stream will scan each interface from the start again, in the order in which they are listed, and connect using the first available interface that will successfully dial and connect. The dialing order of the interfaces in the TieLink panel can be adjusted by clicking a dragging interfaces on the left side of the panel when it is in **Edit** mode.



iv. **Fuse-IP**: Connect using Fuse-IP. This is configured in the **TieLink panel** as displayed in the following example. Only one interface is available on the codec, however codecs bonding data using multiple interfaces over Fuse-IP can connect to the codec. Please note that this is the simplest way to configure a Fuse-IP connection.



Important Note: Ensure an answering program is configured for Fuse-IP on the destination codec and locked. Also ensure TieLink and relevant interfaces are enabled on both codecs.

Helpful Hints for Troubleshooting

If TieLink Does Not Connect:

1. Ensure TieLink and at least one relevant interface is enabled and 'stunned' on both codecs; i.e. the **STUN Status** for the interface is **Connected**.

2. Check STUN server settings.

If a TieLink Fuse-IP Connection Does Not Connect:

1. Ensure Fuse-IP is enabled in the **TieLink panel** in Toolbox, or in the codec's **TieLink Settings screen**, for both the dialing and answering codecs
2. Check at least one interface has been added to the Fuse-IP configuration in the **TieLink panel** in Toolbox, or in the codec's **TieLink Settings screen**, for both the dialing and answering codecs
3. Verify that the **STUN Status** for at least one interface in the Fuse-IP configuration is **Connected** on both the answering and dialing codecs.
4. Check that the answering codec is not already connected using Tielink Fuse-IP. Only one Fuse-IP connection is currently supported.

If SmartStream PLUS Connections do not connect:

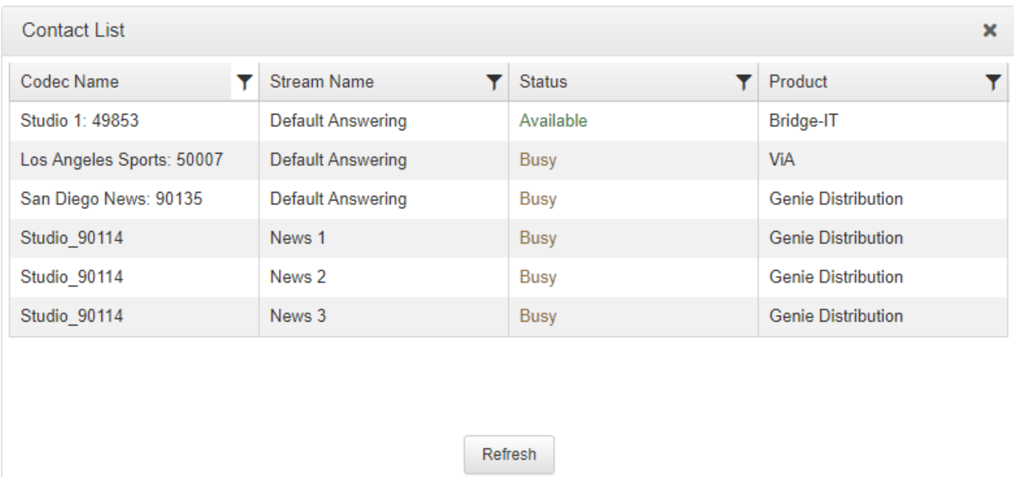
1. Based on the priority order of interfaces in the TieLink panel, a SmartStream PLUS connection may automatically use TieLink Fuse-IP as one of the connections if Fuse-IP is enabled at the dialing or answering codec. If TieLink Fuse-IP is chosen, then that connection will only succeed if the Fuse-IP stream is available at both ends. Disable Fuse-IP as an interface in TieLink to avoid Fuse-IP being selected in this scenario.
2. When "Discrete once only" or "Discrete recurring" is selected, with TieLink Fuse-IP enabled, it may be that TieLink Fuse-IP is selected to transmit one of the streams. In this situation, if the answering codec is already connected to another codec using TieLink Fuse-IP, then the connection will fail for this stream because only one Fuse-IP connection is supported using TieLink.
3. When "First Available" is selected, TieLink Fuse-IP may be used if it is the first interface "ready" in the **TieLink panel** interface list. In this situation, the connection will fail if the answering codec is already connected to another codec using TieLink Fuse-IP.

Configuration Restrictions for TieLink

Tielink only supports one SmartStream PLUS redundant streaming connection on each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.

Updating TieLink Contacts

A TieLink administrator updates codecs included in, and excluded from, Contact Lists. Codec members of a Contact List shared with a codec are displayed from the **Menu Bar** by selecting **Connect > Contact List** to open the **Contact List** panel.



Codec Name	Stream Name	Status	Product
Studio 1: 49853	Default Answering	Available	Bridge-IT
Los Angeles Sports: 50007	Default Answering	Busy	ViA
San Diego News: 90135	Default Answering	Busy	Genie Distribution
Studio_90114	News 1	Busy	Genie Distribution
Studio_90114	News 2	Busy	Genie Distribution
Studio_90114	News 3	Busy	Genie Distribution

If a contact list is adjusted in TieLink it is necessary to click the **Refresh** button in the **Contact List panel** to display the latest contact info. This updates the **Contact List panel**, selectable contacts in Toolbox, and the codec **TOUCH SCREEN** when dialing and creating programs.

Stream Status in TieLink

The **Contact List panel** displays the availability of a TieLink audio stream. The status may display:

1. **Available**: The TieLink Traversal Server Network has provided reachable streaming interfaces and indicates the audio stream is not busy.
2. **Busy**: The codec is connected to the TieLink Traversal Server network and TieLink indicates the stream is busy.
3. **Offline**: The audio stream is unavailable within the TieLink Traversal Server Network. Possible reasons an audio stream may be offline include:
   a. The codec is offline.
   b. An audio stream is not advertised on TieLink.
   c. TieLink and/or TieLink interfaces are disabled in the codec.
   d. Ports are not open.
   e. The **Contact List panel** in the Toolbox Web-GUI may need 'refreshing'.
   f. DNS - Not resolving the STUN server set in the **TieLink panel**. Note: If a codec has a statically configured IP address, ensure that DNS Server settings are also configured.
   g. DNS - Not resolving https://tieserver.com
   h. The codec firmware version is not compatible.

20.9 Configure Fuse-IP Bonding

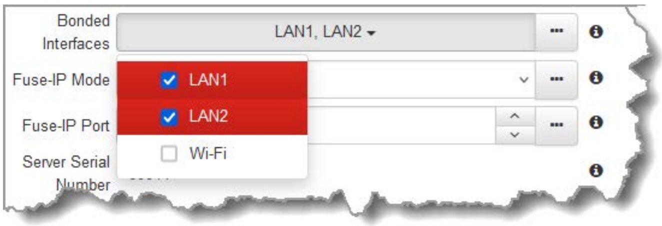
Fuse-IP is a proprietary Tieline IP bonding technology which aggregates data by bonding multiple IP interfaces (peers) and establishing a “tunnel” between two Tieline codecs. A streaming connection can be established after the tunnel is created. Fuse-IP automatically distributes data over any two bonded interfaces, e.g. dual Ethernet LAN ports. There are several benefits in using Fuse-IP to aggregate data from multiple IP interfaces, including:

- The ability to create more stable connections with higher overall data bandwidth.
- Greater choice of encoding algorithms because of higher available bandwidth.
- Redundancy in case one IP connection is lost.

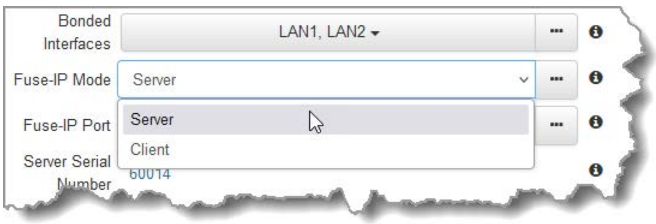
For more details on setting up Fuse-IP using the codec front panel see Configuring Fuse-IP Bonding.

Configuring a Fuse-IP Server at the Studio

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Fuse-IP**.
2. Click the **Bonded Interfaces** drop-down menu to select the interfaces to be bonded.

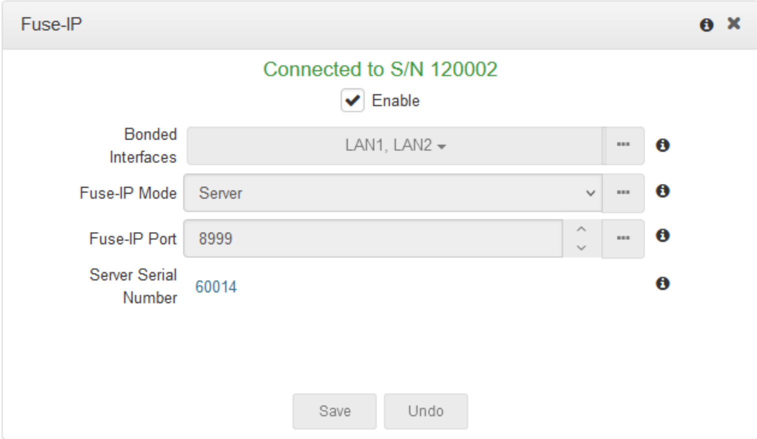


3. Click the **Fuse-IP Mode** drop-down menu and select **Server** if the codec is not initiating the connection.



4. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.

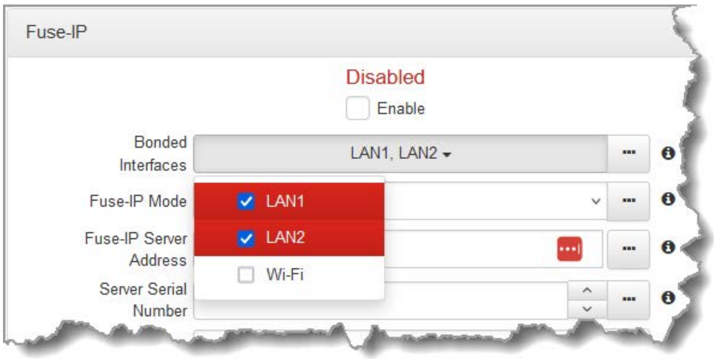
5. Click the **Save** button to store all settings.
6. Click the **Enable** check-box to create a Fuse-IP tunnel between the server and client codecs. Note: **Connected** should be displayed after a few seconds if both codecs are configured correctly and Fuse-IP is enabled on both codecs.



6. Double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after turning Fuse-IP **On**.

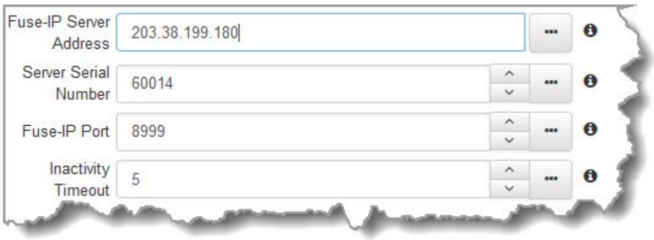
Configuring a Fuse-IP Remote Client

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Fuse-IP**.
2. Click the **Bonded Interfaces** drop-down menu to select the interfaces to be bonded.

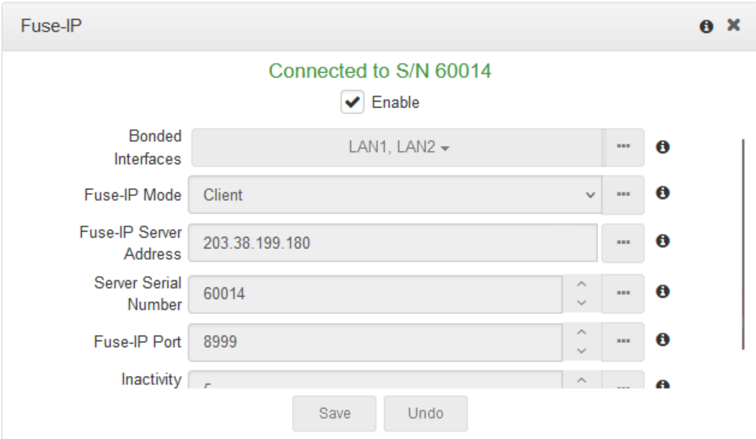


2. Click the **Fuse-IP Mode** drop-down menu and select **Client** as the Fuse-IP mode if you are dialing the studio codec. Note: the studio codec should be configured in server mode.

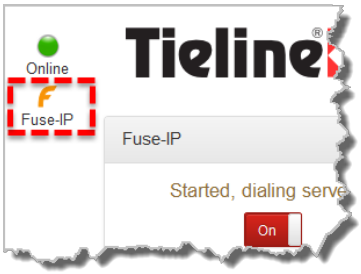
3. Enter the **Fuse-IP Server Address**, which is the public static IP address of the server codec at the studio. Then enter the server codec's serial number in the **Server Serial Number** text box. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same. Configure the **Inactivity Timeout** if you want to turn the Fuse-IP tunnel off after a predetermined time period to save data, then click **Save**. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter **0** to disable the timeout.



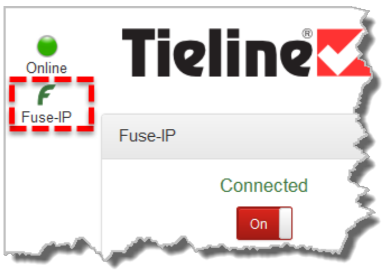
4. Click the **Enable** check-box to create a Fuse-IP tunnel between the server and client codecs. Note: **Connected** should be displayed after a few seconds if both codecs are configured correctly and Fuse-IP is enabled on both codecs.



5. The status indicator is orange when Fuse-IP is enabled but no tunnel is created. It turns green when a tunnel is active. Note: Double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after disabling Fuse-IP.



Fuse-IP Enabled & no tunnel created



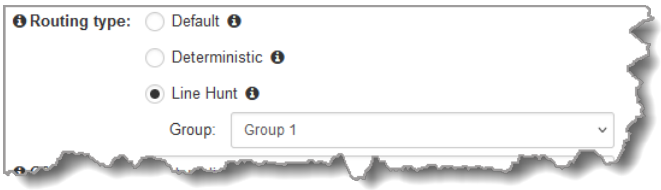
Fuse-IP enabled & tunnel created

Important Notes:

- Data is sent by the codec over the newly created 'tunnel' as soon as Fuse-IP is enabled, even if a connection has not been configured and dialed. Depending on the number of interfaces being used, the codecs may transmit and receive up to 24MB of data per hour at each end of the link.
- The codec remembers the Fuse-IP enabled/disabled state on power up.
- For additional stability it is recommended that a fixed jitter buffer is configured when streaming using Fuse-IP. The actual jitter buffer depth should account for the difference in delay between the interfaces and the maximum jitter experienced. To determine the jitter over each link you can connect and stream audio over each interface separately and look at the jitter reading displayed on the **Connection Statistics** screen.
- Use a dotted quad IPv4 address when configuring the Fuse-IP Server Address.

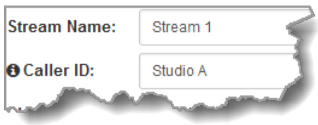
20.10 Line Hunt Call Answering

The codec supports line hunt call answering, whereby line hunt groups can be assigned for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. As an example, when dialing a codec using a multi-stream program with line hunt groups configured, select **Line Hunt** as the routing type. Then select the group to which the audio stream should be routed by the answering codec, e.g. **Group 1** in the following example.

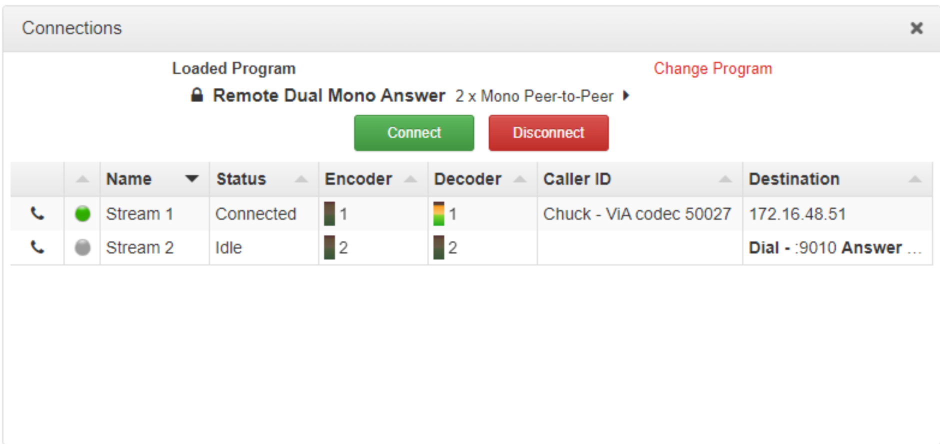


Incoming Caller ID

Tieline codecs also support incoming caller IDs, so you can uniquely identify codecs or Report-IT users when they call in. This is particularly useful for identifying inbound callers when using line hunt answering mode.

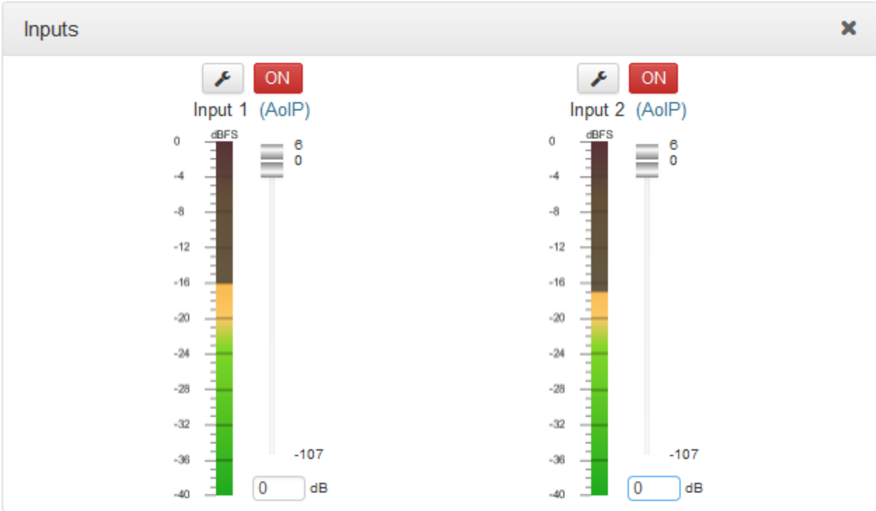


Any Tieline G5 or G6 codec can display a designated **Caller ID** in the **Connections panel** when dialing. In the following example, a ViA codec has called into the codec using the caller ID **Chuck - ViA 50027**.



20.11 Configuring Input Settings

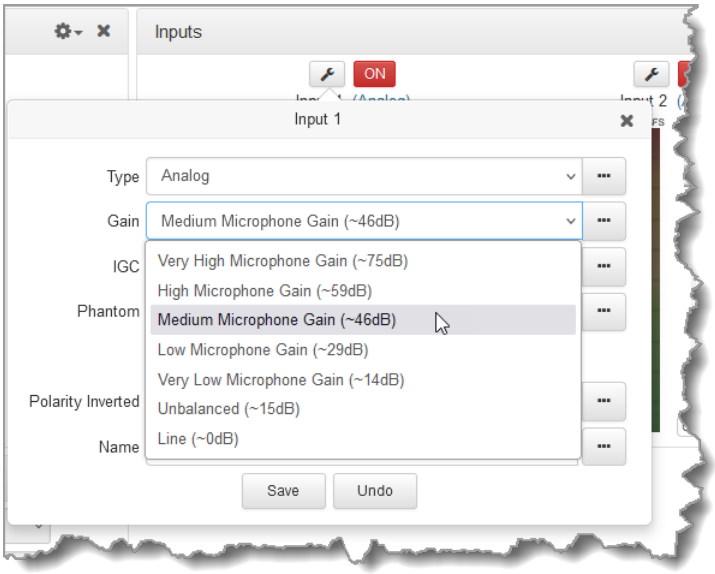
Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Inputs** to display the **Inputs panel**.



Important Note: 15 volt phantom power can only be supplied on analog input 1; this is disabled by default.

Adjusting Analog and Digital Audio Levels

Gain on **Input 1** can be configured for mic, unbalanced or line level sources. **Input 2** accepts line level only.



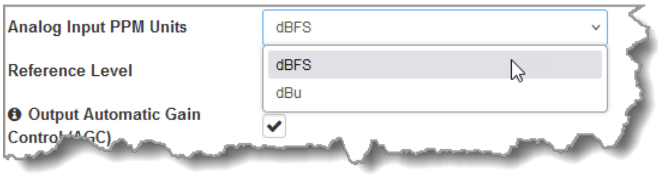
To adjust input audio levels, click on the input slider and drag it to the desired input gain level. Alternatively, type a value below a PPM meter to increase or decrease the input level value. Note: 0.5dB increments are accepted for increasing and decreasing audio levels. Input levels on the **Inputs panel** should be set to ensure audio peaks average at the first yellow indications on the PPM meters, which represents nominal 0 VU at -20dBFS. Audio levels should also be verified using the meters in the PPMs panel.

Important Note: There is a maximum of 6dB of additional gain available when adjusting a digital input.

Changing the Input PPM Meter Units from dBFS to dBU

It is also possible to switch the analog input PPM meter unit of measurement from dBFS (default) to dBU:

1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**, next click **Edit**.
2. Click the **Analog Inputs PPM Units** drop-down menu and select **dBU**, then click **Save**.

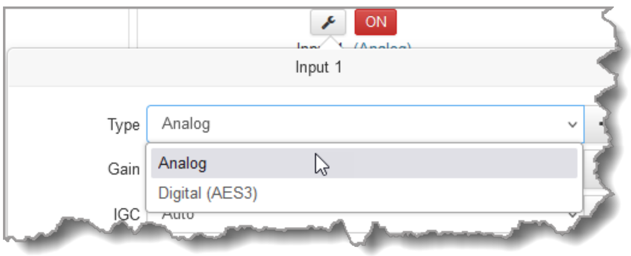


Configuring Input Settings

Selecting Analog and Digital Audio Sources

Codec inputs can be configured for AES3, digital (e.g. AES67, ST 2110-30, Livewire+, RAVENNA), or analog line level audio sources. Use the **Audio Options panel** in the HTML5 Toolbox web-GUI to configure digital inputs globally for either AES3 or AoIP input sources. See Input Configuration, Levels and PPMs for more details. After setting the **Digital Input Type** in the **Options panel**, configure the codec for either analog or digital input types using the **Inputs panel** as follows.

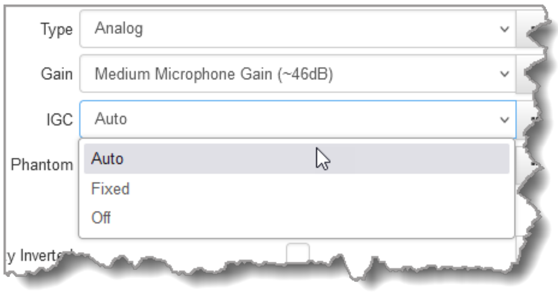
1. Click the **Input Settings** symbol.
2. Select **Type** and then **Analog** or **Digital** (**AES3** or **Native AoIP Protocols**).



3. Click **Save** to confirm the new setting.
4. See Configuring AES3 Audio for more information about AES3 digital inputs and outputs.

Adjusting IGC

1. Click the **Input Settings** symbol.
2. Select **IGC** (Intelligent Gain Control) and then **Auto**, **Fixed** or **Off** as required.



3. Click **Save** to confirm the new setting.

Phantom Power

Input 1 is a balanced mic/line input with the ability to connect high, medium and low gain mics, as well as an unbalanced source. It has switchable phantom power of 15 volts that is turned off by default and can also be used as an AES3 (AES/EBU) digital input. To turn phantom power on:

1. Click the **Input Settings** symbol.
2. Select the **Phantom** check-box to enable 15 volt phantom power on input 1.

3. Click **Save** to confirm the new setting.

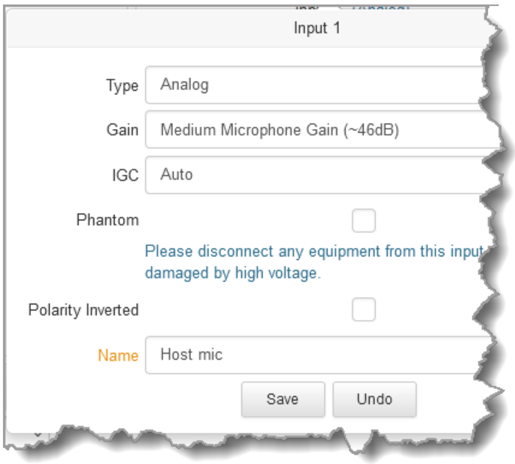
Invert Polarity

1. Click the **Input Settings** symbol.
2. Click to select the **Polarity Inverted** check-box.

3. Click **Save** to store the new setting.

Renaming Inputs

1. Click the **Input Settings** symbol on the input being renamed.
2. Click in the **Name** text box to enter a new name, or edit an existing name.

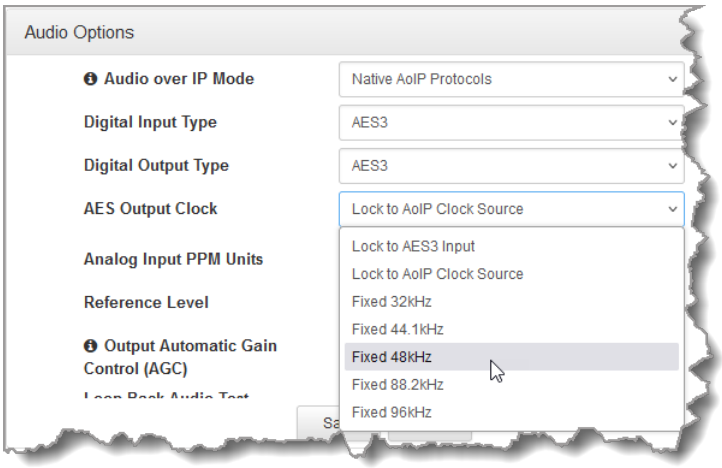


3. Click **Save** to confirm the name change.

AES3 Output Sample Rate Configuration

The AES3 output sample rate can be configured using the HTML5 Toolbox Web-GUI. See Configuring AES3 Audio for more information about AES3 input and output sample rate clocking.

1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**.
2. Click the **AES Output Clock** drop-down menu to select your preferred setting, then click **Save**.



Audio Reference Levels

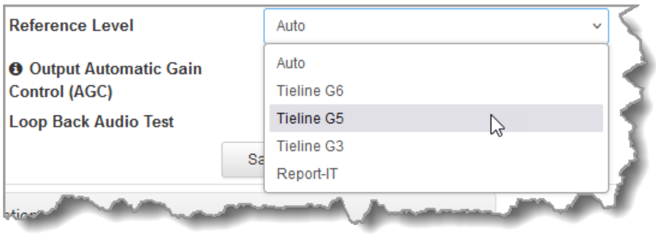
By default, the **PPM METERS** on the **OLED SCREEN**, or the HTML5 Toolbox Web-GUI, use dBFS to express nominal operating, headroom and noise floor levels. The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G5** reference scale setting.

The default Tieline G6 audio reference scale displayed on the PPMs when connecting to a Tieline G6 codec is -40dBFS to 0dBFS. Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 20dB of headroom from the nominal 0vu point.

Important Note: If a codec supports multiple stream programs and the **Auto** (default) reference level is selected, the first codec to connect will configure the reference level used for all subsequent connections. I.e. If a G5 codec connects first, then the G5 Audio Reference Level will be configured for all connections.

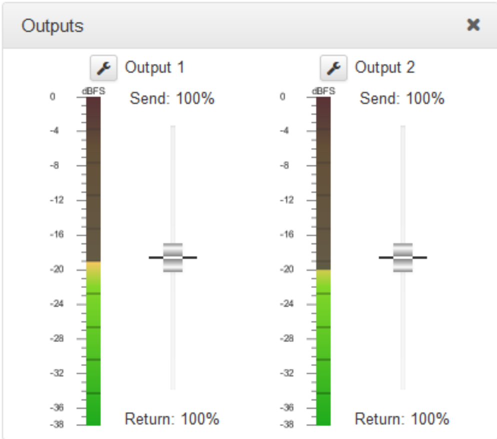
To adjust this setting:

1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**.
2. Click the **Reference Level** drop-down menu and select the correct option, then click **Save**.



20.12 Configuring Output Settings

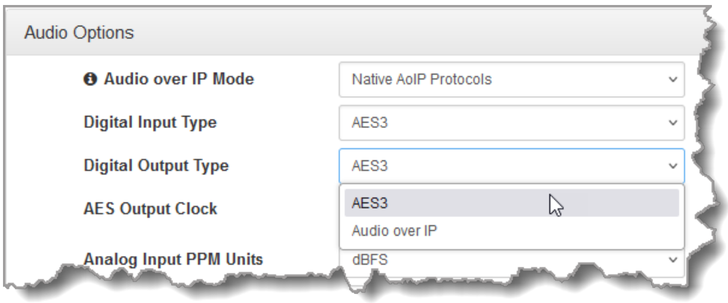
The **Outputs panel** can be configured to support either analog or digital output types for outputs 1 and 2. Configuration of the digital output type, either AES3 or AoIP, is configured for all outputs using the **Audio Options panel**.



Selecting the Digital Output Type

To select the digital output type for both outputs:

1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**.
2. Click the **Digital Output Type** drop-down menu and select either **Audio over IP** or **AES3**, then click **Save**.

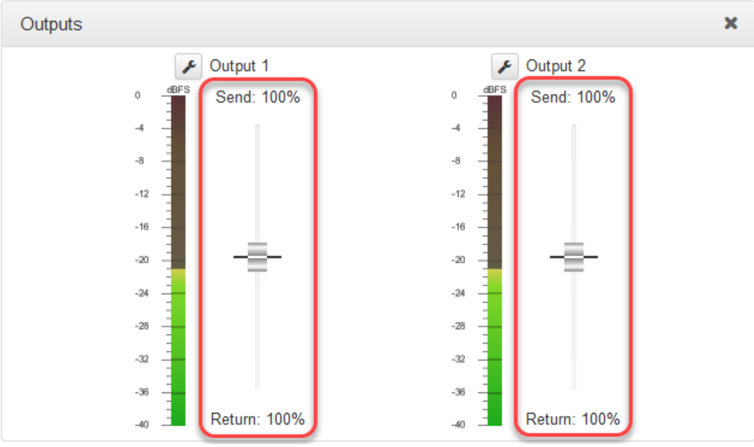


Important Notes:

- Audio is fed simultaneously to both the analog and digital outputs on the codec.
- AES3 audio is sent to the outputs, as well as AoIP audio, regardless of the Digital Output Type setting. However, there is an important point to note. When both the Digital Input Type and Digital Output Type on the codec is set to AES3, the AES Output Clock can be changed. However, when either the Digital Input Type or Digital Output Type is set to AoIP, the AES Output Clock is locked to the AoIP clock (i.e. PTP in AES67, WNIP clock in WNIP mode etc). Since the AES Output Clock also sets the synchronization of the AES67 sources, it is important that the AoIP clock is selected when using AES67 output. If it is not set to use the AoIP Clock, then the AES67 output will usually appear to be ok initially, but the timestamps will eventually deviate from the master clock reference (e.g. PTP) and some devices may stop receiving the streams when the timestamp deviates too far. So, when using AoIP outputs, it's always best to set the Digital Output Type to AoIP. If AES3 outputs are also being used, then the AES3 output sampling clock will be locked to the AoIP reference clock.

Output Send / Return Balance

The balance between send and return audio signals fed to each output can be individually adjusted using each output slider. This provides flexibility and the ability to feed input sources and decoded incoming signals to outputs as required.



In many situations incoming **Return** decoder audio will be the only audio fed from an output, however there may also be a requirement to mix input audio with return audio. Or input (**Send**) audio may be fed directly to an output for AES67/ST 2110-30 streaming.

Output Assignments for AoIP Source Audio Streams

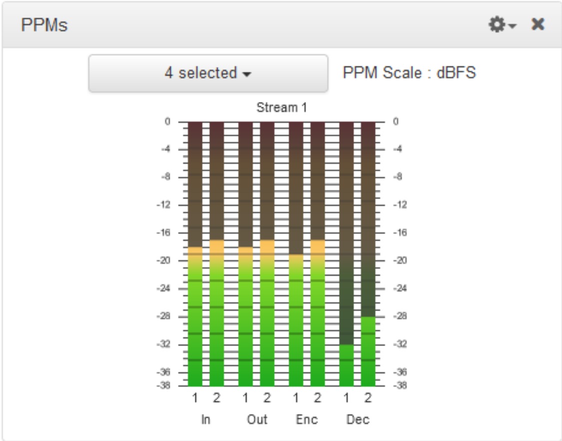
When streaming in AoIP mode (e.g. AES67, ST 2110-30, Livewire+, RAVENNA), the hard coded **Matrix Editor** output channel assignments depend on the stream configuration mode, as displayed in the following table.

AoIP Audio Stream	Mode	Mix Outputs Used
Source 1	Mono	1
	Stereo	1 and 2
	8 Channel	1 to 8
Source 2	Mono	2

Verify audio is displayed on the PPMs in the **Sources** panel before commencing AES67 streaming.

20.13 Monitoring PPMs

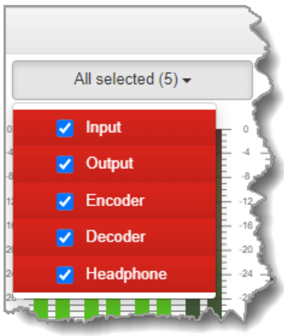
The **PPMs panel** is able to monitor incoming and outgoing audio streams.



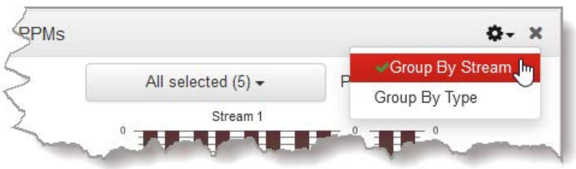
Click the drop-down PPM selector button to select whether the panel will display PPMs for:

- Inputs
- Outputs
- Encoders
- Decoders
- Headphone (output)

Select the check-boxes to select or deselect the PPMs displayed for each audio stream.

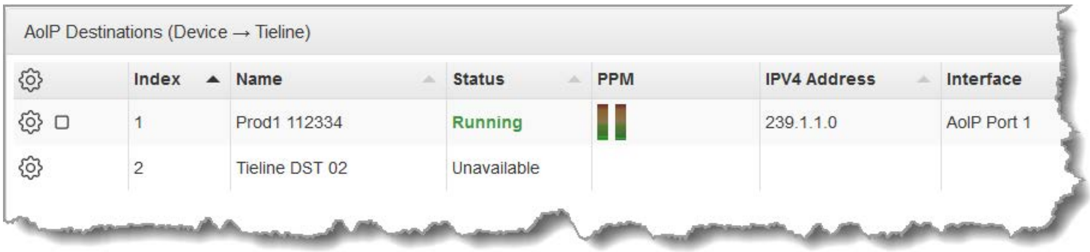


Click the **Options** menu in the panel to choose whether PPMs are grouped by stream or by type.



AoIP Stream PPMs

PPMs for AoIP (e.g. AES67, ST 2110-30, Livewire+, RAVENNA) streams are displayed in the **Destinations panel** and **Sources panel** in the Toolbox HTML5 Web-GUI and the AoIP Web-GUI. For more details please download the AoIP User Manual.



AoIP Destinations (Device → Tieline)

	Index	Name	Status	PPM	IPV4 Address	Interface
	1	Prod1 112334	Running		239.1.1.0	AoIP Port 1
	2	Tieline DST 02	Unavailable			

Toolbox Destinations panel displaying incoming codec audio

Important Note: PPMs on the AoIP **Destinations panel** are inactive if the **Digital Input Type** in the **Audio Options panel** is not set to **Audio over IP** and the corresponding **Inputs panel** input **Type** is not set as **Digital**.

20.14 Headphone Monitoring

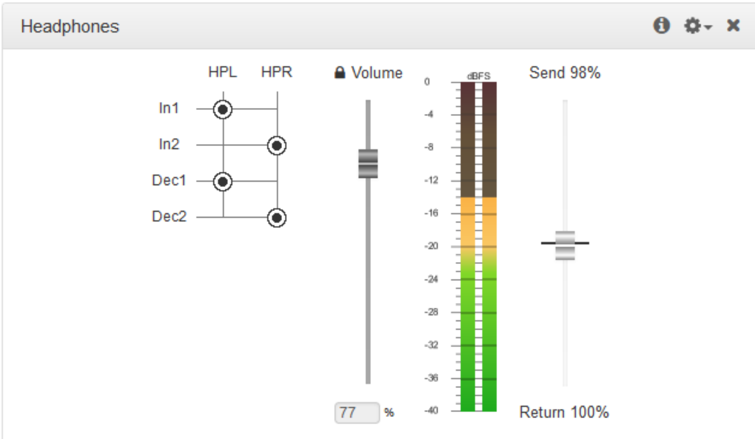
The **Headphones panel** in the HTML5 Toolbox provides customization of audio source monitoring for the 6.35mm (1/4") RTS stereo headphone output. The volume and send/return balance between incoming and outgoing audio sources can also be adjusted.

Adjusting the Headphone Mix and Monitoring

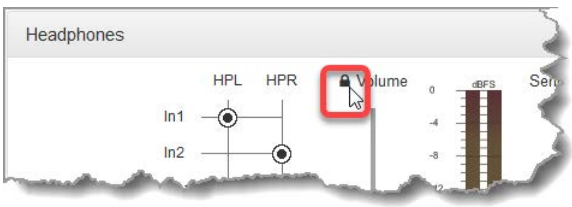
WARNING: LISTENING TO AUDIO AT EXCESSIVE VOLUMES CAN CAUSE PERMANENT HEARING DAMAGE. REDUCE VOLUME AS LOW AS POSSIBLE.

Use extreme caution when remotely adjusting the headphone volume using the Web-GUI. This should always be done in consultation with users at the remote codec to ensure they are not adversely affected by excessive volumes. A headphone volume limiter can be employed to protect hearing when monitoring loud sources and/or when using low impedance headphones.

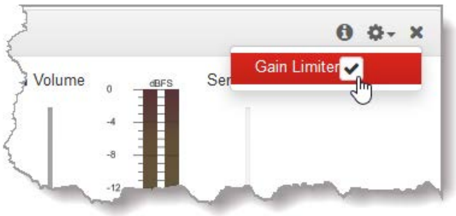
To adjust settings open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Headphones** to open the **Headphones panel**.



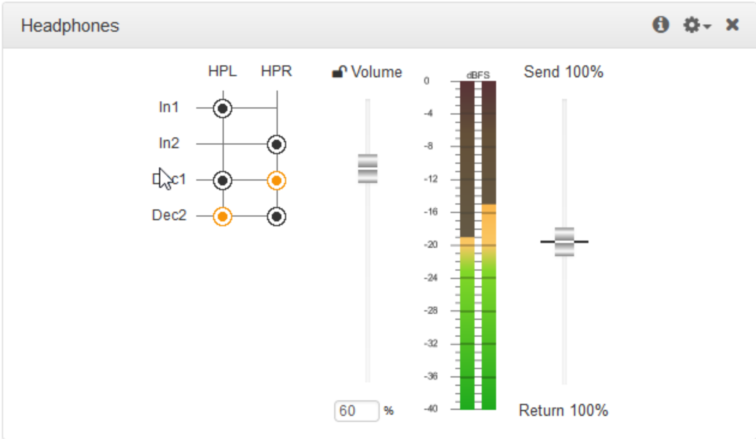
Click the **Headphone Adjust Lock symbol** to enable adjustment of headphone output volume. **WARNING: This may adjust the headphone volume of someone listening to a headphone output at a remote location. Extreme caution is recommended when this setting is enabled. High headphone levels can cause hearing loss!**



Click the **Options symbol** to enable or disable the headphone **Gain Limiter**. **WARNING: This may adjust the headphone volume of someone listening to a headphone output at a remote location. Extreme caution is recommended when this setting is adjusted. High headphone levels can cause hearing loss!**



To edit headphone routing click a crosspoint to select or deselect it. Routing edits are also reflected automatically in the **Matrix Editor**. Click and drag the **Send/Return** slider to adjust the balance between incoming and outgoing audio sources. Note: Routing changes not yet saved as a custom mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.



20.15 Configure Mono or Stereo Peer-to-Peer Programs

The **Program Manager panel** incorporates a wizard to configure a new program and all audio stream settings. Before you configure a new codec program consider if:

- You want your codec to be capable of dialing and answering, dialing only or answering only.
- A backup connection is required.

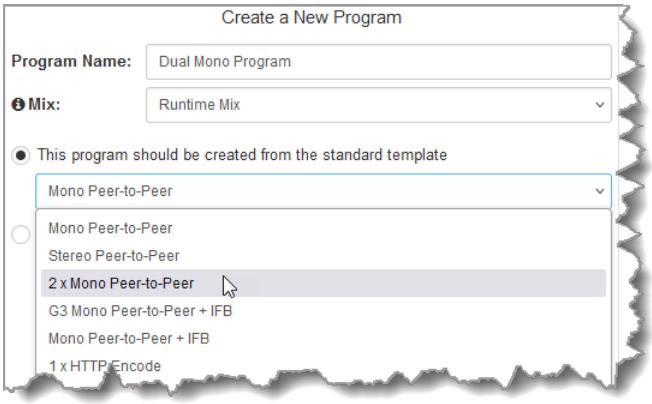
This section contains instructions for:

1. Configuring Peer-to-Peer Programs
2. Configuring a Failover Connection or Auto Reconnect
3. Configuring the Codec to Answer Connections
4. Configuring Output Source Failover options, e.g. File Playback on Silence Detection.

For more information about programs and audio streams within programs see the section titled IP Streaming, Programs and TieLink. Note: The following instructions will display how to configure a dial and answer program, with a backup connection and file playback. If the codec will dial and answer, select the **Dial or answer** option and the wizard will automatically display relevant screens to allow you to configure the codec correctly.

Creating Multistream Programs

When configuring multiple stream programs, e.g. 2 x Mono Peer-to-Peer, use the same process to configure each audio stream in turn. Port numbers are automatically assigned and can be adjusted as required.

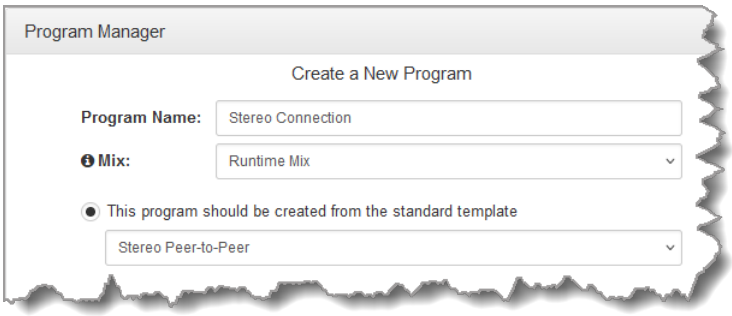


Configuring Peer-to-Peer Programs: Dialing

Important Notes: Before you start program configuration please note:

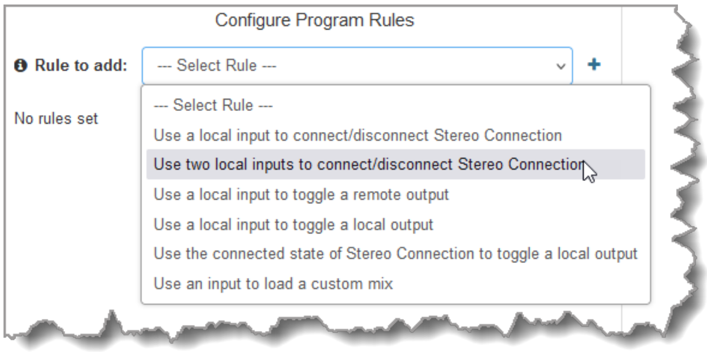
- A program cannot be edited in the **Program Manager** when loaded in the codec.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different program type. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- A program can be saved at several points throughout the program wizard and use default settings to save configuration time.
- SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard and:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.



Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

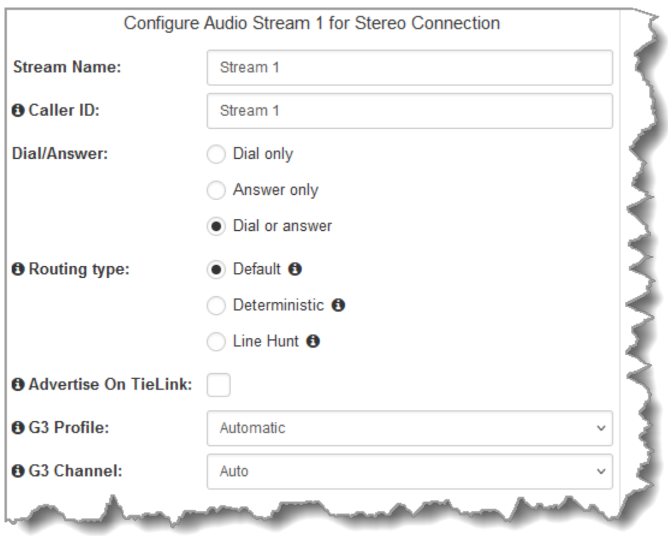
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



Important Notes for Rules:

- Bridge-IT II and Bridge-IT XTRA II have 4 physical **CONTROL PORT GPIOs**; 7 Tieline virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer, or dial and answer. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists.



Routing Type Options:	

Default	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic	Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed.
Line Hunt	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

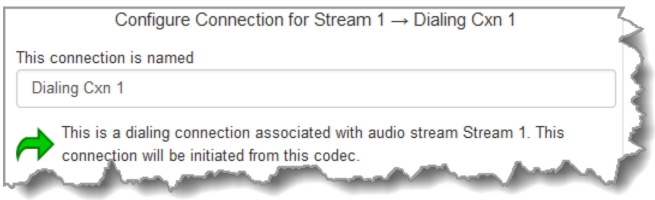
Important Notes on G3 Profile Settings: The G3 profile setting supports maintaining specific G3 codec settings.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. Note: When connecting in dual mono (2 x Mono Peer-to-Peer) mode to a G3 codec over IP both audio streams must encode using the same algorithm and sample rate or the G3 codec will not connect.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from the codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

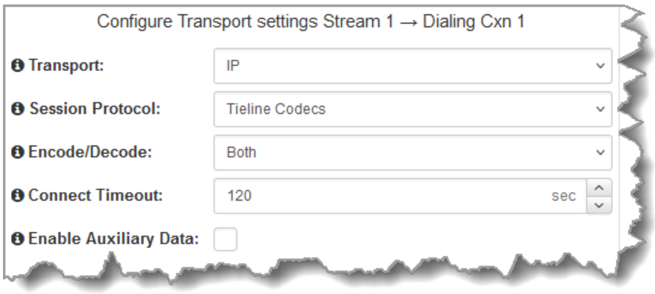
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



6. Configure the transport settings for the connection, then click **Next**. Note: Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm.

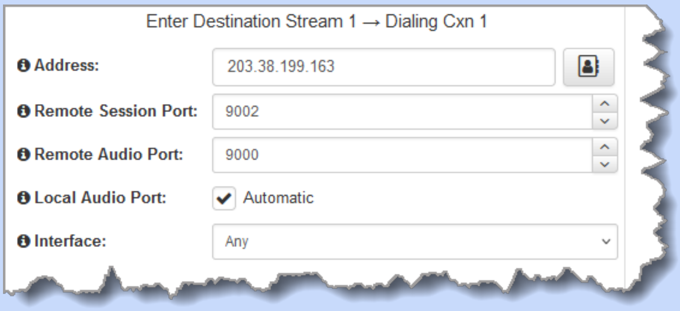


Important Notes:

- If you select **Sessionless** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.
- See RS232 Data Confizguration for detailed information on RS232 data and see Creating Rules for more information on relay operations.

7. Configure destination codec dialing and encoding settings:

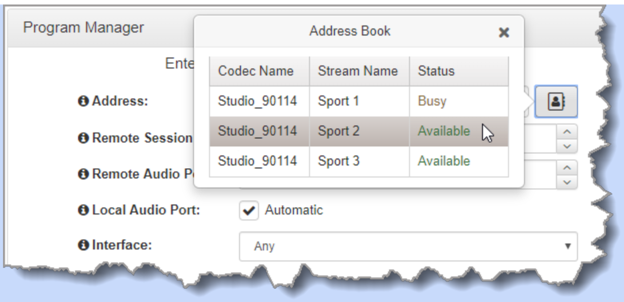
Configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN1**) or **Secondary** (**LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable. **Session Port 9002** and **Remote Audio Port 9000** are used by default. VLANs can be used to configure separate control and streaming interfaces if required. Note: By default **Any** will select **LAN** if it is available.



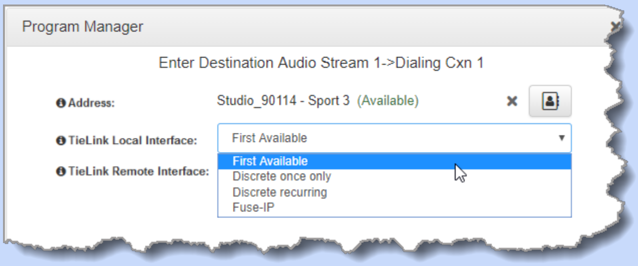
Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Session Port** and **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting. When you select **Sessionless** as the **Session Protocol**, the **Session Port** is not configurable and you can manually configure the **Remote Audio Port** and **Local Audio Port**.

If TieLink Contact List dialing is configured:

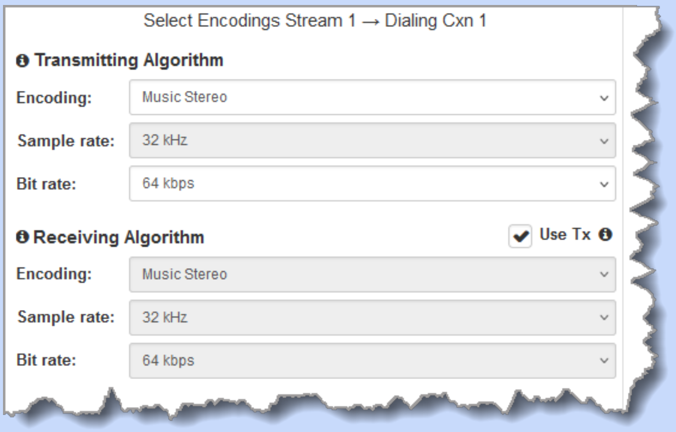
1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.



2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see Configuring TieLink Settings for more info.

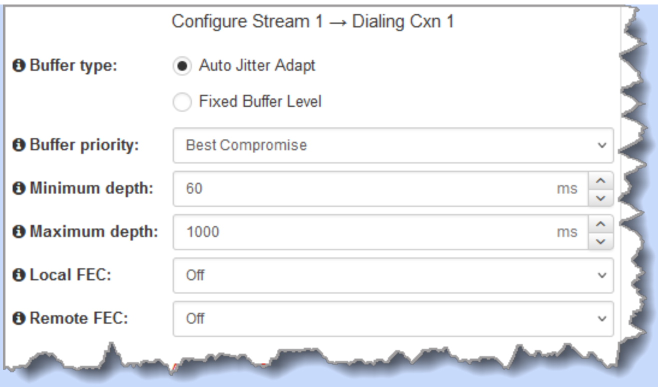


Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings, and configure SmartStream PLUS for this audio stream (recommended). Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.



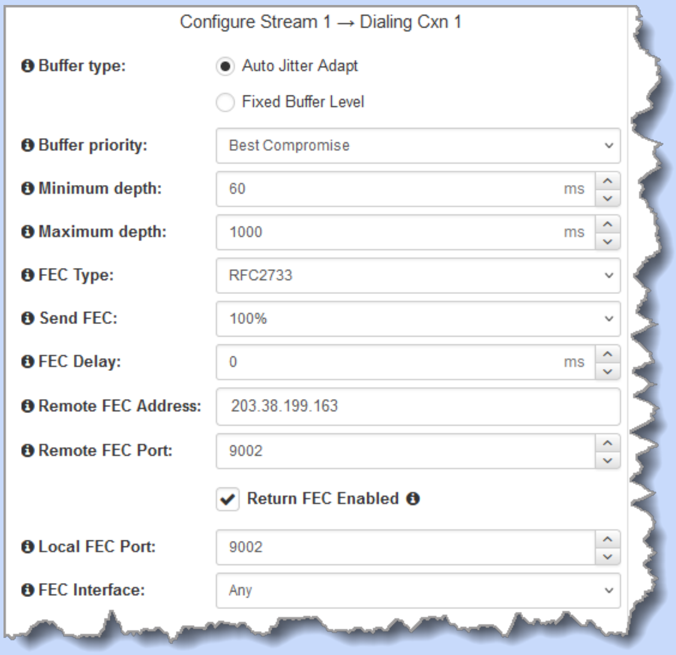
For IP connections click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.



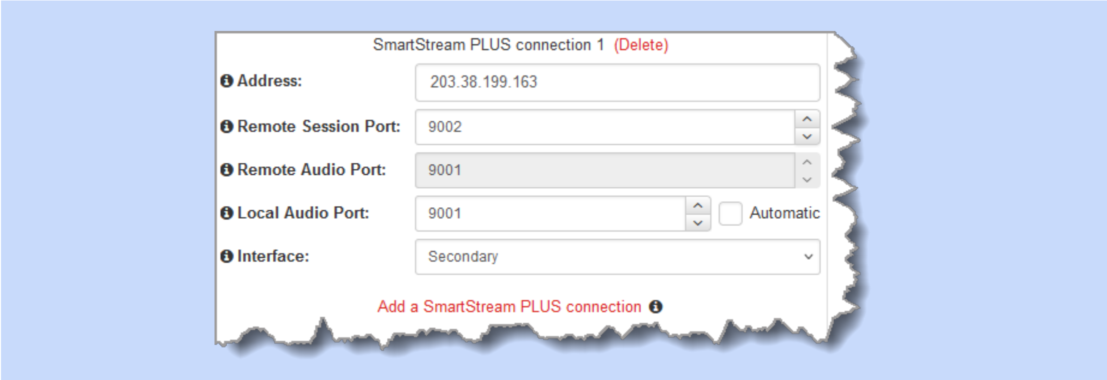
Important Notes:

- If you select **Sessionless** as the **Session Protocol** then RFC-compliant FEC is available. Configuration instructions are displayed in the right-hand pane.
- **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically delayed based on the ratio of primary packets to FEC packets sent at the selected setting.



Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. (To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/). Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.

By default, primary IP streaming is via the **LAN1** port. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from **LAN2** (or select another **Interface**). The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing.



Important Notes:

- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to two SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.
- TieLink only supports one SmartStream PLUS redundant connection for each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.
- To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/

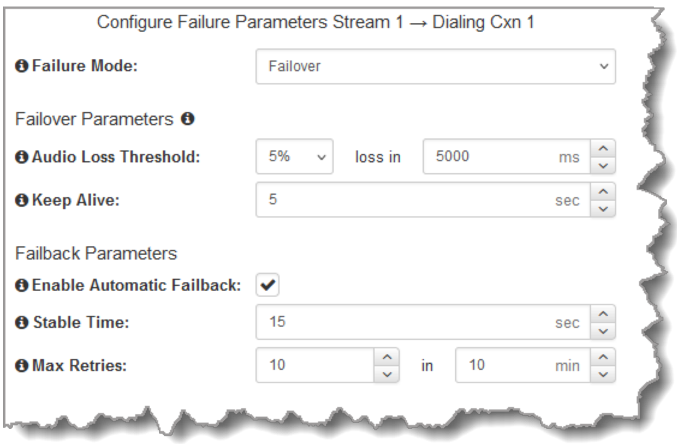
Configuring a Failover Connection or Auto Reconnect

At this point in the wizard you can choose to configure **Auto Reconnect** or create a failover connection for the audio stream.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the Web-GUI.

To configure a backup connection:

1. Click the drop-down **Failure Mode** arrow and select **Failover** to create a failover connection. Adjust **Failover Parameters** and click **Next**.



Important Note: When Failover is enabled, the codec will fail over to a backup connection under the following circumstances:

- There is sudden loss of connectivity to the primary destination.
- The remote codec disconnects the primary connection.
- The user manually attempts to connect to the failover connection.
- When the conditions for failover are triggered by the **Failover Parameters**.

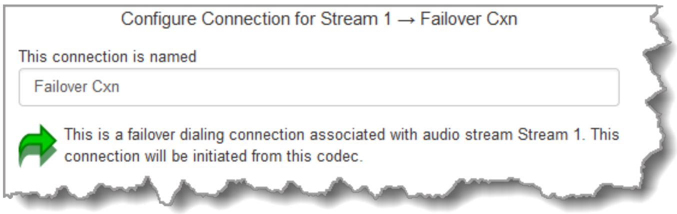
The codec will always attempt Failback if any of the following conditions are met:

- There is sudden loss of connectivity to the backup destination.
- The remote codec disconnects the backup connection.
- The user manually attempts to connect to the primary connection.

The explanations in the following table define configurable failover parameters.

	Failover Parameter	Description
1	Audio Loss Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame in milliseconds against which lost data is measured
3	Keep Alive (Time Frame)	The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity
4	Enable Automatic Failback	Select the check-box to fail back to a higher priority connection when failback parameters are met
5	Stable Time	The amount of time a primary connection must remain stable before attempting to fail back from the backup connection
6	Maximum Retries	The maximum number of fail back retries a codec can try before ending fail back attempts
7	Time Frame (Maximum Retries)	The time frame used to measure the number of fail back retries attempted

2. Enter a name for the failover connection and click **Next**.



3. Click **Next** to continue through the wizard and configure the failover connection in a similar manner to how you configured the primary connection.

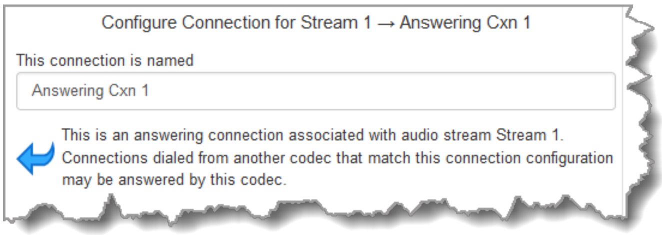
Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the HTML5 Toolbox Web-GUI.

4. Click **Next** to continue through the wizard and configure an answering connection, or click **Save** to save the program with default settings at this point.

Configuring the Codec to Answer Connections

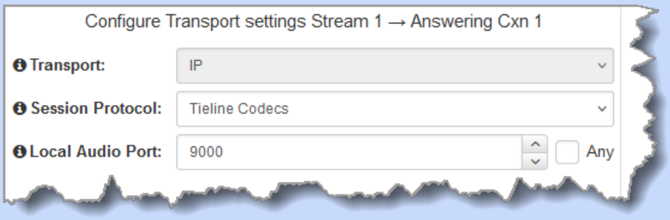
If you are configuring the codec to answer an incoming audio stream connection:

1. Enter a name for the answering connection and click **Next**.



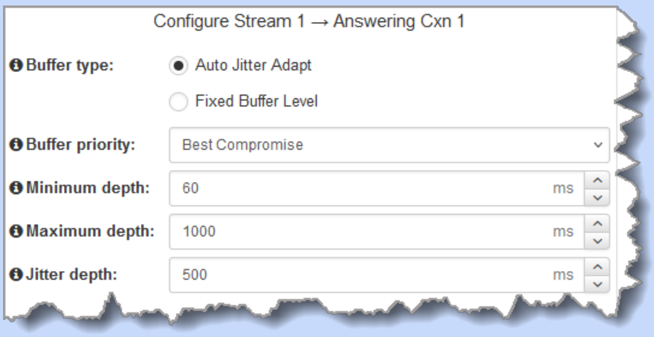
2. Configure the IP transport settings.

IP Select the **Session Protocol** and **Local Audio Port**.

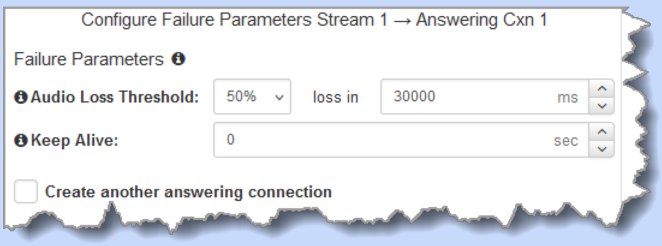


Click **Next** to specify jitter buffer settings, or create another answering connection. Click to configure:

- Auto Jitter Adapt and the preferred auto jitter setting using the drop-down arrow for Buffer priority. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.



Click **Next** to configure **Failure Parameters** for the answering connection if required. **Please note**: In most situations the default answering **Failure Parameters** do not need adjustment. These settings may be useful to troubleshoot certain connections, e.g. satellite IP links.



4. After configuring all settings there are 3 options:

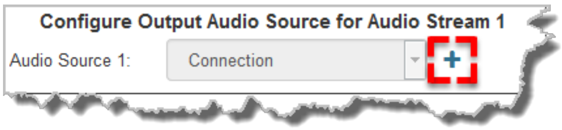
i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
ii. Click **Save** to save the program at this point.
iii. Click **Next** to configure **Output Audio Source** backup options.

Configuring Output Audio Source Options

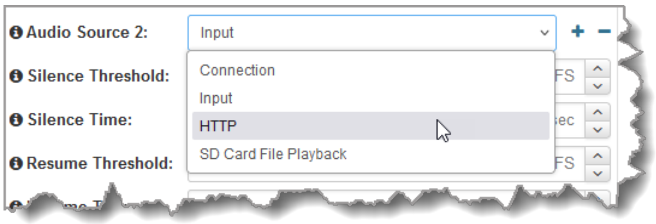
1. Click **Next** to configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program audio at transmitter sites. **Output Audio Source** options include:

   - **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
   - **Input**: Input audio looped to the physical codec outputs.
   - **HTTP**: Icecast client mode to allow media server streaming from a specified URL.
   - **SD Card File Playback**: Audio file playback from an SD card.

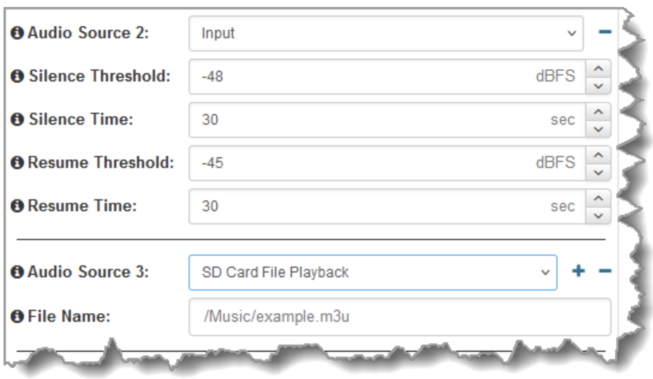
2. Click the blue **Plus symbol** ✚ to add a backup **Output Audio Source**, or click the **Minus symbol** ━ to remove an **Output Audio Source**.



3. Click the drop-down arrow to select an **Output Audio Source** option.



4. Configure silence threshold parameters to enable a preferred backup option, as well as resume thresholds for reactivating a previous source. Then click **Save Program** to save program settings.



5. After configuring **Output Audio Source** options you can:

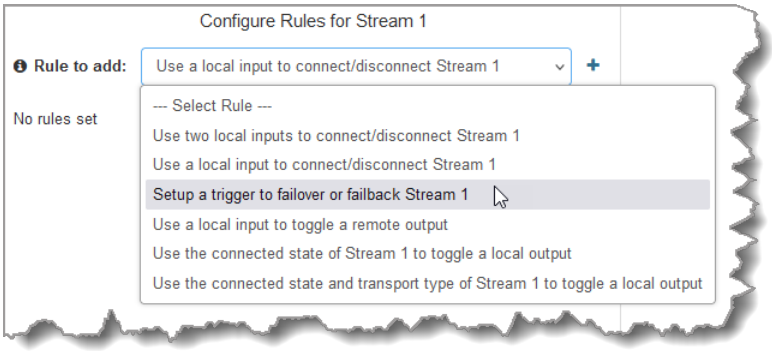
   i. Click **Save** to save the program at this point.
   ii. Click **Next** to configure rules options.

Important Notes for File Playback:

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings used are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
- File playback will occur automatically if the silence threshold parameters are breached. To manage file playback open the **Connections panel** in the web-GUI.

Configuring Rules

1. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



Important Note: Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

2. Click **Save Program** to save the program.
3. Click **Finish** to exit the wizard.
4. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**, or the **Program Loader panel** (in the Quick Connect web-GUI). Select and connect audio streams in a program using the **Connections panel**, or connect the program manually using the codec front panel.

20.16 Configure Mono Peer-to-Peer and IFB Programs

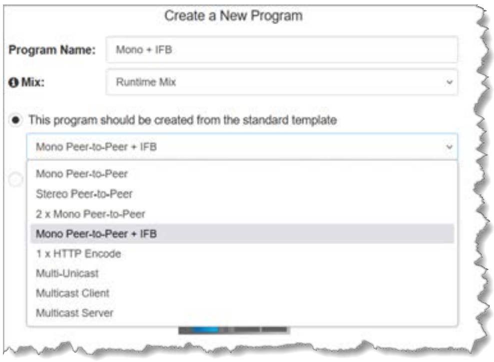
This program is designed to allow ViA codecs to dial a Bridge-IT II or Bridge-IT XTRA II codec at the studio and transmit:

1. A bidirectional mono audio stream for program audio.
2. A separate bidirectional mono IFB audio stream for communications.

This program can also be used in situations where a Bridge-IT II or Bridge-IT XTRA II codec is located at a remote broadcast site and will send program and IFB audio streams back to the studio.

Creating Mono + IFB Programs

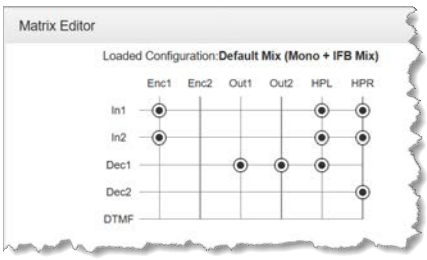
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard and:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Mono Peer-to-Peer + IFB**.



When configuring multiple stream programs, e.g. Mono + IFB, use the same process to configure each audio stream in turn, as described in the section Configure Mono or Stereo Peer-to-Peer Programs. Port numbers are automatically assigned and can be adjusted as required.

Matrix Settings for Mono + IFB Programs

The default Mono + IFB program's **Matrix Editor panel** routing is as follows:



Note that by default:

1. Program audio is routed to Encoder 1.
2. Return mix-minus program is routed to the codec outputs and left headphone output.
3. Return IFB communications is routed to the right headphone output.

These settings can be adjusted as required by selecting or deselecting routing crosspoints. For more details see Matrix Editing.

20.17 Configuring Multi-Unicast Dialing Programs

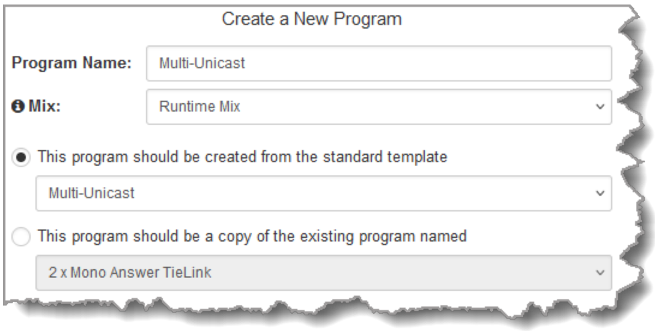
The codec can transmit a mono or stereo multi-unicast audio stream to a maximum of 6 endpoints in total using Bridge-IT II, or a maximum of 10 endpoints using Bridge-IT XTRA II. The first connection in a multi-unicast program is capable of bidirectional audio. Multi-unicast connections can only be created using the ToolBox web-GUI.

Important Notes: Before you start program configuration please note:

- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- When dialing a multistream program to both G3 and G5 codecs, by default the Audio Reference Level will be configured for the compatibility of the codec that connects first. I.e. if you dial a G3 codec first then the G3 Audio Reference Level will be configured for all connections.
- All streams must be dialed initially for a multi-unicast program. Streams can be disconnected and dialed individually after this.
- Connections are "dial only" for multi-unicast programs.
- SmartStream PLUS is supported for multi-unicast connections. Each additional redundant stream is included in the stream limit of 6 for Bridge-IT II and 10 for Bridge-IT XTRA II.
- TieLink supports one SmartStream PLUS redundant connection for each connection in a multi-unicast program.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time. The first connection in each multi-unicast stream determines default settings (e.g. algorithm, sample rate, bit rate), for all subsequent connections in that stream, except for dialing settings.
- All algorithms are supported for multi-unicast connections, however only one can be used for each audio stream.
- Bidirectional audio is only available on the first connection dialed for each audio stream.
- Renegotiation of connection bit rates is not possible when connected.
- Ensure you have sufficient connection bandwidth at the local codec to support all the connections to which you are connecting.
- If a codec is answering more than one mono or stereo multi-unicast connection it is necessary to create an answering program to suit the answering configuration and lock this program in the codec.

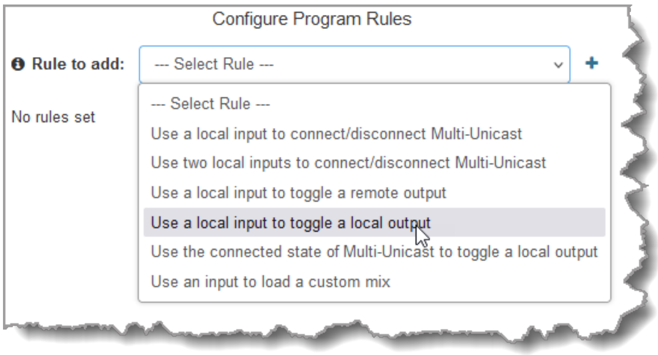
Creating a Multi-Unicast Program

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** symbol to open the wizard and then:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Multi-unicast**, or if you want to use an existing program as a template, select this option. Then click **Next**.



Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

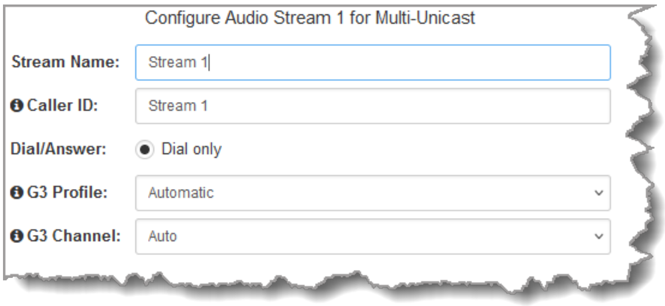
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Notes for Rules:

- Bridge-IT II and Bridge-IT XTRA II have 4 physical **CONTROL PORT GPIOs**; 7 Tieline virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name** and add a **caller ID**. Then click **Next**. Note: The caller ID is used to identify calls.



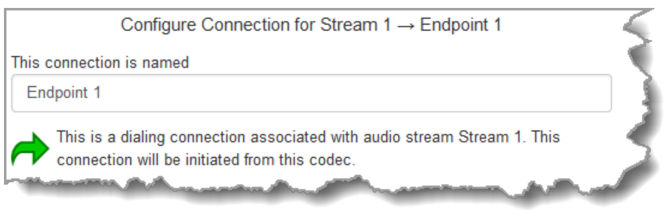
Important Notes on G3 Profile Settings: The G3 profile setting supports maintaining specific G3 codec settings.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. Note: When connecting in dual mono (2 x Mono Peer-to-Peer) mode to a G3 codec over IP both audio streams must encode using the same algorithm and sample rate or the G3 codec will not connect.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from the codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

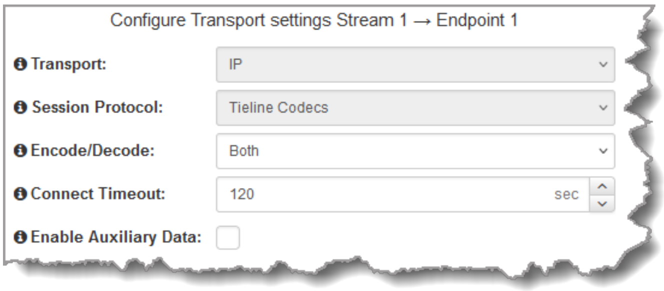
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



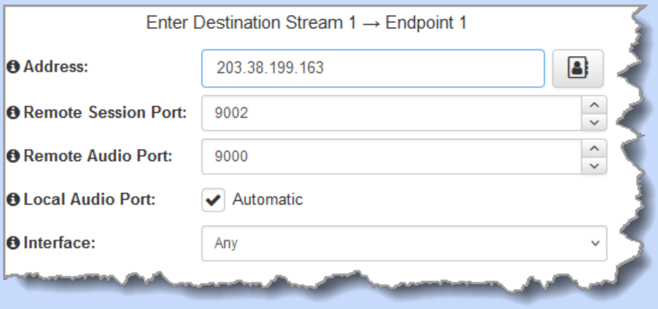
6. Configure the transport settings for the connection, then click **Next**. Note: only the first connection dialed in a multi-unicast program can encode and decode audio. All other connections are unidirectional and encode only.



Important Note: See RS232 Data Configuration for detailed information on RS232 data. Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm. Only in-band auxiliary data using the Music or MusicPLUS algorithms is possible when connecting to G3 Commander and i-Mix codecs.

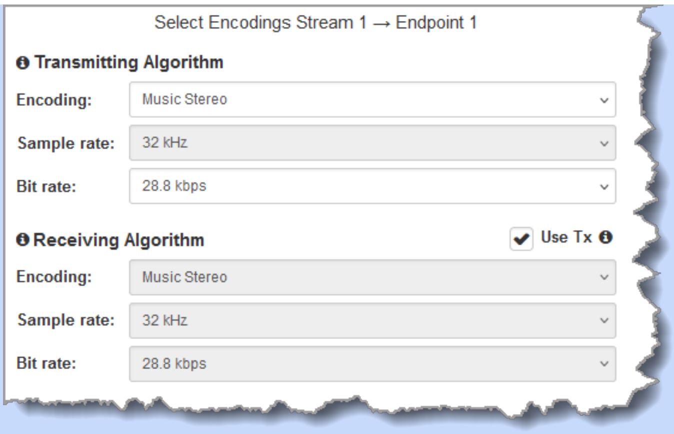
7. Configure destination codec dialing and encoding settings:

IP For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.



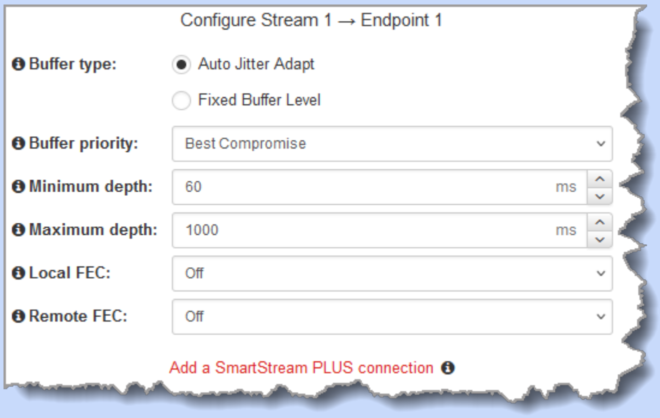
Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting.

Click **Next Stream** to configure the first audio stream with default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings (recommended) and more connection endpoints. Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** or **Bit rate** options.

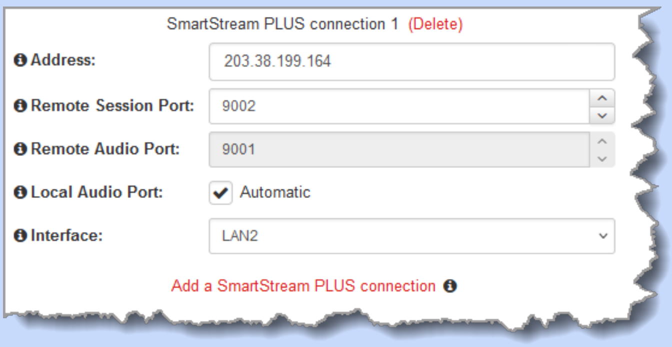


Click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.



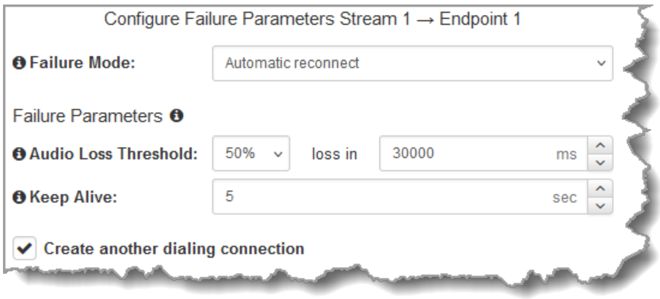
Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. By default, primary IP streaming is via **LAN1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **LAN2**. The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing.



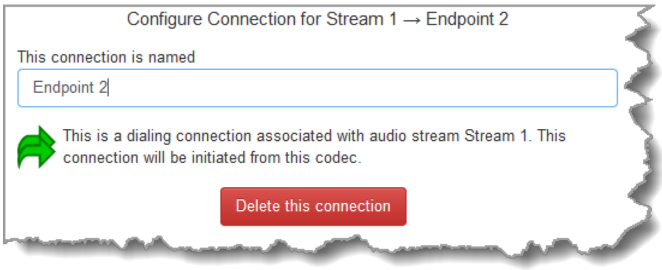
Important Notes:

- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to two SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.
- TieLink only supports one SmartStream PLUS redundant connection for each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.

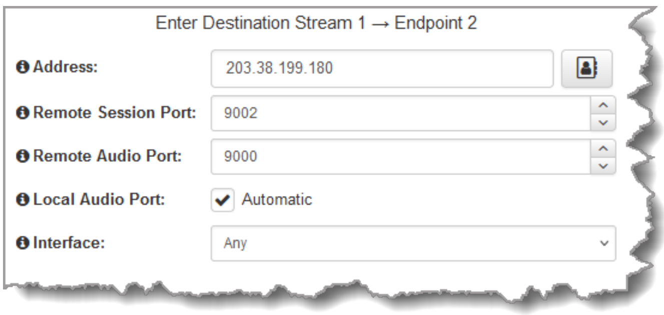
8. Click **Next** and then click the drop-down **Failure Mode** arrow and select **Automatic reconnect** to enable this feature and configure **Failure Parameters**. To add another stream endpoint select the check-box for **Create another dialing connection**, then click **Next**.



9. Name the connection and then click **Next**.



10. Enter destination settings for the dialing connection, then click **Next**.

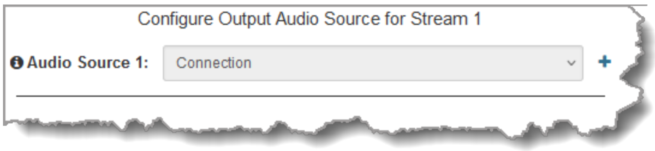


11. There are several options from this point:
    - Click **Add a SmartStream PLUS connection** to add a redundant stream for the current connection.
    - Select **Create another dialing connection** to create another stream endpoint.
    - Click **Next** to configure the **Output Audio Source** for the audio stream.
    - Click **Next** to configure stream **Rules**.
    - Click **Next Stream** to configure the second multi-unicast audio stream.

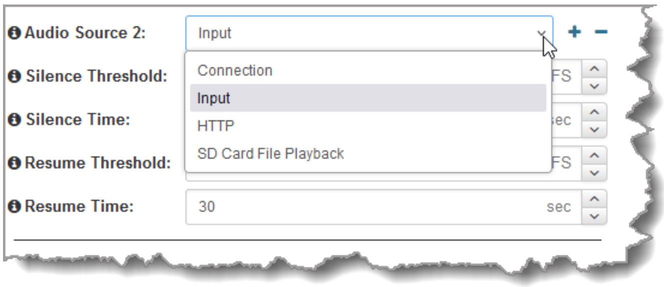
Configuring Output Audio Source Options

1. Click **Next** to configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program audio at transmitter sites. **Output Audio Source** options include:

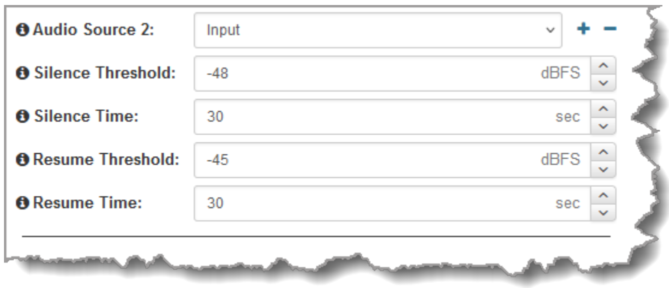
   - **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
   - **Input**: Input audio looped to the physical codec outputs.
   - **HTTP**: Icecast client mode to allow media server streaming from a specified URL.
   - **SD Card File Playback**: Audio file playback from an SD card.



2. Click the blue **Plus symbol** ➕ to add a backup **Output Audio Source**, or click the **Minus symbol** ➖ to remove an **Output Audio Source**.
3. Click the drop-down arrow to select an **Output Audio Source** option.



4. Configure silence threshold parameters to enable a preferred backup option, as well as resume thresholds for reactivating a previous source.



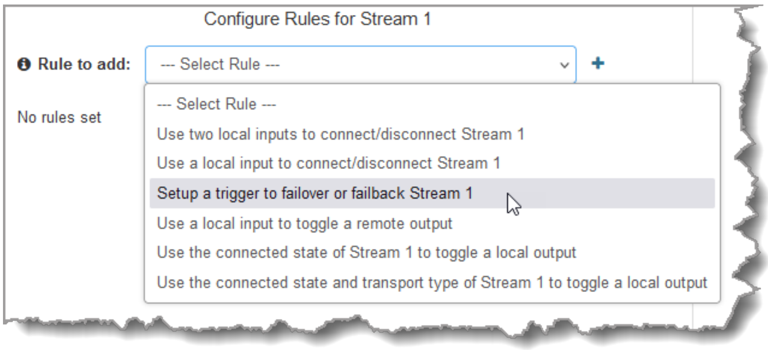
Important Notes for File Playback:

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB. Note: The Windows built-in formatting tool cannot format a SD card larger than 32GB with the FAT32 file system.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.

- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
- File playback will occur automatically if the silence threshold parameters are breached. To manage file playback open the **Connections panel** in the web-GUI.

5. After configuring **Output Audio Source** options you can:

- Click **Next** to configure stream level rules for this audio stream.



- Click **Save** to save the currently configured settings.
- Click **Finish** or **Load** to load the new program.

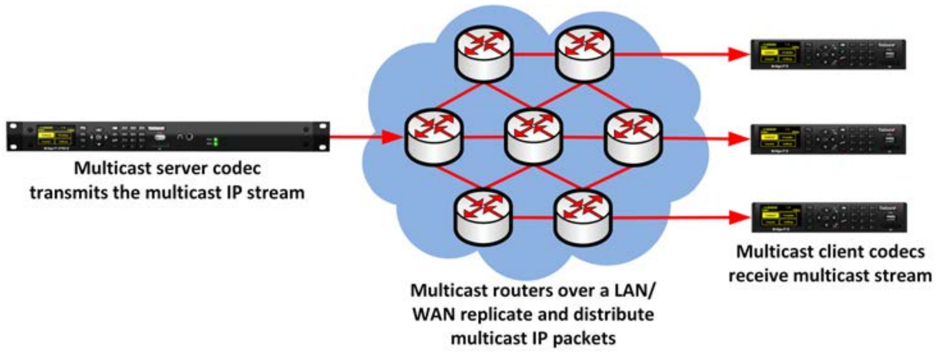
The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**. Select and connect audio streams in a program using the **Connections panel**, or connect the program manually using the codec front panel. Note: When you initially dial using a multi-unicast program all audio streams are dialed simultaneously. It is then possible to connect and disconnect audio streams individually.

20.18 Configuring Multicast Server Programs

How Multicasting Works

Multicast transmissions are sent using a dedicated IP multicast address that looks similar to a regular IP address and multicast subscribers request transmissions from this address. This unique address allows multicast routers to identify multicast requests from a group of codecs interested in a particular transmission and packets are replicated depending on demand. This can create large demands on network bandwidth if the multicast group is significant in size.

As a result, only small sections of the internet are multicast enabled and many internet service providers (ISPs) block multicast traffic over wide area networks. This restricts most multicast broadcasts to private local area networks. Some ISPs provide quality of service (QoS) priority to multicast streams for an increased service charge. You need to check with your ISP to find out what multicast services, if any, are available over WANs.



Important Notes:

- When a connection is dialed Tieline codecs normally use session data to configure settings like the algorithm, connection bit rate and sample rate etc. Multicast connections do not use session data and it is imperative that all codecs are configured with the same connection settings prior to connecting, or they will not be able to join multicast streaming sessions.
- Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

Multicast Server versus Multicast Client Programs

Two different types of multicast programs need to be created when multicasting:

- A multicast server program is used by the broadcasting codec to send multicast IP packets to multicast routers on a network.
- A multicast client program is used by codecs to receive multicast IP audio packets.

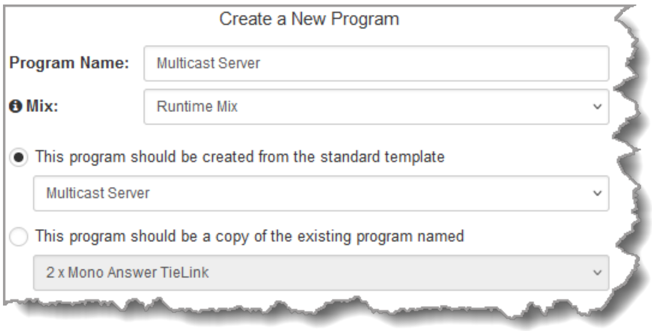
A multicast server codec sends audio packets only and a multicast client codec receives audio packets only. Codecs using the client program request multicast packets (sent from the server codec), which are distributed by multicast routers.

Creating Multicast Server Programs

Important Notes: Before you start program configuration please note:

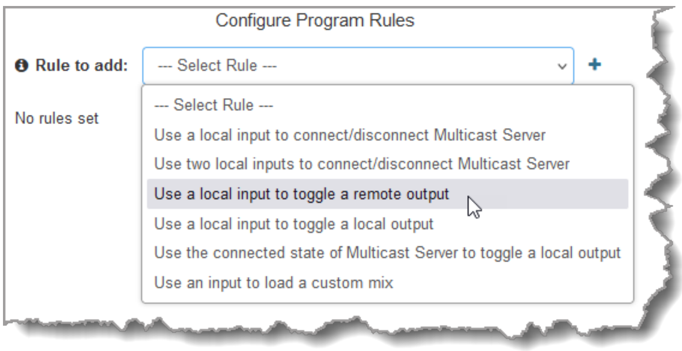
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not connect successfully.
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
- The default UDP audio port setting is 9000 for a multicast and the client and server port settings must match.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- Relay functionality is available for multicast connections between Tieline codecs.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Multicast Server** or if you want to use an existing program as a template, select this option. Then click **Next**.



Important Notes: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

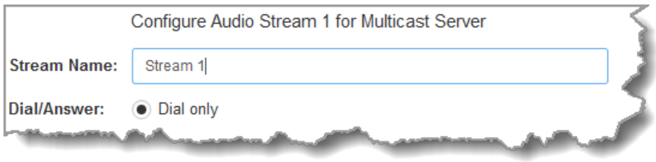
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



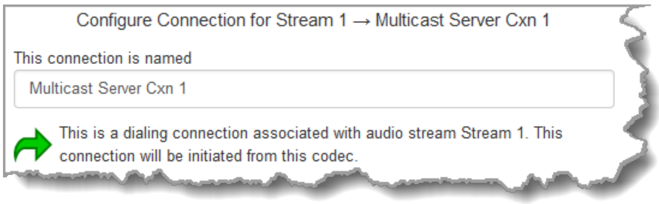
Important Notes for Rules:

- Bridge-IT II and Bridge-IT XTRA II have 4 physical **CONTROL PORT GPIOs**; 7 Tieline virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

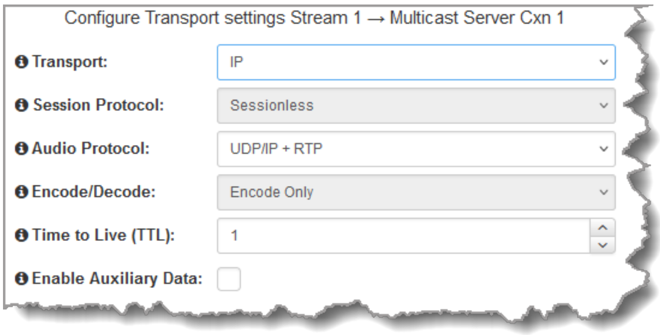
4. Enter a name for the **Audio Stream**, then click **Next**.



5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



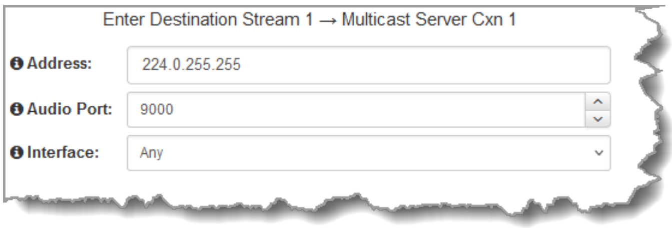
6. Configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC-compliant streaming. If auxiliary data is enabled the audio stream will not be RFC-compliant.



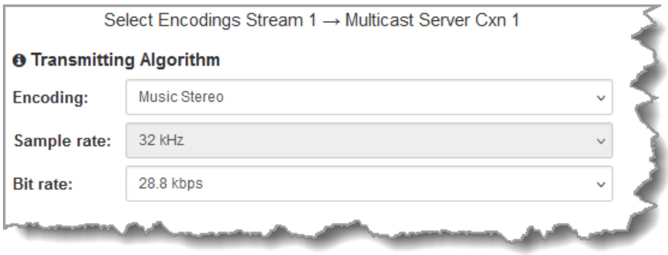
Important Notes:

- The encode and decode direction is configured automatically for **Encode Only** (server program) or **Decode Only** (client program). This setting is configured when you select either **Multicast Server** or **Multicast Client** when you first create the program in the wizard.
- The TTL value you need to use is dependent upon your network infrastructure. Please consult your network administrator if you are unsure about how to configure this setting.

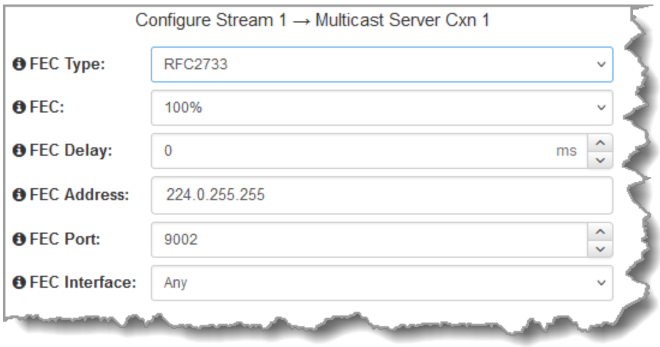
7. Configure the multicast IP address and **Remote Audio Port** (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**), then click **Next**. Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.



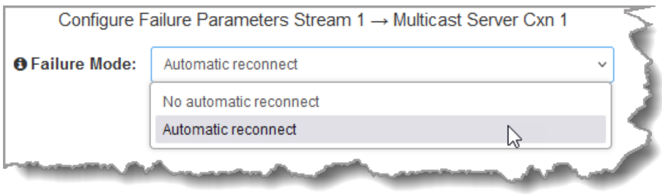
8. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.



9. Click the drop-down arrow for **Send FEC Type** to enable RFC-compliant FEC. Configuration instructions are displayed in the right-hand **Program Manager panel** pane. **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically delayed based on the ratio of primary packets to FEC packets sent.



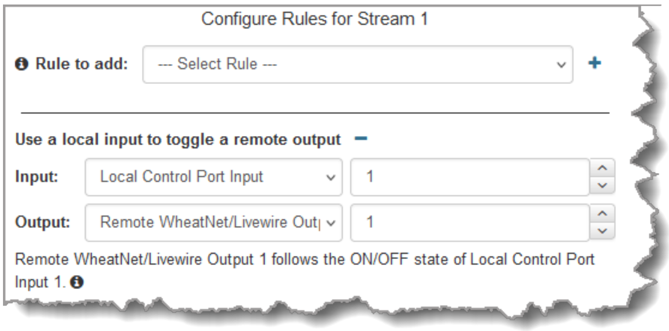
10. Click **Next** to select configure **Enable Auto Reconnect** if required.



11. Next you can either:

    i. Click **Next** to configure **Rules** options.
    ii. Click **Save Program**.

12. To configure new stream-level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



13. After configuring all streams in the multicast server program click **Save Program** to save the program settings, then click **Finish**.

Important Notes: There is no jitter buffer setting on the server codec because it never receives audio packets.

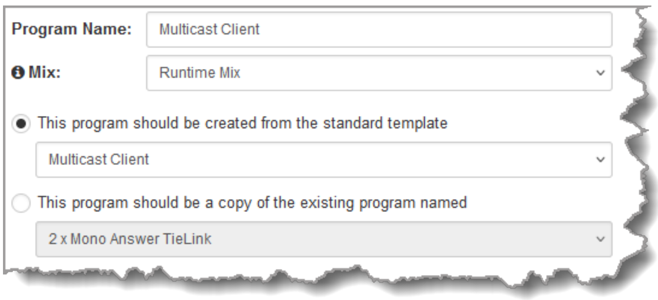
14. Configure multicast server and multicast client programs and load all codecs with the appropriate program. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast audio packets. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

20.19 Configuring Multicast Client Programs

Important Notes: Before you commence program configuration please note:
- Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not connect successfully.
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different program type.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- Always dial the multicast server codec connection first before connecting multicast client codecs.
- Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
- The default UDP audio port setting is 9000 for a multicast and the client and server port settings must match.
- It is not possible to connect to a G3 codec and receive multicast IP audio streams.
- Relay functionality is available for multicast connections between Tieline codecs.

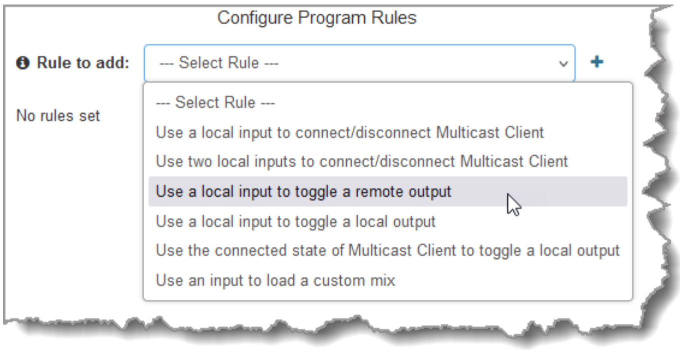
Configuring Multicast Client Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Multicast Client** to configure a multicast program, or if you want to use an existing program as a template, select this option. Then click **Next**.



Important Notes: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

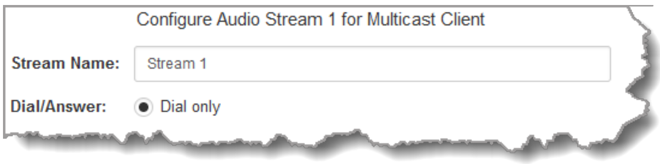
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



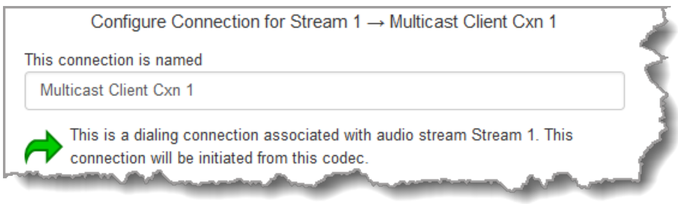
Important Notes for Rules:

- Bridge-IT II and Bridge-IT XTRA II have 4 physical **CONTROL PORT GPIOs**; 7 Tieline virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

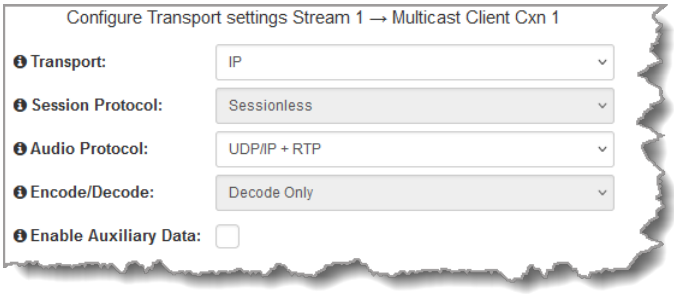
4. Enter a name for the **Stream Name**, then click **Next**.



5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



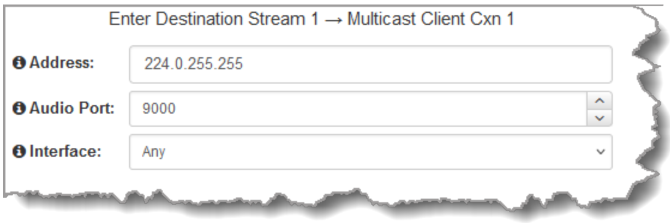
6. Configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC compliant streaming. If auxiliary data is enabled the audio stream will not be RFC-compliant.



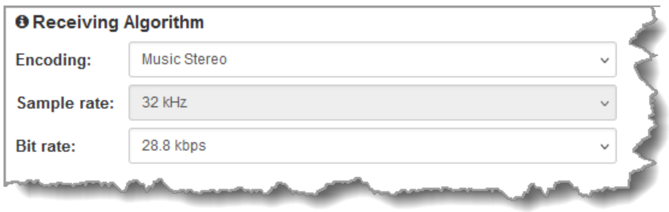
Important Notes:

- The encode and decode direction is configured automatically for **Encode Only** (server program) or **Decode Only** (client program). This setting is configured when you select either **Multicast Server** or **Multicast Client** when you first create the program in the wizard.
- See RS232 Data Configuration for detailed information on RS232 data.

7. Configure the multicast IP address and **Local Audio Port** (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**), then click **Next**. Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.

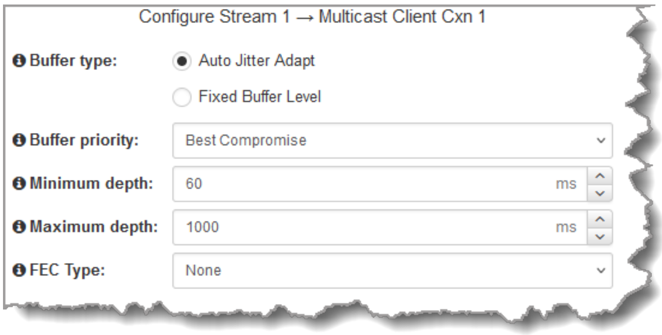


8. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.

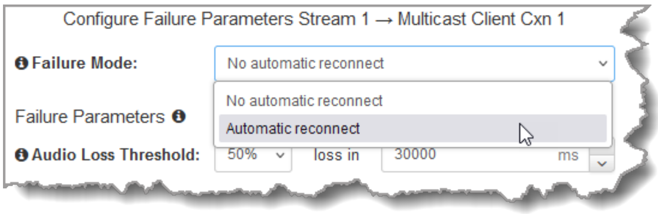


9. Click to configure:
   - **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
   - Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.

Next, select **Return FEC Enabled** to enable RFC2733 compliant Forward Error Correction, which is available because multicast connections are sessionless. Follow the instructions in the right-hand pane. When configuration is complete click **Next**.



10. Click the **Failure Mode** drop-down arrow to select **Automatic reconnect** if required, then click **Next**.



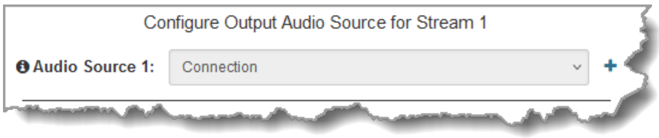
11. After configuring all settings there are 2 options:

    i. Click **Next** to configure **Rules** and/or **Output Audio Source** options.
    ii. Click **Save** to save the program.

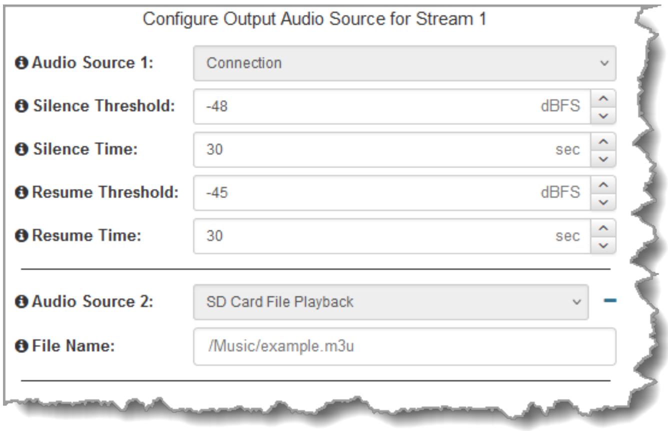
Configuring Output Audio Source Options

1. Click **Next** to configure **Output Audio Source** options and automatically switch between backup audio sources to maintain program audio. **Output Audio Source** options include:

    - **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
    - **File Playback**: Audio file playback from a supported storage medium.



2. Click the blue **Plus symbol** ➕ to add a backup **Output Audio Source**, or click the **Minus symbol** ➖ to remove an **Output Audio Source**.
3. Configure silence threshold parameters to enable a preferred backup option, as well as resume thresholds for reactivating a previous source. Then click **Save Program** to save program settings.



5. After configuring **Output Audio Source** options you can:

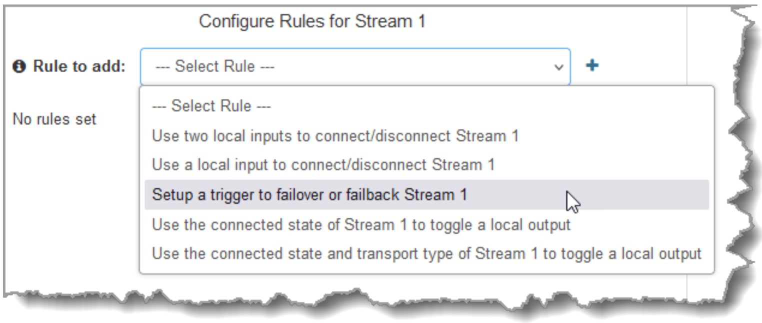
   i. Click **Save Program** to save the program at this point.
   ii. Click **Next** to configure rules options.

Important Notes for File Playback:

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).
- The codec supports SDHC cards which have a physical capacity of up to 32GB. Note: The Windows built-in formatting tool cannot format a SD card larger than 32GB with the FAT32 file system.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
- File playback will occur automatically if the silence threshold parameters are breached. To manage file playback open the **Connections panel** in the web-GUI.

Configuring Rules

1. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.

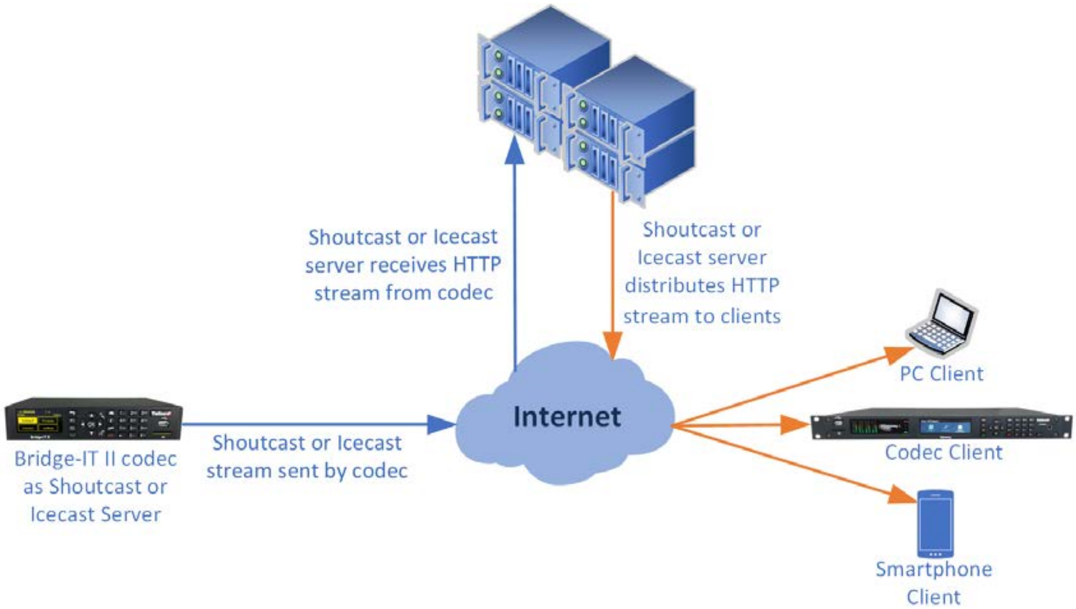


Important Note: Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

2. Click **Save Program** to save the program.
3. Click **Finish** to exit the wizard or **Load** to load it.
4. Configure multicast server and multicast client programs and load all codecs with the appropriate program. The newly created program will be displayed in the left pane within the **Program Manager panel** and in the **Connections panel**. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

20.20 Configure HTTP Streaming Programs

To allow internet radio station streaming, the codec supports a single Icecast or Shoutcast HTTP server encode stream which is configured as a connection within a program. As displayed in the following image, two major components are involved: an Icecast or Shoutcast streaming server and source client/s.



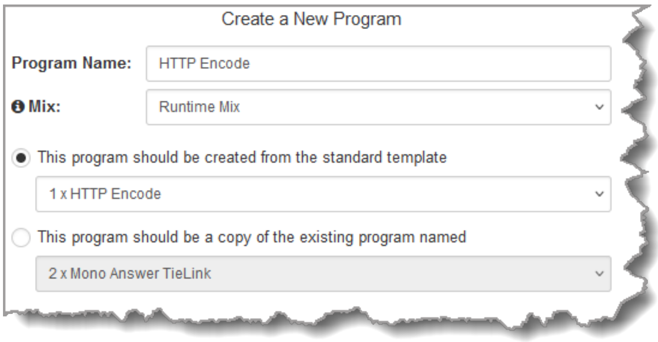
Important Notes:

- Only one HTTP encode stream is supported to an external Icecast or Shoutcast server.
- A local Icecast server can run on a codec and the recommended client limit is 10.
- The codec currently does not support HTTPS Icecast streams.

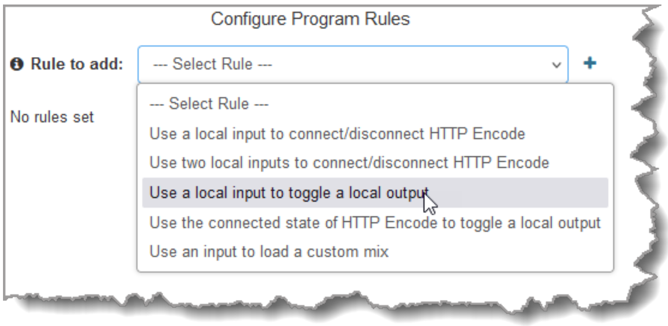
Configuring a Server Program

An HTTP streaming program can be a standalone HTTP streaming connection, or one of several connections in a multi-stream program. To get started:

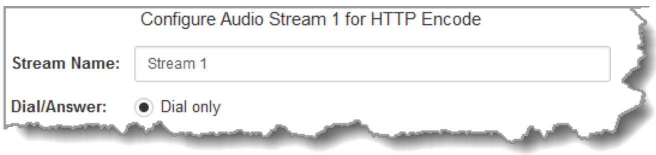
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard and:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **1 x HTTP Encode,** or if you want to use an existing program as a template, select this option. Then click **Next**.



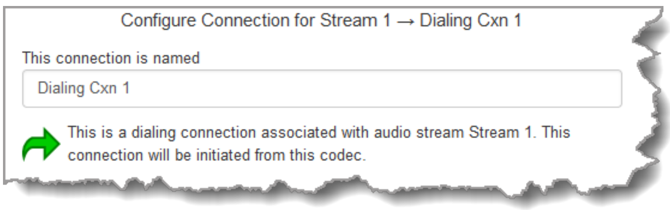
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule, then click **Next**.



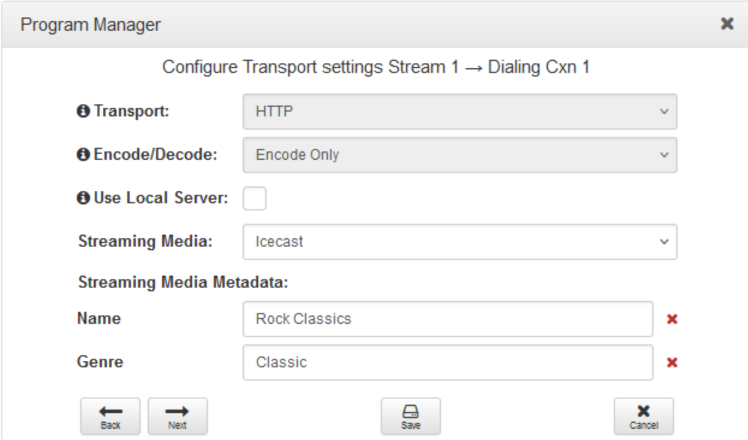
4. Enter a **Stream Name**, then click **Next**.



5. Enter the name of the connection in the text box, then click **Next**.



6. Configure the transport settings for the connection, then click **Next**. Note: Select **Use Local Server** to run a local Icecast server on the codec with a recommended client limit of 10. Then click **Next**.

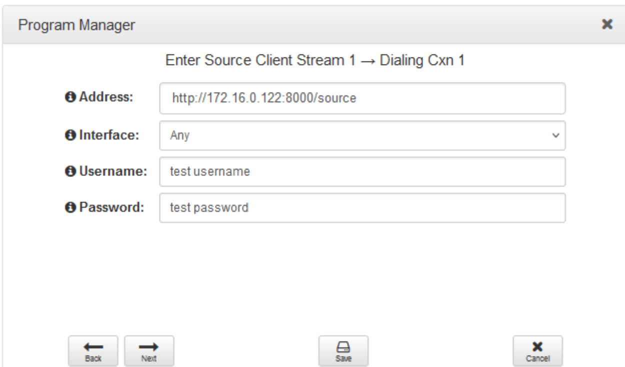


Important Notes:

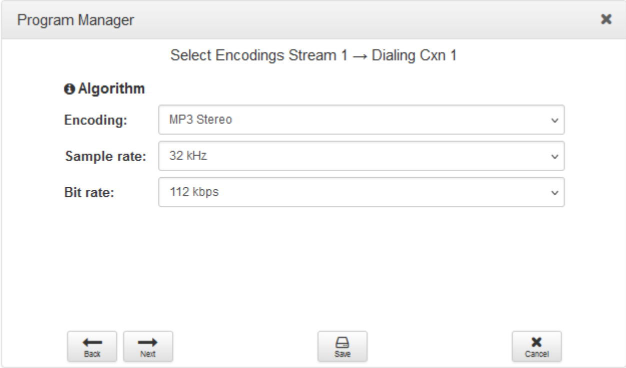
- Use the **Streaming Media panel** to enable and configure the local **Icecast Server Port** when **Use Local Server** has been selected. The UDP port number that the local Icecast server listens on must be between 1024 and 65535. When configured a program pushes audio to the internal Icecast server.

- Only static **Name** and **Genre** metadata is supported.
- The codec firewall will automatically open the selected UDP port when the server is running and close the port when the server isn't running.

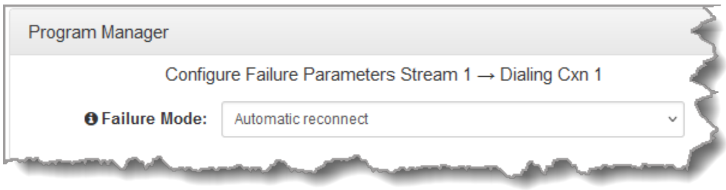
7. Configure the **Source Client** stream settings:
   - Server Destination Address ("http://172.16.0.122" in the example shown)
   - UDP Port ("8000" in the example shown)
   - Mountpoint: a unique name on the server identifying a particular stream ("source" in the example shown)



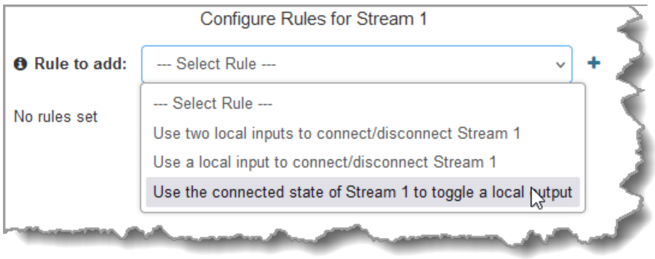
8. Select the preferred **Encoding** option for the stream. A range of sample rate and bit-rate options are available for MP2 and MP3 encoding. Then click **Next**.



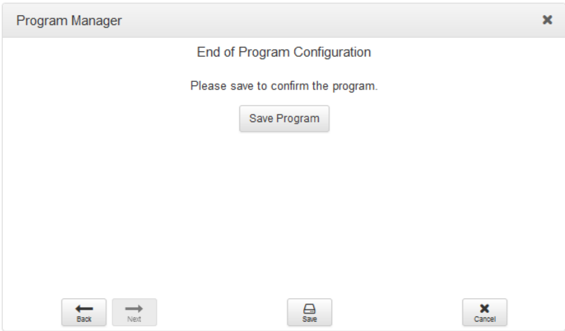
9. Configure **Automatic reconnect** if required, then click **Next**.



10. To configure new stream level rules, click the drop-down arrow and select the preferred option from those available. Then click **Next**.



11. Click **Save Program** to save all settings and then click **Load** to load the program if required.



20.21 Configuring SIP

The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs. For more background on SIP connections and the differences between registered and unregistered peer-to-peer SIP connections see About SIP. To configure the codec to dial over SIP using a SIP Server:

1. Register the codec to a SIP server using SIP account credentials.
2. Configure a SIP interface in the codec. Note: This **SIP1** or **SIP2** interface will include the proxy and port settings, as well as the selected IP interface used to make the connection, e.g. **LAN1** or **LAN2**.
3. Create a SIP program using the codec front panel or the HTML5 Toolbox Web-GUI.

Important Notes:

- The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**.
- SIP dialing is only supported over point-to-point connections.
- RFC5109 and RFC2733 compliant FEC over SIP is supported.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.
- Tieline G3 codecs do not support connections using algorithms like AAC, aptX Enhanced and Opus and will default to MPEG Layer 2 if an incoming call is configured to use these algorithms.
- SmartStream PLUS redundant streaming is not available with SIP or sessionless IP..
- For detailed information about connecting with other brands of codec using SIP visit www.tieline.com.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level. To do this from the **Home screen** select **Audio > Reference Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
  - Select either a mono or stereo profile
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

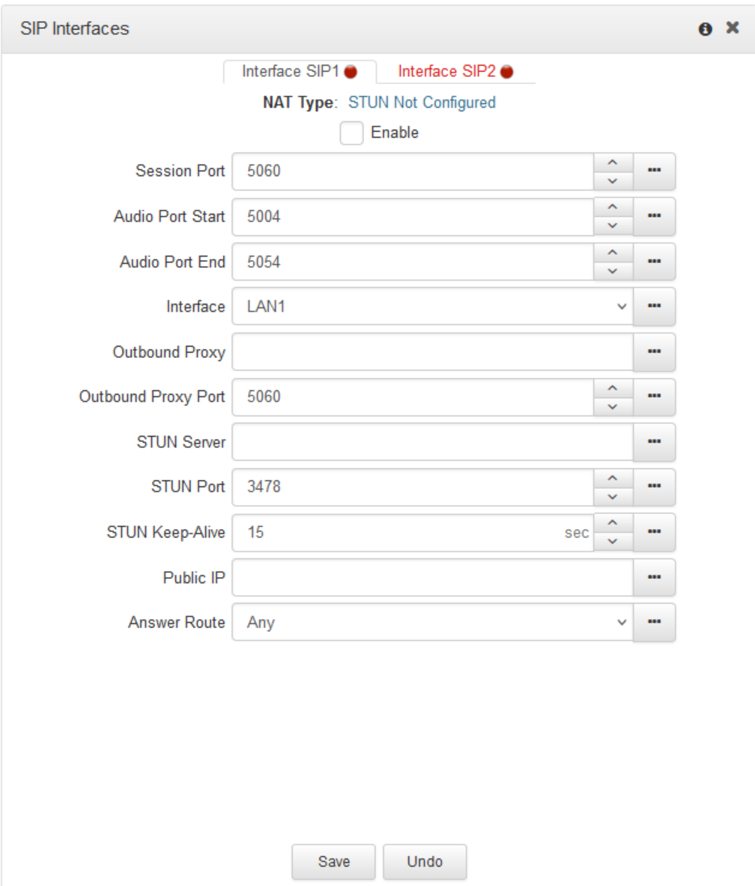
20.21.1 Configuring SIP Interfaces

The codec supports dialing over two SIP interfaces simultaneously.

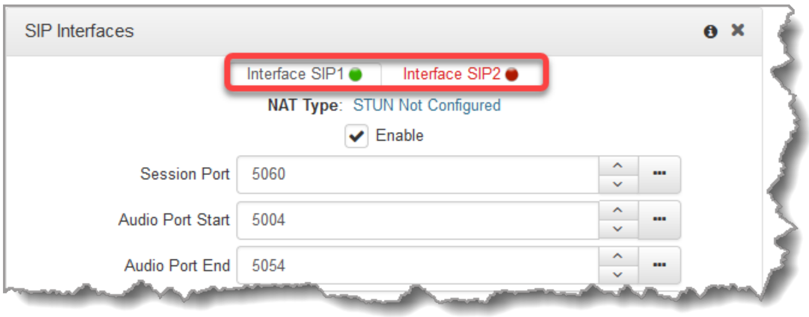
Important Notes:
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN1** by default, which is mapped to the **Primary** interface by default.
3. **SIP2** is configured to use **LAN2** by default, which is mapped to the **Secondary** interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN1** or **LAN2**.
5. **SIP1** and **SIP2** can each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface has been enabled.
7. Enter a public IP address in the **Public IP** text box if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.

To configure **SIP1** or **SIP2**:

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Interfaces** to view and configure SIP interface settings.
2. Default SIP settings are configured and select **Interface SIP1** or **Interface SIP2** to adjust each interface. Note: Ensure each SIP interface uses a unique IP **Interface** because they cannot share one, e.g.**LAN1**.



3. Select the **Enable** check-box and then click the **Save** button to confirm settings.
4. The SIP interface indicator is green when an interface is enabled and red when it is disabled.



20.21.2 Configuring SIP Accounts

Up to 6 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, select the SIP server to which you will register your codec. On a LAN this may be your own server, or it could be one of the many internet servers available. We recommend that you use your own SIP server and configure it to use G.711, G.722, MP2 and AAC algorithms. This is because most internet SIP servers are for VoIP phones and are only configured for G.711 and GSM algorithms.
When you register an account with a SIP server you will be provided with:

- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port
- Password
- Registration Timeout (this shouldn't need to be adjusted from the default setting).

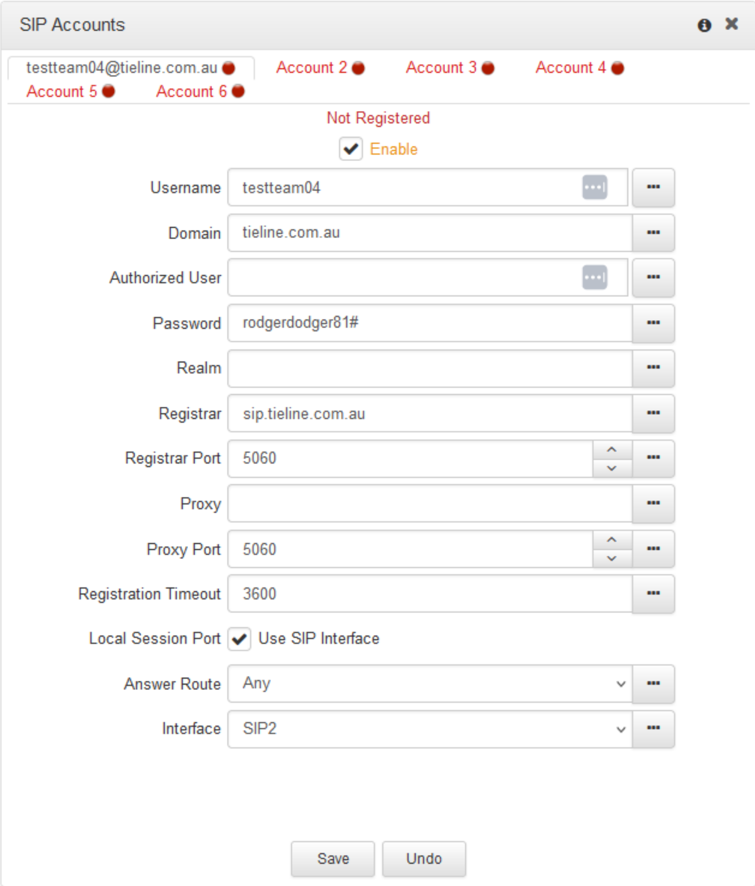
Important Notes:
- In most situations it is best to configure a SIP account when the codec is configured with a public IP address.
- Each SIP account can only be mapped to a single SIP interface, i.e. **SIP1** or **SIP 2**.
- Up to 6 SIP accounts can be added to the codec.
- To configure a SIP Account using the codec **OLED SCREEN** see Configuring SIP Accounts.

Adding a SIP Account

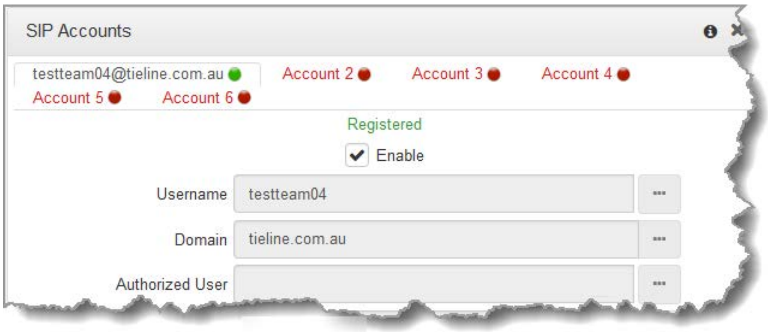
Enter SIP account details and register the account in your codec. Once configured, the codec will contact the SIP server automatically to acknowledge its presence over a wide area network when connected to a public IP address.

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Accounts** to view and configure SIP account settings.
2. Click to select one of the unused **Accounts** at the top of the **SIP Accounts panel**.
3. Enter the SIP account details into the relevant text boxes, including the registration **Timeout** (which shouldn't need to be adjusted from the default setting). Also ensure a SIP

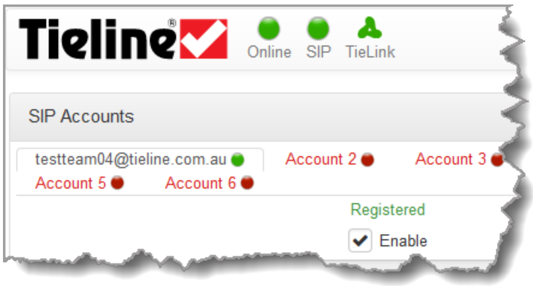
Interface is selected (e.g. **SIP1** or **SIP2**.) The SIP interface contains settings related to ports and the selected interface, e.g. **LAN1** or **LAN2**. See Configuring SIP Interfaces for more details.



4. Click the **Enable** check-box at the top of the panel and then click the **Save** button to register the codec to the server.
5. If an account is registered successfully, the account registration indicator changes from red to green, and **Not Registered** (above the **Enable** check-box) becomes **Registered**.



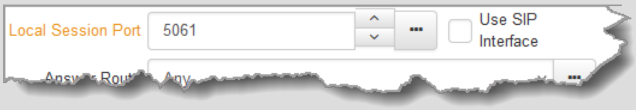
6. In the Toolbox Web-GUI the red **SIP** indicator adjacent to the codec **Online** indicator also changes to green when an account is currently registered in the codec and ready to be used when dialing over SIP.



7. Once enabled, the SIP account can be selected when creating a new SIP connection.

Important Notes:

- Some ISPs may block SIP traffic over UDP port 5060.
- By default, the session port used for each SIP account is the associated SIP interface session port. The default session port is the registered UDP port number 5060. It is also possible to configure a custom local session port for each SIP account for compatibility with Cisco Unified Communications Manager (CUCM). Ensure your firewall has the required TCP and UDP ports open when receiving multiple SIP calls.



Troubleshooting SIP Registration

If a SIP account is not registering successfully please check the following:

1. Confirm all account registration information has been entered correctly.
2. Confirm the SIP interface (**SIP1** or **SIP2**) configured in the account is enabled.
3. Verify that the **Interface** selected in **SIP1** or **SIP2** corresponds with the network interface used by the codec to register the account. E.g. **LAN1**, **LAN2**.

20.21.3 Configure SIP Allow and Block Lists

The **SIP Filter Lists panel** allows filtering of SIP URIs and User Agents to provide greater security for your codec connections. For example, add trusted network codecs to the **URI Allow List** in this panel and only codecs using these SIP URIs will be able to connect. This is like saying, "if you have the key you can open the door" and is perhaps the easiest way to filter outside access to your codec's "front door".

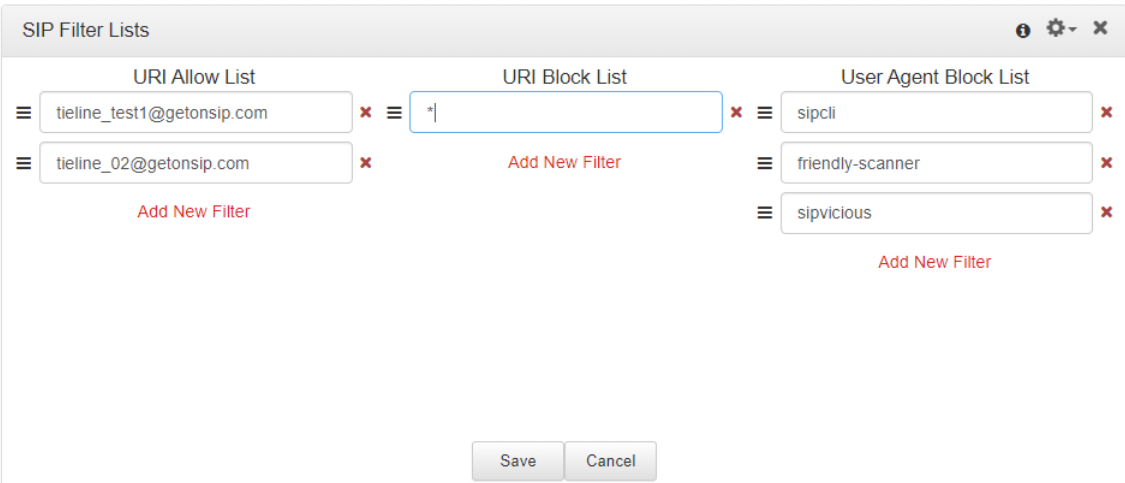
It is also possible to add SIP URIs to the **URI Block List** and add user agents to the **User Agent Block List** to deny them access to the codec. These block lists also filter unwanted traffic and increase the likelihood of rejecting unwanted traffic. Note: If an incoming SIP caller is not on the **URI Allow List** it will be scanned using the **URI Block List**. If there is no match it will be scanned using the **User Agent Block List**. A connection will be established if there is no match on either Block List.

Important Notes:

- To only allow a predefined list of codecs to connect, add them to the **URI Allow List** and add a wildcard (asterisk) * to the **URI Block List**: all incoming calls will be blocked except for codecs in the **URI Allow List**. Add a URL to the **URI Block List** to block traffic from a specific URL.
- The SIP filters operate in the following order:
  1. Deny if User Agent Block List matches
  2. Allow if URI Allow List matches
  3. Deny if URI Block List matches
- Four simple wildcards for matching are allowed by default:
  1. c Any character represents itself apart from those mentioned below.
  2. ? Matches any single character.
  3. * Matches zero or more of any characters.
  4. [...] Sets of characters can be represented in square brackets.

Filter URIs and User Agents

1. Open the HTML5 Toolbox Web-GUI and click **Transport** in the **Menu Bar**, then click **SIP Filter Lists** to launch the **SIP Filter Lists panel**.



2. Click the **Plus symbol** for **URI Allow List**, **URI Block List** or **User Agent Block List** to add a new item to the list.
3. Enter the new item in the text box, click to select the check-box and then click **Save** to store the new setting.

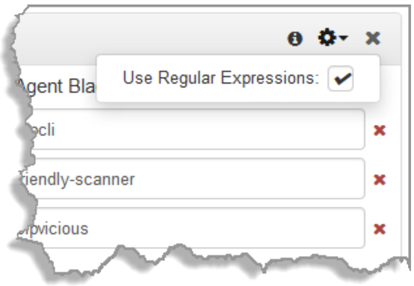
4. Click the **Undo symbol** to undo editing and click and drag the **List symbol** ≡ to shift the position of whitelist and blacklist items.

Important Note:
Some codec manufacturers allow calls based on 'User Agent' identification. It may be necessary to enter a Tieline codec user agent into a non-Tieline codec to connect to a Tieline SIP-enabled codec.

- From firmware v2.16.xx the user agent in the codec is configured as "Tieline <Product Name> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68
- The Bridge-IT II and Bridge-IT XTRA II user agent in the codec is configured as "Tieline <Product Name> <Firmware Version>".
  - **a.** E.g. for Bridge-IT II it will appear like **Tieline Bridge-IT II v3.10.08**
  - **b.** E.g. for Bridge-IT XTRA II it will appear like **Tieline Bridge-IT II XTRA v3.10.08**
- User agent for Report-IT SIP connections: **Tieline Report-IT EE (3.5.6_2894)**. Note: In this example "3.5.6" is the Report-IT version number and "2894" is the build number.
- In Tieline G3 codecs the user agent is configured as "Tieline <ProductName> <Serial Number>". E.g. **Tieline TLR350 8972**. The model numbers for Tieline G3 codecs are as follows:
  - a. Commander G3 Rack TLR300 = Model Number TLR300
  - b. Commander G3 Rack TLR300B = Model Number TLR350
  - c. Commander G3 Field TLF300 = Model Number TLF300
  - d. i-Mix G3 TLM600 = Model Number TLM600

Using Regular Expressions

To filter using regular expressions in the **SIP Filter Lists panel**, click the **Options symbol** in the top right-hand corner of the panel and then click to select the **Use Regular Expressions** check-box.



Important Note: Regular expressions should not use ^ and $ anchors because searches implicitly try to match anywhere in the line.

20.22 Configure Peer-to-Peer SIP Programs

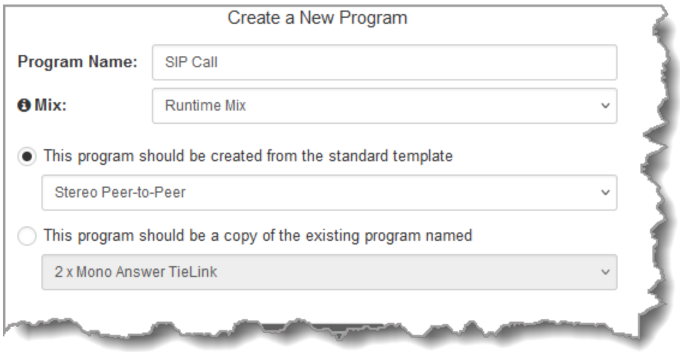
SIP programs are very similar to how a normal Tieline IP program is configured. The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**. It is also necessary to select SIP as the **Session Protocol**.

To configure a SIP multiple stream program, e.g. two mono SIP connections, simply create a new program and configure each SIP audio stream like a single SIP Peer-to-Peer program stream. The codec is capable of registering up to 6 SIP accounts, each of which has an associated **Answer Route** field, which can be matched to a loaded answering program's audio stream Answer Route. Without using SIP accounts, each SIP interface also has an **Answer Route** which can be used for routing 2 audio streams in an answering codec. Note: An account's Answer Route setting is applied first.

Important Notes: Before you start program configuration please note:

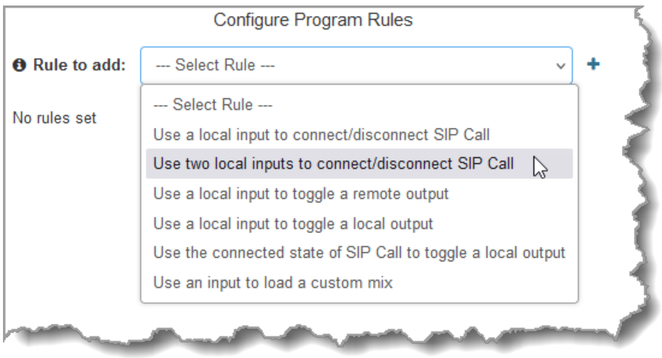
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- Ensure the appropriate TCP and UDP audio ports are open in your firewall to allow SIP audio streams to connect. See Installing the Codec at the Studio for more information.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. Then click **Next**.



Important Notes: When you use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.

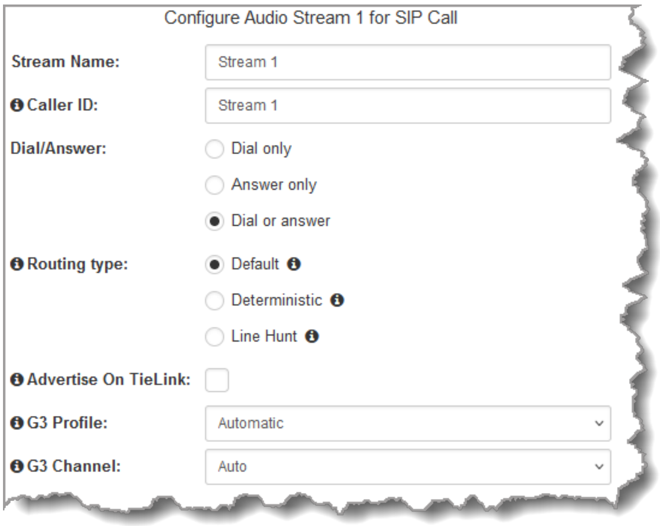


Important Notes for Rules:
- Bridge-IT II and Bridge-IT XTRA II have 4 physical **CONTROL PORT GPIOs**; 7 Tieline virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).

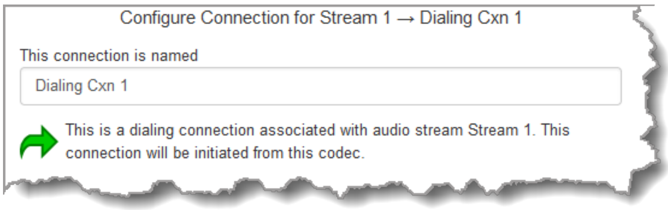
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter the **Stream Name** and configure the codec to dial, answer or dial and answer. Then click **Next**.

Note: The following example displays how to configure a dial and answer program. If you want the codec to either dial or answer only, select the option and the wizard will automatically display screens to allow you to configure the codec correctly. Please note that caller ID, dial routes and G3 profile or G3 channel information can not be used for SIP connections because Tieline session data is replaced by SDP for SIP connections.



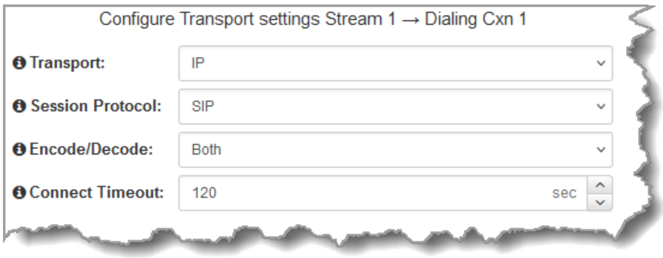
5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



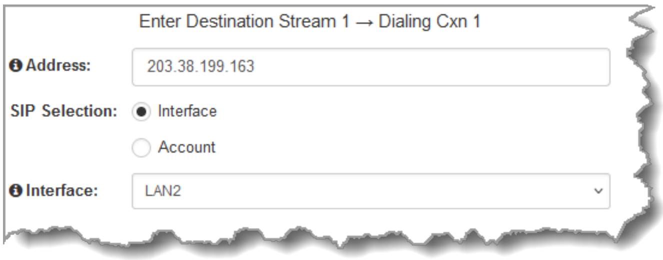
6. Configure the transport settings for the connection. Ensure that you select:

- **IP** as the **Transport**.
- **SIP** from the **Session Protocol** menu option.

Then click **Next**.

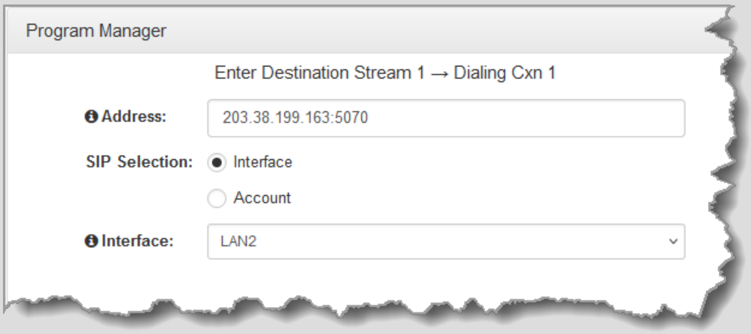


7. Configure the destination codec **Address** if you are dialing peer-to-peer, then specify the network interface used to dial the connection, e.g. **Primary** or **Secondary** (LAN port 1 or 2).

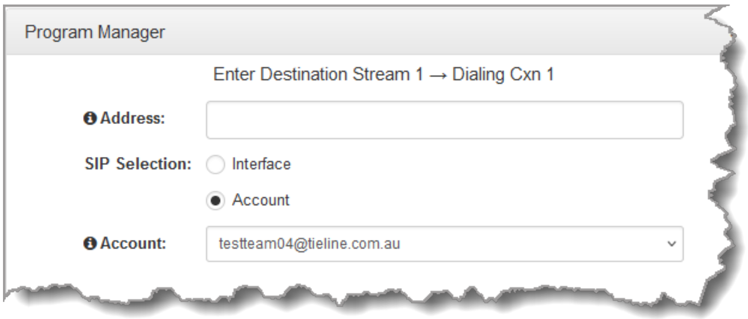


Important Notes:

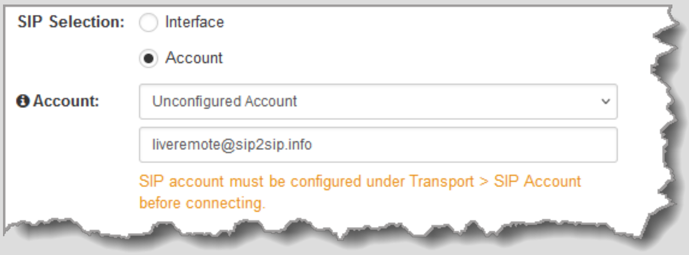
- The default UDP audio port when using SIP for a peer-to-peer connection is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open and forwarded to the other device.
- Tieline codecs automatically add "**sip:**" to the address you enter in the **Address** field when dialing, so it's not necessary to add this.
- Enter the IP address or SIP URI, then a full colon and the session port number to change the session port from the default setting 5060.



If you want to dial from a registered SIP account then the normal workflow is to add the registered account using the **SIP Accounts panel**. Next, select the **Account** radio button in **SIP Selection** and then select the registered account using the **Account** drop-down menu.

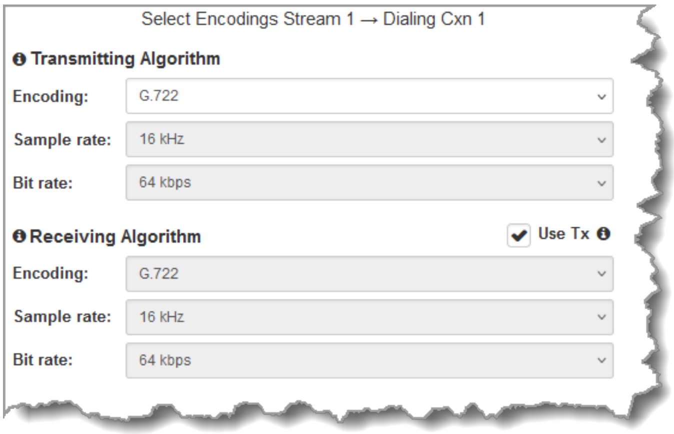


Important Note: A message will be displayed if an account has been added but is not currently registered in the codec. This means it is necessary to register the account using the **SIP Accounts panel** before it can be used to dial a connection.



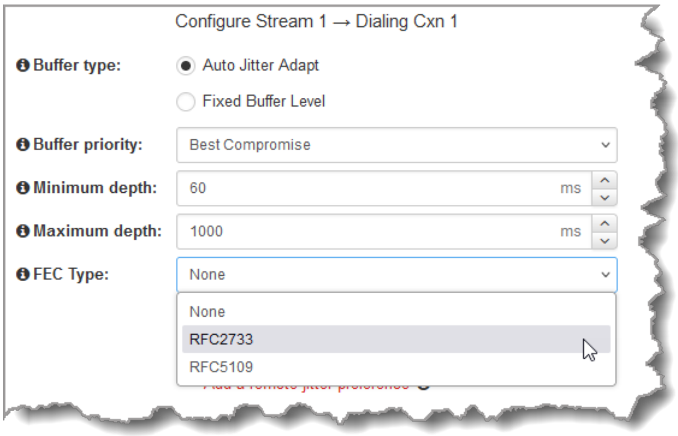
At this point you can click **Save Program** and save the program with default algorithm and jitter settings. Alternatively, click **Next** to confirm and specify algorithm and jitter settings for this connection and configure backup audio settings (recommended).

8. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click **Next** to continue.

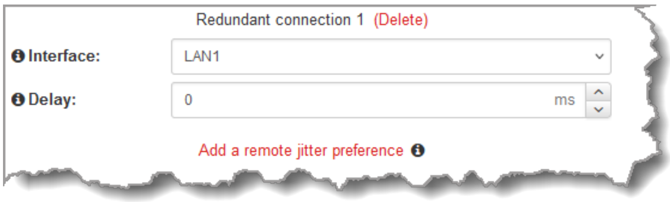


9. Click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- RFC-compliant FEC can also be configured if required and the percentage is configurable.



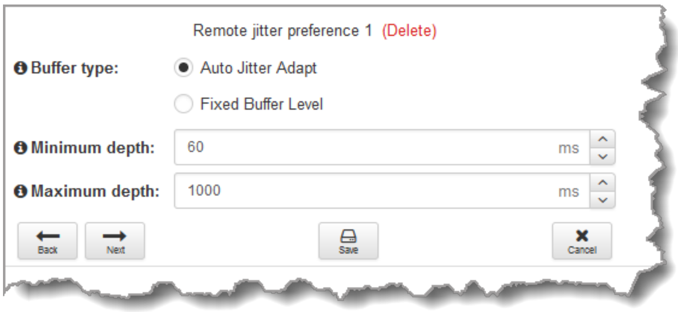
10. Select **Add a Redundant Connection** to configure a redundant stream.



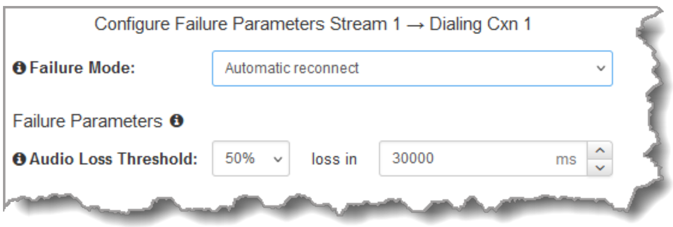
Important Note: Redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if one IP link is lost.

11. Click **Add a remote jitter preference** to send preferred jitter settings to a remote codec. Note: this is just a preference as per EBU Tech 3368 and there is no guarantee that the remote codec will accept or support these jitter configuration settings. Verify configuration settings on the remote codec to ensure settings are correct. Recommended jitter buffer limits are as follows:

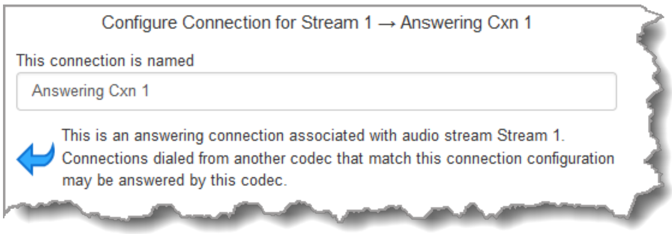
- 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
- 2,500ms for AAC ELD, AAC LD.
- 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS.



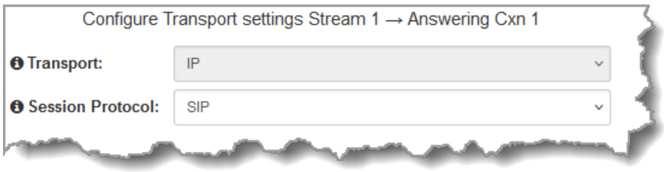
12. Click **Next** and select the drop-down **Failure Mode** arrow to configure **Automatic Reconnect** if required.



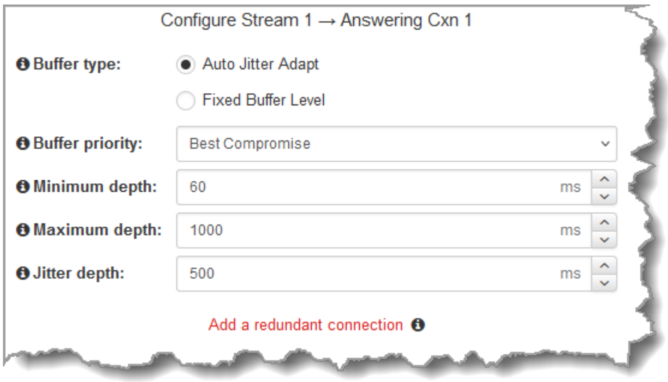
13. Click **Next** to name the answering connection for when calls are received by the codec.



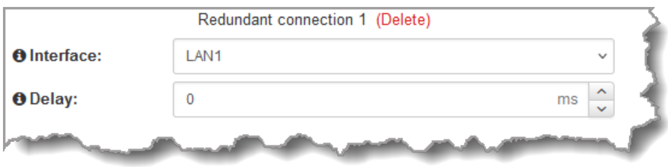
14. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.



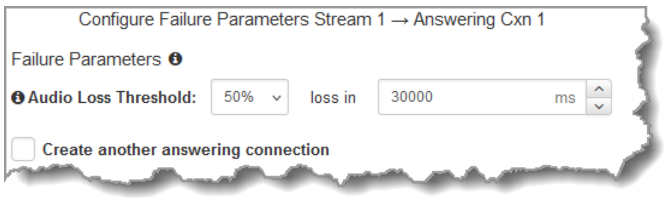
15. Click **Next** to configure the jitter buffer settings for the answering connection.



16. It is also possible to create a redundant connection to accept a redundant stream sent from a remote codec.



17. Click **Next** to configure **Failure Parameters** for the answering connection if required. **Please note**: In most situations the default answering **Failure Parameters** do not need adjustment. These settings may be useful to troubleshoot certain connections, e.g. satellite IP links.



18. After configuring all settings there are 3 options:

    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.

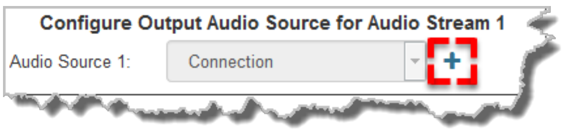
ii. Click **Save** to save the program at this point.
iii. Click **Next** to configure **Output Audio Source** options.

Configuring Output Audio Source Options

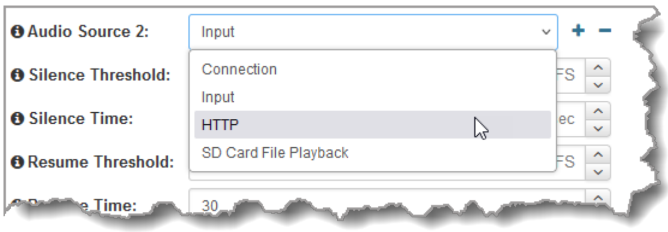
1. Click **Next** to configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program audio at transmitter sites. **Output Audio Source** options include:

   - **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
   - **Input**: Input audio looped to the physical codec outputs.
   - **HTTP**: Icecast client mode to allow media server streaming from a specified URL.
   - **SD Card File Playback**: Audio file playback from an SD card.

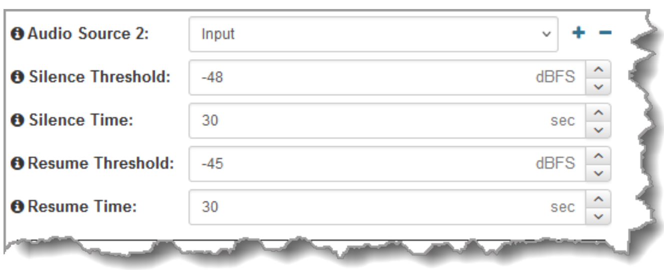
2. Click the blue **Plus symbol** to add a backup **Output Audio Source**, or click the **Minus symbol** to remove an **Output Audio Source**.



3. Click the drop-down arrow to select an **Output Audio Source** option.



4. Configure silence threshold parameters to enable a preferred backup option, as well as resume thresholds for reactivating a previous source. Then click **Save Program** to save program settings.



5. After configuring **Output Audio Source** options you can:

   i. Click **Save** to save the program at this point.
   ii. Click **Next** to configure rules options.

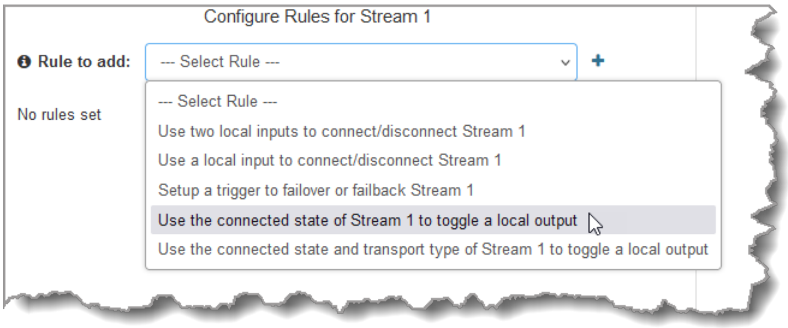
Important Notes for File Playback:

- A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).

- The codec supports SDHC cards which have a physical capacity of up to 32GB. Note: The Windows built-in formatting tool cannot format a SD card larger than 32GB with the FAT32 file system.
- Create MP2 or MP3 files using a 32kHz, 44.1kHz or 48kHz sample rate.
- Ensure recordings are not variable bit rate files.
- SDHC file audio is not sent to codec encoders and cannot be transmitted via an audio stream to another codec.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- If you create a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you create a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
- File playback will occur automatically if the silence threshold parameters are breached. To manage file playback open the **Connections panel** in the web-GUI.

Configuring Rules

1. To configure new stream level rules rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



Important Note: Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

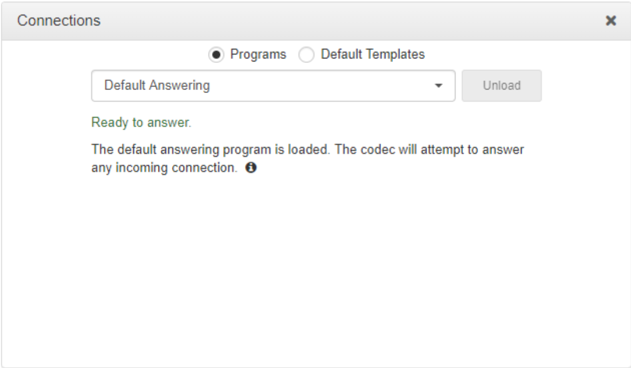
2. Click **Save** to save the program.
3. Click **Finish** to exit the wizard.
4. The newly created program can be loaded from within the **Program Manager panel**, **Connections panel** and the **Program Loader panel** (in the Quick Connect web-GUI). Select and connect audio streams in a program using the **Connections panel**, or connect the program manually using the codec front panel.

20.23 Load, Unload and Dial a Program

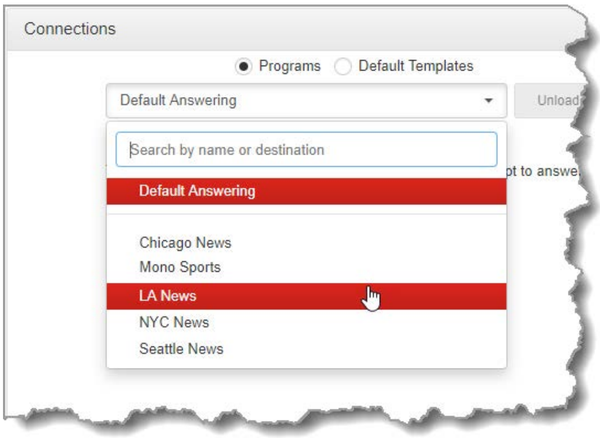
A program can be loaded, unloaded and edited using the **Program Manager panel** or the **Connections panel**. Audio stream dialing settings can be edited without unloading a program. Please note: If you need to use an existing program as a template for a new program, it is necessary to use the **Program Manager panel** to create a program.

Load and Unload a Program

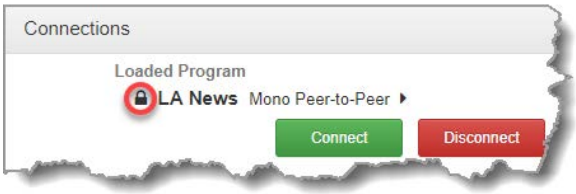
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**.



2. Select **Programs** and then click the drop-down arrow to select one of the programs in the codec, then click **Load**.



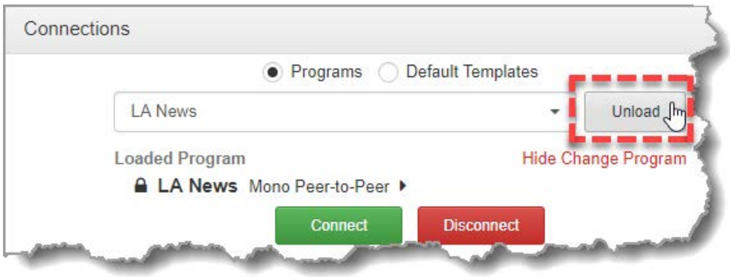
3. If **Lock Loaded User Program** has been configured in the **Options panel**, a black **Padlock** symbol appears next to the program name in the **Connections panel**, to indicate a program is locked in the codec.



Unload a Program

1. To unload a loaded program click **Change Program**.

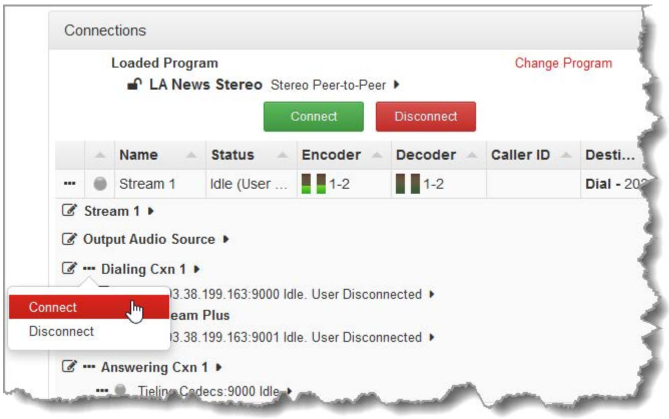
2. Click **Unload**.



Connecting a Program

To connect audio streams and connections within an existing program there are three options:

1. Click the **Connect**  button to connect all audio streams and connections configured in a program.
2. Click the audio stream **3 dot** ··· icon and then click **Connect** to connect all connections associated with this audio stream.

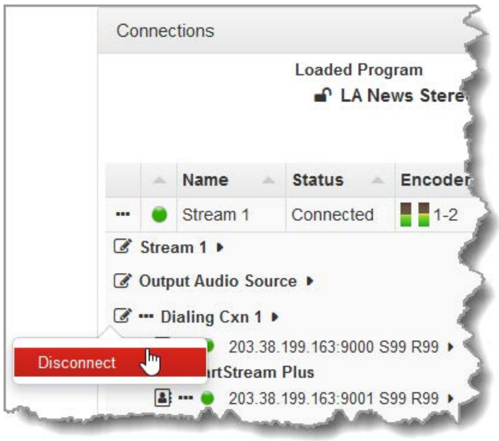


3. Click the connection **3 dot** ··· icon and then click **Connect** to connect an individual audio stream connection.

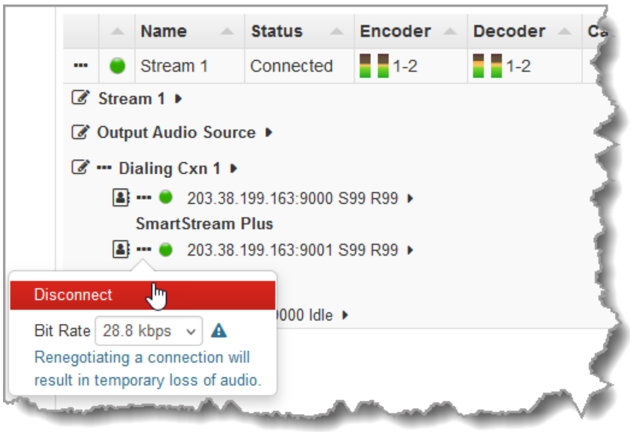
Disconnecting a Program

To disconnect audio streams and connections within an existing program there are three options:

1. Click the **Disconnect** Disconnect button to disconnect all audio streams and connections configured in a program.
2. Click the audio stream **3 dot** ··· icon and then click **Disconnect** to disconnect an individual audio stream and all associated connections.



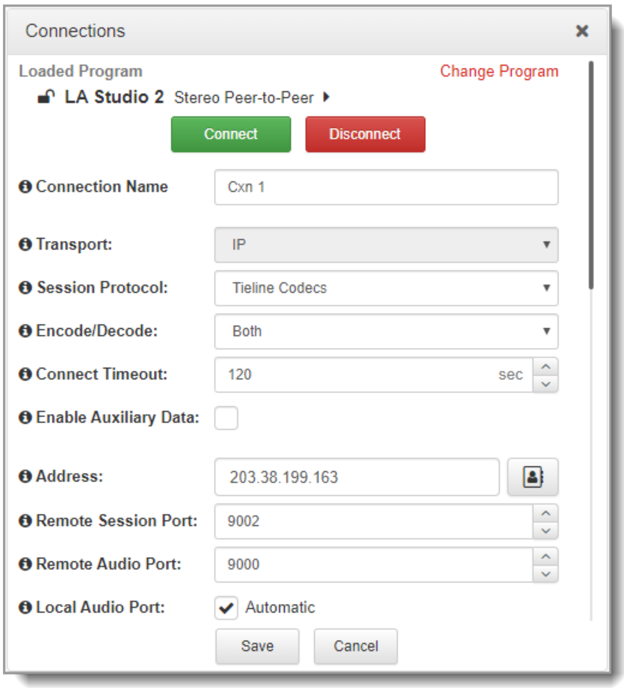
3. Click the connection **3 dot** ··· icon to disconnect an individual audio stream connection.



Change Dialing Settings

To edit destination dialing settings:

1. Click the **Edit symbol** adjacent to a connection.
2. Adjust dialing and connection settings and then click **Save** to change edited settings in the program. Note: The IP address can be changed, or a **TieLink contact** can be selected if Traversal Server Contact Lists have been configured. See Configuring TieLink Settings for more info.



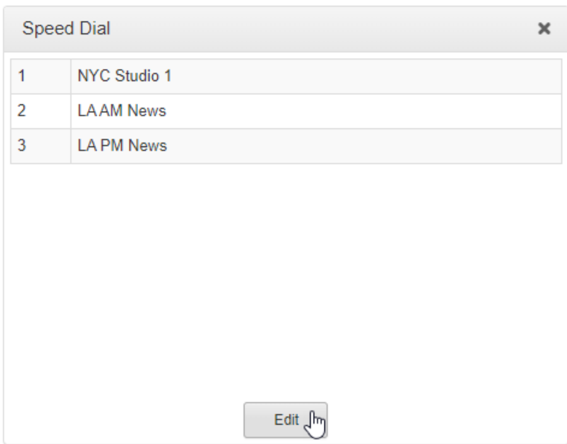
20.24 Configure Speed Dialing

The Toolbox HTML5 web-GUI supports dialing programs using preconfigured speed dials. This is configured and managed using the **Speed Dial panel**.

Create a New Speed Dial

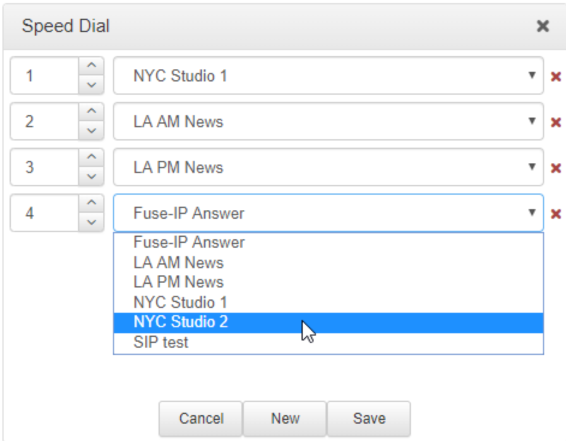
To configure speed dialing:

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Speed Dial** to open the **Speed Dial panel**.
2. Click **Edit**.



3. Click **New** to create a new speed dial and then click the arrows on the left side of the panel to select the speed dial number.

4. Click the drop-down arrow on the right side of the panel to select the program to associate with the speed dial number.

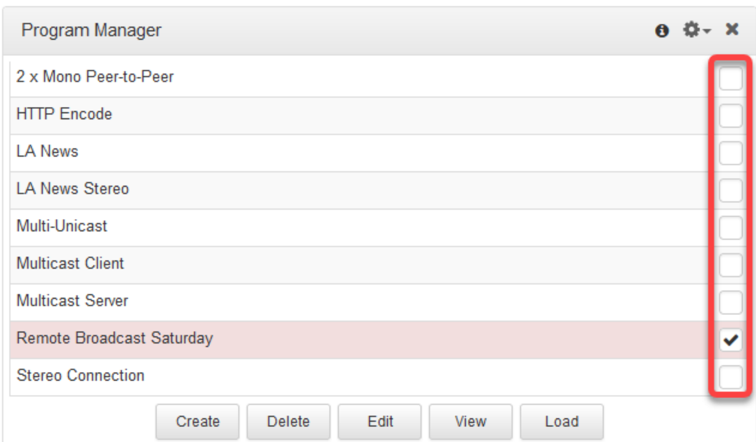


5. Click **Save** to store newly configured Speed Dials.

Important Note: To view speed dial configurations using the front panel of the codec select **SETTINGS** **> Speed Dial** and press the button.

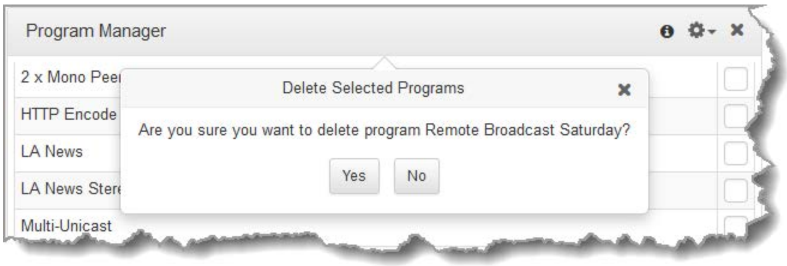
20.25 Delete a Program

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Program Manager** to open the **Program Manager panel**.
2. Click to select the check-box for each program to be deleted. Note: multiple programs can be selected and deleted simultaneously.



3. Click the **Delete** button.

4. Click **Yes** in the **Delete Selected Programs** confirmation dialog to delete all selected programs.



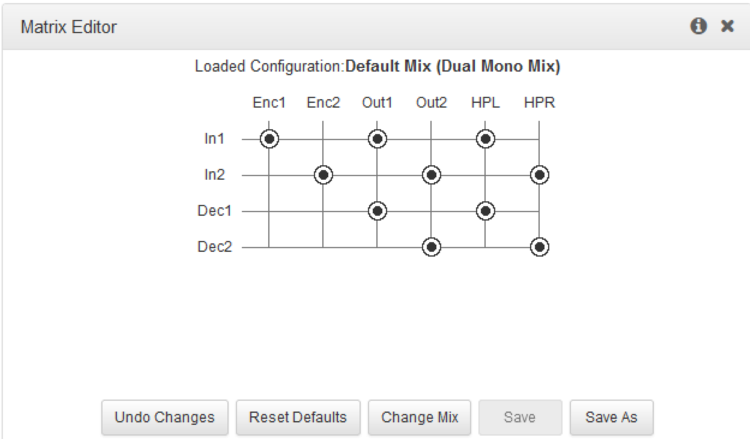
Important Notes: Any program that is currently loaded or listed in the **Scheduler Events panel** cannot be deleted.

20.26 Matrix Editing

The Matrix Editor in the codec allows any input to be routed to any output. Default routing settings are configured for each program type and these default matrices can be edited, saved and recalled as required. All saved custom matrices are available if a compatible program is loaded. If a matrix is not compatible with a program type it will not be visible in the menu, e.g. a saved stereo matrix is not visible when a mono program is loaded. Custom matrices can be created, saved and then backed up with program and scheduler data. This allows them to be copied between codecs by using the Backup and Restore feature.

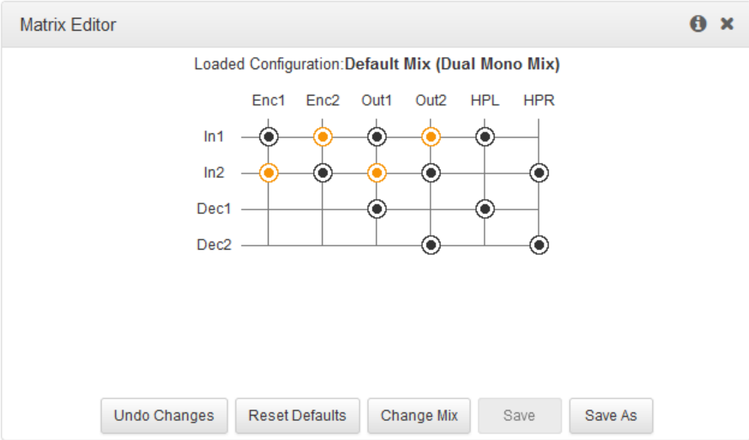
Viewing the Matrix Editor

Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Matrix Editor** to open the **Matrix Editor panel**. In the following image the default dual mono matrix is loaded, as indicated at the top of the panel.



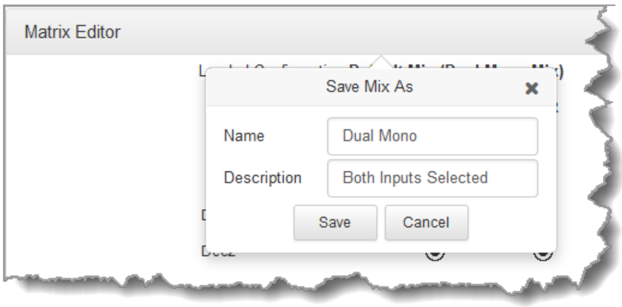
Using the Matrix Editor

Routing can be adjusted very simply in the **Matrix Editor**. To edit default matrix settings click a crosspoint to select or deselect it. Note: Changes not yet saved as a custom mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.



Edits can be saved as a custom mix as follows:

1. Click **Save As**.
2. Enter a new **Name** and **Description** (if required) and click **Save**.



3. The new custom mix stays loaded in the codec until a new mix is loaded, an incompatible program is loaded, or program defaults are restored via **Reset Defaults**.

Important Notes:
- If a new program is loaded and it is compatible with the current custom mix then it will remain loaded. If an incompatible program type is loaded, the last compatible mix to suit the new program type is loaded, including any previous runtime changes. E.g. If a custom stereo mix is loaded and then a mono program is loaded, the last mono mix used will be loaded.
- If you make runtime matrix edits to a loaded mono program, then load a stereo program, and subsequently reload the original mono program, the runtime matrix edits are recalled.

Matrix editor options include:

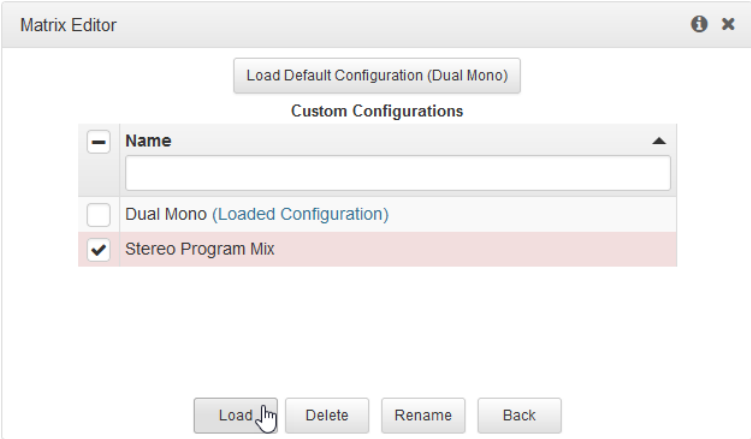
1. **Undo Changes**: Clears any changes that have not been saved.
2. **Reset Defaults**: Resets the matrix to defaults for the currently loaded program, e.g. mono, stereo, mono/stereo plus IFB.
3. **Change Mix**: Allows the selection of an alternative saved matrix. Select and then **Load**, **Rename** or **Delete** the matrix.
4. **Save as**: Save the current matrix settings as a new **Custom Mix** with a unique name.

Important Note: Click the **Information symbol** at the top of the panel to view details about the matrix.

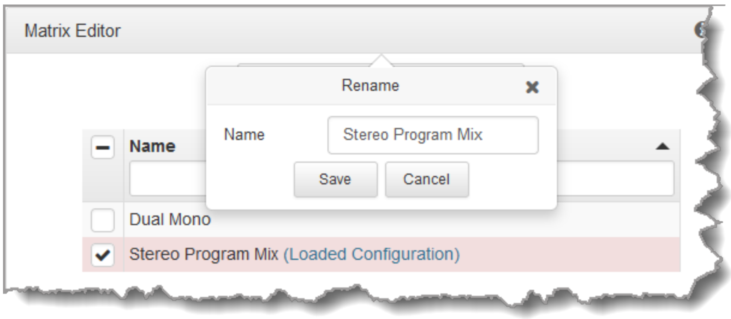
Load, Rename and Delete a Custom Matrix Editor Mix

When a new program is loaded it may be necessary to load a new custom mix. It is also possible to rename or delete mixes that are no longer required.

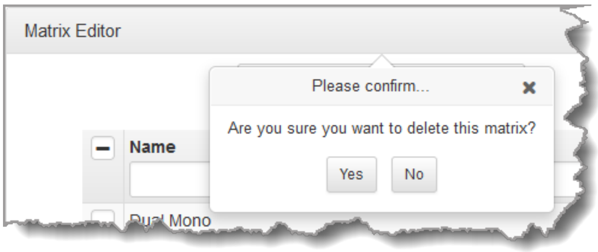
1. To load a custom mix click **Change Mix**, select a mix and click **Load**.



2. To rename a custom mix, click to select a saved mix and then click **Rename**. Edit the name and click **Save**.

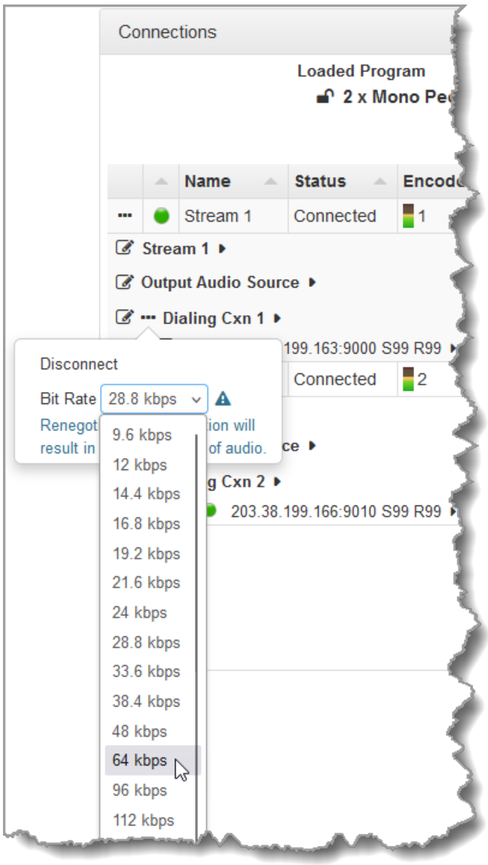


3. To delete a custom mix, click to select a custom mix and then click **Delete**. Note: A loaded mix cannot be deleted.



20.27 Adjusting the Connection Bit Rate

1. Open the HTML5 Toolbox Web-GUI and click **Connect**, then select **Connections** to open the **Connections panel**.
2. Click the **3 dot** ··· icon for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit-rate.

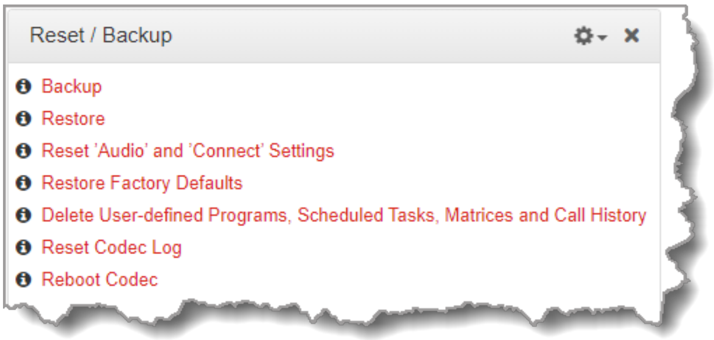


Important Notes:
- It is not possible to renegotiate the connection bit rate of a SIP connection.
- Renegotiating a connection will result in the temporary loss of audio.

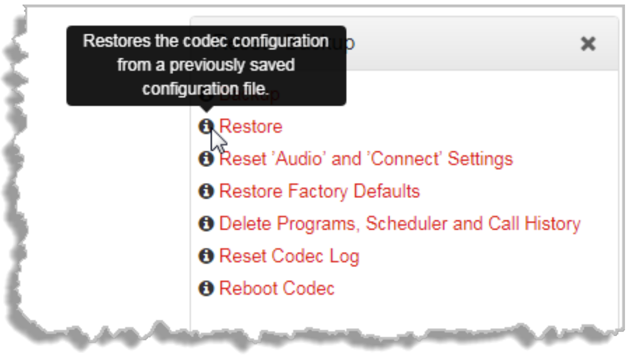
20.28 Reset Factory Default Settings

There are several options which allow you to restore factory default settings within the codec. See Reset and Restore Factory Defaults for more details on each option.

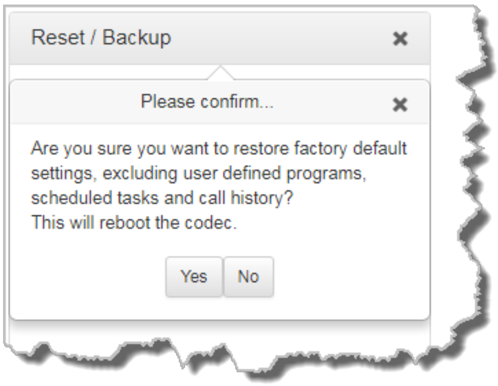
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset / Backup** to display the **Reset / Backup panel**.



2. Click one of the available reset options to adjust codec settings, or reboot the codec. Note: Hover with the mouse pointer over the **Information** symbol to view a tool-tip for each reset option.



3. A confirmation dialog appears for each option, then click **Yes** to proceed.



Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

20.29 Backup and Restore Functions

The HTML5 Toolbox Web-GUI can be used to backup and restore codec settings, including:

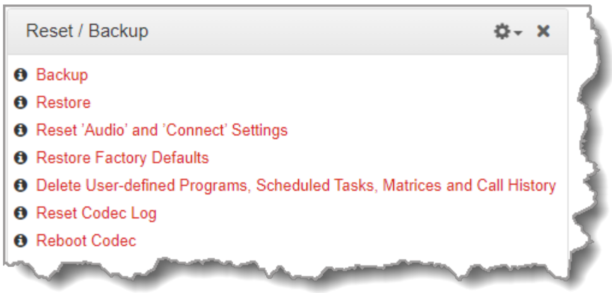
- Programs containing a variety of connection settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

Files can also be used to copy configurations onto other similar codecs. Programs are essentially connection profiles that may include:

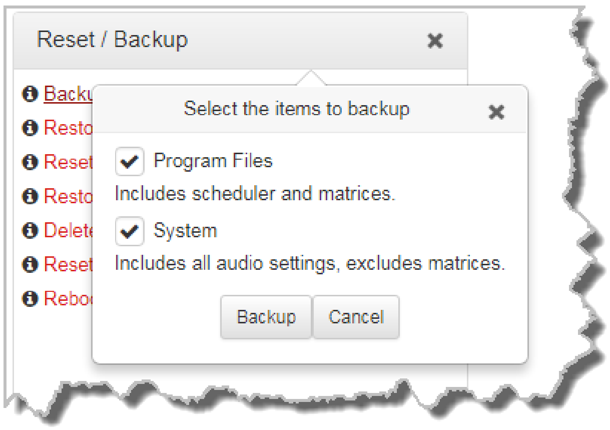
- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

Creating Backup Files

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset / Backup** to display the **Reset / Backup panel**.



2. Click **Backup**.
3. Click to select the check-boxes to confirm your backup requirements, then click **Backup**.

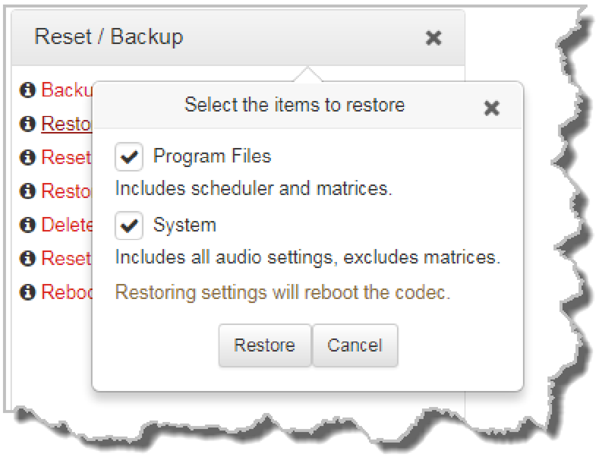


4. Select a location on your PC to save the configuration file. Note: You may need to "allow" your browser to display the pop-up dialog.

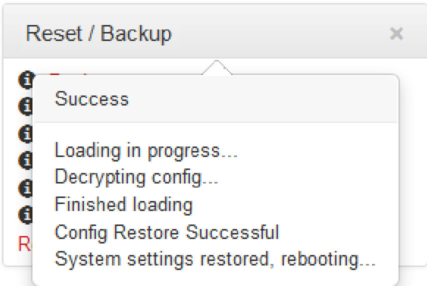
Restoring Configuration File Settings

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset / Backup** to display the **Reset / Backup panel**.
2. Click **Restore**.

3. Click to select the check-boxes and confirm your restore settings. For example, you could select the **Program Files** check-box and deselect the**System** check-box if you are only copying programs onto codecs.



4. Click **Restore** and select the .tgz file to load onto the codec. A **Success** dialog confirms the files have been restored.



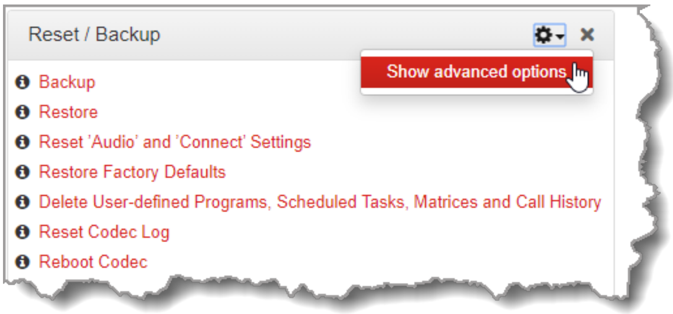
Note: The codec will automatically reboot when restoring system settings.

Advanced Settings: XML Config

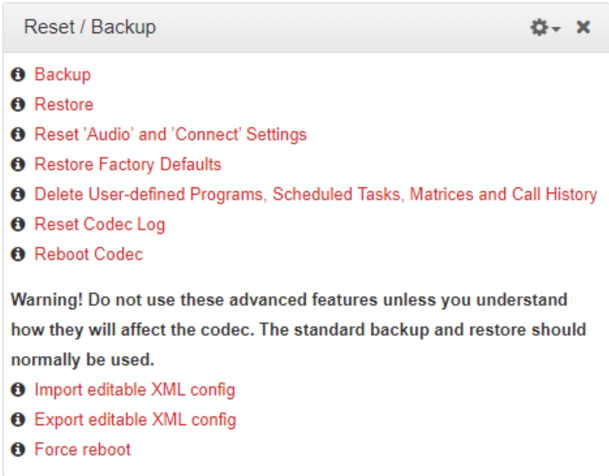
Caution: DO NOT use advanced XML configuration features unless you fully understand how they will affect the codec. The standard backup and restore function should normally be used. Damage to the codec may occur if this feature is used without fully understanding how it will affect the codec.

XML Config is a highly advanced feature which should only be performed by suitably qualified personnel. To import or export XML config files:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset/Backup** to display the **Reset/Backup panel**.
2. Click the **Options symbol** to view **Show Advanced Options**.



3. Click to select **Import/Export editable XML config** as required, or force the codec to reboot.



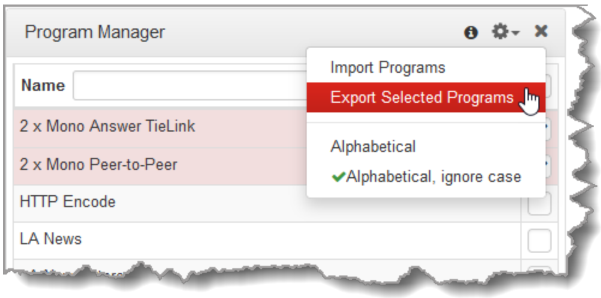
20.30 Import and Export Programs

It is possible to import and export individual programs via the **Program Manager panel**.

Exporting Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click to select the check-box for the programs you wish to export.

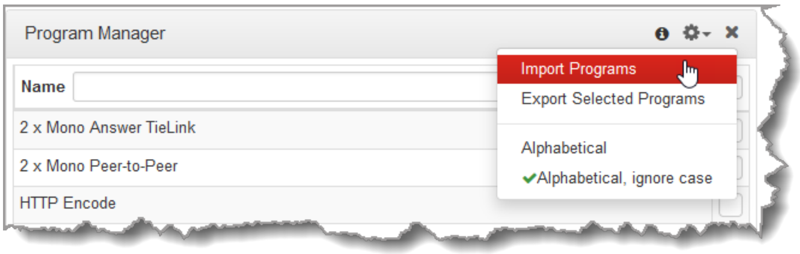
3. Click the **Options symbol** in the top right-hand corner of the **Program Manager** and select **Export Selected Programs**.



4. Navigate to a file folder and save the program .zip file.
5. Click **Save** to save the program file.

Importing Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Options symbol** ⚙ in the top right-hand corner of the **Program Manager** and select **Import Programs**.

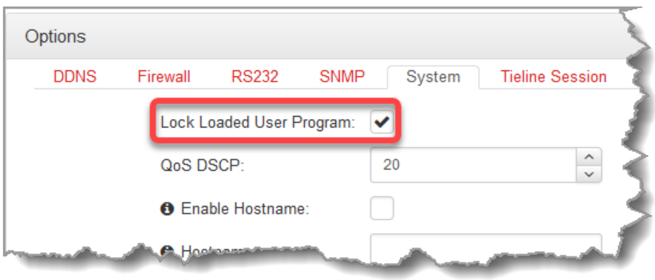


3. Navigate to the file folder containing the program .zip file to be imported.
4. Click to select the .zip file and click **Open** to select and import required programs.

20.31 Lock or Unlock Programs

It is possible to lock a loaded custom program in a codec to ensure the currently loaded program type, e.g. mono, cannot be unloaded by a codec dialing in with a different program type, e.g. stereo. For example, if you require the codec at the studio to always connect in mono, simply load and lock a mono program in the codec. Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will be rejected.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **System** and click **Edit**.
3. Click the **Lock Loaded User Program** check-box to lock or unlock user programs in the codec, then click **Save**.



Important Note:

- A black **Padlock** symbol appears next to the program name in the **Connections panel** and in the **Program Loader panel** (in the Quick Connect web-GUI), to indicate a program is locked in the codec.
- It is only possible to lock custom programs in a codec.
- If **Lock Program** is enabled and you load a new custom program in the codec, **Lock Program** remains enabled and locks the most recently loaded custom program.

20.32 HTML5 Software License Installation

Perform an Automatic Software License Install with the HTML5 Toolbox Web-GUI

Prior to installation it is necessary connect the codec to the internet in order to update codec licenses via TieServer. This may be requested during customer support when troubleshooting an issue.

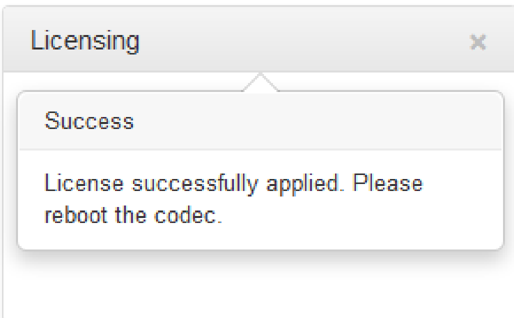
1. Open the HTML5 Toolbox web-GUI in a browser on your PC by typing either the IP address of the codec (LAN connection), or the USB address of the codec (USB connection) into the address bar.
2. Ensure you have unloaded any currently loaded program in the codec via the **Program Loader panel**.
3. Click **Settings** in the **Menu Bar**, then click **Licensing** to display the **Licensing panel**.
4. Click **Update from TieServer**.
5. A **Success** dialog in the web-GUI **Licensing panel** confirms when installation is complete and the codec screen should display a confirmation message.

6. Reboot the codec via **Settings > Reset > Reboot Codec** and press the  button. Note: do not reboot by removing the power cable from the codec.

Download a License File and Install Manually

Prior to installing any new software license manually it is necessary to connect the codec to a PC and save the license file on the computer.

1. Open the HTML5 Toolbox web-GUI in a browser on your PC.
2. Click **Settings** in the **Menu Bar**, then click **Licensing** to display the **Licensing panel**.
3. Click **Upload a selected file**.
4. Navigate to the ".lcf" license file on your PC, then click the **Open** button to commence license installation.
5. A **Success** dialog in the web-GUI **Licensing panel** confirms when installation is complete and the codec screen should display a confirmation message.



6. Reboot the codec using the **Reset / Backup panel** in the web-GUI, or via the codec front panel by selecting **Settings > Reset/Backup > Reboot Codec** and press the OK button. Note: do not reboot by removing the power cable from the codec.

20.33 Configuring SNMP in the Codec

The codec supports Simple Network Management Protocol (SNMP) for managing devices on IP networks. There are two elements to configuring SNMP in your codec:

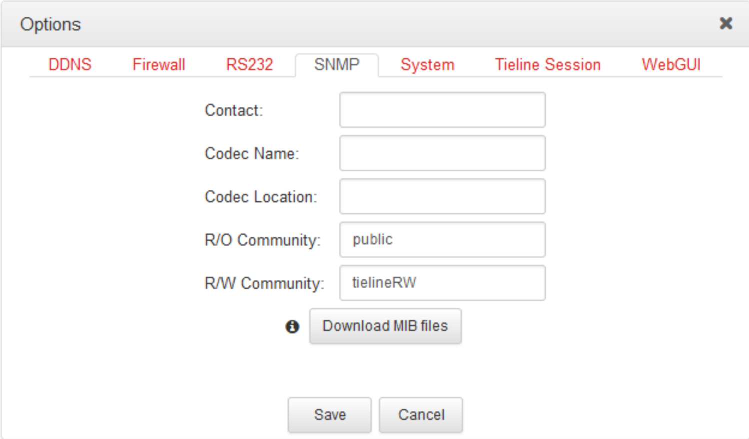
1. Configure SNMP Device settings in your codec.
2. Configure SNMP Traps via the **Alarms panel** in the Web-GUI (see SNMP Trap Configuration, or to configure using the codec front panel see Configuring SNMP Settings).

Description of SNMP Settings in the Codec

Features	Operation Button Descriptions
Codec Name	A user-specified alphanumeric identifier which may be used by third-party SNMP software to identify a device. The device name corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysName" SNMP attribute and is completely independent of DNS, NIS, WINS or other device naming and identification schemes, though convention is to use the device's fully-qualified domain name.
Codec Location	A user-specified alphanumeric string which may be used by third-party SNMP software to identify a device. Device location corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysLocation" SNMP attribute.
Contact	A text identifier for the contact person for this managed node, together with information on how to contact this person.
R/O Community	SNMP provides two types of access, namely Read-Only access and Read-Write access. The R/O Community identifier allows Read Only level access.
R/W Community	The R/W Community identifier allows Read/Write level access.

Configuring SNMP Settings in the Codec

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **SNMP** and click **Edit**.
3. Click in the text boxes to enter SNMP configuration settings, then click **Save**.



4. Click **Save** to store the new configuration.

MIB Files for SNMP Configuration

Management Information Base (MIB) files are required for SNMP applications to interact with your Tieline codec and interpret SNMP data. The codec supports SNMPv1 and SNMPv2 MIB protocols. Click the **Download MIB files** button to download the MIB .zip file to a PC and import the contents into the MIB browser used to manage SNMP-enabled network devices.

MIB files can also be downloaded from the codec using the following link in a web browser on a device connected to the same network as the codec:

- http://<YOUR_CODEC_ADDRESS>/mibs/tieline-mibs.zip

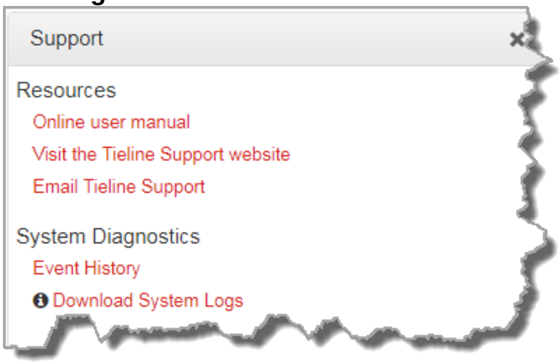
Important Notes:
- The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213. An example of a free MIB browser is available at http://www.ireasoning.com/.
- Tieline recommends SNMP is disabled if a codec is connected to a public network like the internet. Adjust settings using the Toolbox HTML5 Web-GUI **Options panel** in the **Firewall** tab, or see Firewall Configuration.

20.34 Download Logs

The codec is capable of providing diagnostic information via user logs, which can either be sent to Tieline support, or downloaded for user diagnostics.

Procedure for Sending Logs to Tieline

1. Open the HTML5 Toolbox Web-GUI and click **Help** in the **Menu Bar**, then click **Support**.
2. Click **Download System Logs**.



3. Save the file to your computer and then send the .zip file to Tieline support at support@tieline.com

Download Event Logs

Event logs can be downloaded from the codec and viewed in your browser.

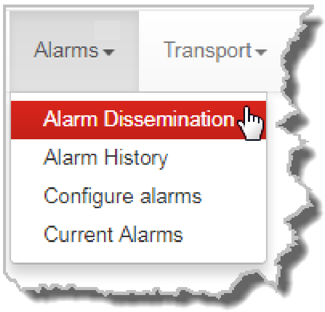
1. Open the HTML5 Toolbox Web-GUI and click **Help** in the **Menu Bar**, then click **Support**.
2. Click **Event History** to view the event log in a new web-browser window.

Clearing Logs

This option should only be used if instructed to by Tieline support staff. To clear all event and other logs in the codec via the front panel see the Reset and Restore Factory Default Settings section of this manual, or see Reset Factory Default Settings to clear recent log history using the Web-GUI.

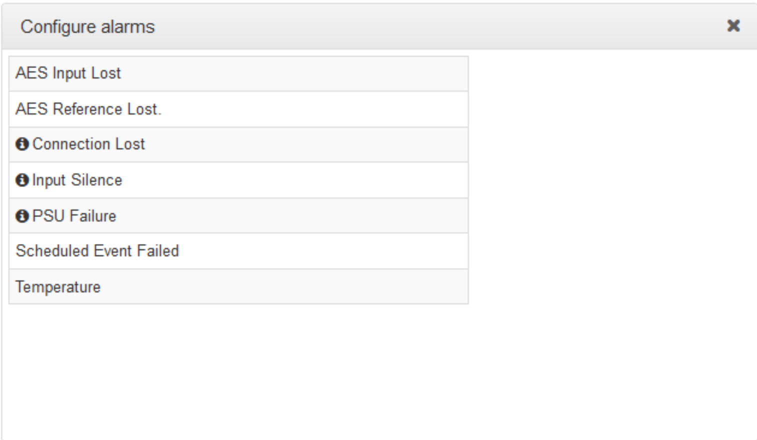
20.35 Configuring Alarms

Open the HTML5 Toolbox Web-GUI and click **Alarms** in the **Menu Bar** to open and view panels used to configure and monitor a range of alarms.

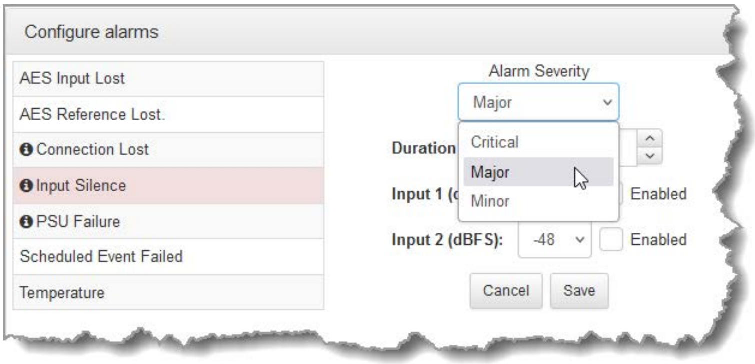


Configure and Enable Alarms

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Configure alarms** to open the panel.



2. Click to select an alarm from the list on the left side of the panel.
3. Click **Edit** to configure alarm settings.
4. Click the **Enabled** check-box to activate the alarm and then select an **Alarm Severity** level from the drop-down menu.



5. Click **Save** to store the new settings.

The following table outlines **System** and **Audio** alarms:

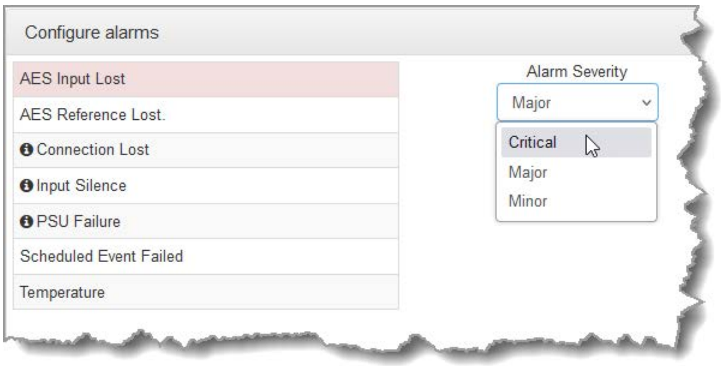
Alarm	Alarm Type	Explanation
AES Input Lost	Audio	Raises an alarm if the AES input signal is lost
AES Reference Lost	Audio	Raises an alarm if the AES reference clock signal is lost
Connection Lost	Audio	Triggers and alarm whenever a streaming connection is lost
Input Silence	Audio	Raises an alarm if input audio is lost based on whether silence is detected on a single input or pair of inputs as selected (according to preconfigured silence detection threshold parameters)
PSU Failure	System	Raises an alarm if one or both PSUs fail in Bridge-IT XTRA II
Scheduled Event Failed	Audio	A scheduled event could not be started or stopped at the specified time. Note: the alarm is only present for the duration of the event.
Temperature	System	Raises an alarm if the temperature is too low or too high

Configuring an Alarm's Severity Level

Codec alarms can be configured for three different severity levels:

1. Click to select an alarm from those displayed in the **Configure alarms** panel.

2. Click **Edit** to configure alarm settings.
3. Click the **Alarm Severity** drop-down menu and select the preferred severity level.

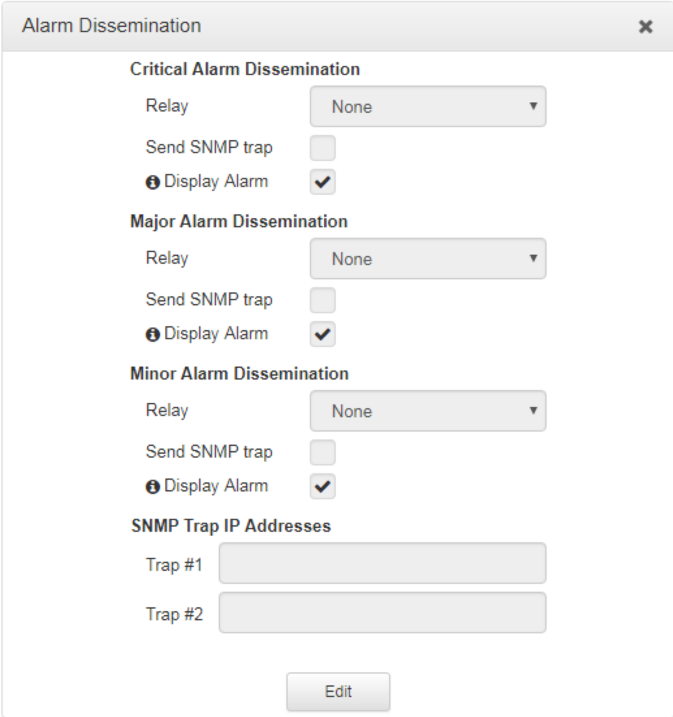


4. Click **Save** to store the new settings for the selected alarm.

Configuring Alarm Dissemination Severity Alerts

Alerts for each alarm severity level are configured using the **Alarm Dissemination** panel.

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Alarm Dissemination** to open the panel.



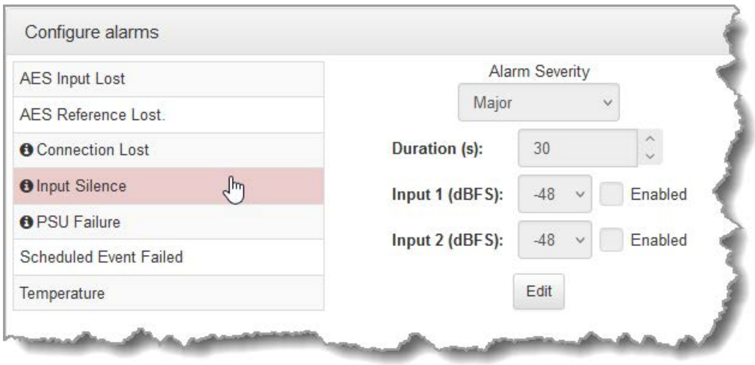
2. Click **Edit** to configure notification settings.
3. Select and configure relay, SNMP trap and alarm display settings for each **Alarm Severity** level. Enter **SNMP Trap IP Addresses** as required at the bottom of the panel.
4. Click **Save** to store the new settings.

Important Note: Simple Network Management Protocol (SNMP) is a protocol used to manage devices on IP networks. SNMP provides the ability to send traps (notifications or alerts), which are packets containing data relating to a system component that may be either statistic or status related. Please see your system administrator if you require more information.

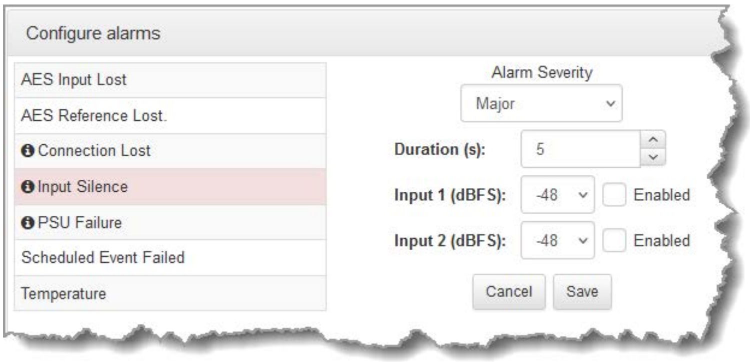
Configuring Input Silence Detection Parameters

When configuring an **Input Silence** alarm it is also necessary to configure the audio silence thresholds and timeout duration.

1. Click **Input Silence** to select the alarm.



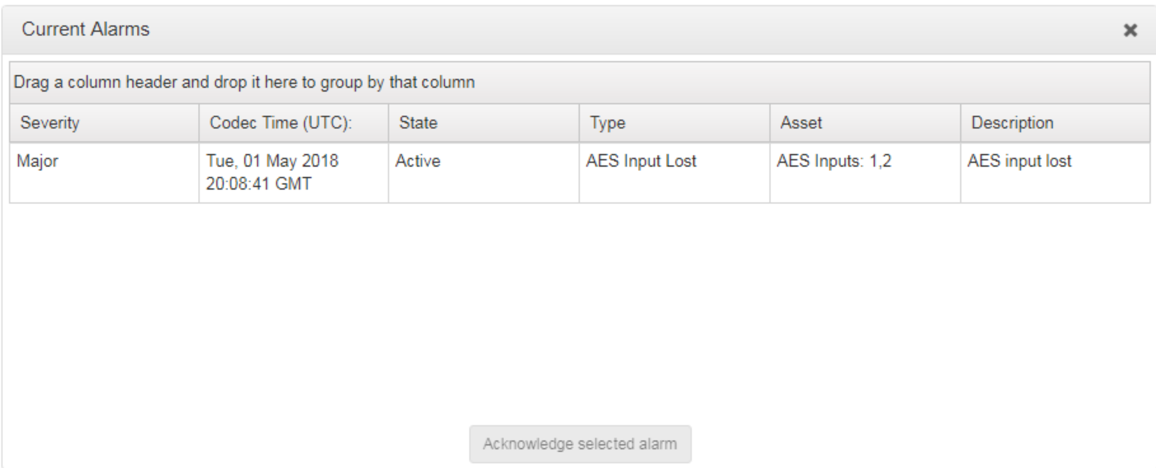
2. Click **Edit** to configure alarm settings.
3. Configure the dBFS threshold and timeout duration in seconds and ensure the input **Enabled** check-boxes are selected. An alarm will be raised when these thresholds are breached.



4. Click **Save** to store the new input silence alarm settings.

20.35.1 Managing Alarms

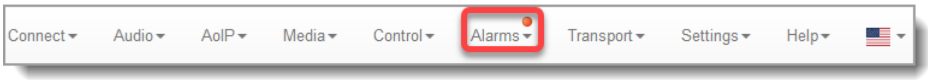
Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Current Alarms** to view active alarms.



Viewing Current Alarms

Active alarms are indicated by:

1. The red **Alarm Symbol** flashing in the toolbar of the HTML5 Toolbox Web-GUI screen.



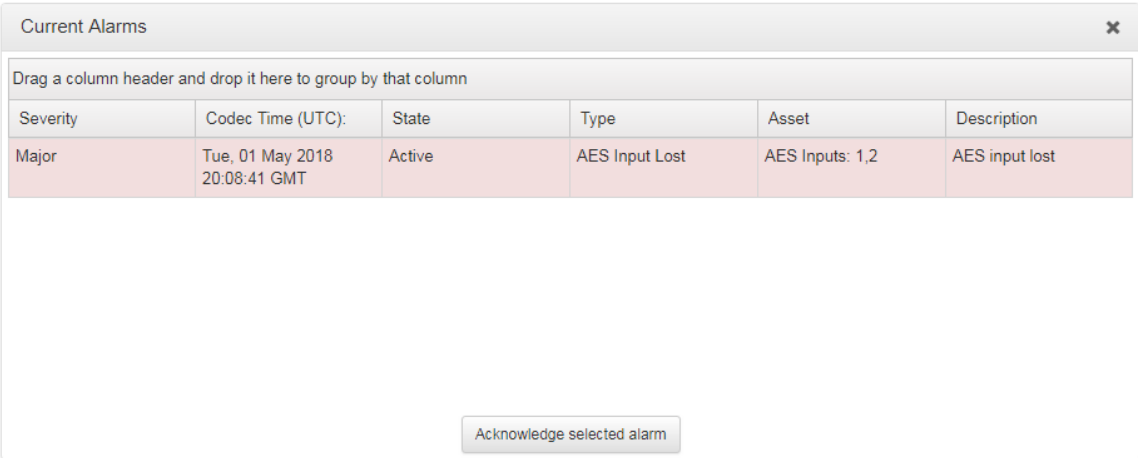
2. All new alarms being listed in the **Current Alarms panel**.
3. Other alerts as per **Alarm Dissemination panel** settings.
4. The codec front panel **ALARM LED** flashing red.

Important Note: When a connection is active the front panel **CONNECTED LED** is illuminated solid green. Illumination ceases if a connection is lost.

Acknowledging Alarms

To acknowledge an alarm in the **Current Alarms panel**:

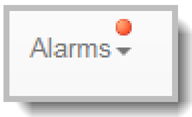
1. Click to select the alarm in the **Current Alarms panel**.



2. Click **Acknowledge selected alarm**.

After acknowledging the alarm:

1. The **State** will change from **Active** to **Acknowledged**.
2. The red **Alarm Symbol** will stop flashing but remain visible in the toolbar of the HTML5 Toolbox Web-GUI screen.



3. The **Alarm Symbol** in the top right-hand corner of the **OLED SCREEN** will stop flashing.

4. The state of other alerts may change, as per **Alarm Dissemination panel** settings.

Alarm State	Front Panel OLED Alert	HTML5 Toolbox Web-GUI
Active	Flashing Alarm Symbol	Flashing Red Toolbar

Alarm State	Front Panel OLED Alert	HTML5 Toolbox Web-GUI
		Alarm Symbol
Acknowledged	Stops flashing & remains visible	No symbol displayed

Summary of Alarm Indications

Deactivating Alarms

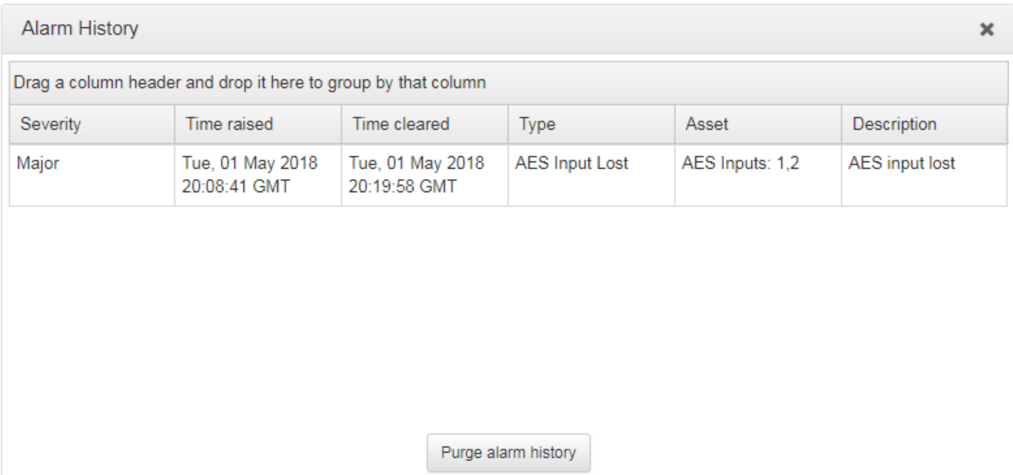
An alarm is deactivated automatically when the alarm state is reversed. E.g. if audio is restored after an **Input Silence** alarm.

Deactivating Input Silence Alarms

An **Input Silence** alarm is activated when the configured audio and duration thresholds have been breached. To recover from this alarm state the codec must detect input audio higher than the failure threshold. When audio at this level is detected, the codec monitors input audio to ensure it doesn't drop below the recovery threshold setting more than 5 times within the nominated **Input Silence** duration time. The alarm is then deactivated automatically.

Alarm History

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Alarm History** to display a record of all system alarms which have been raised.



2. Click the **Purge alarm history** button to clear all alarms from the **Alarm History** panel.

20.36 RS232 Data Configuration

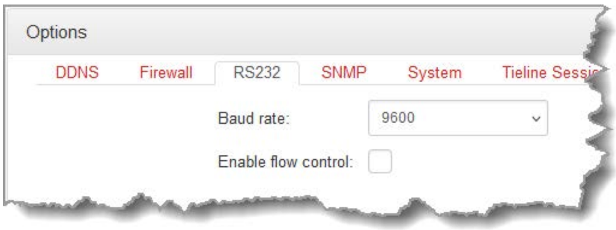
The codec supports both in-band and out-of-band data depending on the connection transport and algorithm used. In-band RPTP data is automatically included within an audio stream when using Tieline Music and Music PLUS algorithms. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP Transport
Tieline Music and MusicPLUS	• In-band RPTP data is enabled automatically • Synchronized out-of-band data can be enabled and disabled as required • Using out-of-band data with rules between G5 or G6 codecs employing relay reflection minimizes latency • These algorithms must be used when connected to G3 codecs as they don't support out-of-band data
All other algorithms	• No in-band data available; synchronized out-of-band data can be enabled and disabled

Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** or **Connections panel** to enable out-of-band data and activate rules employing relay reflection over a connection. This will allow the codec to connect to external devices and send RS232-compatible data via the serial port on the rear panel. Alternatively, enable auxiliary data using the front panel **OLED SCREEN** menus via **Settings > System > RS232** (see GPIOs and RS232 Data). See Appendix B for Control Port and RS232 connector pinouts.

Setting RS232 Data Rates and Flow Control

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **RS232**, then click **Edit** and click the **Baud rate** drop-down menu arrow to select the serial port baud rate. Ensure this matches the baud rate of the external device connected to the RS232 port on the codec.
3. Click to select the **Enable flow control** check box and enable flow control, then click **Save** to store the new settings.



Important Notes:

- When connecting over IP to G3 codecs only in-band data is available via the Music and MusicPLUS algorithms.
- Use firmware higher than 2.8.xx in the G5 Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data over multicast connections.
- It is important to enable serial port flow control as it regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you configure the serial port baud rate to match the setting of the external device to which you are connecting. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Session data sent from the dialing codec will configure all other compatible codecs (non-G3) when you connect.

- RS232 data can be sent from the dialing codec to all endpoints of a multi-unicast or multicast connection if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

20.37 Creating Rules

Codec 'rules' configure events based on specific codec actions. A range of default rules are preprogrammed into the codec to facilitate activation of the most common events required by broadcast engineers. Typically rules are based on a change in the state of a physical **CONTROL PORT** GPIO, or a logic IO, or a codec program being connected or disconnected. There are three categories of rules:

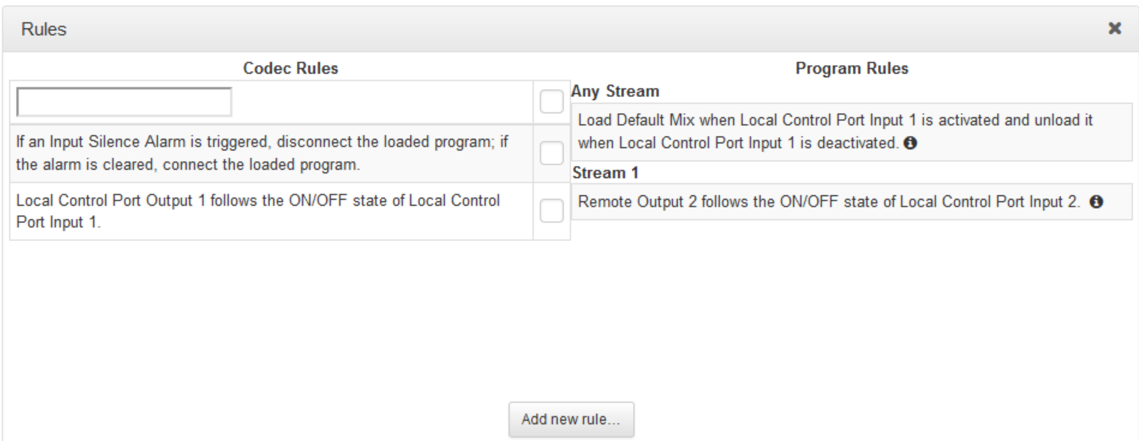
1. Codec level rules: Rules based on programs or codec hardware and software I/O states, e.g. Connect or disconnect a program when an input is toggled, or synchronize a local input to a remote relay.
2. Program level rules: Rules based on codec behaviors at the program level, e.g. Connect and disconnect program A when an input is toggled, or synchronize a local input to a remote relay.
3. Stream level rules: Rules based on codec behaviors at the stream level, e.g. Synchronize a local input to a remote relay.

There are three ways to create rules in the HTML5 Toolbox Web-GUI:

1. **Rules panel**: Configure codec level rules related to programs and/or hardware and software I/O states.
2. **Program Manager panel**: Configure program level rules early in the **Program Manager panel** wizard.
3. **Program Manager panel**: Configure stream level rules for each audio stream as you proceed through the **Program Manager panel** wizard.

Important Notes:
- Rules can only be created with the Web-GUI while the codec is disconnected.
- Program and stream level rules configured in the **Program Manager panel** are only active when the program is loaded.

Following is a summary of how codec, program and stream level rules are displayed in the **Rules panel** when configured.



Enabling Data

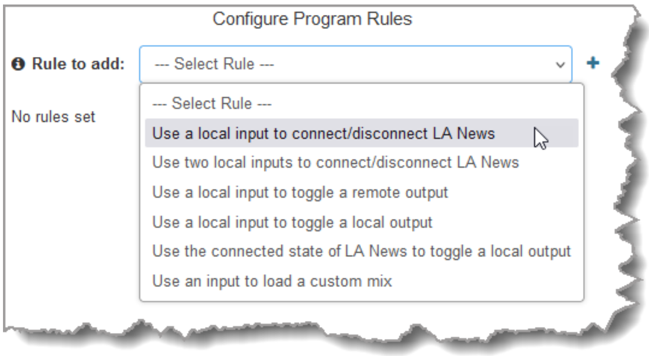
Data is disabled by default and must be enabled to allow contact closure operation and transmission of RS232 data. Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** or **Connections panel** to enable out-of-band RS232 data and activate rules employing relay reflection over a connection. It is also possible to enable data using the front panel of the codec. See GPIOs and RS232 Data.

Configure Rules with the Program Manager Panel

To configure program or stream level rules follow the instructions in this user manual for setting up connections.

Program Level Rules

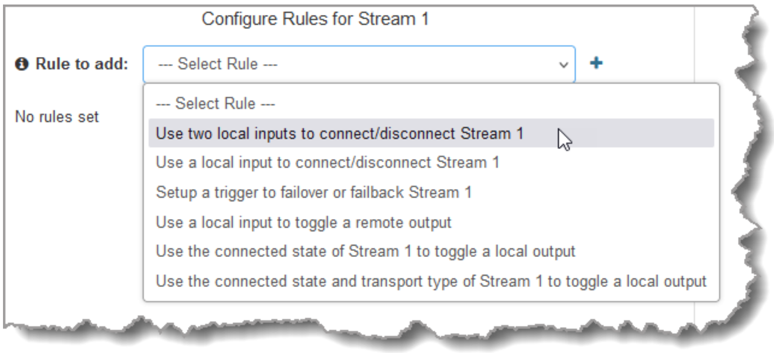
In the **Program Manager panel** wizard use the **Configure Program Rules** screen to configure program level rules. The rules available are displayed in the following image.



Note: Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.

Stream Level Rules

In the **Program Manager panel** wizard use the **Configure Rules for Audio Stream** screen later in the wizard to configure stream level rules. The rules available are displayed in the following image.

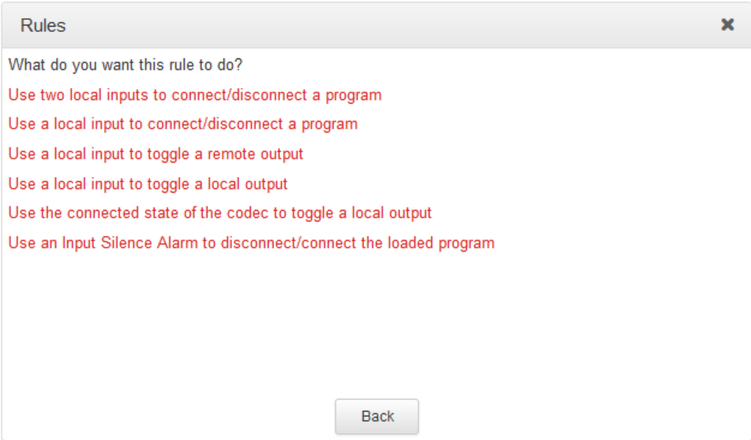


Note: A subset of filtered rules will be displayed for **Answer only** audio stream connections.

Configure Rules using the Rules Panel

Use the **Rules panel** to configure codec level rules related to programs and/or hardware and software I/O states.

1. Open the Toolbox HTML5 Web-GUI and click **Control** in the **Menu Bar**, then click **Rules** to display the **Rules panel**.
2. Click **Add New Rule**.
3. Click to select the appropriate rule for your requirements. See the **Rules panel** section in Using the Toolbox HTML5 Web-GUI for an explanation of the action each rule can perform.

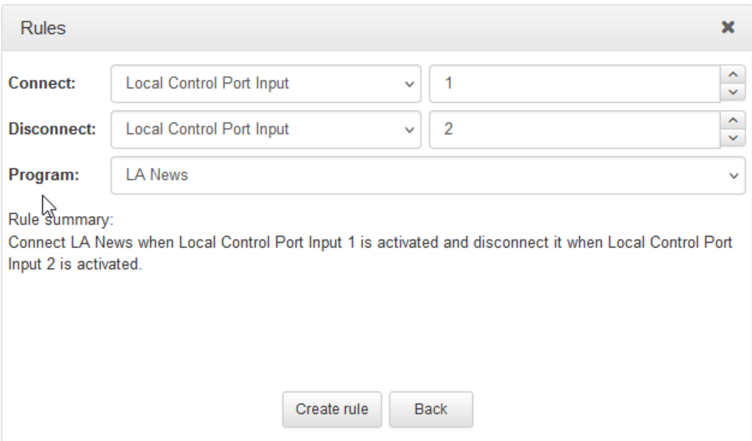


Note: When rules have been configured previously they are displayed when the **Rules panel** is opened.

Rule 1: Use Two Local Inputs to Connect/Disconnect a Program

This rule is used to connect and disconnect a selected program when different codec control port inputs or virtual inputs are activated.

1. Click the rule in the **Rules panel** titled **Use two local inputs to connect/disconnect a program**.
2. Click the drop-down arrows to select the control port input used to connect the selected program, and then select the alternative input used to disconnect the program.
3. Click the drop-down **Program** arrow to select the program to be connected.

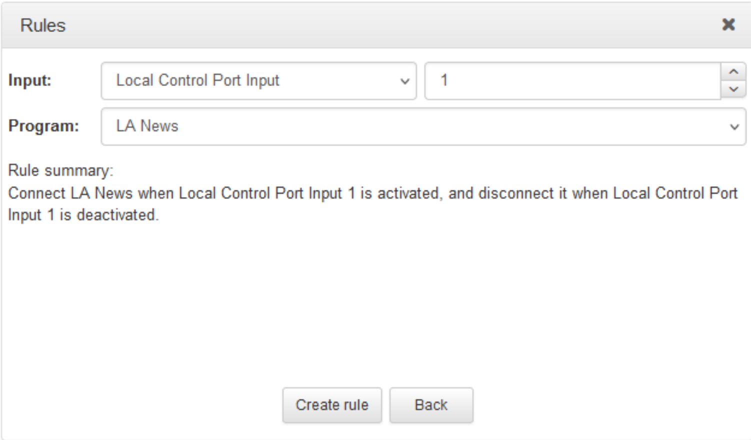


4. Check the **Rule Summary** and click **Create Rule** to save the settings.

Rule 2: Use a Local Input to Connect/Disconnect a Program

This rule is used to connect and disconnect a selected program when a control port or virtual input is toggled.

1. Click the rule in the **Rules panel** titled **Use a local input to connect/disconnect a program**.
2. Click the drop-down arrows to select the control port input or virtual input used to toggle connecting and disconnecting a program.
3. Click the drop-down **Program** arrow to select an individual program which will connect and disconnect when the input is toggled.

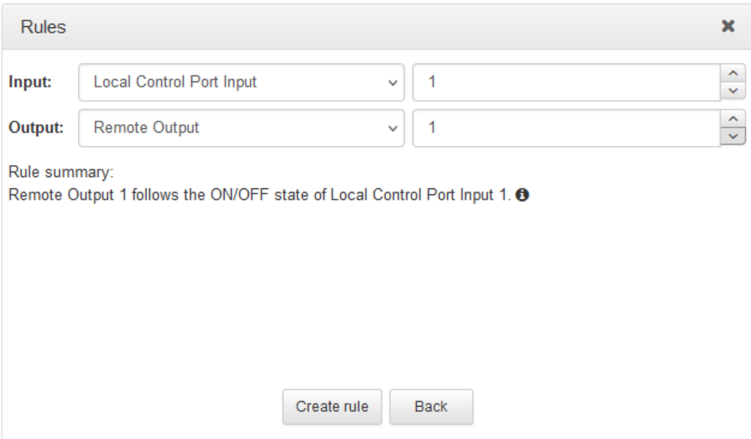


4. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 3: Use a Local Input to Toggle a Remote Output

Use this rule to allow a local codec's control port input or virtual input to change the state of a remote output.

1. Click the rule in the **Rules panel** titled **Use a local input to toggle a remote output**.
2. Click the drop-down arrow to select the local input used to control a remote output.

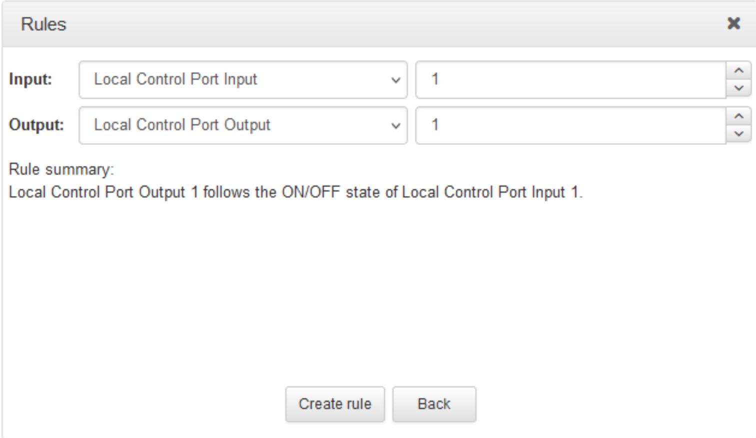


3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 4: Use a Local Input to Toggle a Local Output

Use this rule allow a local control port input or virtual input to change the state of a local relay output.

1. Click the rule in the **Rules panel** titled **Use a local input to toggle a local output**.
2. Click the drop-down arrow to select the local control port input used to control a local control port output.

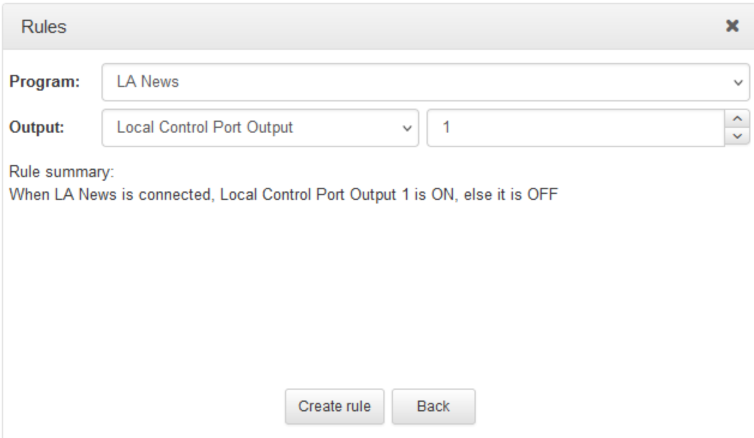


3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 5: Use the Connected State of the Codec to Toggle a Local Output

This rule is used to toggle a codec control port relay output each time a program connects and disconnects.

1. Click the rule in the **Rules panel** titled **Use the connected state of the codec to toggle a local output**.
2. Click the drop-down **Program** arrow to select the program which will affect the relay toggle function, or use the default setting whereby any program connecting will toggle the relay output.
3. Click the drop-down arrow and select the relay output you want to toggle.



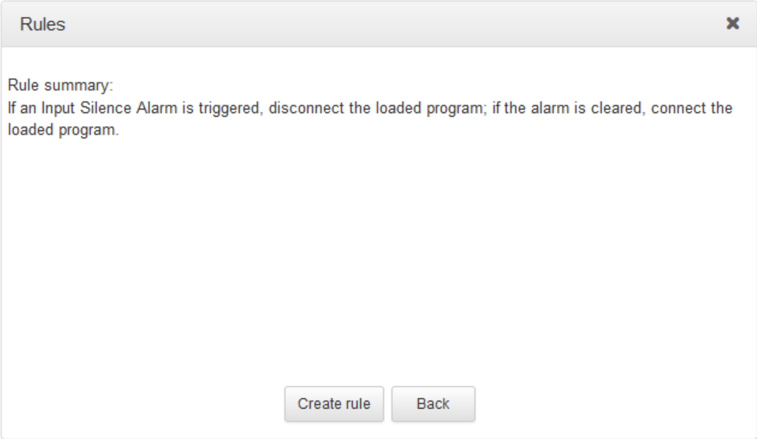
4. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 6: Use an Input Silence Alarm to Disconnect/Connect the Loaded Program

Use this rule to configure the codec to disconnect when input silence is detected and reconnect when input audio is subsequently restored. For this rule to be activated an **Input Silence** alarm must be enabled. This rule is activated according to the silence threshold configured in the **Alarms panel** for an **Input Silence** alarm. To enable the rule:

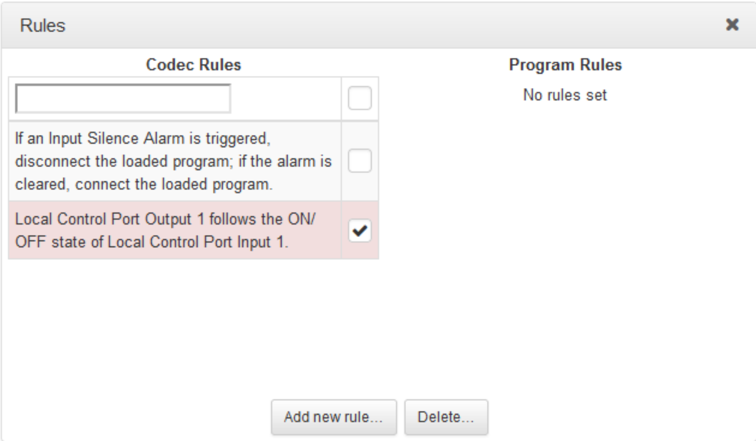
1. Click the rule in the **Rules panel** titled **Use an Input Silence Alarm to disconnect/connect the loaded program**.

2. Click **Create Rule**.



Deleting Rules

1. Open the Toolbox HTML5 Web-GUI and click **Rules** in the **Menu Bar** to display the **Rules panel**.
2. Click to select the rule you want to delete.
3. Click the **Delete** button.

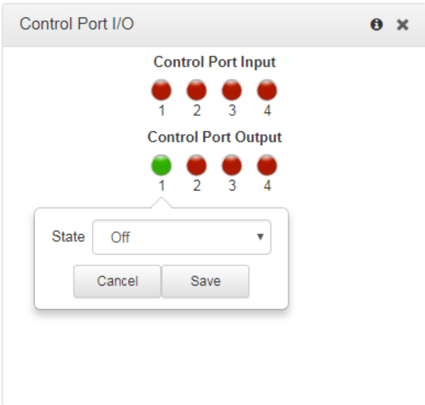


4. Click **Yes** in the confirmation dialog.

20.38 Monitoring Control Port I/O Status

It is possible to monitor the status of the four control port inputs and four opto-isolated outputs available via the DB15 **CONTROL PORT** connector. To monitor status:

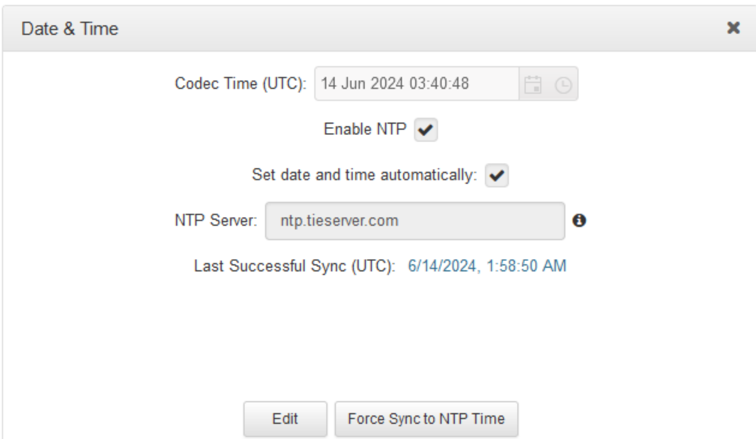
1. Open the Toolbox HTML5 Web-GUI and click **Control** in the **Menu Bar**, then click **Control Port I/O** to display this panel.
2. Click on an output to change the state from **Off** to **On**, then click **Save**. Note: Input states cannot be changed.



20.39 Adjusting Codec Time and Date

Network Time Protocol (NTP) is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency data networks. By default **Use NTP** time is enabled in the codec and it will synchronize with **ntp.tieserver.com**. When the codec is attached to a network it will automatically ping the selected NTP server every two hours and update the time in the codec. It is also possible to manually synchronize the time. Note: The codec will not ping a server while connected. To manually synchronize time settings in the codec:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Date & Time** to display the **Date & Time panel**.
2. Click **Force Sync to NTP Time** to manually synchronize the codec to NTP time.



Important Notes:

- It may take more than one attempt to **Force Sync to NTP Time**.

- When NTP address settings are configured and enabled, the codec will immediately jump to the new time when it synchronizes with the server. This may cause scheduled events to be missed.
- Ensure DNS settings are configured correctly as this is required for NTP.

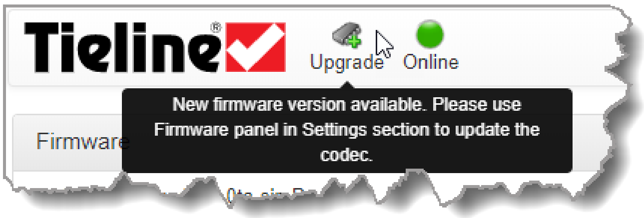
20.40 Upgrading Codec Firmware

To download the latest codec firmware visit www.tieline.com. See Upgrading Firmware via SD to upgrade codec firmware using an SD card with new firmware copied onto it.

New Firmware Notifications

By default the HTML5 Web-GUI integrates with TieServer to automatically update users when a firmware upgrade is available.

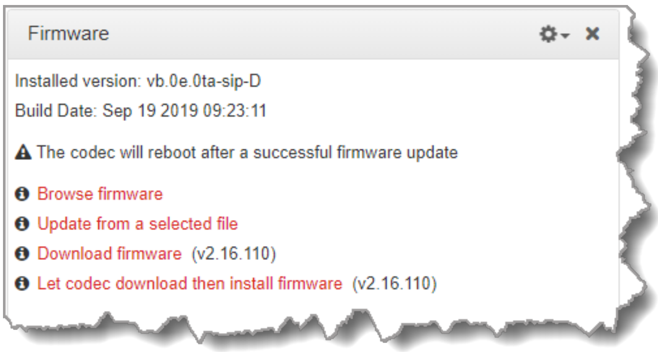
1. Connect the codec to a PC using either a LAN or USB connection and open the HTML5 Toolbox Web-GUI.
2. If new software is available the **Upgrade** symbol appears in the top-left of the screen.



3. Click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel** to perform the firmware upgrade.

Performing a Firmware Upgrade

There are several firmware upgrade options available:

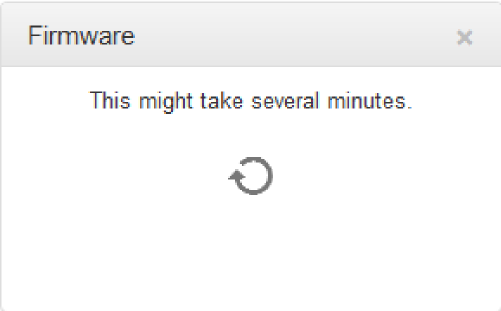


1. **Browse Firmware**: Click to navigate to the Tieline website and download the latest firmware for the codec.
2. **Update from a Selected file**: Click to navigate to a firmware file saved on a computer or network drive. Note: If you click the **Options symbol** ⚙ and select **Show Advanced Options**, it is possible to select **Force update from a selected file** to commence a firmware update in the codec even if it is still connected.
3. **Download firmware**: Click to download a previous reliable firmware version. Note: Only visible when a new release is available.
4. **Let codec download and install firmware**: Click to download a previous reliable firmware version directly into the codec and then complete the update. Note: Only visible when a new release is available.

5. Install from HTTP sources from within private networks: Click to select the **Options symbol** and select **Show Advanced Options** to install official firmware versions when internet access is unavailable.

The following procedure explains how to perform codec firmware upgrades with a downloaded firmware file saved to your PC.

1. Open the Toolbox HTML5 Web-GUI and click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel** options.
2. Click **Browse firmware** to search for the latest firmware and download it to your computer. Note: the **Download firmware** and **Let codec download then install firmware** links are only visible if firmware more recent than the currently installed version is available.
3. Once the firmware has been saved locally, click **Update from a selected file** in the **Firmware panel**.
4. Select the **.bin** file to use in performing the upgrade and click **Open** to start the upgrade. **IMPORTANT**: The codec will reboot automatically after the firmware upgrade. DO NOT remove power or reboot the codec before the update has completed and the codec has rebooted itself.



Refresh Browser Cache After Firmware Upgrade

Tieline recommends clearing your browser cache after the upgrade is complete. The short cuts for this are:

- Google Chrome: shift+Ctrl+delete
- Mozilla Firefox: Ctrl+shift+delete
- Internet Explorer: Ctrl+shift+delete
- Safari: Ctrl+alt+e

21 Front Panel Configuration Tasks

The following sections explain how to configure codec settings using the front panel **OLED** screen and **KEYPAD**.

21.1 Configuring IP via the Front Panel

Checking IP Address and Unit Details in the Codec

1. Press the **Home** button to return to the home screen.
2. Use the navigation buttons on the front panel to select **Settings** and press the button.
3. Select **Unit** and press the button.

4. Select **LAN1** or **LAN2** and press the button.
5. IP address details and other relevant information is listed. Use the up and down arrow buttons to scroll and view all details listed.

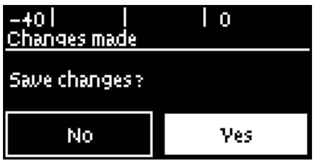
Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

Ethernet and VLAN Configuration Options

Bridge-IT II and Bridge-IT XTRA II codecs have two Gigabit (10/100/1000) RJ-45 Ethernet **LAN** ports for to support IP redundancy. The Bridge-IT XTRA II codec has dual AoIP ports in addition to dual Gigabit LAN ports. LAN2 on the Bridge-IT II codec can optionally be configured as an AoIP port for AoIP streaming (AES67, ST 2110-30, Livewire+, RAVENNA).

Both codecs also support up to four additional VLAN interfaces. VLAN interfaces have features similar to physical Ethernet ports. However, your network administrator will need to configure VLAN support throughout your network for VLANs to be supported in the codec. As an example, if only one physical Ethernet interface is available, VLANs can be used to operate SmartStream PLUS, or to separate codec Control and Streaming functions if required.

Following are a range of Ethernet and VLAN settings which can be configured in the **Network** menu. After completing configuration you are prompted to save any changes. Select **Yes** and press the button to confirm menu adjustments.



Configure LAN2 as an AoIP Port

To configure LAN2 as an AoIP port and allow it to connect to networks supporting AES67, ST 2110-30, ST 2022-7 (Bridge-IT XTRA II only), Livewire+ and RAVENNA:

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN2**.
4. Select AoIP and if it **Disabled** press the button.

5. Confirm the codec is to use LAN2 as an AoIP port.
6. To view the AoIP network address assigned to the codec navigate to **SETTINGS** **> Unit > AoIP**.

Download the AoIP user manual from www.tieline.com for more information about how to configure sources and destinations and AoIP streaming more generally.

Configure an IPv4 DHCP Address

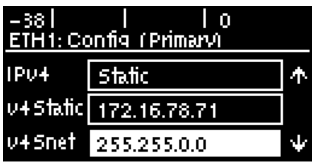
By default the codec is programmed for DHCP-assigned IP addresses. DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider or by a router on your local area network (LAN).

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Config** and then **Usage** and the the appropriate control and/or streaming mode for the connection. Next, press the button.
5. Select **IPv4** and press the button.
6. Select **DHCP** and press the button.
7. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

Configure a Static IPv4 Address

Static IP addresses are fixed addresses which are recommended for studio installations. Using a static IP address ensures remote codecs can connect reliably using the same IP address over time. Note: DNS settings must be specified when a static IP address is configured.

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Config** and then **Usage** and then the appropriate control and/or streaming mode for the connection. Next, press the button.
5. Select **IPv4** and press the button.
6. Select **Static** and press the button.
7. Navigate to **v4 Static** and enter the IP address, then press the button.
8. Navigate to **v4 Snet** and enter the Subnet Mask, then press the button.



9. Navigate to **v4 Gway** and enter the Gateway details, then press the button.

10. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.
11. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

IPv6 Address Assignment

There are three IPv6 settings available for the Ethernet ports and any VLANs configured.

1. Auto: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. Manual: Select to manually enter IPv6 address details.
3. Off: Select to ignore IPv6 address details.

Important Note: Select **Off** if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

To adjust this setting:

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Config** and then **IPv6** and press the button.
5. Select **Auto**, **Manual** or **Off** and press the button.
6. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

By default the codec is configured to allow the codec to automatically receive IPv6 address information from an IPv6 enabled router.

Manual IPv6 Address Assignment

Select **Manual** mode using the previous procedure and enter information into the **v6 Static** (Address)**, v6 Prefix** and IPv6 **v6 Gway** fields in the codec to manually configure address details.

Specifying DNS Server Settings

It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains** section within the web-GUI. This feature can be turned on or off in the LAN codec menu. Note: When installing codecs at the studio it is important to ensure that DNS settings are configured correctly. This is particularly important when using tools like the Cloud Codec Controller and. Correct DNS settings are also required for NTP, DDNS, use of host names, and if the **Download and Install** option is used to upgrade codec firmware.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **Network** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface and press the button.
4. Select **Config** and press the button.
5. Use the down navigation button to scroll to **DNS**.
6. Press the button to toggle between **Auto** and **Manual**.
7. Enter DNS Address and Domain details as required.

8. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

Link Mode Configuration

It is possible to configure the Ethernet link speed (10/100/1000/Auto) and whether each available interface operates in Full-Duplex or Half-Duplex modes.

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface, then press the button.
4. Select **Config** and press the button.
5. Use the down navigation button to scroll to **Link Mode**.
6. Press the button to select a preferred setting. Note: Default setting is **Auto**.
7. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

VLAN ID (VLAN configuration only)

The **VLAN ID** is encapsulated in IP packets to facilitate routing throughout your network.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **Network** and press the button.
3. Use the down navigation button to select a **VLAN** interface.
4. Select **Config** and then **Usage** and press the button.
5. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
6. Use the down navigation button to scroll to **VLAN ID**.
7. Press the button to enter a number between 1-4094 inclusive.
8. Press the button to confirm this setting.
9. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

VLAN Priority (VLAN configuration only)

The **VLAN Priority** setting represents a prioritization scheme for forwarding data packets throughout Virtual Local Area Networks.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **Network** and press the button.
3. Use the down navigation button to select a **VLAN** interface.
4. Select **Config** and then **Usage** and press the button.
5. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
6. Use the down navigation button to scroll to **Priority**.
7. Press the button to enter a number from 0 to 7 inclusive.
8. Press the button to confirm this setting.
9. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

VLAN Interface (VLAN configuration only)

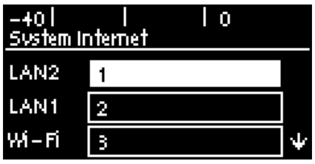
This setting applies the VLAN settings to a physical Ethernet port in the codec. Select either Ethernet port 1 or 2.

21.2 System Internet Setting

In some situations when multiple interfaces are connected to the codec it may be necessary to select the preference for interfaces used to connect to the internet. For example, when using DDNS services. Or perhaps where the default setting is LAN1, however this may be connected to a closed network and LAN2 is the interface connected to a WAN, i.e. the internet. To adjust the default settings:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Network** and press the button.

3. Navigate to **System Internet** and press the button.
4. Navigate to an interface and press the button to adjust the numeric order as required. Then navigate to **Enter** on the screen and press the button. In the following example **LAN 2** has been selected as the main interface to use when connecting to the internet.



5. The new setting is reflected in the adjusted interface order.

Important Notes:

- The **System Internet** setting is only related to internet connectivity. It is not related to the IP Interface Setup for IP streaming interfaces.
- If interface 1 is not connected to the internet, the codec will attempt to use interface 2 and other interfaces thereafter to access the internet.
- If the codec is connecting to the Cloud Codec Controller and has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. See Enabling the Cloud Codec Controller for more details.

21.3 Configuring DDNS

Dynamic DNS (DDNS) can be configured in the codec to allow the use of a host name to connect when a dynamic IP address is used. To facilitate this a DDNS service is used to register a device's IP address to a host name, allowing a remote codec to dial to a host name. The host name will be updated automatically periodically as required, or when the dynamic IP address changes.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **DDNS** and press the button.

3. Select and configure each DDNS setting and enable DDNS. The codec is then able to accept a call using the specified DDNS Host Name.

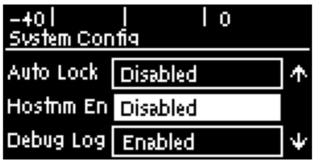
Important Notes:

- Supported DDNS providers are listed in the **DDNS Provider** menu.
- Devices should be DDNS registered to public IP addresses.
- The codec will utilize the **System Internet** interface order when contacting DDNS service providers.
- Does not support remote G3 codecs connecting in this mode.
- DDNS **Host Name** settings are unrelated to the **Hostname** setting accessed via the **Options panel** under **Settings**.
- Codec DNS settings must be specified when configuring DDNS.

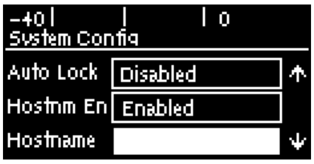
21.4 Configuring a Hostname

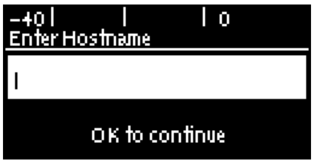
It is possible to assign a hostname to the codec to provide a flexible way of identifying the codec on a network.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.
3. Navigate down to **Hostname Enable** and press the button to enable this feature.



4. Navigate to **Hostname** and press the button to display the **Enter Hostname** screen and enter the hostname. Next, navigate to **Enter** and press the button to save the settings.





Important Notes:
- Modifying hostname settings requires a codec restart before they take effect
- In the **Hostname** only enter the characters a-z, A-Z, 0-9 and - and the first or last character cannot be a hyphen/dash.

21.5 Configure Default Interfaces

When dialing over IP you can select the preferred interface to use when establishing a connection. By default **Any** is selected, which means the first available interface will be used to dial a connection. The default interfaces in order of use when available are:

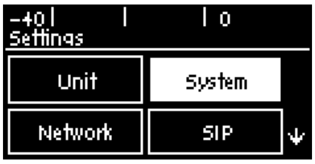
1. **LAN1** Ethernet port (default **Primary** interface)
2. **LAN2** Ethernet port (default **Secondary** interface)
3. **USB** (default **Tertiary** interface)
4. **VLAN1** (default **Quaternary** interface)

Important Notes:
- If an interface is not available it is not listed in the interface selection screen. E.g. **Fuse-IP**.
- VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.

Reconfigure Default Primary, Secondary, Tertiary and Quaternary Interfaces

It is possible to reconfigure the default **Primary** (LAN1), **Secondary** (LAN2), **Tertiary** (USB) and **Quaternary** (VLAN1) IP interfaces in the codec. As an example, you may want to select **Primary** as the dialing interface in a program and then copy this program onto multiple codecs. However, the actual primary interface used at each location can vary for each codec. For one codec it may be an Ethernet port and for another it may be a Wi-Fi interface. This allows you to configure site-specific settings to suit available network interfaces at different remote locations.

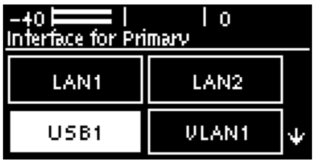
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.



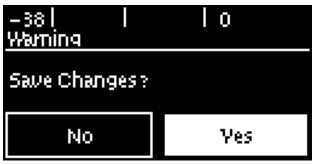
3. Navigate to **Interface Set** and press the button.

4. Navigate to the **Primary**, **Secondary**, **Tertiary** or **Quaternary** interface setting and press the button.

5. Select an alternative default interface and press the button.



6. After making adjustments, press the **Return button** to navigate out of the menu, then navigate to **Yes** in the **Warning** dialog and press the button to save all changes.



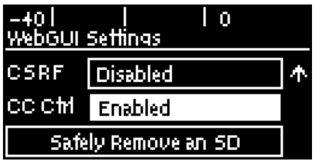
Important Note: Fuse-IP cannot be configured as a default Primary, Secondary, Tertiary, or Quaternary interface.

21.6 Enabling the Cloud Codec Controller

To allow the codec to be configured and managed by Tieline's Cloud Codec Controller over the public internet it needs to be enabled for CCC management:

1. Press the **SETTINGS**  button.
2. Use the navigation buttons to select **Web-GUI** and press the  button.

3. Navigate down to **CC Controller** and press the  button to toggle between **Enabled** and **Disabled**.



Important Notes:

- Ensure **CSRF** is disabled in the codec or it will not be able to connect to the CCC. This setting is **[OFF]** by default and is also available in the codec menu via **Settings > WebGUI**, and in the **Options** panel in Toolbox.
- Locally Defined Codecs over a private network do not need to be enabled for CCC operation. Only codecs that require internet access need to be enabled.
- The CCC needs to continually send and receive data between codecs to update information displayed. If the CCC is left open on a computer and is not used for more than 4 hours, the Codec Viewer is placed in 'sleep' mode to save on data use.

DNS and the Cloud Codec Controller (CCC)

If a codec has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. When using the Toolbox HTML5 Web-GUI to configure DNS network settings:

1. Select **Transport** and then **Network** to open the **Network panel**.
2. Select an interface and then select the **DNS** tab to enter details into the **DNS Addresses** fields as required.
3. Ensure that the **Specify DNS Settings** check-box is selected. Note: This will ensure the DNS servers are used.

Note: These settings can also be configured using the front panel of the codec.

Where can I find my DNS settings?

DNS settings for static public IP addresses can be sourced from your internet provider/Telco. Sometimes the Gateway setting in the TCP/IP tab in the Network panel will work as the DNS Address to enter. Another common scenario encountered is that static DNS settings are configured to override DHCP settings when they are not compatible with the network being used. When using DHCP IP settings ensure the **Specify DNS Settings** check-box is disabled in the **Network panel**.

21.7 Configuring Fuse-IP Bonding

Fuse-IP is a proprietary Tieline IP bonding technology which aggregates data by bonding multiple IP interfaces (peers) and establishing a “tunnel” between two Tieline codecs. A streaming connection can be established after the tunnel is created. Fuse-IP automatically distributes data over any two bonded interfaces, e.g. dual Ethernet LAN ports. There are several benefits in using Fuse-IP to aggregate data from multiple IP interfaces, including:

- The ability to create more stable connections with higher overall data bandwidth.
- Greater choice of encoding algorithms because of higher available bandwidth.
- Redundancy in case one IP connection is lost.

How does Fuse-IP work?

Fuse-IP is another interface you can use to dial, similar to selecting a LAN interface. Fuse-IP requires one codec to be a server and the other codec is the client. Normally the remote codec is configured as the client and the studio codec is the server, because it's generally easier to dial static IP addresses configured at the studio than interfaces at the remote site. Like SmartStream PLUS redundant streaming, you can use two IP interfaces at the studio for additional redundancy.

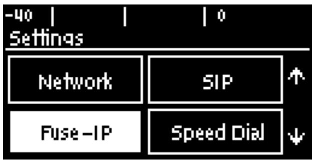
Prerequisites

Before configuring Fuse-IP you need to know:

- The IP address (or addresses) for the codec acting as the server at the studio.
- The serial number of the server codec to which you are connecting using Fuse-IP.

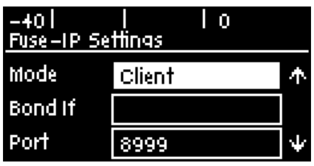
Configuring a Fuse-IP Server at the Studio

1. Press the **SETTINGS** button.
2. Use the down navigation button to select **Fuse-IP** and press the button.



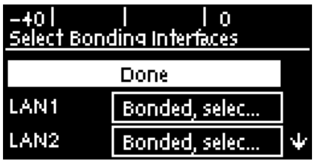
Important Note: Ensure Fuse-IP is disabled prior to configuration. Navigate to **Settings > Fuse-IP > Status** and press the button to disable Fuse-IP if it is enabled.

3. Navigate down to **Mode** and press the button to select **Server** if the codec is at the studio and not initiating the connection. Note: the server codec serial number is displayed and needs to be entered into the Fuse-IP client codec.

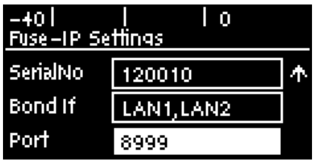


4. Navigate to **Bonded If** (Bonded Interfaces) and press the button, then navigate to each interface in turn and press the button to select or deselect interfaces.

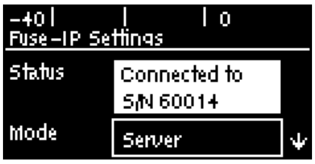
5. Confirm all changes by selecting **Done** and then press the OK button before exiting the menu.



6. Leave the default Fuse-IP **Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.



5. The codec is now configured to connect with a client codec over a **Fuse-IP** tunnel. Navigate up to **Status** and then press the OK button to toggle the selection to **Enabled** and create a Fuse-IP tunnel between the server and client codecs. **Status** on the **OLED SCREEN** displays as **Connected** when Fuse-IP is configured successfully.

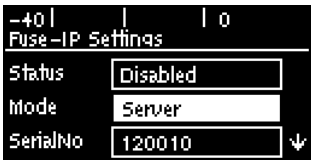


Configuring a Fuse-IP Remote Client

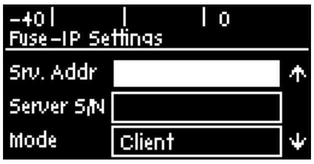
1. Press the **SETTINGS** button.
2. Select **Fuse-IP** and press the OK button.

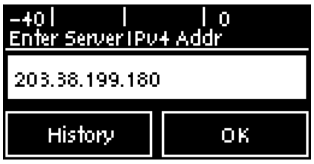
Important Note: Ensure Fuse-IP is disabled prior to configuration. Navigate to **Fuse-IP Status** and press the OK button to disable Fuse-IP if it is enabled.

3. Navigate to Fuse-IP **Mode** and press the OK button to select **Client** if it is a remote codec.

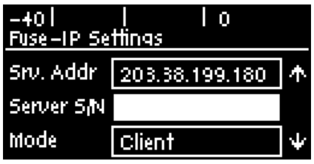


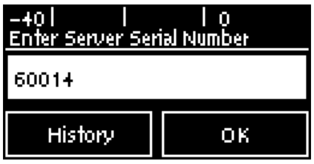
4. Navigate to **Srv. Addr** (server address) and press the OK button to enter a public static IP address associated with the bonded interfaces at the studio, then press the OK button. Note: if the bonded interfaces have private addresses behind a firewall then port forwarding needs to be configured. See Installing the Codec at the Studio for more details on port forwarding.



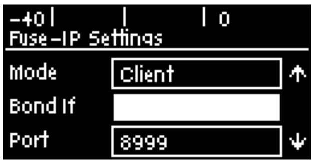


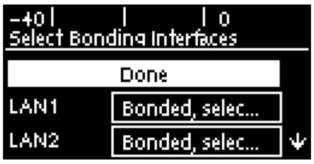
5. Navigate to **Server S/N** and press the button to enter the serial number of the server codec to which you are connecting (or select a previously entered serial number via History), then press the button.





6. Navigate to **Bonded Interfaces** and press the button, then navigate to each interface in turn and press the button to select or deselect interfaces. Confirm the changes when exiting the menu.

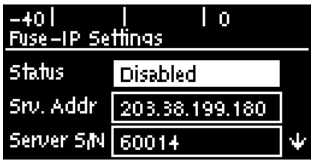


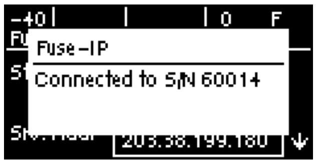


6. Leave the default Fuse-IP **Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.

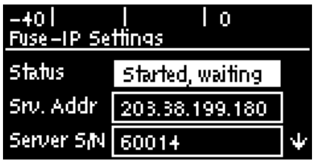
7. Navigate to inactivity **Timeout** and press the button if you want to adjust the predetermined time period for turning the Fuse-IP tunnel off. Adjust the setting and press the button to store the new setting. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter **0** to disable the timeout.

8. Navigate up to **Status** and then press the button to toggle the setting and select **Enabled** and create a Fuse-IP tunnel between the server and client codecs. Remember Fuse-IP must be enabled on both codecs. A dialog on the screen confirms when two codecs have connected and created a tunnel.





Please note: double-check all settings on both the server and client codecs if the message **Started, waiting** persists after enabling Fuse-IP.



9. Select **Fuse-IP** as the interface with which to connect when creating a program using the front panel codec menus, or the HTML5 Toolbox Web-GUI **Program Manager panel**.

Important Notes:

- Data is sent by the codec over the newly created 'tunnel' as soon as Fuse-IP is enabled, even if a connection has not been configured and dialed. Depending on the number of interfaces being used, codecs may transmit and receive up to 24MB of data per hour at each end of the link.
- The codec remembers the Fuse-IP enabled/disabled state on power up.
- For additional stability it is recommended that a fixed jitter buffer is configured when streaming using Fuse-IP. The actual jitter buffer depth should account for the difference in delay between the interfaces and the maximum jitter experienced. To determine the jitter over each link you can connect and stream audio over each interface separately and look at the jitter reading displayed on the **Connection Statistics** screen.
- Use a dotted quad IPv4 address when configuring the Fuse-IP Server Address.
- For more details on setting up Fuse-IP using the codec HTML5 Web-GUI see Configure Fuse-IP Bonding.

21.8 Selecting an Algorithm

The codec offers PCM uncompressed linear audio as well as Opus, aptX® Enhanced, LC-AAC, HE-AAC v.1 and HE-AAC v.2, AAC-LD, AAC-ELD, AAC-ELDv2, MPEG Layer 2, G.711 and G.722, Tieline Music and MusicPLUS algorithms. Pre-configured programs simplify codec configuration.

Overview of Tieline Algorithms

1. The Tieline Music algorithm is optimized for audio bit rates as low as 19.2kbps with only a 20 millisecond encode delay. It offers 15 kHz mono from 24Kbps to 48Kbps.
2. Tieline MusicPLUS delivers up to 20 kHz mono from 48kbps upwards. It can also deliver up to 20 kHz stereo from 96kbps upwards, offering huge savings on your IP data bills and outstanding audio quality.

Overview of AAC Algorithms

AAC-LC

LC-AAC is optimised for audio bit-rates of 64Kbps per channel or higher using a sample rate of 48kHz. Tieline recommends using LC-AAC instead of HE-AAC if bandwidth of 64Kbps or higher per channel is available, to optimise audio quality. If lower than 64Kbps is available, consider using HE-AAC, Tieline Music or Tieline MusicPLUS.

AAC-HE

Codecs include both HE-AAC v.1 and HE-AAC v.2, which are optimized for low bit rate connections. Selection of HE-AAC v.1 and v.2 is automatically managed within the codec, so only **AAC-HE** is displayed on the screen. When used for mono connections, HE-AAC v.1 performs best at bit rates of 24kbps per channel or higher. HE-AAC v.1 is also used for stereo connections when audio connection bandwidth is 48kbps or higher.

HE-AAC v.2 is used for stereo connections when audio connection bandwidth is below 48kbps and is capable of delivering 15kHz quality stereo audio at audio bit rates as low as 24kbps.

A sample rate of 32kHz is used in the codec's default profiles to achieve ultra-low bit-rate connections, but this is adjustable to 44.1kHz or 48kHz if required.

AAC-LD

AAC-LD (Low Delay AAC), AAC-ELD (Enhanced Low Delay AAC) and AAC-ELDv 2 are optimized for low latency real-time communication. AAC-LD is suited to bit rates of 96kbps or higher for stereo audio.

AAC-ELD

AAC-ELD is optimised for high quality stereo connections from 48 - 96kbps and performs better at these bit rates when compared with AAC-LD.

AAC-ELD v 2

For stereo connections below 48kbps AAC-ELD v2 will deliver better performance than AAC-ELD down to 24kbps.

Overview of aptX® Enhanced Audio Coding

aptX® Enhanced audio coding is used by thousands of radio stations to deliver very low delay audio for IP broadcasts and is ideal for high quality studio-to-transmitter links and audio distribution. It delivers outstanding audio quality with exceptionally low delay across a range of IP networks.

32kHz or 48kHz sample rates are available at either 16 bit or 24 bits per sample. aptX Enhanced has a minimum connection bit rate of 128kbps per channel and offers 10Hz to 24kHz frequency response. 24 bit, 48kHz aptX Enhanced at the maximum bit rate of 576kbps delivers >120dB of dynamic range.

Overview of Opus Algorithm

Opus is a highly versatile open source audio coding algorithm. It incorporates technology from the well-known SILK and CELT codecs to create a low latency speech and audio codec. It is a variable bit rate algorithm ideal for live broadcast situations because of its capacity to deliver high quality, real-time Audio over IP (AoIP) at low bit rates. Visit http://www.opus-codec.org for more info. There are three Opus encoding configurations available:

Algorithm	Recommended connection for on-air use
Opus Voice	High quality low bit rate remotes (9.6kbps -64kbps)
Opus Mono	Very high quality mono remotes, STLs and audio distribution (48kbps -128kbps)
Opus Stereo	Very high quality stereo remotes, STLs and audio distribution (64kbps -256kbps)

Configuring an Algorithm in the Codec

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Cxns** and press the button.
3. Select a connection and press the button.

4. Navigate down to **Algor'm** and press .

5. Select the mono or stereo algorithm that you want to connect with, then the bit-rate, and press .

Stream Encoding Limits

G5/G6 products:

- High frame-rate algorithms are limited to one SmartStream PLUS redundant stream (e.g. PCM, aptX Enhanced, G.711 and G.722).
- Music PLUS encoding limited to two SmartStream PLUS redundant streams
- All other algorithms support three SmartStream PLUS redundant streams (Note: only 2 SmartStream PLUS streams supported in Bridge-IT II and Bridge-IT XTRA II)

How do I choose the right algorithm?

The algorithm you select will not only affect the quality of the broadcast but it will also contribute to the amount of latency or delay introduced. For example, if MP2 algorithms are used, program delays will be much longer than when using Tieline Music or MusicPLUS algorithms. This is due to the additional inherent encoding delays involved when using MP2 algorithms. This can be a major consideration for live applications that integrate remote-crosses into a broadcast. The algorithm you choose to connect with will also depend upon:

- The codecs you are connecting to (Tieline versus non-Tieline)
- Whether you are creating multi-unicast connections
- Whether you are connecting using SIP
- The uplink bandwidth capability of your broadband connection

Important Notes: Music and MusicPLUS algorithms cannot be used over SIP connections. Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP, or use G.711 and G.722 if required. Tieline G3 codecs do not support connections using AAC, aptX Enhanced and Opus algorithms and will default to MPEG Layer 2 if an incoming connection is configured to use these algorithms.

It can be a good idea to listen to the quality of your program signal using each algorithm and to see how it sounds when it is sent at different connection bit-rates (as well as different FEC and jitter-buffer millisecond settings). This will assist you to determine what the best algorithm setting is for the connection you are setting up. Please see the following table for details on the connection requirements of the different algorithms available.

Algorithm	Audio Band-width	Algor-ithmic Delay	IP bit rate per channel	IP over-head per connection	Audio Quality and Features	Recommended applications for on-air use
PCM/Linear (Uncom-pressed)	16/24 bit up to 24kHz	0ms	sample rate x bits per sample x no. channels	80kbps	• Full bandwidth, perfect audio quality for voice and music • No error concealment/correction or artefacts	• Extremely high quality PCM linear uncompressed audio for STLs and audio distribution. • Ideal for fiber or high bandwidth links.
Tieline Music	Up to 15kHz	20ms	24 kbps minimum	16kbps	• High quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth (e.g. 3G wireless) • Suitable when bidirectional communication between announcers is required
Tieline Music-PLUS	Up to 22kHz	20ms	48 kbps minimum (Optimised for 64kbps per audio channel)	16kbps	• Very high quality voice and music • Very low delay at low to moderate bit-rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel • Suitable when bidirectional communication between announcers is required
G.711	3kHz	1ms	64kbps minimum	80kbps	• Low quality 3kHz POTS phone quality audio • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Low quality and used generally for compatibility
G.722	7kHz	1ms	64kbps minimum	80kbps	• Good quality 7kHz voice • Better quality than a standard POTS phone call • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Good voice quality audio for remotes and other voice quality applications
MPEG Layer 2	Up to 22kHz	24 to 36ms	64kbps minimum	8.5 - 13.3kbps	• Very high quality voice and music • Low to moderate delay at moderate to high bit rates	• Highly compatible with other brands of audio codec • Very high quality audio for remotes, STLs and audio distribution
MPEG Layer 3	Up to 15kHz	100ms	64kbps	8.5 - 13.3kbps	• High quality voice and music • Moderate bit rates • High delay	• High quality remotes, STLs and audio distribution • Use when bidirectional communication

						between announcers is not required
LC-AAC	Up to 15kHz	64ms	64kbps	15kbps	• High quality voice and music at lowest bit rate; better quality at higher bit rates • Moderate delay at moderate to high bit rates	• Voice or music remotes as well as STLs and audio distribution where some delay is tolerable • Tieline Music or MusicPLUS deliver lower delay
HE-AAC v.1	Up to 15kHz	128ms	48kbps	7.4kbps	• High quality voice and music at the lowest bit rate; better quality at higher bit rates • Low to Moderate bit rates • High delay	• Live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth • Use when bidirectional communication between announcers is not required
HE-AAC v.2	Up to 15kHz	128ms	Minimum 16kbps (Mono); 24kbps (stereo)	7.4kbps	• High quality voice and music • Low bit rates • High delay	• Used for DAB+ radio streaming • Ideal for low bit rate remotes • Use when bidirectional communication between announcers is not required
AAC-LD	Up to 20kHz	20ms at 48kHz	48kbps minimum	30kbps	• Very high quality voice and music • Very low delay at low to moderate bit rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel requiring • Suitable when bidirectional communication between announcers is required
AAC-ELD	Up to 20kHz	15-30ms	24 kbps minimum	15-30kbps	• Very high quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes • Suitable when bidirectional communication between announcers is required
aptX Enhanced	10Hz-24kHz	2.5ms at 48kHz	128kbps minimum (16bit; 32kHz) to 288kbps (24bit;48kHz)	80kbps	• Very high quality voice and music • Extremely low delay at high bit rates • Highly cascade resilient	• Ideal for STLs and audio distribution where high connection bandwidth is available and very low delay is highly desirable. • Resilient with multiple encodes/decodes when required
Opus	4Hz-20kHz	20ms	9.6-256kbps	16kbps	• Very high quality voice and music • Very low delay at low bit rates	• "Opus Voice" is ideal for high quality, and low delay voice quality remotes at extremely low bit rates. • "Opus Mono" and "Opus Stereo" are perfect for high fidelity remotes, STLs and audio distribution at higher bit rates

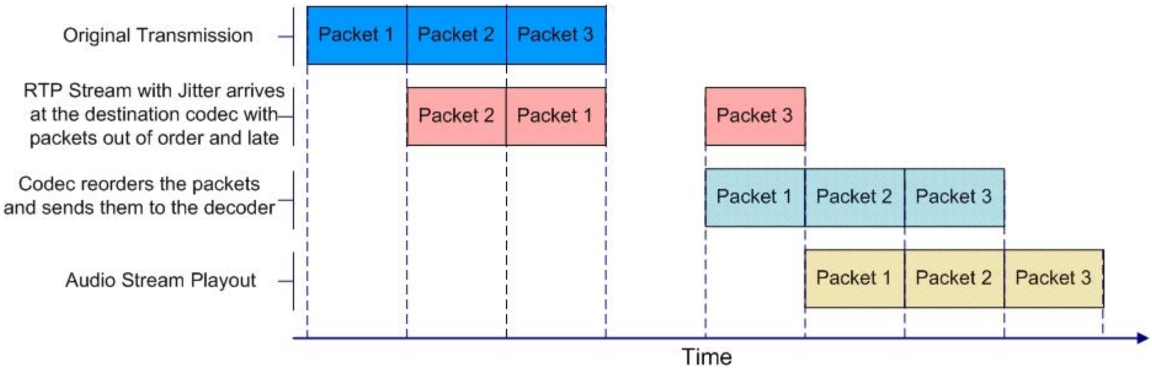
Algorithm Selection Guide

Algorithm	Very Low Delay	Moderate to High Delay	Excellent Performance at Low Bit rates	Preferred for Live Remotes	Preferred for STLs and Audio Distribution	Highly Compatible with other Codecs
Linear/PCM	✓				✓	✓
Opus	✓		✓	✓	✓	✓
Tieline Music	✓		✓	✓		
Tieline MusicPLUS	✓		✓	✓	✓	
aptX Enhanced	✓				✓	✓
LC-AAC		✓			✓	✓
HE-AACv1		✓			✓	
HE-AACv2		✓	✓	✓ *		
AAC-LD	✓			✓	✓	
AAC-ELD	✓		✓	✓		
MPEG Layer 2	✓				✓	✓
MPEG Layer 3		✓				✓
G.722	✓					✓
G.711	✓					✓

***** Use with caution for remotes due to high delay; not suitable when bidirectional communications is required.

21.9 Configuring the Jitter Buffer

Jitter, (also known as latency or delay), is the amount of time it takes for a packet of data to get from one point to another. A jitter buffer is a temporary storage buffer used to capture incoming data packets. It is used in packet-based networks to ensure the continuity of audio streams by smoothing out packet arrival times during periods of network congestion. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, i.e. LAN versus wireless networks. The concept of jitter buffering is displayed visually in the following image.



Jitter buffer management is encompassed within Tieline's SmartStream IP technology which can:

- Remove duplicate packets.
- Re-order packets if they arrive out-of-order.
- Repair the stream in the event of packet loss (error concealment).
- Manage delay dynamically based on current network congestion.
- Manage forward error correction (FEC).

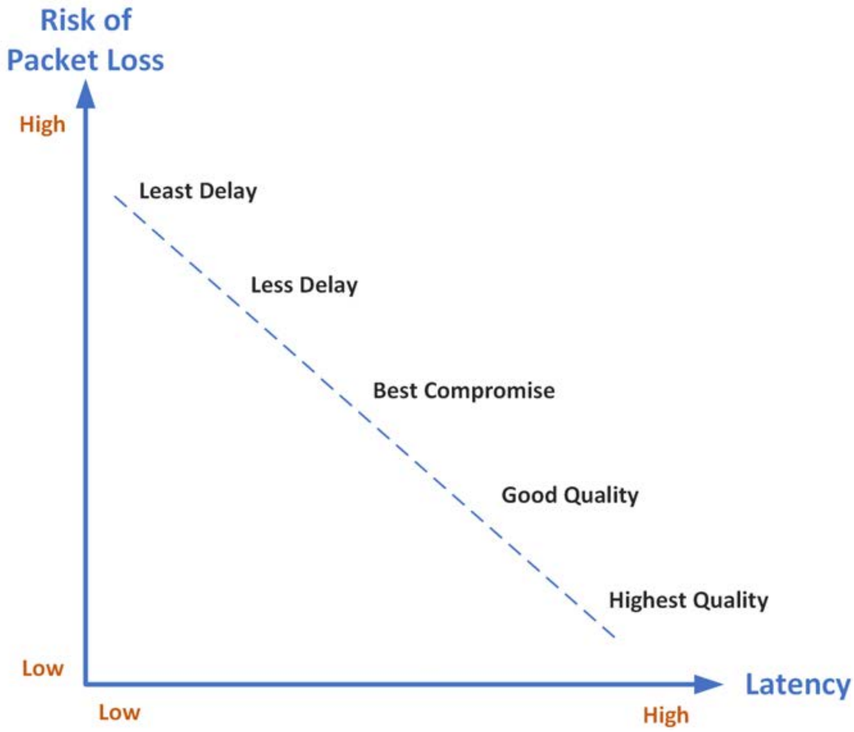
Tieline codecs can be used to program either a fixed or automatic jitter buffer and the setting you use depends on the IP network you are connecting over. Over LANs, WANs and wireless networks the automatic jitter buffer generally works well. It adapts automatically to the prevailing IP network conditions to provide continuity of audio streaming and minimizes delay.

A fixed jitter buffer is preferable over satellite connections to ensure continuity of signals.

⚠ **CAUTION:** If a Tieline codec connects to a device that is using non-compliant RTP streams then the last fixed setting entered into the codec will be enabled (default is 500ms). Non-compliant devices include some other brands of codec, web streams and other devices.

Tieline 'Auto Jitter Buffer' Settings

The following automatic jitter buffer settings range from the most aggressive "Least Delay" setting, which endeavors to minimize delay as much as possible while adapting to prevailing network conditions, to the "Highest Quality" setting, which is the most conservative delay setting to minimize packet loss when packet latency is not as critical. Best compromise is the most popular setting used by most broadcasters as it achieves an excellent balance that minimizes latency for bidirectional communications, as well as minimizing packet loss over most wired and wireless IP networks.



Jitter Buffer Settings and Relationship of Latency and Packet Loss

Least Delay: This setting attempts to reduce the jitter buffer to the lowest possible point, while still trying to capture the majority of data packets and keep audio quality at a high level. This setting is the most aggressive in adapting to prevailing conditions, so the jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant, or occurs in bursts. (E.g. cellular/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

Less Delay: This setting lies between "Best Compromise" and "Least Delay". It may assist in reducing latency over a connection without incurring packet loss.

Best Compromise: This default setting is the midpoint between the jitter buffer settings applicable for "Highest Quality" and "Least Delay." It is designed to provide the safest level of good audio quality without introducing too much latency. In most situations it will deliver very high quality and low delay to support live bidirectional communications over cellular and wireless networks.

Good Quality: This setting lies between "Best Compromise" and "Highest Quality." It may assist in achieving higher quality connections without incurring extreme delays in transmission or significant packet loss.

Highest Quality: This setting is the most conservative in terms of adapting to prevailing network conditions to reduce delay. The jitter-buffer will remain higher for a longer period after a jitter spike is detected – just in case there are more spikes to follow. This setting is best to use where audio quality is the most important factor and delay is not as critical. Unless delay is irrelevant, this setting is not recommended over peaky jitter networks (e.g. cellular networks) and is best used on more stable networks where fluctuating jitter bursts are not common.

Jitter Depth

The jitter **Depth** setting allows you to select predetermined minimum and maximum jitter settings within the auto jitter buffer's minimum and maximum jitter limitations. The default setting of **60 to 1000ms** is a good starting point for most networks. It may be necessary to increase the maximum auto jitter latency setting for networks experiencing higher packet latency, or the minimum depth depending on the reliability of the network.

Which Algorithms can use Automatic Jitter Buffering?

The following table provides an overview of which algorithms are capable of using the automatic jitter buffer feature over SIP and non-SIP connections.

Algorithm	Tieline Session Data Connections	SIP Connections
PCM Linear (Uncompressed)	✗	✗
Tieline Music	✓	✗
Tieline MusicPLUS	✓	✗
G.711	✗	✓
G.722	✗	✓
MPEG Layer 2	✓	✓
LC-AAC	✓	✓
HE-AAC v.1	✓	✓
HE-AAC v.2	✓	✓
AAC-LD	✗	✗
AAC-ELD	✗	✗
Opus	✓	✓
aptX Enhanced	✗	✗

Configuring Automatic Jitter Buffering (Default Setting)

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Cxns** and press the button.
3. Select a connection and press the button.

4. Navigate down to **Jitter** and press .

5. Select **Auto Adapt** and press .
6. Select your preferred jitter buffer setting and press .

How to get the Best Jitter Buffer Results

When configuring automatic jitter buffer settings, establish the IP connection for a while before ‘going live’, to let the codec evaluate the prevailing network conditions. The initial jitter buffer setting when a codec connects is 500ms and it is kept at this level for the first minute of connection (as long as observed delay values are lower than this point).

After the initial connection period the jitter buffer is adapted to suit the current network conditions and is usually reduced. Establish a connection for at least 5 minutes prior to broadcasting, so that the codec has been provided with enough jitter history to ensure a reliable connection.

There are five states or stages that jitter buffer may display and these can be observed in the connection status screen by selecting **HOME** **> Cxns > [Select a connection]** while connected. Use the navigation buttons to scroll through connection statistics displayed. The first four stages are observed in “auto” jitter buffer mode.

1. **Stabilization period (a1):** A few seconds at the start of a connection where no action is taken at all while the establishment of a stable connection means analysis of jitter data is not valid.
2. **Stage 2 (a2):** A compatibility check to ensure the RTP connection is compliant and RTP clocks are synchronized enough to perform jitter analysis.
3. **Stage 3 (a3):** If the compatibility check is successful, this is the analysis hold-off period. During a minute, the jitter buffer is held at a safe, fixed value of 500ms while enough history is recorded to start jitter buffer adaptation.
4. **Stage 4 “live” (A)**: This is where the codec determines it is safe enough to start broadcasting using the auto-jitter buffer level. We recommend running the codec for a few more minutes to obtain a more comprehensive history of the connection’s characteristics.
5. **Fixed (F):** This state is displayed if the jitter buffer is fixed.

Important Note: The jitter buffer setting in the codec can only be adjusted when a connection is off-line. Automatic jitter buffering is disabled for a PCM (linear uncompressed) audio connection.

Fixing Jitter Buffer Settings

The default jitter-buffer setting in Tieline codecs is 500 milliseconds. This is a very reliable setting that will work for just about all connections. However, this is quite a long delay and we recommend that when you set up an IP connection you test how low you can set the jitter-buffer in your codec.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Cxns** and press the button.
3. Select a connection and press the button.

4. Navigate down to **Jitter** and press .

5. Select **Fixed Buffer** and press the OK button.
6. Use the numeric **KEYPAD** to enter the fixed buffer value in milliseconds and press the OK button. Note: Recommended maximum fixed jitter limits are as follows:
   - 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
   - 2,500ms for AAC ELD, AAC LD.
   - 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS.

Configure the Jitter Buffer on the Answering Codec

Create an answering program to independently configure the jitter buffer settings on an answering codec. This will ensure specific fixed or auto jitter settings can be configured to suit the IP network to which the codec is connected. To do this:

1. Create a new answering program on the answering codec.
2. Configure preferred jitter buffer settings in this answering program.
3. Lock the answering program in the codec.

Please note that with the implementation of EBU N/ACIP 3368 SIP configuration the dialing codec can configure the jitter setting on the answering codec. This will override the jitter buffer settings in a locked and loaded answering program in a Tieline codec.

If you change the jitter buffer setting in a codec it will only adjust to the new level when link quality is high (e.g. above 70%). This is done to ensure audio quality is not compromised. When manually configuring the jitter-buffer delay in a codec it is necessary to think carefully about the type of connection you will be using. Following is a table displaying rule of thumb settings for configuring jitter-buffer delays into your codec.

Connection	Jitter-Buffer Recommendation
Private LAN	60 milliseconds
Local	100 - 200 milliseconds
National	100 - 300 milliseconds
International	100 – 400 milliseconds
Wireless Network	250 - 750 milliseconds
Satellite IP	500 - 999 milliseconds

Important Note: The preceding table assumes Tieline Music is the algorithm in use. Do not use PCM (uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit-rates required.

Relationship between the Jitter Buffer and Forward Error Correction (FEC)

If forward error correction is configured then additional data packets are sent over a connection to replace any data packets lost. There is no need to modify jitter buffer settings if you are sending FEC data, only if you are receiving FEC data. The jitter buffer depth on the receive codec needs to be increased if forward error correction is employed. We recommend you add 100ms to the jitter buffer on a codec receiving FEC at a setting of 20% and 20ms at a setting of 100%. Tieline's auto jitter buffer detects the amount of FEC that is being used and automatically compensates to increase the codec jitter buffer if FEC is being used.

21.10 Configuring Forward Error Correction

There are two modes of Forward Error Correction (FEC) available in the codec:

1. In-band FEC: Transmits a secondary stream of audio data packets over a single in-band connection. This is relevant for Tieline Music and MusicPLUS encoding as well as Opus. Note: Opus encoding bit-rates are variable and therefore FEC percentages will vary slightly compared to those outlined in the following table.
2. RFC 2733 compliant FEC (Sessionless connections only): Transmits audio packets over a separate connection.

FEC is designed to increase the stability of UDP/IP connections in the event that data packets are lost. FEC works by sending a secondary stream of audio packets over a connection so that if your primary audio stream packets are lost or corrupted, then packets from the secondary stream can be substituted to replace them. The amount of FEC required depends on the number of data packets lost over the IP connection.

Both the local and remote codec FEC settings can be configured in the codec before dialing. FEC should only be used if link quality displayed on the codec is below **S:99 R:99**, as it is of no benefit otherwise. Tieline and RFC2733 compliant FEC settings are outlined in the following table with their bit rate ratios.

FEC Setting	Bit rate Ratios	Connection Use	In-band FEC	RFC 2733 FEC
100% (Lowest delay)	A simultaneous dual-redundant stream (1:1 ratio) is sent from the codec. Twice the connection bit rate is required to operate the codec using the 100% setting. E.g. if your connection is 14,400Kbps, you will require an additional 14,400 Kbps of bandwidth to allow for the FEC data stream.	Recommended to be used over wireless and international connections.	Yes	Yes
50%	Additional data is sent by FEC in a ratio of 2:1.	Recommended for international & national connections	Yes	Yes
33%	Additional data is sent by FEC in a ratio of 3:1.	Recommended for national and local connections.	Yes	Yes
25%	Additional data is sent by FEC in a ratio of 4:1.	Recommended for national and local connections.	No	Yes
20%	Additional data is sent by FEC in a ratio of 5:1.	Recommended for local and LAN connections.	Yes	Yes
10%(Highest delay)	Additional data is sent by FEC in a ratio of 10:1.	Recommended for local and LAN connections.	No	Yes
Off	FEC is off in the codec and the connection bandwidth is equal	Recommended for wired LAN connections &	Yes	Yes

	to the connection bit rate setting in the codec.	managed T1 & E1 connections for STLs with connections that aren't shared & have quality of service (QoS).		

Important Notes:

- The **FEC Delay** configured should take into account the packet arrival (jitter buffer) strategy at the remote codec. For example, if the maximum jitter buffer at the remote codec is 1000 ms, the FEC Delay setting should be lower, to ensure there is enough time for FEC packets to arrive and replace lost packets prior to audio playout.
- By default, the codec will use the audio stream IP address as the remote FEC IP address as well. This can be adjusted in the **Program Manager panel** in the HTML5 Toolbox web-GUI.
- The default local and remote UDP audio FEC ports are 9002.
- Any of the available algorithms can be selected when configuring RFC 2733 FEC in the codec.

How does FEC work?

If you enter a FEC setting of 20% and you are losing one packet in every five sent, the lost packet will be replaced by FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

Note: There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

So why not use 100% FEC every time? The answer is because you need twice the bit rate to achieve full redundancy and depending on the link conditions, this could potentially cause more dropouts because of network congestion than it fixes. Here is a simple rule to remember: Your maximum uplink speed is the maximum bandwidth you have to play with. As a rule of thumb, try not to exceed more than 80% of your maximum bandwidth. If your link is shared, be even more conservative.

You should also consider the maximum upload speed at the remote end too. Is the connection shared at either end? Your bit rates, FEC settings and buffer rates must be pre-configured at both ends before you connect, so it's always better to set your connection speed and balance your FEC according to the available uplink bandwidth at each end for best performance.

As an example, if you want 15 kHz mono (using the Tieline Music Algorithm) you will need at least a 24kbps connection for audio. Adding 100% FEC will add another 24kbps making your bit rate 48kbps plus some overhead of around 10kbps is required. If you're on a 64kbps uplink, you should consider reducing your FEC to minimize the likelihood of exceeding your bandwidth capacity.

Here is another example, if you want 15 kHz stereo, you need at least 56kbps for the audio. 100% FEC requires at least 112kbps and 50% FEC requires at least 84kbps. If your uplink speed is 256kbps and you're on a shared connection, then choosing a lower FEC setting of 20%-33% may give you better results.

Conserving Bandwidth with FEC

There is a trade-off between the quality and the reliability of an IP connection – particularly when FEC is activated on your codecs. However, it is possible in certain situations to configure different FEC settings on each codec to match connection bandwidth capabilities at either end of the link, conserve bandwidth and create more stable IP connections.

For example, if your broadcast is a one-way broadcast from a remote site, i.e. you are not using the return path from the studio, or only using it for communications purposes, it is possible to reduce or turn off FEC at the studio codec. This effectively reduces the bandwidth required over the return link and increases the overall bandwidth available for the incoming broadcast signal from the remote site.

21.11 Configuring Encode/Decode Direction

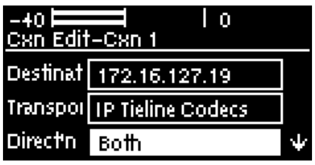
By default the codec is configured to both encode and decode data. However, it is possible to encode or decode audio data only. This is useful for:

- Conserving connection bandwidth when only unidirectional data streaming is required
- Lowering data costs
- Increasing connection reliability

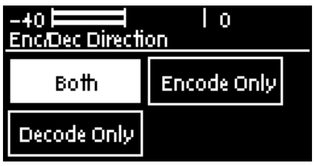
To use this feature configure the transmitting codec to encode only and the receive codec to decode only. To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Cxns** and press the OK button.
3. Select a connection and press the OK button.

4. Navigate down to **Jitter** and press OK.
5. Navigate to **Direct'n** and press OK.



6. Select **Both**, **Encode Only** or **Decode Only** and press OK.



21.12 GPIOs and RS232 Data

Data must be enabled to activate contact closure operation and RS232 data. The codec supports both in-band and out-of-band data depending on the connection transport and algorithm you are using. In-band RPTP data is automatically included within an audio stream when using Tieline Music and Music PLUS algorithms. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP
Tieline Music and MusicPLUS	• In-band RPTP data is enabled automatically • Synchronized out-of-band data can be enabled and disabled as required • Using out-of-band data with rules between G5 or G6 codecs employing relay reflection minimizes latency • These algorithms must be used when connected to G3 codecs as they don't support out-of-band data
All other algorithms	• No in-band data available; synchronized out-of-band data can be enabled and disabled

Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** or **Connections panel** to enable out-of-band RS232 data and activate rules employing relay reflection over a connection. This will allow the codec to connect to external devices and send RS232-compatible data via the serial port on the rear panel. Alternatively, enable auxiliary data using the **Setup** menu as follows:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the button.
3. Select a connection and press the button.
4. Navigate down to **Data** to enable out-of-band RS232 data and activate rules employing relay reflection over a connection. Note: default setting is **Disabled**.

Important Notes:
- When connecting to G3 codecs only in-band data is available via the Music and MusicPLUS algorithms.
- Codecs using G5 Bridge-IT firmware lower than v2.8.xx cannot activate relays on Tieline G3 codecs or send RS232 data to them.
- It is important to enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Session data sent from the dialing codec will configure all other compatible codecs (non-G3) you connect with.
- RS232 data can be sent from the dialing codec to all end-points of a multi-unicast connection. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

Configuring Control Port Contact Closure Operation

The **Rules panel** in the Web-GUI can be used to configure switch inputs and relay outputs. Codec 'rules' configure events based on specific codec actions. Typically rules are based on a change in the state of a physical **CONTROL PORT** GPIO, or a WheatNet-IP logic IO, or a codec program being connected or disconnected. Rules can only be created with the Web-GUI while the codec is disconnected. There are three ways to create rules in the HTML5 Toolbox Web-GUI:

1. **Rules panel**: Configure codec level rules related to programs and/or hardware and software I/O states.
2. **Program Manager panel**: Configure program level rules early in the **Program Manager panel** wizard.
3. **Program Manager panel**: Configure stream level rules for an audio stream when proceeding through the **Program Manager panel** wizard..

Important Notes: A non-WheatNet-IP Tieline codec can be configured to trigger a logic IO in a Tieline Genie Distribution and Merlin PLUS WheatNet-IP codec, as well as physical **CONTROL PORT** GPIOs. The codec has:

- Bridge-IT II and Bridge-IT XTRA II codecs have 4 physical **CONTROL PORT** GPIOs; 7 virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network).
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** buttons 1-3.
- Relay reflection is not available for SIP and Multicast Client programs.

Relays

A DB15 connector provides four CMOS solid state relays for the control of equipment, consisting of four relay closures and four opto-isolated outputs.

Inputs

The input signal is referenced to chassis ground, i.e. the ground reference terminal on the connector is connected the chassis. The input device is a high impedance CMOS device with a 330 ohm pull-up resistor to +5 volts. Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

Outputs

CMOS field effect transistors switch a low impedance path between the two pins when activated. These are opto-isolated and floating above ground. It is important to current-limit the source as damage will result where the current exceeds 100mA peak-to-peak. No more than 48 volts peak-to-peak should be used as a safety precaution. The resistance of the CMOS element is approximately 25 ohms in the ON state.

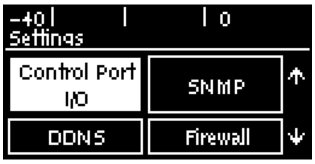
Important Notes:

- See Creating Rules for more information about configuring Control Port rules in the HTML5 Toolbox Web-GUI.
- See Appendix B for Control Port and RS232 Pinouts.

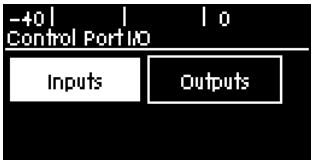
21.13 Monitor Control Port I/O Status

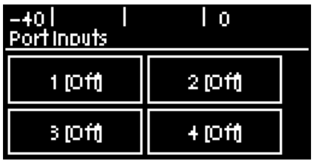
To monitor the status of the four DB15 **CONTROL PORT** relay inputs and opto-isolated outputs:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Control Port I/O** and press the button.

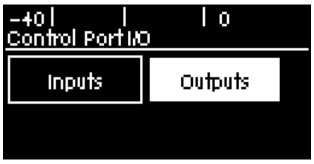


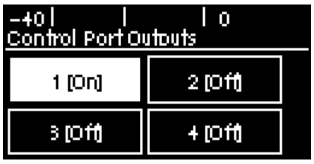
3. Select **Inputs** and then press the button to view input status.





4. Select **Outputs** and then press the button to view output status. Select an output and press the button to toggle the output state from **Off** to **On**. Note: Input states cannot be changed in this way.





21.14 Configuring TCP/UDP Ports

In TCP and UDP networks the codec port is the endpoint of your connection. Software network ports are doorways for systems to communicate with each other. For example, several codecs in your studio may use the same public static IP address. Unique port numbers can be used to port forward audio to each codec.

Tieline Codec Default Port Settings

By default, the codec uses a TCP session port to send session data and a UDP port to send audio. The session port uses the TCP protocol because it is most likely to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably.

The default session and audio port settings in Tieline codecs, for both TCP and UDP connections, are outlined in the Installing the Codec at the Studio section of the manual. This section also contains useful information for configuring port forwarding and troubleshooting IP connections.

Changing Codec Port Numbers

Reasons for adjusting the port setting on your codec include:

- Creating a path through gateways and firewalls.
- Another IP device is already using a codec's port number.

- More than one studio codec is in use and each codec requires a different port number.

Configuring the Session and Audio Port Numbers used when Dialing a Program

Codecs require matching port numbers to connect successfully. When you create a program the session and audio ports can be adjusted from the defaults as required. Note: If there is a need to change codec port settings please consult your organization’s resident IT professional. To adjust port settings:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Cxns** and press the OK button.
3. Select a connection and press the OK button.
4. Select **Destination** and press the OK button.
5. Navigate to either **Session** (session protocol) or **Audio** (audio protocol) and press OK.
6. Select **Auto** or **Manual** and press OK.
7. Use the numeric **KEYPAD** to configure the port number and press OK.

Configuring Tieline Session Ports when Answering

To adjust the local Tieline session data port used by your codec to answer incoming calls:

1. Press the **HOME** button to return to the **Home** screen.
2. Select **Settings** and press the OK button.
3. Select **Tieline Session** and press OK.
4. Navigate to **Sess. Port** (session port) or **Alt. Port** (alternative session port) and press OK.
5. Use the numeric **KEYPAD** to configure the port number and press OK to store the new setting.

Audio Port Settings for Tieline Session Data and Sessionless IP Calls

The codec supports sessionless IP streaming, whereby the codec does not send Tieline session data when attempting to connect. When using this mode you need to configure the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec). It is also possible to configure the remote and local audio ports for a codec using Tieline session data to establish IP connections. This may be required because some firewalls require symmetric port configuration.

Sessionless Audio Port Configuration

When you select **Sessionless** as the **Session Protocol**:

- The default value for both the **Send** and **Return** (audio) **Ports** is 9000
- The range of values for the audio ports is 2000 to 65535
- The audio port values can be set independently
- Both audio ports can always be configured, i.e. there is no dependency on encode/decode direction

"Tieline Codec" Port Configuration

If using the **Tieline Codec** setting for call establishment (i.e. Tieline session data is enabled), you can also change the default audio ports if required.

- The default value for the **Send** (audio) **Port** is 9000
- The range of values for the **Send Port** is 2000 to 65535
- The default port value for the **Return** (audio) **Port** is **Automatic**. Note: **Automatic** indicates that the codec will allocate the return port value and send this information to the codec to which you are dialing
- The range of values for the **Return Port** is 2000 to 65535

Sessionless Multicast Connections

For a sessionless multicast server connection:

- Only the **Send Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

For a sessionless multicast client connection:

- Only the **Return Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

21.15 Configuring QoS

It is possible for IP networks to differentiate between and prioritise data packets being transmitted through routers across networks. This is useful because in modern data networks many different IP services like email, voice, web pages, video and streaming music coexist within the same network infrastructure.

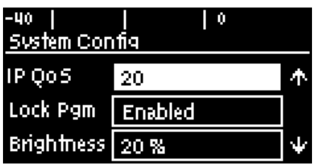
Prioritizing IP Data Packets when Broadcasting

Broadcast IP audio data packets can be configured for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks. Routers can also be configured to ignore these forwarding priorities so they are not assured across all networks.

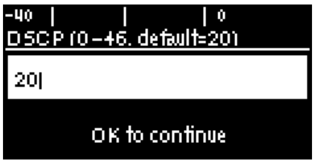
The codec can be configured to prioritise IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted by the codec over the network. Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

Configuring for QoS

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the OK button.
3. Navigate to **IP QoS**, then press the OK button.



4. Use the **RETURN** button to delete the DSCP value entered, then use the numeric **KEYPAD** to enter the new setting.



5. Press the button to save the new setting.

Important Note: To ensure the continuous and regular flow of tagged data packets along the path from point-to-point, all routers and switching equipment must allow the QoS DSCP setting. Any bandwidth partitioning schemes should partition over a small interval to ensure the codec jitter buffer does not empty and audio remains continuous.

See Configuring IP Settings for instructions on configuring this setting using the HTML5 Toolbox web-GUI.

21.16 Reset and Restore Factory Default Settings

There are several options in the **Reset/Backup** menu which allow you to restore factory default settings within the codec.

	Function	Description
1	Backup	Select to backup custom Program and/or System data.
2	Restore	Select to restore custom Program and/or System data.
3	Reset Audio and 'Connect' Settings	Select to restore factory default settings for Audio and Connect menu settings
4	Restore Factory Defaults	Select to restore factory default settings, excluding user defined programs and call history
5	Delete Programs & Call History	Deletes custom programs and recent calls in the codec; speed dial contacts are retained
6	Reboot Codec	Select to restart the codec
7	Clear Logs	Deletes codec event and log history. Note: This should only be performed if instructed to by Tieline support staff.

Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

1. Press the **SETTINGS** button.
2. Navigate to **Reset/Backup** and press the button.
3. Navigate to the preferred option from those available and press the button.

4. Confirm the change and press the button.

Reset and Restore Factory Defaults using the Web-GUI

See Reset Factory Default Settings to use the HTML5 Toolbox Web-GUI to reset and restore factory defaults.

21.17 System Backup and Restore

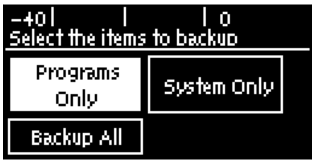
The **Reset / Backup** menu allows users to backup and restore Program information and/or system data. Backup or restore data using an SDHC card inserted into the SD slot on the front panel of the codec. Note: A single partition FAT32 formatted SDHC Card is required (SD cards may be less reliable and are not recommended).

Creating a Backup File

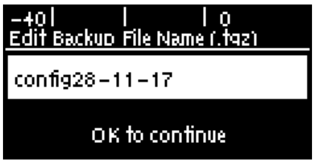
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset/Backup** and press the button.

3. Insert a single partition FAT 32 formatted SDHC card into the SD card slot on the front panel of the codec.
4. Select **Backup** and press the button.

5. Select the preferred backup option and press the button.

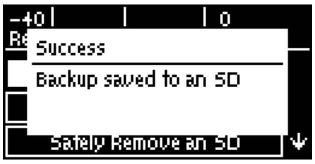


6. Use the **KEYPAD** to edit the file name and press the button.

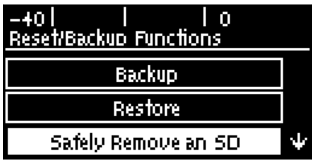


7. Navigate to a directory, or create a new directory in which to save the backup .tgz file. Then navigate to **Save** and press the button to save the file.

8. A confirmation dialog is displayed when the file has been saved successfully.



9. Select **Safely Remove an SD** before removing the SD card from the codec.

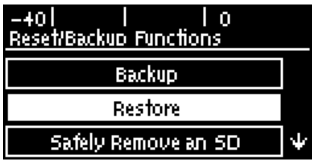


10. A confirmation dialog appears when it is safe to remove the SD card from the codec.

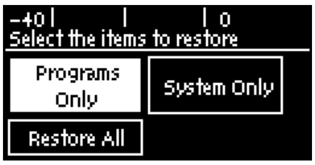
Restoring Data from an SD Backup File

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset/Backup** and press the OK button.

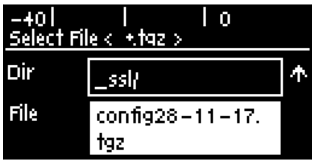
3. Insert a single partition FAT 32 formatted SDHC card into the SD card slot on the front panel of the codec.
4. Select **Restore** and press the OK button.



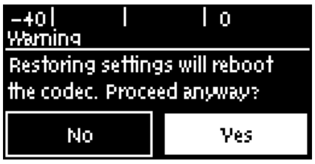
5. Select the preferred restore option and press the OK button.



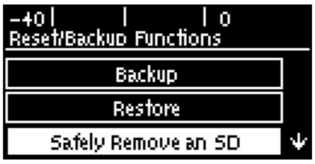
6. Select the file to restore from the SD card and press the OK button.



7. Select **Yes** to perform the system restore. Note: The codec will automatically reboot after settings have been restored.



8. Select **Safely Remove an SD card** before removing the SD card from the codec.



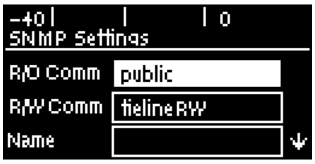
9. A confirmation dialog appears when it is safe to remove the SD card.

21.18 Configuring SNMP Settings

The codec supports Simple Network Management Protocol (SNMP) for managing devices on IP networks. To configure SNMP settings:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **SNMP** and press the OK button.

3. Navigate to each setting in turn and press the OK button to adjust and save each new setting.



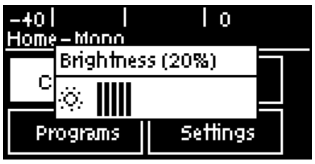
Important Note:

- Tieline recommends SNMP is disabled if a codec is connected to a public network like the internet. Adjust settings using the Toolbox HTML5 Web-GUI **Options panel** in the **Firewall** tab, or see Firewall Configuration.
- For more information on SNMP codec settings and downloading MIB files, see Configuring SNMP in the Codec.

21.19 Adjusting the OLED Screen Display

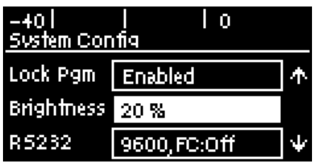
Adjusting OLED Screen Brightness Levels

1. Press and hold the F1 button and then press the arrow up button to display the **Brightness** adjustment screen.
2. Use the left and right arrow buttons to adjust the brightness until viewing is optimized.
3. Press OK when completed.



It is also possible to adjust brightness using codec **System** menus:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the OK button.
3. Navigate to **Brightness** and press the OK button.

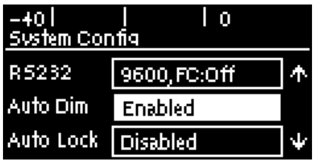


4. Use the left and right navigation buttons to adjust the brightness level up and down.

OLED Screen Auto Dim Mode

By default the codec **OLED SCREEN** has **Auto Dim** mode enabled. This dims the intensity of the display 30 secs after inactivity and is designed to maximize the working life of the screen. Disable this mode if you want the screen to be illuminated at all times.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons on the front panel to select **Settings** and press the button.
3. Navigate to **System** and press the button.
4. Navigate to **Auto Dim** and press the button to toggle between **Enabled** and **Disabled**.



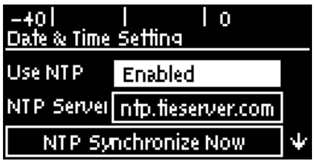
Important Note: The default **Auto Dim** time-out is reduced from 30 seconds to 10 seconds when the **Auto Lock** function is enabled (to lock the front panel controls). Disabling **Auto Dim** mode will override all time-out periods and the **OLED SCREEN** will remain fully illuminated at all times.

21.20 Adjusting Time Settings

By default **Use NTP** time is enabled in the codec. When the codec is attached to a network it will automatically ping the selected NTP server every two hours and update the time in the codec. It is also possible to manually synchronize the time. Note: The codec will not ping a server while connected. Ensure DNS settings are configured correctly as this is required for NTP.

To adjust time settings in the codec:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Time** and press the button.
3. Navigate to **Use NTP** and press the button to enable or disable this feature.



4. Navigate to **NTP Synchronize Now** and press the button to synchronize the codec time with the designated **NTP Server**.

21.21 Installing Software Licences

Checking Installed Licenses

The codec **License Manager** is used to view licenses installed in each codec. To view installed licenses:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Licenses** and press the button.

Update and Install Licenses from the Codec

1. Ensure the codec is connected to the internet and then navigate to **Update from TieServer** in the **License Manager** screen and press the button.
2. The codec will contact TieServer and automatically install all valid licenses.
3. The screen will indicate the update is in progress and then confirm it has been completed successfully.
4. Press the **RETURN** button a few times to return to the **Home** screen.
5. Use the navigation buttons to select **Settings** and press .
6. Navigate to **Reset/Backup** and press .
7. Navigate to **Reboot Codec** and press .
8. Select **Yes** and press the button to reboot the codec.

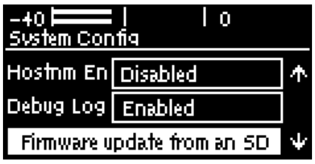
To install a software license using the HTML5 Toolbox web-GUI see Web-GUI Software License Installation.

21.22 Upgrading Firmware via SD Card

To download the latest codec firmware visit www.tieline.com. Copy the firmware file onto an SD card and then use the following procedure to perform a firmware upgrade.

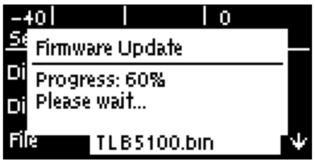
Important Note for SDHC Card: A single partition FAT32 formatted SDHC card must be used to perform the firmware upgrade.

1. Insert an SDHC card with the latest firmware into the SD card slot on the front panel of the codec.
2. Press the **SETTINGS** button.
3. Navigate to **System** and press the button.
4. Navigate down to **Firmware update from an SD** and press the button. Note: it can take a few seconds for the SDHC card to be detected.



5. Navigate to the firmware file after the SDHC card has been detected, then press the button.

6. The codec will automatically reboot after the upgrade is complete.



7. To safely remove the SD card, press the **SETTINGS** button and select **Reset/Backup > Safely Remove an SD** before removing the SD card from the codec.

Important Note: We recommend clearing your browser cache after the upgrade is complete when using the HTML5 Toolbox web-browser GUI to control codec functions. The short cuts for this are:

- Google Chrome: shift+Ctrl+delete
- Mozilla Firefox: Ctrl+shift+delete
- Internet Explorer: Ctrl+shift+delete
- Safari: Ctrl+alt+e

21.23 Installing a Security Certificate

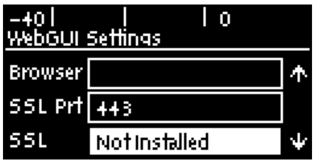
Tieline codecs support the installation of TLS/SSL (hereafter referred to as SSL) security certificates to deliver an additional layer of security when connecting to IP networks. The digital SSL security certificate authenticates the codec and provides more secure encrypted HTTPS browser connections. The codec supports installing a private key as well as an intermediate and SSL certificate.

Certificate Installation

To install certificates purchased from a reputable vendor:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Web-GUI** and press the button.

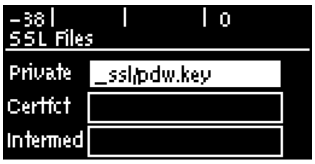
3. Navigate down to **SSL** and press the button.



4. Select **Configure SSL** and press the button.

5. Ensure the Private Key, digital SSL Certificate and Intermediate Certificate (if required), are loaded onto an SDHC card and then insert it into the SD card slot on the front panel of the codec. Note: A single partition FAT32 formatted SDHC card must be used.

6. Select **Private Key** and navigate to the correct directory and .key (Private Key) file to install from the SDHC card and press the button.



7. Select **Certificate** and navigate to the SSL Certificate (.crt) file on the SDHC card and press the button.

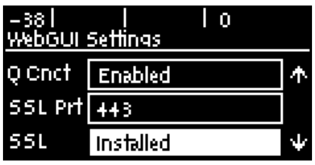
8. If an Intermediate Certificate has been supplied, select **Intermediate** and navigate to the Intermediate Certificate (.crt) file on the SDHC card and press the OK button.

9. After adding the private key, SSL certificate, and intermediate certificate (if supplied), navigate up to **Install** and press the OK button .

10. Select **Yes** to confirm installation of the certificates.

11. A dialog confirms the certificates have been installed correctly.

12. The **SSL** menu also confirms the files are successfully installed.



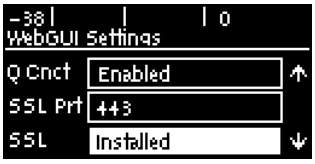
13. To safely remove the SD card, press the **SETTINGS** button and select **Reset/Backup > Safely Remove an SD** before removing the SD card from the codec.
14. To access a codec via the HTML5 Toolbox Web-GUI in a browser after installing SSL security certificates ensure you type "https://" before the codec IP address. For example, https://172.16.0.100.

Remove SSL Security Certificates

To remove installed SSL security certificates from a codec:

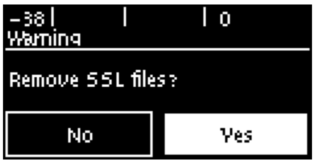
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Web-GUI** and press the button.

3. Navigate down to **SSL** and press the button.



4. Select **Remove SSL** and press the button.

5. Confirm removal of the SSL files.



6. A dialog confirms the certificates have been removed succesfully.

Changing the Default SSL Port

The codec uses the standard TCP port 443 for SSL communications. The port number can be adjusted by navigating to **SETTINGS** **> Web-GUI > SSL Prt**.

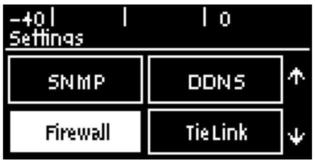
Troubleshooting Certificate Installation

If you type "https://" before the codec IP address and can't open the Toolbox web-GUI, first uninstall the certificates and then reinstall them. Also double-check you are installing the correct certificates. If you continue to have issues, contact your certificate vendor to ensure the certificate is valid.

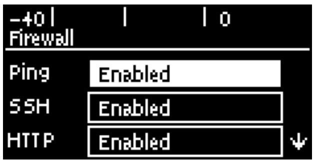
21.24 Firewall Configuration

The **Firewall** menu can be used to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Tieline also recommends that SNMP is disabled if a codec is connected to a public facing network like the internet.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Firewall** and press the button.



3. Select **Ping**, **SSH**, **HTTP**, **HTTPS**, **NTP** and **SNMP** firewall options.



21.25 Enabling CSRF Security

CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

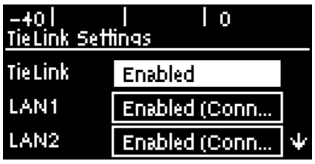
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Web-GUI** and press the button.

3. Navigate down to **CSRF** and press the button to toggle between **Enabled** and **Disabled**.

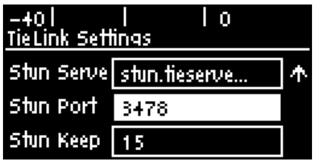
21.26 TieLink Configuration

Some TieLink settings can be adjusted in the codec using the **TOUCH SCREEN**.

1. Press the **SETTINGS** button.
2. Navigate to **TieLink** and press the button.
3. Connectivity to the TieLink Traversal Server and individual interfaces can be enabled in the TieLink Settings screen.



4. Select **TieLink** and press the OK button to toggle enabling and disabling TieLink functionality.
5. Navigate to an interface and press the OK button to toggle enabling and disabling of each interface.
6. Navigate to the bottom of the screen to adjust STUN server settings as required.



For more detailed information about TieLink configuration see Configuring TieLink Settings.

22 Reference

The following sections contain reference and troubleshooting information.

22.1 Regular Maintenance

Tieline recommends the codec undergoes regular maintenance to ensure operational efficiency and prolong its life.

SAFETY PRECAUTIONS:

- A readily accessible disconnect device shall be incorporated in the building installation wiring.
- Due to the risks of electrical shock, and energy, mechanical, and fire hazards, any procedures that involve opening panels or changing components must be performed by qualified service personnel only.
- To reduce the risk of fire and electrical shock, disconnect the device from the power line before removing the cover or panels.
- This unit has more than one power supply. Disconnect all power supplies before opening to avoid electric shock. Disconnecting one power supply disconnects only one power supply module. To isolate the unit completely, disconnect all power supplies.

SERVICING WARNINGS:

- Do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so.
- All work should be carried out by suitably qualified personnel.
- This unit has more than one power supply. Disconnect all power cables before maintenance to avoid electric shock.

HIGH VOLTAGE WARNINGS:

- Any adjustment, maintenance, and repair of the opened instrument under voltage must be avoided as much as possible and, when inevitable, must be carried out only by a skilled person who is aware of the hazard involved.
- Capacitors inside the instrument may still be charged even if the instrument has been disconnected from its source of supply.

GROUNDING:

Before connecting this device to the power line, where required, the protective earth terminal screws of this device must be connected to the protective earth in the building installation.

LINE VOLTAGE:
Before connecting this device to the power line, make sure the voltage of the power source matches the requirements of the device. Refer to the device specifications for information about the correct power rating for the device.

FUSES:

Make sure that only fuses with the required rated current and of the specified type are used for replacement. The use of repaired fuses and the short-circuiting of fuse holders must be avoided. Whenever it is likely that the protection offered by fuses has been impaired, the instrument must be made inoperative and be secured against any unintended operation.

Maintenance Schedule

Tieline recommends a three year maintenance schedule which includes the following procedures to be completed:

1. Evacuate all dust from the unit and clean vents.
2. Replace both PSUs in the Bridge-IT XTRA II. LEDs on the front panel of the Bridge-IT XTRA II, if extinguished, will indicate a failed PSU.

Controlled rack environments may allow a longer maintenance cycle. Uncontrolled environments, where temperatures are elevated, may require a shorter maintenance cycle.

Tieline recommends that the racks in which codecs are installed are thoroughly evacuated to ensure proper airflow from the bottom to the top. Where space is available, a 1RU gap between codecs will assist in minimizing internal temperature build up. Tieline has incorporated dual internal PSUs in the Bridge-IT XTRA II to assist in maintaining reliable operations. Maintaining the internal rack temperature below 25 degrees Celsius (77 degrees F) will greatly extend the codec working life.

22.2 Installing the Codec at the Studio

Studio IP Streaming Setup for Tieline Audio Codecs

The following instructions are intended to help you configure your internet connection and Tieline codecs at the studio to enable incoming calls over the internet from a remote Tieline codec. It is assumed that you have a basic understanding of your IP network and how to configure IP devices. If you have limited IT network knowledge, we recommend you engage the services of an IT professional to install the public IP address and perform the Network Address Translation (NAT) and port forwarding between the public internet and your private Local Area Network (LAN) at the studio.

Getting Started at the Studio

To perform a typical codec installation at the studio you will need to:

1. Contact your Internet Service Provider and organize a dedicated high speed broadband connection at the studio for your codec with a public static IP address. Do not share this connection with other devices.
2. Install your codec at the studio and attach an active RJ-45 LAN cable to the “LAN” or “Ethernet” port on the rear of the codec. Please note:
   - The green LED underneath the “LAN” or "Ethernet" port will illuminate and the orange LED will flash steadily if you are connected to an active LAN connection.
   - The Genie and Merlin families of IP codecs support two simultaneous Ethernet connections.

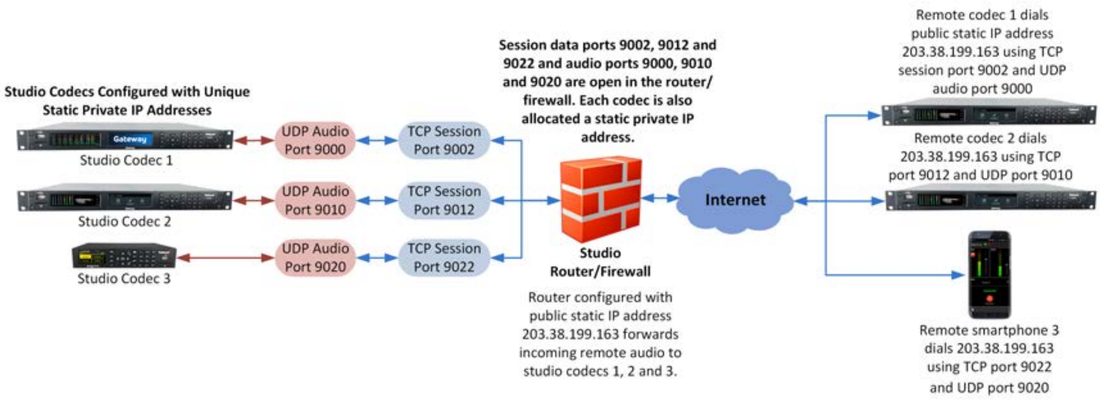
3. If you are connecting a single codec to a router without a firewall you can enter the public IP address, Subnet Mask and Gateway directly into the codec and your work is done. Note: your Telco should be able to provide this information.
4. Alternatively, if you are connected to a router with a firewall, configure Network Address Translation (NAT) in your router. NAT is performed between the public internet and your private Local Area Network (LAN) by your router. Your remote codec sends IP data packets to the studio router's public static IP address and the router performs NAT, which forwards these data packets to the private IP address allocated by the router to your codec. As part of this process we recommended you:
   - Connect to your router using a web-browser.
   - Configure it to allocate a static private IP address for each codec.

Important Note: The IP address may change if the codec is allocated a DHCP IP address by the router and it loses power or is temporarily disconnected from the LAN. This will cause problems for remote codecs attempting to dial and connect.

5. Ensure your router's firewall is configured with the relevant TCP and UDP IP ports open to allow data traffic between your codec and the remote codec. The process is fairly simple if you use the following procedure:

   a. Connect to your router using a web-browser.
   b. Navigate to http://portforward.com/english/applications/port_forwarding/Tieline-G5/default.htm (Note: when configuring a Commander or i-Mix G3 codec at the studio use http://portforward.com/english/applications/port_forwarding/Tieline-G3/default.htm)
   c. Click to select your router manufacturer from the list.
   d. Next, click to select your router model from the list.
   e. Follow the instructions to complete port forwarding

6. Visit www.portforward.com and download the port checking application to verify your router's ports are open.
7. Configure the static IP address in your codec using the instructions in the next section. To allow multiple codecs to share a single public static IP address behind a firewall and route the calls correctly, your codecs and the firewall need to be configured similarly to the example diagram which follows. Ensure the port, IP address, Subnet Mask and Gateway settings in your codecs match those configured in your router.



Port Forwarding to 3 Studio Codecs Sharing a Public Static IP Address

Important Note:

- The most common studio configuration issue is a firewall which blocks the incoming and/or outgoing TCP and UDP ports, or not configuring NAT and port forwarding correctly. The following table lists the firewall ports you need to open for each model of Tieline codec if they are dialing your router at the studio. If the remote codec is also connected to a LAN with a firewall you may also need to open the ports at the remote end of the link to connect successfully.

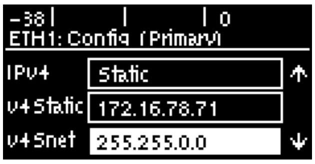
- Some firewalls require symmetric port configuration. The codec supports configuration of the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

Firewall Ports														
Commander G3 / i-Mix G3		Bridge-IT / Bridge-IT XTRA		Merlin and Genie Codec Families		ViA Codec		Gateway / Gateway 4 Codecs		MPX I / MPX II Codecs		Bridge-IT II / Bridge-IT XTRA II		Cloud Codec Controller
TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP	TCP
IP1 Session Port: 9002	IP1 Audio Port: 9000	Session Port (Sess): 9002	Audio (Proto): 9000	Session Port: 9002	Audio Port Stream 1: 9000	Session Port: 9002	Audio Port Stream 1: 9000	**See Table Below**		Session Port: 9002	Audio Port Stream 1: 9000	Session Port: 9002	Audio Port Stream 1: 9000	HTTP 80
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alt. Session: 9012	Audio Port Stream 2: 9010	Alt. Session: 9012	Audio Port Stream 2: 9010			Alt. Session: 9012	Audio Port Stream 2: 9010	Alt. Session: 9012	Audio Port Stream 2: 9010	HTTPS 443
Toolbox Software: 5550	Toolbox Software: 5550	Alt. Session: 9012	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060			Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060	
	SIP Session: 5060	Alt. Web-GUI: 8080	Fuse-IP 8999	Alt. Web-GUI: 8080	Audio Port Stream 4: 9030	Alt. Web-GUI: 8080	SIP Audio: 5004-5054			Alt. Web-GUI: 8080	Audio Port Stream 4: 9030	Alt. Web-GUI: 8080	SIP Audio: 5004-5054	
	SIP Audio: 5004	TLS/SSL 443		TLS/SSL 443	Audio Port Stream 5: 9040	TLS/SSL 443	Fuse-IP 8999			TLS/SSL 443	Audio Port MPX1 Link 8854	TLS/SSL 443	Fuse-IP 8999	
					Audio Port Stream 6: 9050						Audio Port MPX2 Link 8874			
					SIP Session: 5060			NMOS 8081	SIP Session: 5060	NMOS 8081		NMOS 8081		
					SIP Audio: 5004-5054			Ember+ 9000	SIP Audio: 5004-5054	Ember+ 9000				
					Fuse-IP 8999				Fuse-IP 8999					

Configure a Static IPv4 Address in Bridge-IT II and Bridge-IT XTRA II

Static IP addresses are fixed addresses which are recommended for studio installations. Using a static IP address ensures remote codecs can connect reliably using the same IP address over time.

1. Press the **SETTINGS** button.
2. Select **Network** and press the OK button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Config** and then **Usage** and then the appropriate control and/or streaming mode for the connection. Next, press the OK button.
5. Select **IPv4** and press the OK button.
6. Select **Static** and press the OK button.
7. Navigate to **v4 Static** and enter the IP address, then press the OK button.
8. Navigate to **v4 Snet** and enter the Subnet Mask, then press the OK button.



9. Navigate to **v4 Gway** and enter the Gateway details, then press the button.
10. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.
11. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

Critical DNS Network Setup to Complete when Installing Codecs

When installing codecs at the studio it is important to ensure that DNS settings are configured correctly. This is particularly important when using tools like the Cloud Codec Controller and TieLink. Correct DNS settings are also required for NTP, DDNS, use of host names, and if the Download and Install option is used to upgrade codec firmware.

DNS and the Cloud Codec Controller (CCC) and TieLink

If a codec has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC and TieLink servers in order to be able to contact them. When using the Toolbox HTML5 Web-GUI to configure DNS network settings:

1. Select **Transport** and then **Network** to open the **Network panel**.
2. Select an interface and then select the **DNS** tab to enter details into the **DNS Addresses** fields as required.
3. Ensure that the **Specify DNS Settings** check-box is selected. Note: This will ensure the DNS servers are used.

Note: These DNS settings can also be configured using the front panel of the codec.

Where can I find my DNS settings?

DNS settings for static public IP addresses can be sourced from your internet provider/Telco. Sometimes the **Gateway** setting in the TCP/IP tab in the **Network panel** will work as the **DNS Address** to enter.

Another common scenario encountered is that static DNS settings are configured to override DHCP settings when they are not compatible with the network being used. When using DHCP IP settings ensure the **Specify DNS Settings** check-box is disabled in the **Network panel**.

Getting Connected

Once the studio codec is configured you are now ready to receive an incoming call from the remote codec over the internet. Always dial from the field codec to the studio codec over the internet unless the remote codec is assigned a public static IP address and you know this address.If you dial the studio using a cell-phone data network at the remote site you will not normally experience any firewall or port blocking issues at the remote end of the link using default Tieline ports.

Troubleshooting: How to Determine Where Firewall Port Blocking is Occurring

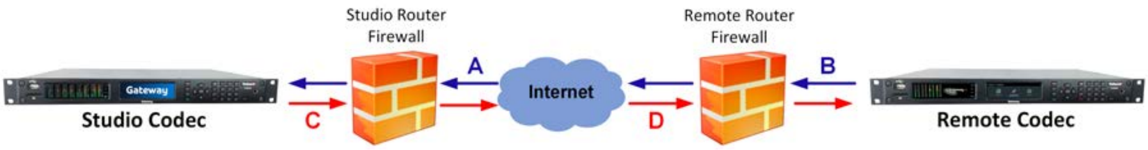
If you find you are unable to either send or receive audio between the studio and remote codecs you can use Tieline's Link Quality reading to diagnose where ports are being blocked. LQ can be displayed on the front screen of Tieline's Bridge-IT, Merlin and Genie codecs by selecting **Cxns**, then select the connection you want to view and press the **OK** button. LQ readings are also displayed on the home screen of all Commander and i-Mix G3 codecs.

Link Quality (LQ) Readings

Send and Return LQ numbers help you to determine if a problem is occurring at either end of a connection. For example, on an IP connection the Return LQ reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the Send LQ reading represents the audio data being sent by the local codec (i.e. being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** LQ reading.

Important Note:
- The **Return** link quality reading is the same as the Local (**L**) setting displayed on a G3 codec.
- The **Send** link quality reading is the same as the Remote (**R**) setting displayed on a G3 codec.

Diagnosing Port Blocking via the Studio Codec LQ
If the studio codec **Return** LQ reading is **01** then incoming audio from the remote codec is being blocked by a firewall at either point A or B in the following diagram. If the studio codec **Send** LQ reading is **01** then outgoing audio from the studio is being blocked by a firewall at either point C or D in the following diagram.



Diagnosing Port Blocking via the Remote Codec LQ
If you attach your Tieline codec at the remote site to a LAN with access to the internet you can often dial and connect to the studio without any problem. It is less likely that a firewall will block outgoing TCP and UDP ports. However, if there is a firewall at the remote site it may block incoming data packets from the studio.

The principle is the same at the remote codec for diagnosing blocked ports. If the remote codec **Return** LQ reading is 01 then incoming audio from the studio codec is being blocked by a firewall at either point C or D in the preceding diagram. If the remote codec **Send** LQ reading is 01 then the outgoing audio from the remote codec is being blocked by a firewall at either point A or B in the preceding diagram.

Troubleshooting TCP Port Blocking

Error messages on the codec screen can help to diagnose TCP port blocking.

1. **"Connection Refused"** usually means that the firewall is configured correctly but the codec is not using the expected port. For example, the firewall is set up to forward via port 9002 but codec is 'listening' to port 10,000. “Connection Refused” is not normally shown if the firewall is not configured correctly because a firewall will by

design silently drop any forwarding requests to ports that it doesn't have open (see next point). Note: "Connection Refused" will also be displayed if the Commander G3 or i-Mix G3 codec you are calling is already connected.

2. **"Connection Timeout"** can mean one of two things:
   - The firewall is not configured correctly and the attempted codec connection is being silently dropped, e.g. a remote codec is dialing to port 9002 but the studio firewall port forwarding is not configured.
   - The UDP port is not port forwarded correctly. Tieline codecs send test data during connection establishment to make sure that the audio path is configured correctly; if this process fails then it will also result in a "Connection Timeout".

How do I determine which end is blocking data flow?

Tieline test codec firewalls have the default Tieline TCP and UDP ports open. You can dial into these test codecs (or other codecs you know are configured correctly) from your recently configured studio and remote codecs and use the LQ readings to diagnose whether your studio or remote codec firewall is blocking your data packets. If one codec connects ok and the other one doesn't, then you will know which end is likely to be causing the problem. As an example:

1. Dial from site 1 to a Tieline test codec.
2. Dial from site 2 to Tieline test codec.

If both of these connect successfully then the "outbound" TCP path for session data is OK, and the inbound UDP audio path is OK.

3. Dial to site 1 from a codec you know is configured correctly.
4. Dial to site 2 from a codec you know is configured correctly.

If either of these calls fail then TCP and/or UDP inbound data is being blocked on the failed connection (see "Troubleshooting TCP Port Blocking" above).

Testing your Codec

- Visit the Tieline website at www.tieline.com and select "Support" and then "Test Lines" for a list of test IP codec addresses you can use to verify your codec is configured correctly.
- See Testing IP Network Connections for more IP test information.

22.3 Tips for Creating Reliable IP Connections

The following 10 tips are provided to help obtain the best possible IP connection between two codecs, without paying for Quality of Service (QoS).

1. Always use the best quality Internet Service Provider (ISP). Tier 1 service providers are best as their infrastructure actually makes up the internet 'backbone'. Wikipedia lists the major service providers that make up the internet backbone at: http://en.wikipedia.org/wiki/internet_backbone. In Australia Telstra is equivalent to one of these service providers.

2. You will get the best quality connection if both the local (studio) and remote codecs use the same ISP. This can substantially increase reliability, audio bandwidth and reduce audio delay. Using the same service provider nationally can give better results than using different local service providers. This is especially true if one of the service providers is a cheap, low-end domestic service provider, which buys its bandwidth from other ISPs. Second and third tier providers sublease bandwidth from first tier providers and can result in connection reliability issues due to multiple switch hops. We also highly recommend using First Tier ISPs if connecting two codecs in different countries.

3. Sign up for a business plan that provides better performance than domestic or residential plans. Business plans typically have a fixed data limit per month with an additional cost for data beyond that limit. In addition, Service Level Agreements (SLA) will often provide better support and response times in the event of a connection failure. Domestic plans are often speed-limited or “shaped” when usage exceeds a predefined limit. These plans are cheap but they are dangerous for streaming broadcast audio.

4. Ensure that the speed of the connection for both codecs is adequate for the job. The minimum upload speed recommended is 256 kbps for a studio codec and 64 kbps for a field unit connection.

5. Use good quality equipment to connect your codecs to the internet. (Tieline successfully uses Cisco® switching and routing equipment.):

- If you are using a DSL or ADSL connection make sure you purchase a high quality modem that can easily meet your speed requirements. This is especially important if you are over 4 kms from an exchange.
- If you have multiple codecs connected to a local area network (LAN) please ensure that your network infrastructure is designed for media streaming and not domestic usage. Tieline has tested several cheap 8-port switches that lose more packets between local computers than an international IP connection between Australia and the USA!
- If using a wireless connection ensure that the antenna signal strength received is strong. The type of antenna used and the amount of output gain also affects connection quality.

Important Note: You should be able to stream audio between two codecs on your LAN and get ‘link quality’ readings of S99R99. If you see anything less than this then you should get a network engineer to investigate the issue.

6. Once your internet connection is installed at the studio check that the connection performance is approximately what you ordered and are paying for. A connection can perform below advertised bit rates if:

- There is an error in ISP configuration
- There is an error in modem configuration
- There is a poor quality line between the studio and the exchange
- There are too may phones or faxes connected to the phone line
- Line filters have been connected incorrectly

You can test your internet connection speed by connecting a PC to the internet and using http://www.speedtest.net/index.php. If the bandwidth detected is low then something is wrong. Get it fixed before going live!

7. Use a dedicated DSL/ADSL line for your codecs. Do not share a link with PCs or company networks. The only exception to this rule is if an organisation has network equipment and engineers that can implement and manage quality of service (QoS) on its network.

8. Use UDP as the preferred audio transport protocol. TCP generally results in lower bit rates and random drop-outs of audio over the internet. Only use TCP if UDP is blocked by firewalls and you are unable to connect.

9. When using UDP ensure the total bit rate (audio bit rate plus header bit rate) is no more than 80% of the ISP connection rate. IP headers require around 20 kbps in addition to the audio bit rate. For example, with a 64 kbps connection the audio bit rate should be (64-20) x 0.8 = 31.2 kbps or lower. For TCP we suggest a limit of 50% or less.

10. Wireless IP connections can easily become congested and result in packet loss and audio drop-outs. It is very difficult to guarantee connection quality when there is no way of knowing how many people are sharing the same wireless connection.

Important Note: Be careful when using cell-phone connections at major events where thousands of people have mobile phones. This can result in poor quality connections and audio drop-outs if cell-phone base stations are overloaded.

IP Connection Checklist

Complete the following check list and aim for a score of at least 8 out of 10 before going live.

Number	Check	Result
1	Using a reputable Tier1 ISP that's part of internet backbone.	
2	The same ISP is being used for both codec connections.	
3	The ISP Plan is a Business Plan or equivalent.	
4	The ISP connection speed is adequate.	
5	Equipment is high quality and suitable for media streaming.	
6	The ISP connection speed has been tested and is suitable.	
7	The ISP connection is not shared with other PCs or devices.	
8	UDP is being used as the audio transport protocol.	
9	No more than 80% of ISP connection bandwidth is being used.	
10	There are no wireless connections being used.	

22.4 Testing IP Network Connections

There are a few very simple tools that you can use to test whether a codec can be reached over an IP network.

- Visit http://www.speedtest.net/ to test the upload and download speed of your IP connections and identify your public IP address.
- Visit http://www.ipfingerprints.com/portscan.php to verify your router's ports are open. Note: Using a port scanner to test a codec will be unsuccessful if you try to scan and the port is already in use, i.e. the codec is connected.
- Visit www.subnetonline.com and use an online port scanner to check for open and closed TCP ports. This site also has numerous other software tools, including an online ping web-tool for IPv4, plus TraceRoute and TracePath software tools.

Ping the Codec

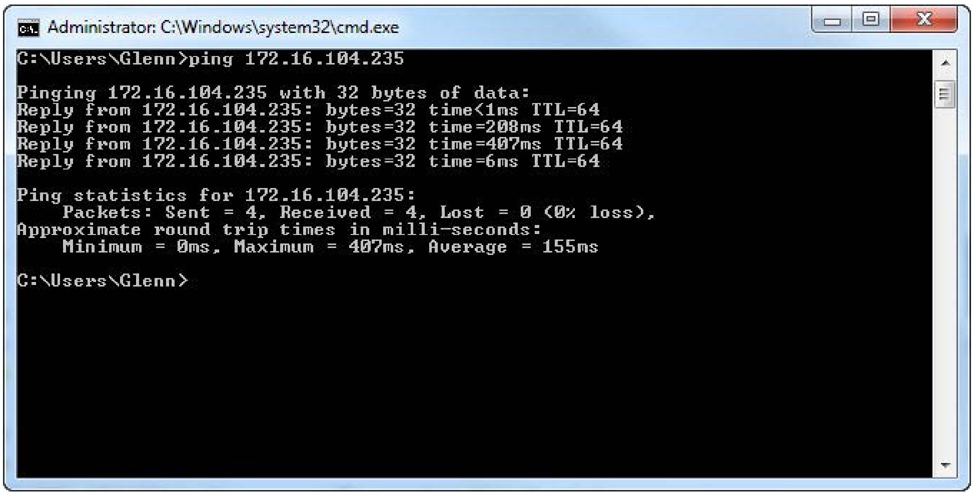
A ping test can be used to test whether it is possible to reach a codec or any device over an IP network. A ping test measures:

- The round-trip time of packets.
- Any packet loss.

There are two types of ping tests:

1. **Short test:** sends 4 packets and delivers statistics.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping** and the IP address of the codec you are pinging (i.e. **ping 192.168.0.159**) and press the **Enter** key on your keyboard.

v. The round trip time of the packets is displayed, as well as any packet loss.



2. **Long test:** sends packets continuously until stopped.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping**, the IP address of the codec you are pinging, and then **-t** (i.e. **ping 203.36.205.163 -t**) and press the **Enter** key on your keyboard.
   v. Let the test run for several minutes and then press **CTRL C**.
   vi. The round trip time of the packets is displayed, as well as any packet loss for the period of time that the test occurred.

Trace the Route of IP Packets

Another utility available on your PC is traceroute. This tool can be used to determine the route and number of hops that data packets are taking to their destination (codec). This is useful because the more routers that packets traverse, the more latency your connection will have, and the less reliable it will be.

i. Point to the **start** menu on your PC and click once.
ii. In the search text box type **Run** and press **Enter.**
iii. Type **CMD** in the **Run dialog** text box and click **OK**.
iv. Type **tracert**, the IP address of the codec you are contacting (i.e. **tracert 203.36.205.163**) and press the **Enter** key on your keyboard.

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22.6 Bridge-IT II Declaration of Conformity

Tieline® The Codec Company

Declaration of Conformity

according to ISO/IEC 17050-1 and EN 17050-1

Manufacturer's Name	Tieline Research Pty Ltd ACN 099-303-045
Manufacturer's Address	4 Bendsten Place Australia Balcatta WA 6021 Australia

declares, that the product

Product Name	Bridge-IT II
Regulatory Model Number	TLB6100
Product Options	ALL

complies with the essential requirements of the following applicable European Directives, and carries the CE marking accordingly:

Low Voltage Directive (2014/35/EU)
EMC Directive (2014/30/EU)

and conforms to the following product standards

EMC	CISPR32-2015 Class A IEC 61000-4-2:2008 / EN 61000-4-2:2008 IEC 61000-4-3:2010 / EN 61000-4-3:2010 IEC 61000-4-4:2012 / EN 61000-4-4:2012 IEC 61000-4-5:2014 / EN 61000-4-5:2014 IEC 61000-4-6:2013 / EN 61000-4-6:2013 IEC 61000-4-11:2004 / EN 61000-4-11:2004

Supplementary Information:

This Device complies with the requirements of CISPR 32:2015 Class A, Electromagnetic compatibility of multimedia equipment – Emission requirements and EN 55035:2017 Electromagnetic compatibility of multimedia equipment – Immunity requirements.

William McLean
CEO: Tieline Pty Ltd
Place: Perth, Western Australia
July 10, 2024

For regulatory topics only:

Tieline Research Pty Ltd
4 Bendsten Place, Balcatta WA 6021, Australia

Ph: 61 8 94132000
Web: www.tieline.com

Tieline
The Codec Company

Supplier's Declaration of Conformity

47 CFR § 2.1077 Compliance Information

Product Name	Bridge-IT II
Regulatory Model Number	TLB6100
Product Options	ALL

Responsible Party

Tieline Research Pty Ltd
ACN 099-303-045
C/- Tieline America LLC
7202 East 87th Street
Suite #116
Indianapolis IN 46256
USA

FCC Compliance Statement

This Device complies with Part 15 of the FCC Rules. Operation is subject to the following two Conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Tieline Pty Ltd

William McLean
CEO: Tieline Pty Ltd
Place: Perth, Western Australia

July 10, 2024

For regulatory topics only:

Tieline Pty Ltd
4 Bendsten Place, Balcatta WA 6021, Australia

Ph: 61 8 94132000

Web: www.tieline.com

Please note: Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021. Ph +61 8 9413 2000 or email info@tieline.com (web page www.tieline.com) for repair and warranty information.

22.7 Bridge-IT II Compliances and Certifications

FCC Compliance Notice

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

CAUTION:
Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

Canadian Department of Communications Radio Interference Regulations

This digital apparatus (Tieline Bridge-IT II) does not exceed the Class A limits for radio-noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

Règlement sur le brouillage radioélectrique du ministère des Communications

Cet appareil numérique (Tieline Bridge-IT II) respecte les limites de bruits radioélectriques visant les appareils numériques de classe A prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

22.8 Bridge-IT XTRA II Declaration of Conformity

Tieline
The Codec Company

Declaration of Conformity

according to ISO/IEC 17050-1 and EN 17050-1

Manufacturer's Name	Tieline Research Pty Ltd ACN 099-303-045
Manufacturer's Address	4 Bendsten Place Australia Balcatta WA 6021 Australia

declares, that the product

Product Name	Bridge-IT XTRA II
Regulatory Model Number	TLB6150
Product Options	ALL

complies with the essential requirements of the following applicable European Directives, and carries the CE marking accordingly:

Low Voltage Directive (2014/35/EU)
EMC Directive (2014/30/EU)

and conforms to the following product standards

SAFETY	AS/NZS 62368.1: 2022 (IEC 62368-1: 2018 (ED.3.0), MOD)
EMC	CISPR32-2015 Class A IEC 61000-4-2:2008 / EN 61000-4-2:2008 IEC 61000-4-3:2010 / EN 61000-4-3:2010 IEC 61000-4-4:2012 / EN 61000-4-4:2012 IEC 61000-4-5:2014 / EN 61000-4-5:2014 IEC 61000-4-6:2013 / EN 61000-4-6:2013 IEC 61000-4-11:2004 / EN 61000-4-11:2004 Canada: ICES-003 USA: FCC Title 47 CFR, Part 15 Class A

Supplementary Information:

This Device complies with the requirements of CISPR 32:2015 Class A, Electromagnetic compatibility of multimedia equipment – Emission requirements and EN 55035:2017 Electromagnetic compatibility of multimedia equipment – Immunity requirements.

This Device complies with the requirements of ICES-003 – Information Technology Equipment (Including Digital Apparatus) Limits and Methods of Measurement – Class A.

Tieline Research Pty Ltd

William McLean
Chief Executive Officer
February 14, 2025

For regulatory topics only:

Tieline Research Pty Ltd
4 Bendsten Place, Balcatta WA 6021, Australia
Ph: 61 8 94132000
Web: www.tieline.com

Tieline
The Codec Company

Supplier's Declaration of Conformity

47 CFR § 2.1077 Compliance Information

Product Name	Bridge-IT XTRA II
Regulatory Model Number	TLB6150
Product Options	ALL

Responsible Party

Tieline Research Pty Ltd
ACN 099-303-045
C/- Tieline America LLC
7202 E.87th Street
Indianapolis IN 46256
USA

FCC Compliance Statement

This Device complies with Part 15 of the FCC Rules. Operation is subject to the following two Conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Tieline Pty Ltd

William McLean
CEO, Tieline Pty Ltd
Place: Perth, Western Australia

February 14, 2025

For regulatory topics only:

Tieline Pty Ltd
4 Bendsten Place, Balcatta WA 6021, Australia

Ph: 61 8 94132000

Web: www.tieline.com

Please note: Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021. Ph +61 8 9413 2000 or email info@tieline.com (web page www.tieline.com) for repair and warranty information.

22.9 Bridge-IT XTRA II Compliances and Certifications

FCC Compliance Notice

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 may cause harmful interference, in which case the user will be required to correct the interference at his/her own expense. There is no guarantee, however, 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 receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio or TV technician for help.

CAUTION:
Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

Declaration of Conformity

The Tieline Bridge-IT XTRA II IP codec meets the requirements of directives for CE and C-Tick certifications. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021.

EU Directive 2014/30/EU (Electromagnetic Compatibility Directive) Statement

This product complies with the essential requirements of the European Union Directive 2014/30/EU (Electromagnetic Compatibility Directive) and conforms to the limits for electromagnetic emissions and immunity specified in CISPR 32 and EN 55035. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021.

Canadian Department of Communications Radio Interference Regulations

This digital apparatus (Tieline Bridge-IT XTRA II) does not exceed the Class A limits for radio-noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

Règlement sur le brouillage radioélectrique du ministère des Communications

Cet appareil numérique (Tieline Bridge-IT XTRA II) respecte les limites de bruits radioélectriques visant les appareils numériques de classe A prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

Safety of Electrical and Electronic Products and Components

The IECEE CB Scheme is an international system for mutual acceptance of CB test reports and certificates covering the safety of electrical and electronic products and components. The IEC CB is a multilateral scheme among participating countries and certification organizations, based on the use of international (IEC) standards.

This product has been tested by an independent certifying company and has been certified to comply with IEC 62368-1: 2018 (ED.3.0), MOD.

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23 Specifications

Common Input/Output Specifications	
Analog Audio Inputs	2 x Female XLR (Channel 1 mic/line; channel 2 line only)
Analog Audio Outputs	2 x Male XLR left and right outputs
AES3 Left/Right Input	1 x female XLR (Channel 1 in; shared with Ch1 analog input)
AES3 Left/Right Output	1 x RJ-45
USB 2.0 Host Port	USB Type-A supports connecting a single device like a cellular modem, or a tethered cellphone; 500mA power for charging
SD/SDHC/SDIO Card Slot	Full size push-push SD card slot for backup/restore config, firmware upgrades, and file failover playback of MPEG Layer 2, MPEG Layer 3, AAC, uncompressed 24-bit stereo PCM
AES3 (AES/EBU)	24 bit inputs with support for sample rates from 32kHz to 96 kHz; 24 bit outputs
Audio Input Impedance	High Impedance > 10k ohm
Output Impedance	<40 ohm Balanced
Clipping Level	+24dBu
A/D & D/A Converters	24 bit
Frequency Response	20Hz to 20kHz
Total Harmonic Distortion	<0.005% at +19dBu, 1kHz
Signal To Noise Ratio	> 96dB at +4dBu, unweighted, 20Hz – 20kHz
Crosstalk	< -80dB between adjacent channels
Bridge-IT II Input/Output Specifications	
Headphones	1 x 6.35mm (1/4") Jack on rear panel
Gigabit LAN Port	Gbit Ethernet port for IP streaming over WANs
Gigabit LAN/AoIP Port	Second Gbit Ethernet port for IP streaming over WANs, or configure as AoIP port for AoIP streaming (AES67, ST 2110-30, Livewire+, RAVENNA)
12 VDC, 1.5A Power Input	2.5mm Jack with Thread Lock
Bridge-IT XTRA Input/Output Specifications	
Headphones	1 x 6.35mm (1/4") Jack on front panel
Gigabit LAN Ports	2 x Gbit Ethernet ports for IP streaming over WANs
Gigabit AoIP Streaming Ports	2 x Gbit Ethernet ports for AoIP streaming (AES67, ST 2110-30, ST 2022-7, Livewire+, RAVENNA)
AC Power Inlet Sockets	Dual 100-240VAC IEC power inlets; 1A/50-60 Hz
Encoding and IP Streaming	
Encoding Formats	Tieline Music, Tieline MusicPLUS, Opus, G.711, G.722, MPEG Layer 2, MPEG Layer-3 LC-AAC, HE-AAC, HE-AACv.2, AAC-LD, AAC-ELD, 16/24 bit aptX® Enhanced algorithm. MPEG audio technologies licensed by Frunhofer IIS (http://www.iis.fraunhofer.de/audio)
Uncompressed IP	Linear PCM16/24 bit 32kHz, 44.1kHz, 48kHz sampling

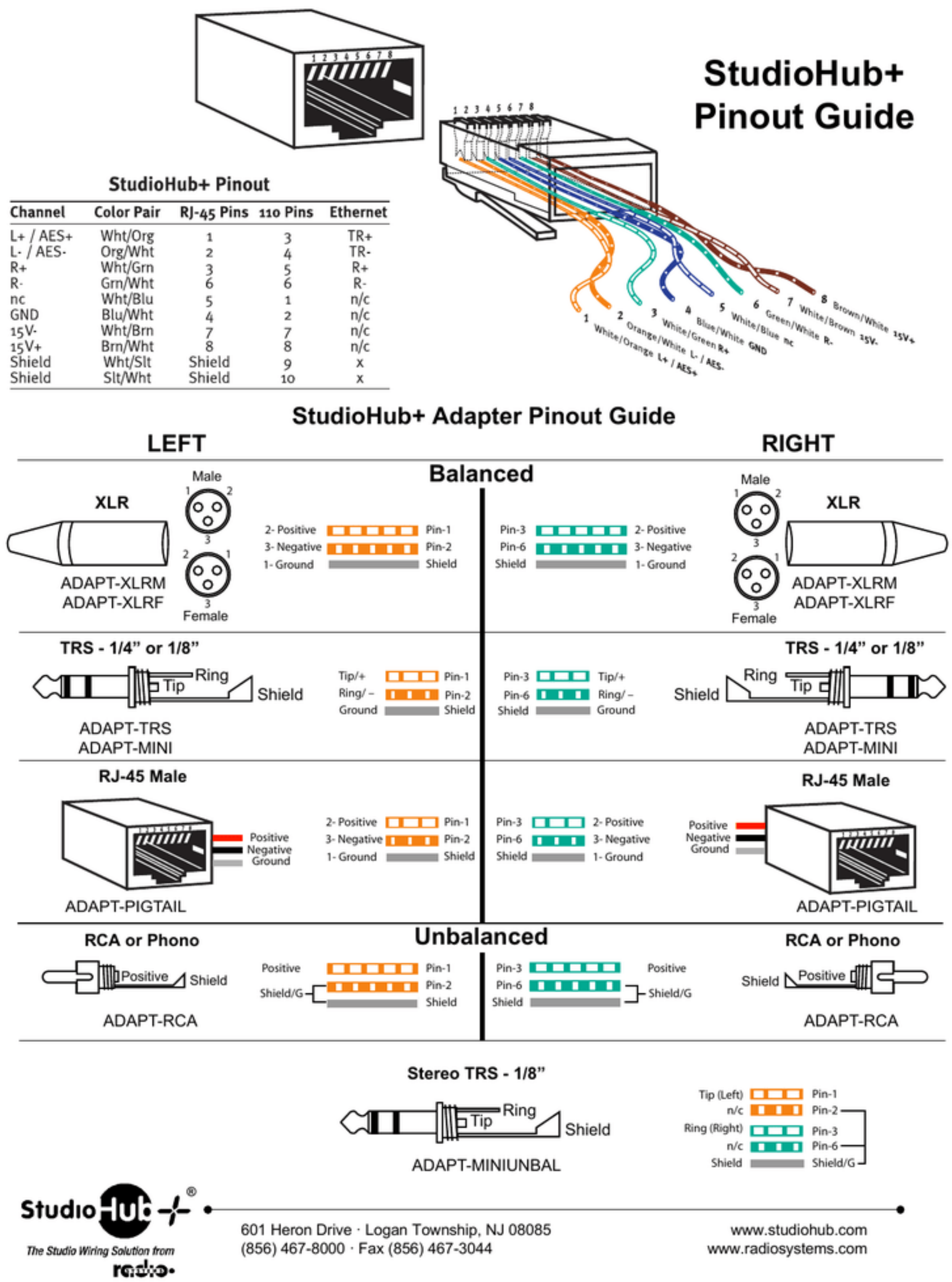
IP Sample Frequencies	8kHz, 16kHz, 32kHz, 44.1kHz, 48kHz
Asymetric Encoding	Supports asymetric multi-format encoding
Protocols	RTP, DHCP, SNMP, DNS, HTTP, IGMP, IPv4/IPv6, RTCP, STUN, SSL Security Certificate, CSRF, RFC5109, RFC5956, RFC5588, RFC4756, RFC3388, RFC5956, RFC 5588, RFC2733, RFC3190
SmartStream PLUS	Primary plus two redundant streams supported
Multi-unicasting	Support for multi-unicasting to up to 6 endpoints in Bridge-IT II and up to 10 endpoints in Bridge-IT XTRA II
Multicasting	Support for Multicasting over compatible IP networks
Fuse-IP	Bond up to 3 IP interfaces to aggregate data
Data and Control	
Configuration and connectivity	HTML5 Toolbox Web-GUI, Cloud Codec Controller (CCC)
GPIO Control Ports	4 relay inputs and 4 opto-isolated outputs via female DB-15 connector
Serial	RS232 up to 115kbps with or without CTS/RTS flow control via female DB9 connector, can be used as a proprietary data channel
AoIP Standards & Specifications	
EBU N/ACIP Tech 3326	Audio Contribution Over IP Compliant
I3P EBU Tech 3347	Intercom Over IP Compliant
N/ACIP 3368	SIP Profiles Compliant
AES67 Compliant	44.1.kHz, 48kHz, 96kHz Sample rates; 16 & 24 Bit, SDP
SMPTE 2110-30 Compliant	Sender and Receiver Compliant
RAVENNA Compliant	Natively supports RAVENNA Stream Discovery and Advertisement
Livewire Compliant	Natively supports Livewire+ for AoIP streaming
ST 2022-7 Seamless Protection Switching	Bridge-IT XTRA II is compliant with requirements for sending multiple redundant streams of RTP packets to enable seamless protection switching
NMOS Compliant	NMOS IS-04 & IS-05 Discovery, Registration and Connection Management
Supported Audio Frames	125us, 250us, 333μs, 1ms, 4ms
Clock Modes Supported	Primary Leader, Follower, Follower Only
Advanced Networking	
VLAN Tagging	IEEE 802.1Q,802.1p
Quality of Service (QoS)	Support for DiffServ (DSCP)
Synchronization	IEEE 1588-2008 (PTPv2)
Multicasting	IGMP v2 and v3
SAP	SAP v2 (Session Announcement Protocol) as defined in RFC 2974
General	
Display	128 x 64 OLED display
Keypad	20 button keypad
Navigation	5 button keypad

Bridge-IT II Dimensions	8 5/8" x 6 31/32" x 1 11/16 " [219mm (W) x 177mm (D) x 43mm (H)] including rear connectors and front panel buttons
Bridge-IT XTRA II Dimensions	19” x 6 27/32" x 1 47/64" [483mm (W) x 174mm (D) x 44mm (H)] including rear connectors and front panel buttons
Bridge-IT II Weight	3lb 2oz / 1.4kg
Bridge-IT XTRA II Weight	5lb 14 oz / 2.66kg
Bridge-IT XTRA II Power Consumption	~12W one PSU, ~18W both PSU at 240VAC dual IEC Power Inlets
Bridge-IT II Operating Temperature	0°C to 45°C (32°F to 113°F)
Bridge-IT XTRA II Operating Temperature	0°C to 45°C (32°F to 113°F)
Humidity Operating Range	30% ≤RH ≤90% (0 to 40°C), non-condensing

24 Appendix A: RJ-45 AES Pinouts

StudioHub+ Adapter Pinout Guide

RJ-45 pinouts for the codec adhere to StudioHub+ pinouts. Please see the following pinouts for RJ-45 adapters that can be used with the codec. Please note the codec does not use +/-15V on pins 7 and 8. Image courtesy of AngryAudio and StudioHub.

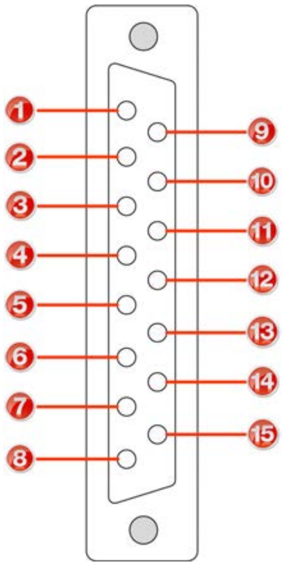


Channel	Color Pair	RJ-45 Pins	110 Pins	Ethernet
L+ / AES+	Wht/Org	1	3	TR+
L- / AES-	Org/Wht	2	4	TR-
R+	Wht/Grn	3	5	R+
R-	Grn/Wht	6	6	R-
nc	Wht/Blu	5	1	n/c
GND	Blu/Wht	4	2	n/c
15V-	Wht/Brn	7	7	n/c
15V+	Brn/Wht	8	8	n/c
Shield	Wht/Slt	Shield	9	x
Shield	Slt/Wht	Shield	10	x

25 Appendix B: Control Port and RS232 Pinouts

Control Port Pinouts

A closing contact across Inputs 1-4 to Ground will provide a closing contact on the remote codec Outputs 1 to 4. If your codec supports multi-unicast connections to multiple codecs, a contact closure will appear on each of the compatible (non-G3) remote codecs' corresponding contacts. I.e. Input 1 shorted, Output 1 contacts on all connected codecs closed.

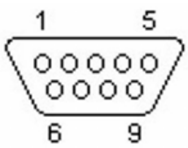


Female DB-15 Codec Connector

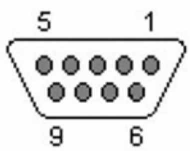
Pins	Pin Function
1	No connection
2	Output 4
3	Output 3
4	Output 2
5	Output 1
6	Ground
7	Input 3
8	Input 1
9	Output 4
10	Output 3
11	Output 2
12	Output 1
13	Ground
14	Input 4
15	Input 2

RS232 Pinouts and Data Connections

Pin	INTERFACE Female DB9 (RS232) DCE	DATA Male DB9 (RS232) DTE
1	No Connection	No connection
2	TX Data	RX Data
3	RX Data	TX Data
4	No connection	No connection
5	Signal Ground	Signal Ground
6	No Connection	No connection
7	CTS	RTS
8	RTS	CTS
9	No connection	No connection



DB9 Male Connector Pins



DB9 Female Connector Pins

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- C -

- D -

- I -

- N -

- O -

- P -

- W -